AUDIO PROJECTS

-FROM
ELECTRONICS TODAY INTERNATIONAL

Volume 1 $2·00

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Record/Playback switch (6 pole) $4.50

Famous SINCLAIR STEREO 60.

Shipment just received
Was $31.30 NOW **$27**

Use with SANKEN 10, 25 or 50 Watt MITY-AMPS. Design your own multi-purpose stereo sound system.
Stereo 60 Pre-Amp and Control Unit.
Features: Input sensitivity = 3 mV. Input impedance = 50 K. Equalization R1AA = ± 1dB. Freq. response ±1dB = 20 Hz = 25 KHz. Distortion = 0.03%. Signal to noise = 70 dB. post $1.00

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Features:—

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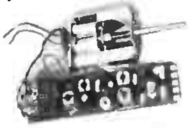
Complete Kit with step-by-step instructions	$22.50
Supplied with stereo VU recording level meters	$29.45
Power Transformer optional, add extra	$10.50
Post	$ 1.00

NEW BASIC TUNER KIT

Cheapest and best
$14.50 post 75c



A compact broadcast tuner with a multitude of uses at a sensible price. Features reliable permeability tuning (suit car radio), high sensitivity, easy construction. Supply 6 to 12V at 5mA. Output 100mV. Uses ext. aerial. Complete kit, step-by-step instructions incl.

10 Watt STEREO Kit with Sanken S1010Y

$33
P & P $3

- Ceramic input.
- Base and Treble controls.
- Rumble and Scratch Filters.
- Mains Power Supply.

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Magnetic Cartridge Kit (Stereo)	$ 9.00
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Stereo Changer in base/cover	$59.00
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As above, complete in a modern factory — made walnut cabinet 11" x 7" x 15½" high — pair $49.00.
Allow extra freight $2 to $3, accordingly.

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Great Value $21.00
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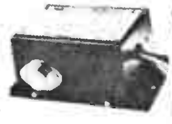

- 12 volt +ve or -ve gnc
- suit all engines to 8 cyl
- more power, greater
- peak performance at (Olims Mk II)
- instant all-weather starting.
- proven commercial design
- changeover to std. ignition.

EVERYTHING included — P.C.B., pre-wound transformer, wire, hardware, decal etc. Mk III KIT — as above, incl. car anti-theft feature $22.50., post $1.00.

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$32.50 post $2.00

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HI-FI Quality TURNTABLE

Supraphon Model HC-12

Direct Import Special only
$29.95
Pack and Post $2.00

SPECIFICATIONS:
3 speeds (16, 33, 45 r.p.m.), 4 pole induction motor, 240V, 50Hz operation, power consumption 16VA, low rumble content, performance complies with standards CSN36 7000, CSN36 8401.

FEATURES:
Includes ceramic cartridge (180mV output), manual operation only, hydraulic lowering device fitted, complete with mains and output leads.
Available in base/cover, ready to connect to amplifier or tape recorder, only $49.95.

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- RF Booster for transistor radios.
- Pulls in stations 100's miles away.

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Only **27.95**
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- High power
- Adjustable
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AUDIO PROJECTS

Volume 1

-FROM ELECTRONICS TODAY INTERNATIONAL

AMPLIFIERS

FOUR CHANNEL INTERNATIONAL 420 6
Simple unit has SQ and ambience decoding plus 15 watts per channel

DISCRETE SQ DECODER — ETI 420E 16
Alternative decoder offers same performance as IC version

PLUS TWO ADD-ON DECODER AMPLIFIER — ETI 423 20
Adapt your stereo hi-fi to full SQ operation

INTERNATIONAL 422 27
This economic amplifier provides a full 50 watts per channel

INTERNATIONAL 413 44
100 watt unit specially suited for pop groups

MIXER PREAMPLIFIER — ETI 419 48
Four-input mixer preamp specifically for ETI's 100 W power amplifier

FOUR CHANNEL — A SIMPLER WAY 63
Simple system to obtain four-channel effect from stereo records

TEST EQUIPMENT

AUDIO SIGNAL GENERATOR 54
15 Hz to 150 kHz, sine or square with less than 1% distortion

AUDIO ATTENUATOR 61
Unit provides 0-59 dB attenuation in 1 dB steps

SPEAKER SYSTEMS

SIMPLE BASS REFLEX CABINETS 92
Simplified design approach provides surprisingly effective results

EXTENSION SPEAKERS 96
How to connect extension speakers of various impedance to your amplifier

GENERAL

GRAPHIC EQUALIZER 37
Inexpensive unit compensates for speaker and room deficiencies

FET FOUR-INPUT MIXER 51
Mix any combination of four audio signals with this simple unit

THE OVER-LED 58
Is your amplifier clipping? This simple unit lets you know

EIGHT CHANNEL MASTER MIXER 64
Versatile mixer/preamplifier has full professional facilities

STEREO RUMBLE FILTER 82
Active filter design improves clarity of bass reproduction

SPRING REVERBERATION UNIT 86
Stereo reverberation unit has built-in mixer

DIN CONNECTORS 95
Standardized pin connections for DIN connectors

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Collyn Rivers
Editorial Director

Brian Chapman
Technical Editor

Barry Wilkinson
Engineering Manager

Clive Davis
Art Director

Bob Izzard
Production Manager

Terry Marsden
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Grant Evans
Advertising Representative
(Sydney)

Clarrie Levy
Manager
(Melbourne)

Ken Morrisey
Advertising Production

Louis A. Challis
and Associates
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Electronics Today International
Ryrie House,
15 Boundary Street,
Rushcutters Bay, 2011
Tel: 33-4282

By far the most popular of all our projects are those which describe Hi-Fi and Audio equipment. To cater for this demand many such projects have been presented in the pages of Electronics Today International. As a result of many requests from enthusiasts these wide ranging, thoroughly engineered projects have now been collected into one volume.

Here you may select an amplifier, or piece of ancillary equipment, which satisfies your individual needs. Full constructional details are given for all projects and ready made front panels, metalwork and woodwork may be purchased from most kitset suppliers. Thus a hobbyist, even one with limited skills, will be able to construct a project of professional appearance and performance at a fraction the cost of a commercial equivalent.

For example, the eight channel master mixer project can be constructed for around $200, and yet, its commercial equivalent may cost $1200 or more!

Unlike most other electronics magazines, we maintain a fully equipped projects laboratory staffed by experienced engineers. All our major projects are designed by our engineering manager, Barry Wilkinson and every project is built and tested before publication. The home constructor is thus assured that the project — if built according to instructions — will not only work first time, but will be a new (rather than outdated) design.

If you find these projects interesting you will find many more in the pages of Electronics Today International. Why not purchase it each month?

INTERNATIONAL 420 4-CHANNEL AMPLIFIER

By BARRY WILKINSON

Simple, yet effective unit has SQ plus ambience decoding — and produces 15 watts per channel.

TWO basic, non-compatible methods are used to obtain four-channel sound from gramophone records.

These are **discrete**, exemplified by JVC/RCA's CD-4 system; and the competing **matrix** systems, of which the Columbia SQ and the Sansui QS are best known.

Opinions differ as to which of the two main techniques has the most to offer.

Broadly, the discrete system has far better separation between channels — especially front-to-rear, but is more complex than the matrix systems. Furthermore a special cartridge and stylus is required.

All matrix systems are basically the same. Their only real differences lie in the proportions and phase relationships in which the original four channels of programme material are mixed down to two — and subsequently expanded out again to four during replay.

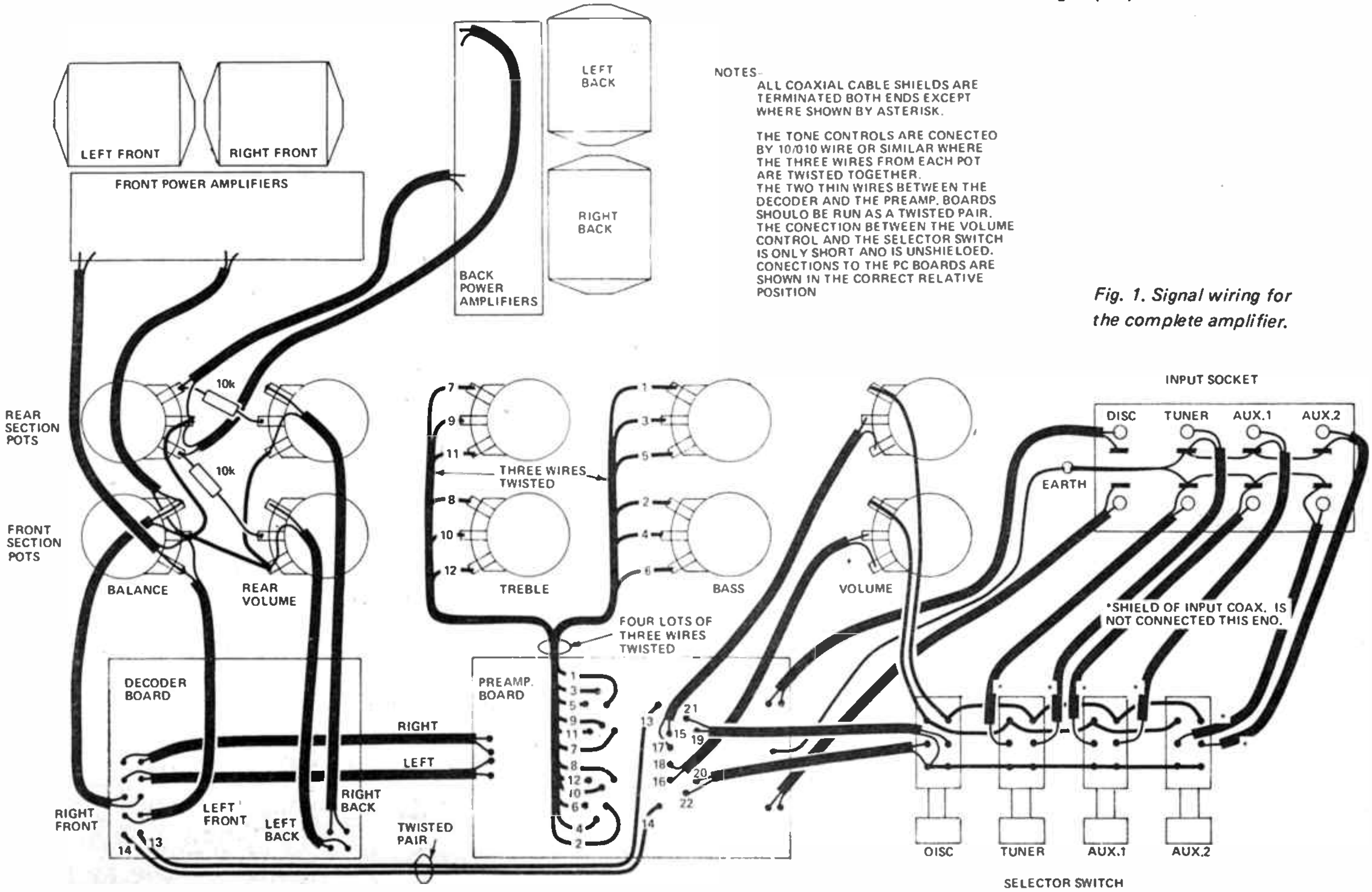


Fig. 1. Signal wiring for the complete amplifier.

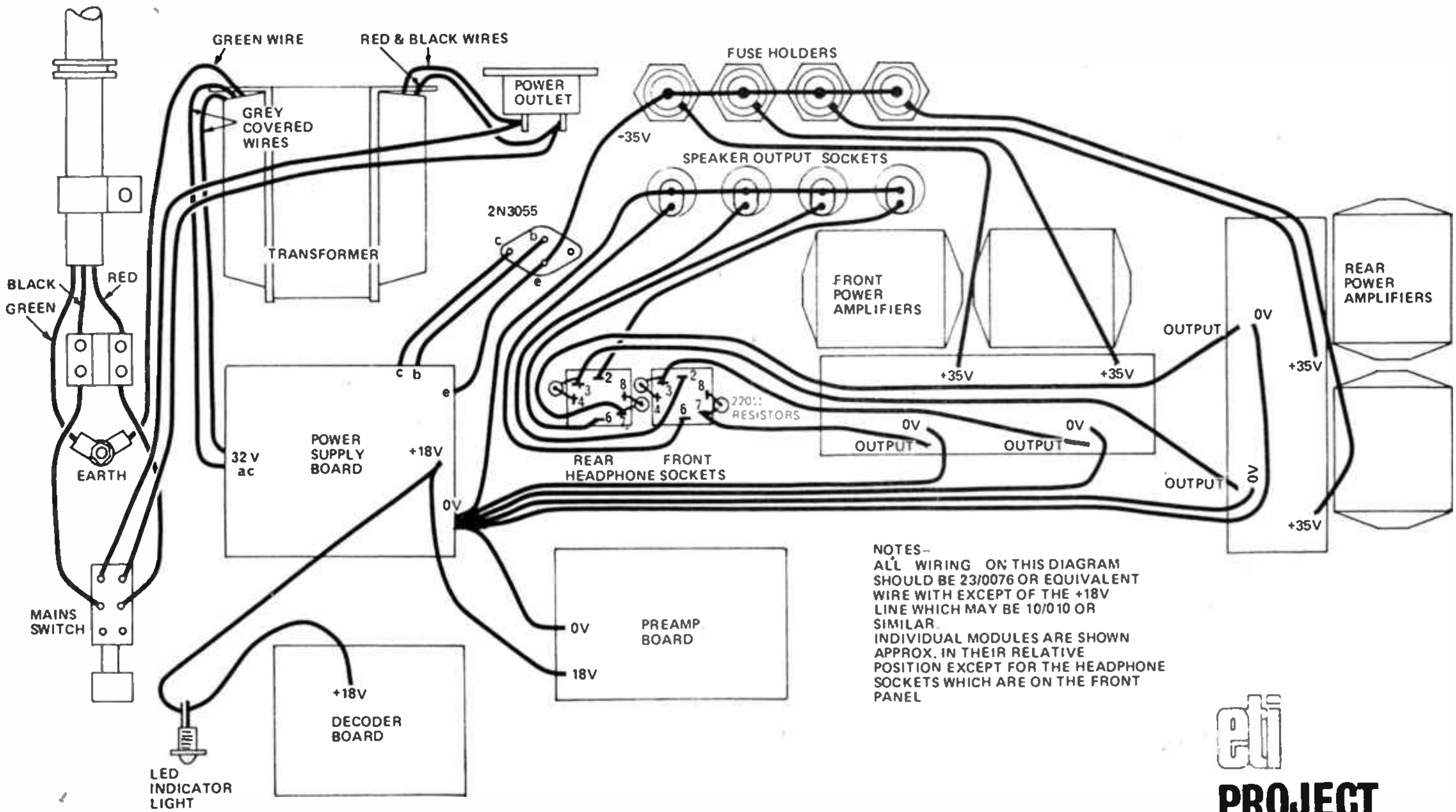


Fig. 2. Power wiring for the complete amplifier.

PROJECT 420

Channel separation of even very sophisticated matrix decoders is not as good as can be obtained from discrete units. Typically it is 10 dB left-to-right, and 3 dB or so front to rear – compared with the 20 dB – 40 dB and 20 dB respectively that is typically obtained from the CD-4 technique.

But against inferior channel separation, matrix systems are cheap, simple and relatively effective. So much so that over 100 makers of audio equipment worldwide now produce matrix four-channel systems, and of these, some 85 per cent use the so-called SQ coding/decoding technique.

The SQ system has also been chosen by the great majority of FM broadcasting stations. These stations transmit SQ matrixed programme material by conventional stereo broadcasting. The programmes are then decoded by the listeners' receiving equipment.

No special turntable, cartridge or stylus is required for use with SQ records.

Considering all the above, it is clear that the SQ system is by far the most commonly used today – hence our decision to use SQ decoding in the International 420 four-channel amplifier. We would however like to stress that this decision does not necessarily imply that we think that the SQ technique is the best

SPECIFICATION

OUTPUT POWER (at 1% distortion)			
2 channels driven	15 W per channel		
4 channels driven	12.5 per channel		
DISTORTION	100 Hz	1 kHz	10 kHz
At 0.1 W output	0.15%	0.13%	0.25%
At 1 W output	0.14%	0.11	0.18%
At 10 W output	0.14%	0.1%	0.15%
SENSITIVITY			
Disc input	1.6 mW at 1 kHz		
All other inputs	160 mV at 1 kHz		
INPUT IMPEDANCE			
All inputs	50 kohm		
SIGNAL TO NOISE RATIO (Stereo)			
Disc re 10 mV input	70 dB		
All other inputs	79 dB		
DAMPING FACTOR			
100 Hz	5		
1 kHz	30		
10 kHz	30		
SQ DECODER PHASE SHIFT			
100 Hz to 10 kHz within ± 10°			
FREQUENCY RESPONSE			
Tone controls flat	+ 0 / – 2	dB at 10 Hz	
	+ 2 / –	dB at 50 kHz	
Treble control	– 10 / + 11	dB at 10 kHz	
Bass control	– 12 / + 11	dB at 100 Hz	

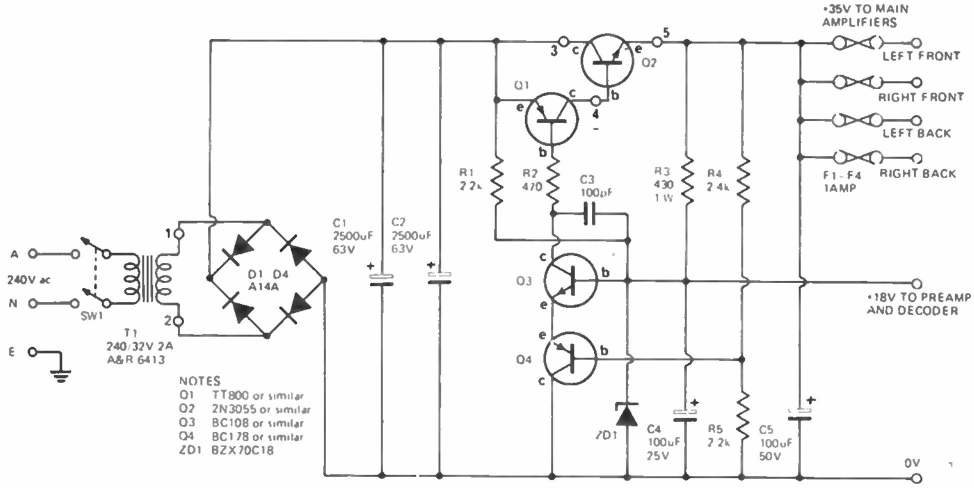


Fig. 3. Circuit diagram of the power supply.

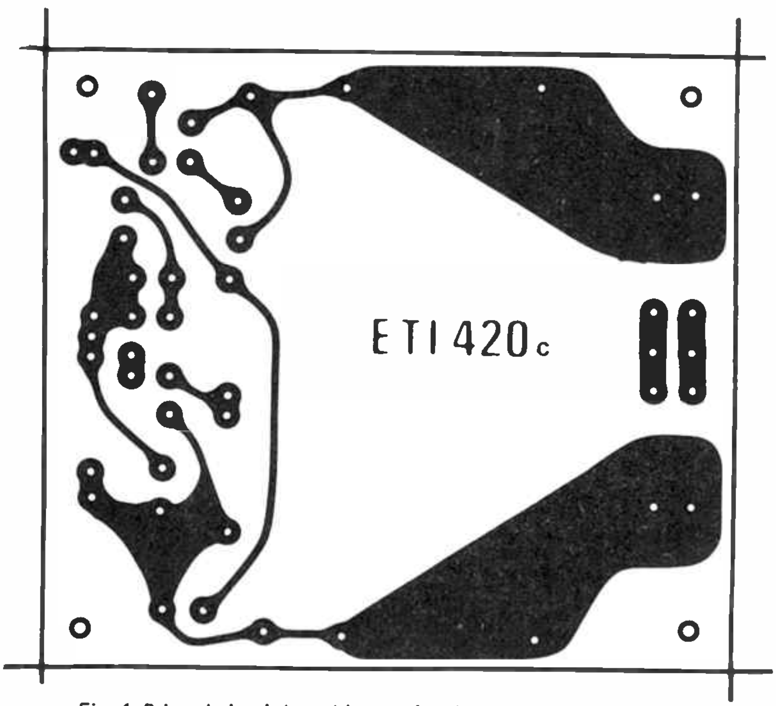


Fig. 4. Printed circuit board layout for the power supply.

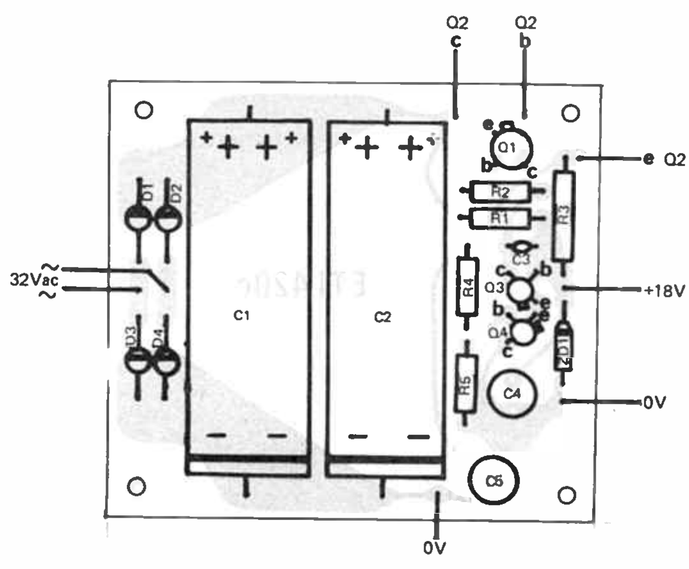


Fig. 5. Component overlay for the power supply.

technically – merely that it is the most successful commercially, and the only one for which programme material is really freely available.

Unlike a number of other four-channel amplifier projects published recently, the International 420 has been designed specifically for home construction. The unit is simple yet rugged – above all it is relatively easy to construct.

Heart of the unit is of course the SQ decoder. Here we have used Motorola's very latest SQ decoder integrated circuit type MC 1312P. This chip is produced under a licencing arrangement from the holders of the world patents, CBS. Price of the chip includes a licence fee which is paid by the seller to CBS.

It is also possible to build the SQ decoder using discrete components. Full details of this are given on page 16.

Apart from the inbuilt SQ decoder, our new four-channel amplifier incorporates 'ambience' circuitry. This enables 'synthesized' four-channel reproduction to be obtained from normal stereo records.

Further to simplify construction, the complete power amplifier stage is based on four hybrid amplifier modules. These modules are again, simple, cheap and effective.

HOW IT WORKS – POWER SUPPLY

Transformer T1 reduces the 240 Vac mains to 32 Vac which is then rectified by D1 – D4 and smoothed by capacitors C1 and C2. This provides a no-load voltage of about 48 V dc.

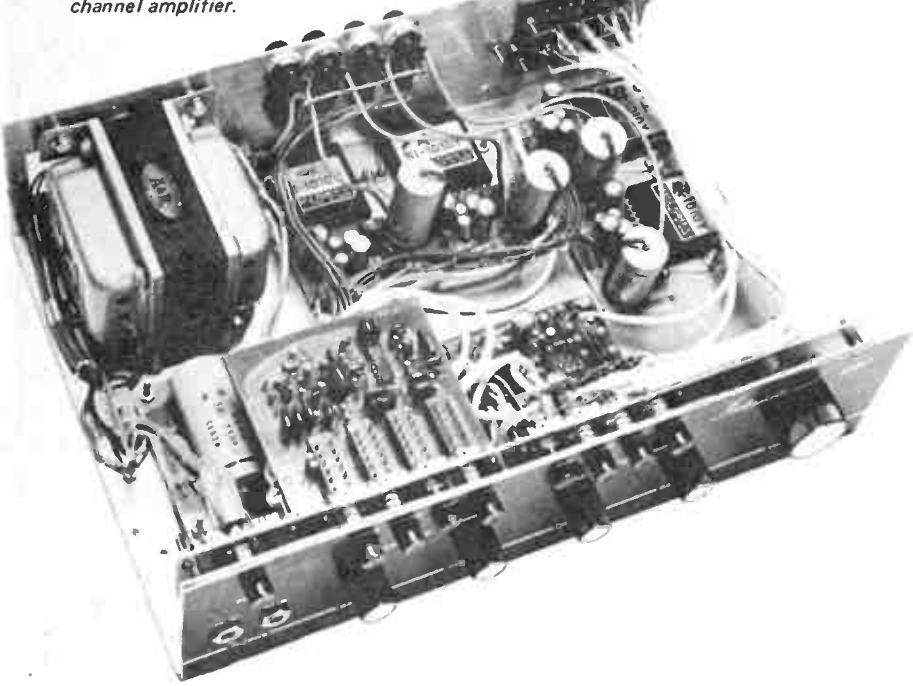
Since this is in excess of the maximum allowable for the power modules, a series regulator is used to provide a maximum of 35 V regardless of load. Transistor Q2 is the series regulator transistor, and as it dissipates a fair amount of heat it is mounted on the rear panel (but insulated from it) and has a plastic cover fitted to prevent damage to the transistor by accidental shorts.

Zener diode ZD1 (18 V) provides the reference voltage for the regulator. Transistors Q3 and Q4 compare the zener voltage with the output voltage (divided by R4/R5) and hence provide a control voltage to Q1 which in turn controls Q2. The final output voltage is thus:–

$$V_o = \frac{(V_z - 1.2) \times (R4 + R5)}{R5}$$

Fuses are incorporated in the

Internal layout of the four channel amplifier.



outputs of the main amplifier modules to protect them from damage due to accidentally shorted speaker leads, and in addition, protect the regulator from overload.

As current limit is not provided for the regulator itself, care should be taken not to cause any shorts on the regulator side of the fuses. Such shorts could damage the regulator.

The 18 volt zener also provides a regulated supply for the pre-amplifier and decoder boards. The 18 volt supply to the decoder board goes via a series LED indicator which thus provides a power-on indication.

Most of the power amplifier circuitry is contained within the Sanken SI-1010 amplifier modules and there is little point in describing the operation of these modules. The gain of the power amplifier is determined by the value of R3. The higher the value, the higher the gain. With 2.2 k the sensitivity is 650 mV P-P for 10 W and with 4.7 k the sensitivity is 350 mV P-P for 10W. The higher gain is used in the rear channel amplifiers to obtain a greater control range for the rear volume potentiometer.

CONSTRUCTION

The individual printed circuit boards should be assembled in accordance with the respective overlays. Care should be taken with regards to the polarities of all polarized components such as integrated circuits, diodes, capacitors and transistors etc.

Before soldering the power modules to the power amplifier boards, fit the boards and modules to the chassis and check alignment of the board and module since it is essential that the completed assembly aligns with the mounting holes later.

The mode selector switch is mounted directly onto the decoder printed circuit board. Press the switch fully home into the printed circuit board and solder all the pins to the board.

ASSEMBLY AND INTERCONNECTION
(Refer to Figs. 1 and 2)

1. Fit the power-cord grommet and the power cord and clamp.
2. Fit the power transformer to the rear panel using ¼ inch (6 mm) spacers (to allow room for the wires to come out). Orientate the transformer so that the red and black mains input leads are on the right-hand side.
3. Mount potentiometers and power switch to the support bracket, Fig. 19, and mount the bracket on the chassis on ½ inch (13 mm) spacers, retained by countersunk screws. Cut the potentiometer shafts to the correct length for the knobs being used.
4. Mount the external power socket, speaker outlets, fuses, input sockets and the power transistor to the rear panel. The power transistor must be mounted with insulating hardware. An insulating cover should be fitted to the transistor to prevent accidental shorts.
5. Fit a two-way terminal block, in the position shown in Fig. 2, in front and to the left of the power transformer. A screw should be fitted to the chassis just in front of the terminal block to which should be clamped the mains earth (green) and the transformer earth (also green). No other leads should be attached to this point.
6. The red and black wires from the power transformer should be terminated into the power outlet socket (one wire to each terminal) and two more wires should be taken from the power outlet socket to the mains power switch. Two further wires are then taken from the power switch to the mains terminal block as shown in Fig. 2.

Note that all wiring in the above section should be insulated for 240 Vac. All exposed 240 V connections, i.e. the power switch and output socket, should be well wrapped with insulating tape as a precaution against electrical shock whilst servicing the unit.

PARTS LIST
POWER SUPPLY

R1	Resistor	2.2k	½W	5%
R2	"	470	"	"
R3	"	430	1W	"
R4	"	2.4k	½W	"
R5	"	2.2k	"	"
C1,2	Capacitor	2500µF 63V electrolytic TYPE RT or similar		
C3	"	100pF ceramic		
C4	"	100µF 25V electrolytic PC mounting		
C5	"	100µF 50V electrolytic PC mounting		
Q1	Transistor	TT800 or similar		
Q2	"	2N3055 or similar		
Q3	"	BC108 " "		
Q4	"	BC178 " "		
D1-D4	Diode	A14A		
ZD1	Zener diode	BZX70C18		
T1	Transformer	240V/32V @ 2A A & R 6413 or equivalent		
SW1	Switch	McMurdo Type 2904-1		
PC Board		ETI 420C		
F1-F4		1 Amp Fuse and panel mounting holders		
		Cover for 2N3055 transistor		
		insulator kit for 2N3055		

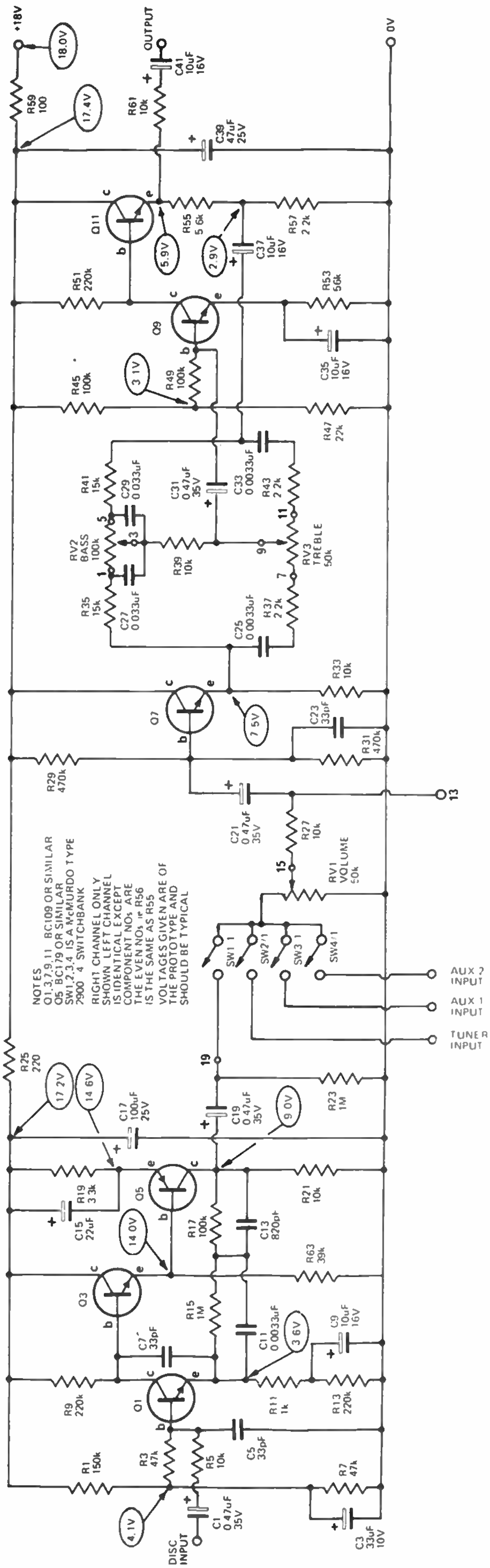


Fig. 6. Circuit diagram of the preamplifier.

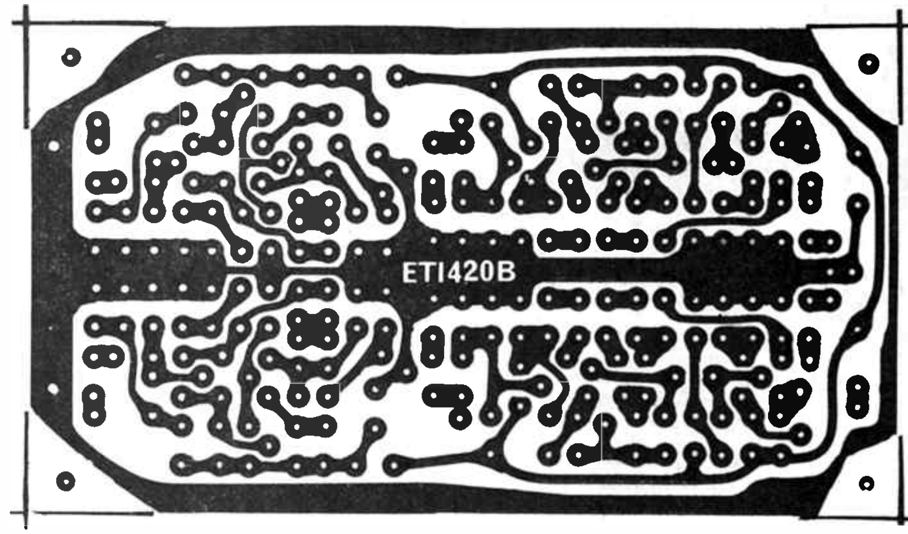


Fig. 7. Printed circuit board for the preamplifier (full size).

PARTS LIST – PREAMPLIFIER

R1,2	Resistor	150k	1/4 or 1/2 W	5%
R3,4,7,8	"	47k	"	"
R5,6,21,22,27,28,33	"	10k	"	"
R34,39,40,61,62	"	10k	"	"
R9,10,13,14,51,52	"	220k	"	"
R11,12	"	1k	"	"
R15,16,23,24	"	1M	"	"
R17,18,45,46,49,50	"	100k	"	"
R19,20	"	3.3k	"	"
R25,26		220	"	"
R29,30,31,32	"	470k	"	"
R35,36,41,42	"	15k	"	"
R37,38,43,44,57,58	"	2.2k	"	"
R47,48	"	22k	"	"
R53,54	"	56k	"	"
R55,56	"	5.6k	"	"
R59	"	100	"	"
R63,64	"	39k	"	"
C1,2,19,20,21,22,31,32	Capacitor	0.47 µF 25 V tag tantalum		
C3,4	"	33µF 10V electrolytic or tag tantalum *		
C5,6,7,8,23,24	"	33pF ceramic		
C9,10,35,36	"	10µF 16V electrolytic or tan tantalum *		
C11,12,25,26,33,34	"	0.0033µF polyester		
C13,14	"	820pF ceramic		
C15,16	"	22µF 16V electrolytic or tag tantalum *		
C17,18	"	100µF 25V electrolytic *		
C27,28,29,30	"	0.033µF polyester		
C37,38,41,42	"	10µF 16V tag tantalum		
C39	"	47µF 25V electrolytic *		

* PC Mounting capacitor

RV1	Potentiometer	50k log dual rotary
RV2	"	100k lin dual rotary
RV3	"	50k lin dual rotary
Q1,2,3,4,7,8,9,10,11,12	Transistor	BC109, BC549 or similar
Q5,6		BC179, BC559 or similar

SW1-4 Switch assembly McMurdo type 2900-4
PC board ETI 420 B

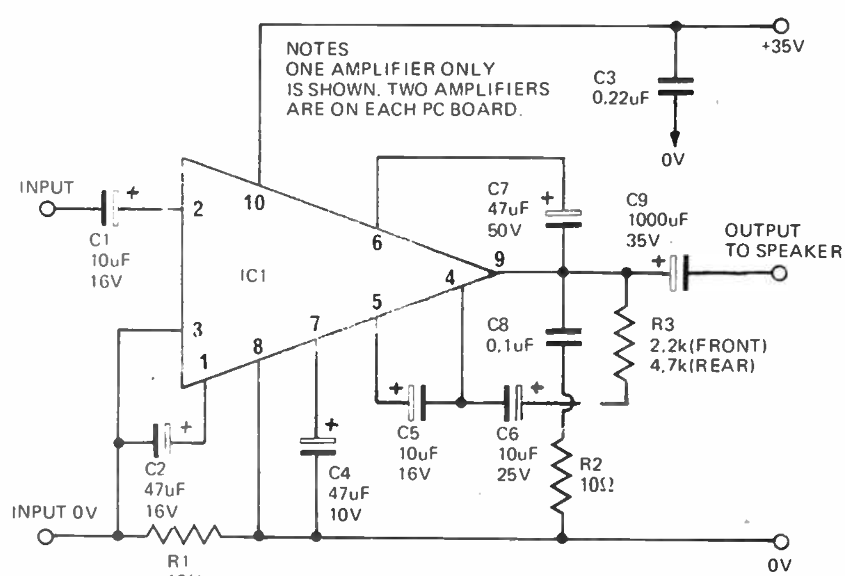
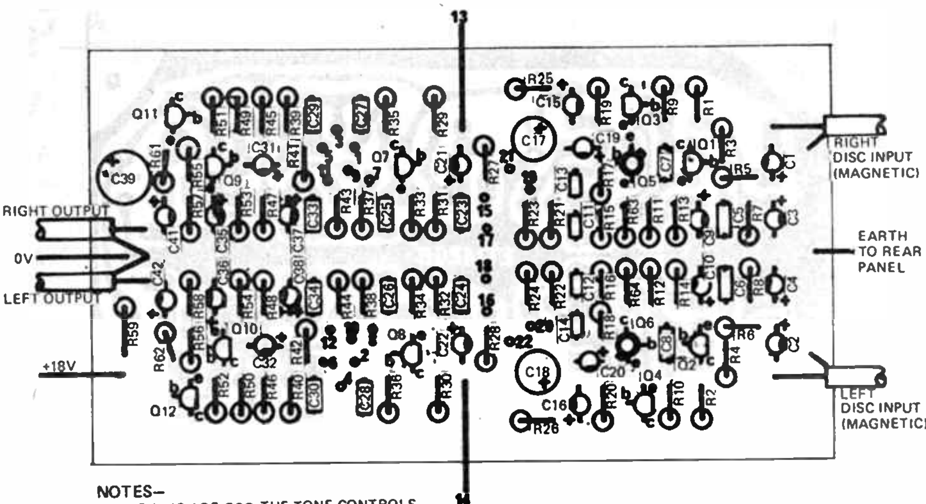


Fig. 9. Circuit diagram of one power amplifier module (two per assembly).



NOTES—
PADS 1–12 ARE FOR THE TONE CONTROLS
WIRES 13 & 14 GO TO THE DECODER BOARD
PADS 19 & 20 ARE THE MAG. AMP. OUTPUT
PADS 21 & 22 ARE THE SHIELD OF 19 & 20
PADS 15 & 16 ARE ARE THE PREAMP. INPUTS
PADS 17 & 18 ARE THE SHIELD OF 15 & 16

Fig. 8. Component overlay for the preamplifier.

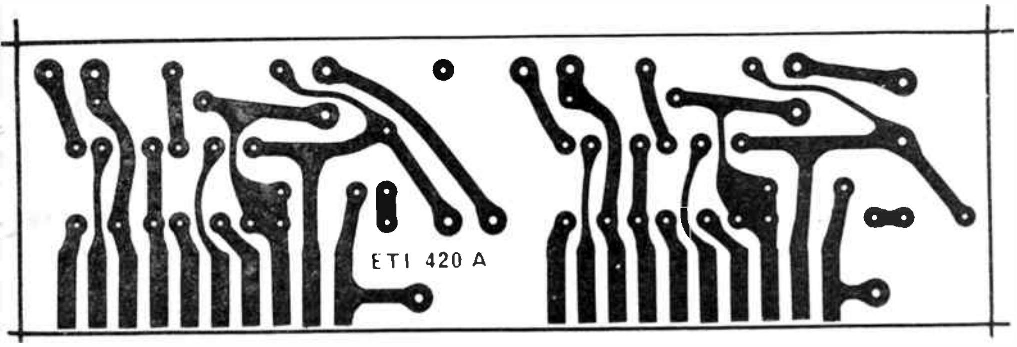


Fig. 10. Printed circuit board for the twin power amplifier assembly.

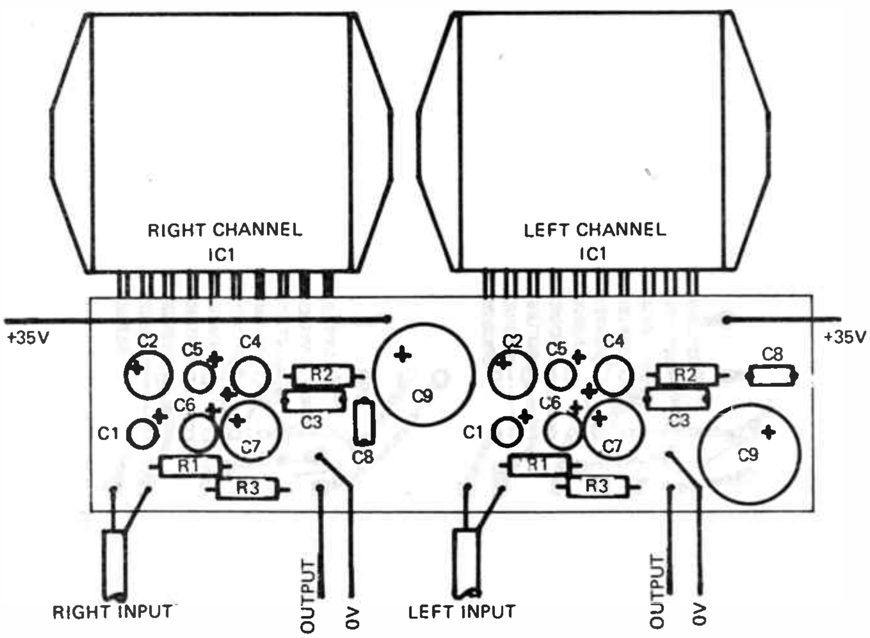


Fig. 11. Component overlay for the twin power amplifier assembly.

HOW IT WORKS – PREAMPLIFIER

The output level of a magnetic cartridge may be as low as 1 mV and this must be amplified and equalized before being applied to the tone controls.

Transistors Q1, 3 and 5 form this equalizing amplifier. The gain is controlled by R11, and the frequency response by R15, R17, C11 and C13. This complex network provides the correct RIAA equalization, the desired signal source and appropriate network being selected by SW1, 2, 3 and 4. The signal is then passed to Q7 which buffers the output of the volume control and drives the tone control network.

Transistor Q9 and Q11 form a high gain amplifier in which the gain is determined by the relative positions of the bass and treble controls. The 1 kHz gain is being approximately 2.

When monophonic mode is selected the outputs are combined after the volume control.

7. On the power amplifier boards, connect leads for +35 V, 0 V and the speaker output. Allow plenty of length for these wires so that they may be loomed up later. The wire gauge used for these leads should preferably be 23/.0076. At this time also connect the input coaxial cables to these boards making sure that the shield is also connected to the board at this end.

8. Mount the two power amplifier boards into the chassis. The modules are bolted directly to the chassis and the boards are supported by ¼" (6 mm) spacers.

9. The power supply may now be installed. Wires may be soldered directly to the printed circuit board, or, to pins inserted in the appropriate places on the board. The power transistor (mounted on the rear of chassis) should be connected to the power supply board using 23/.0076 wire. Connect all the 0V lines to the board at this time, plus the single +35 V line which goes directly to the fuse holders on the rear panel. If it is decided to solder wires direct to the underside of the board, make sure the leads are left long enough to gain access to the bottom of the board later.

10. The speaker sockets, phone jacks and +35 V supply lines to the power amplifier (from the fuses) may now all be connected using 23/.0076 wire.

11. Terminate the power amplifier input cables onto the balance control (RV5). The wipers (centre contact) of these potentiometers are at 0 V and hence the shields of the cables should

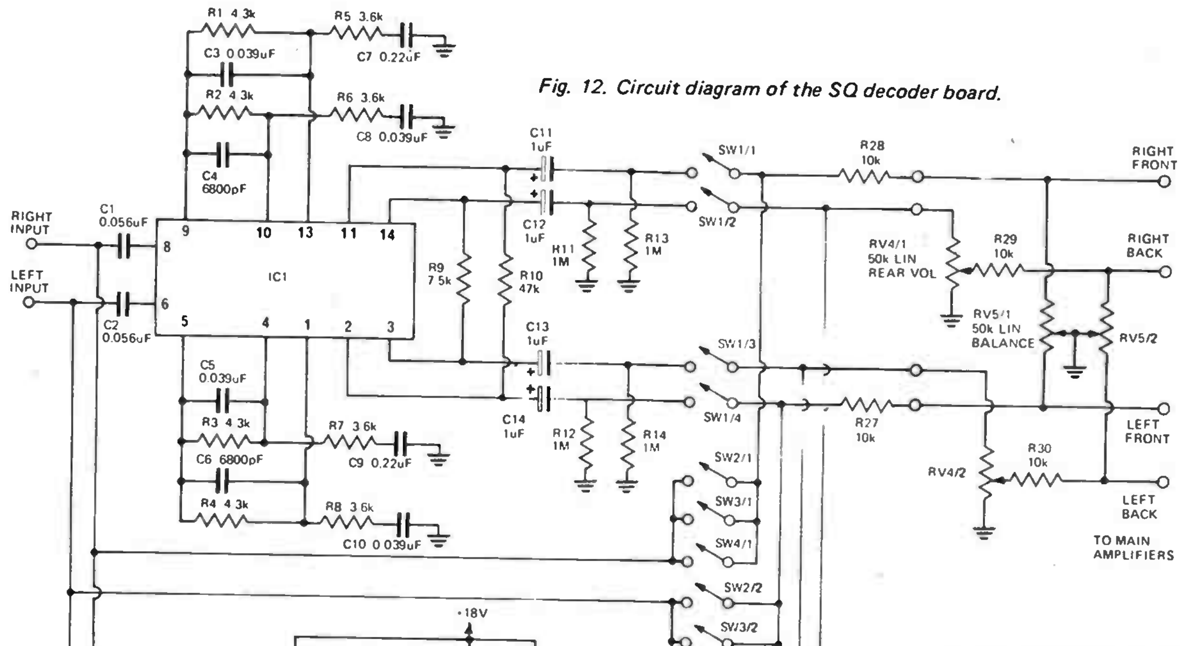
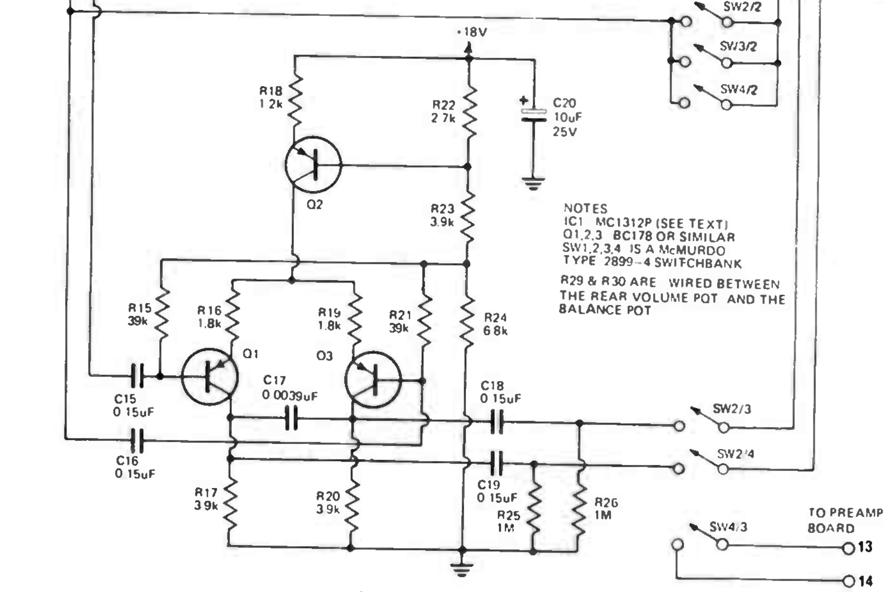



Fig. 12. Circuit diagram of the SQ decoder board.

all be connected to these points. Two resistors R29 and R30, are mounted between RV5 and RV4 (rear volume). Fit 6 inch (15 cm) lengths of coaxial cable to the inputs of these potentiometers. The other ends will be terminated later, on the decoder board.

12. The preamplifier board has a number of coaxial cables terminated on it (four per channel) these are:—

(a) Disc input (12" – 30 cm long)

(b) Disc output to selector switch (6" – 15 cm long)

(c) Preamplifier input (to volume pot) 8" – 20 cm

(d) Preamplifier output (10" – 25 cm long)

The tone control connections are made by 10/.010 wire, the three wires to each pot being twisted together.

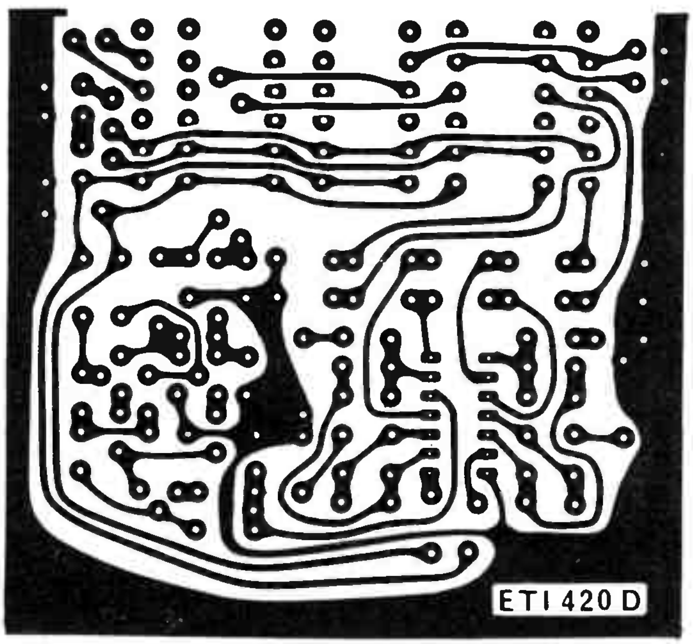


Fig. 13. Printed circuit board for the SQ decoder.

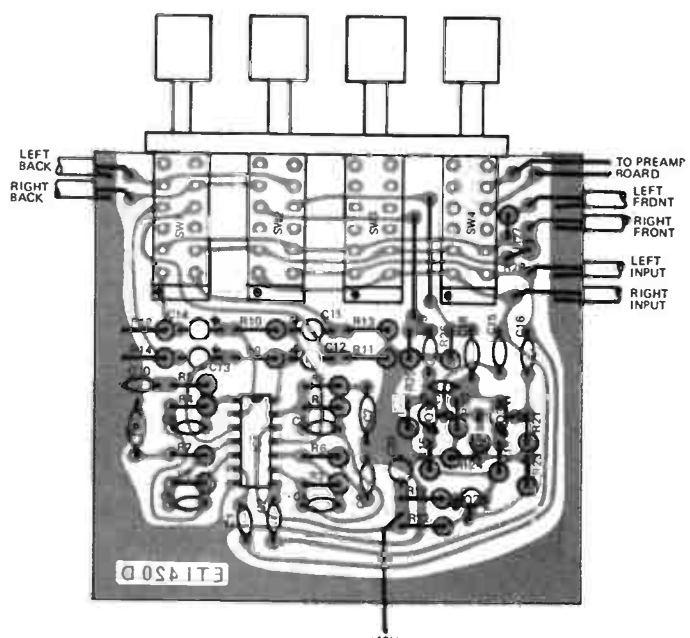


Fig. 14. Component overlay for the SQ decoder.

INTERNATIONAL 420 4-CHANNEL AMPLIFIER

PARTS LIST
AMPLIFIER

R1,2	Resistor	10Ω	½W 5%
R3, (front amps)	"	2.2k	" "
R3 (rear amps)	"	4.7k	" "
C1,5	Capacitor	10μF 16V electrolytic*	
C2	"	47μF 16V "	
C3	"	0.22μF polyester	
C4	"	47μF 10V electrolytic*	
C6	"	10μF 25V "	
C7	"	47μF 50V "	
C8	"	0.1μF polyester	
C9	"	1000μF 35V electrolytic*	

*all electrolytic should be PC mounting type.

IC1	Amplifier Module	Sanken SI-1010Y
PC Board ETI 420A	(pc board holds two amplifiers)	

Note that four sets of the above are needed for each complete 420 amplifier.

PARTS LIST
DECODER

R1,2,3,4	Resistor	4.3k	¼ or ½W 5%
R5,6,7,8	"	3.6k	" "
R9	"	7.5k	" "
R10	"	47k	" "
R11,12,13,14,25,26	"	1M	" "
R15,21	"	39k	" "
R16,19	"	1.8k	" "
R17,20,23	"	3.9k	" "
R18	"	1.2k	" "
R22	"	2.7k	" "
R24	"	6.8k	" "
R27,28,29,30	"	10k	" "
RV4	Potentiometer	50k	Lin Dual Rotary
RV5	"	50k	Lin Dual Rotary
C1,2	Capacitor	0.056μF polyester	
C3,5,8,10	"	0.039μF "	
C4,6	"	0.0068μF "	
C7,9	"	0.22μF "	
C11,12,13,14	"	1μF 25V electrolytic or tag tantalum	
C15,16,18,19	"	0.15μF polyester	
C17	"	0.0039μF "	
C20	"	10μF 25V electrolytic or tag tantalum	
IC1	Integrated circuit	MC 1312P (Kitsets Aust)	
Q1,2,3	Transistor	BC178, BC558 or similar	
SW1,2,3,4	Switch bank	McMurdo type 2899-4	
PC Board	ET1420D		

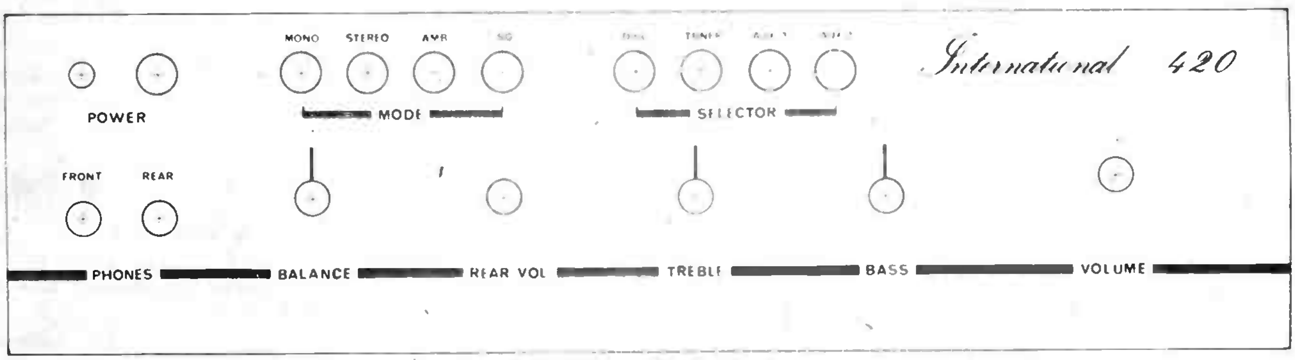


Fig. 15. Artwork for front panel escutcheon of the ETI 420 four channel amplifier.

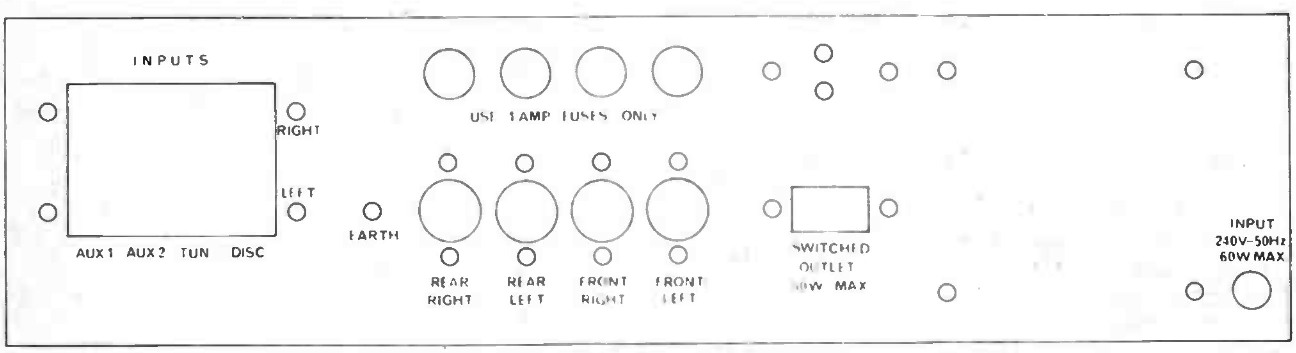


Fig. 16. Artwork for the rear panel.

HOW IT WORKS – DECODER

To properly decode SQ records it is necessary to phase shift the input signals and then selectively mix the phase shifted and original signals to synthesize the four outputs required. The Motorola IC, MC1312P, with the aid of a few external components, provides the four outputs very simply.

With conventional stereo records a mode of synthesizing four channel, known as 'Ambience Mode', has been used. This is quite effective and very simple. Normally in this method the two rear speakers are connected in series, but in anti-phase, across the live outputs of the front speakers. In our case it is difficult to interface this system directly without complex switching. As we must use the rear-channel power amplifiers, we perform the same function by using a differential amplifier Q1, Q2 and Q3, the output of which is simply the difference of the two input signals with two outputs in antiphase. Switches SW1, 2, 3 and 4 select the appropriate output.

Allow about 6 inches (15 cm) for each lead. A further twisted pair is fitted between the preamplifier and decoder boards for 'mono' mode selection (allow about 8 inches (20 cm) of wire) and an earth lead is connected to the rear panel earth lug.

Note that on the input RCA sockets all commons are tied to the rear panel earth lug with the exception of those for the disc input which are individually connected to the preamplifier board.

The preamplifier power rails can now be connected and the preamplifier board bolted into position using ¼

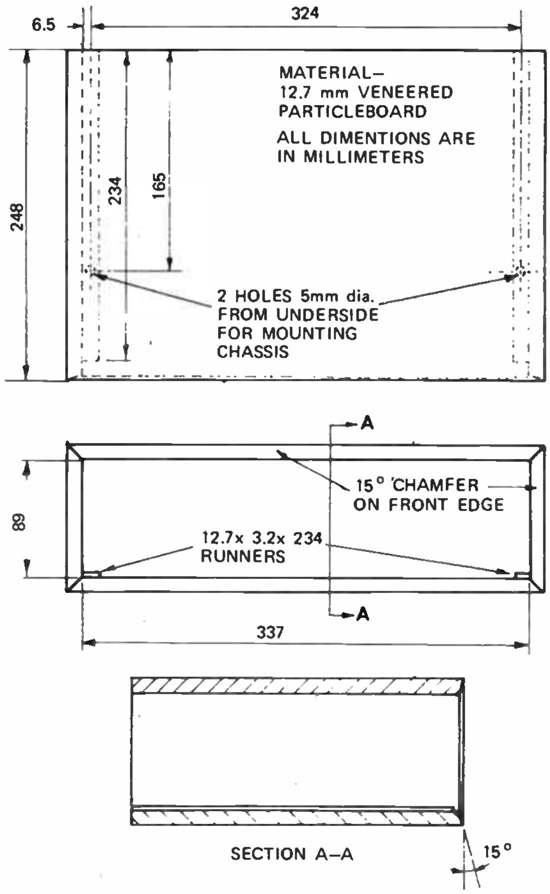


Fig. 17. Constructional details of the cabinet.

PARTS LIST

CHASSIS AND MISCELLANEOUS

14 spacer ¼" long
5 spacer ½" long
LED indicator McMurdo bezel type 3240-01-02 or similar
4 small knobs John Carr type TK196 or similar
1 large knob " " " TK195 or similar
2 stereo phone jack McMurdo type P/N1291-04-02 or similar
4 220Ω ½W resistor phone jack.
metal chassis to Fig. 20
metal bracket to Fig. 19
front panel to Fig. 18
wood box to Fig. 17
4 two-pin DIN sockets
2 4 way RCA sockets JAYCAR type
earth terminal for rear panel
2 pin American type power outlet
Main cord, grommet and clamp
2 way terminal block
nuts bolts and washers
23/0076 wire
10/010 wire
shielded cable

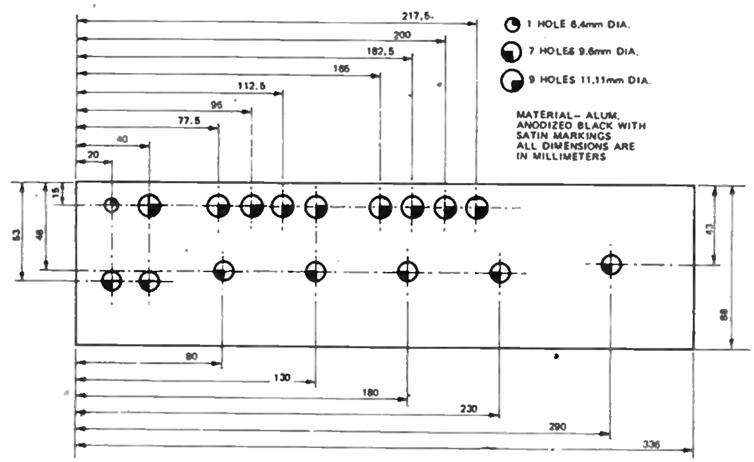
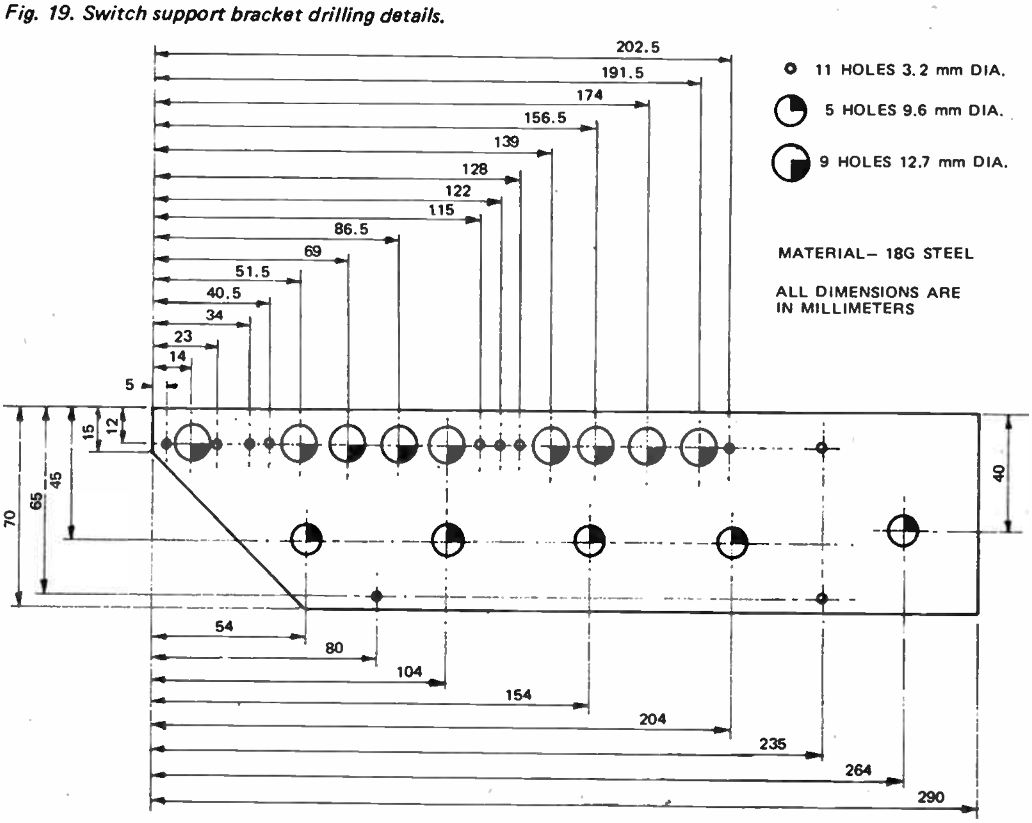


Fig. 18. Drilling details for front panel escutcheon.

Fig. 19. Switch support bracket drilling details.



inch (6 mm) spacers. Check that these spacers do not short any of the copper tracks on the board, if so, use a small insulated washer between the spacer and the board.

13. Wire up the tone controls in accordance with Fig.1. Again using Fig.1 as a guide, wire the selector switch, volume control and all input wiring.

14. The +18 V power rail may now be connected to the decoder via the front panel LED power indicator. Make sure that the supply is connected to the thin lead of the LED and the thick lead is connected to the decoder board. This ensures correct polarity of the diode, which could be damaged under reversed supply conditions.

There is no direct earth connection to the decoder board, earthing is provided by the shields of the preamplifier output coaxial cables. Additionally, the earths to RV4 and RV5 are made via the shields on the coaxial cables from the decoder board. The decoder board may now be bolted onto the support bracket. ●

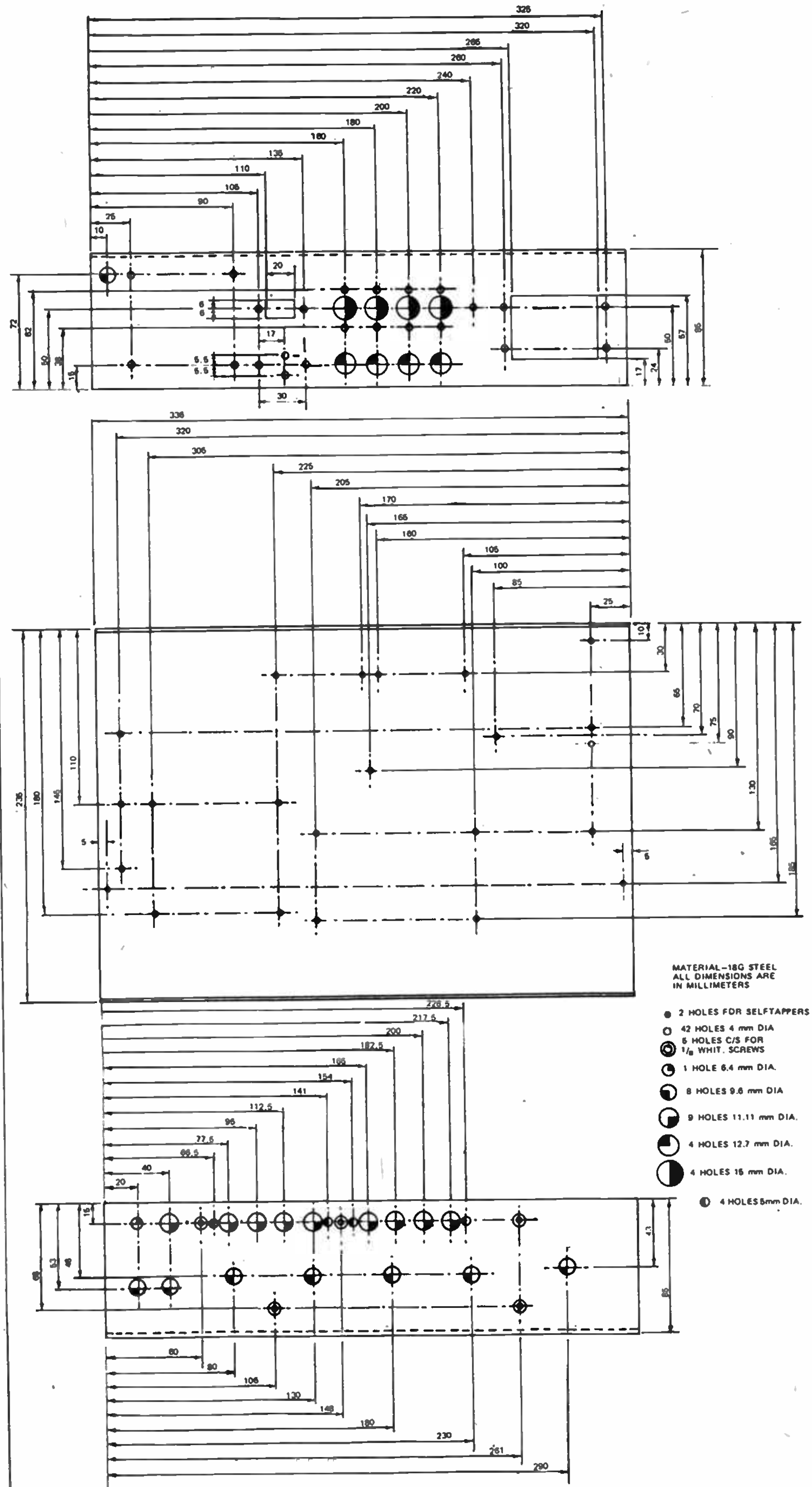
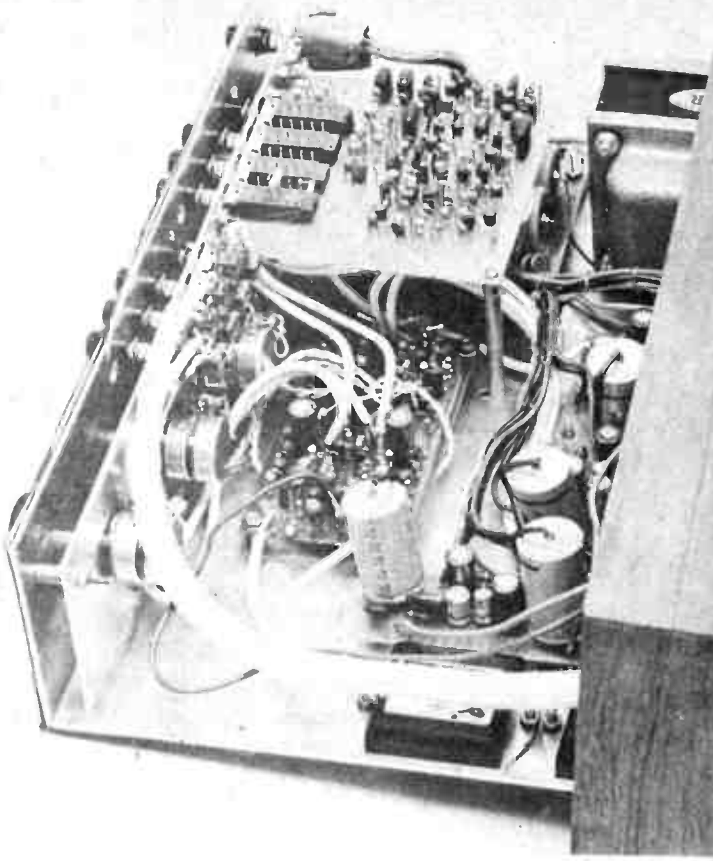


Fig.20. Details of chassis metal work and drilling.

PROJECT 420E

Alternative decoder offers same performance as IC version.

DISCRETE SQ DECODER



This photo shows the ETI 420 amplifier with the discrete decoder fitted. Note the supporting spacer at the rear of the board. ▶

THE 420 4-channel amplifier was first published in the January 1974 issue of ETI and achieved instant popularity. Unfortunately the special SQ decoder IC which we used in the original design subsequently became difficult to obtain.

To overcome this problem we have developed a new decoder board which is a direct replacement of the existing one, but uses transistors to replace the IC. The decoder may of course also be used for building your own four-channel amplifier.

Obviously the printed circuit board is larger and the circuit more complex, but cost is approximately the same and assembly is simple.

The specification of the amplifier, incorporating the discrete SQ decoder, is virtually the same, with the exception that the SQ decoder phase shift is now 90° ±10° from 30 Hz to 20 kHz, rather than from 100 Hz to 10 kHz as with the IC.

CONSTRUCTION

The construction is straightforward. The usual precautions should be taken with all polarized components, transistors and capacitors etc, with regard to polarity. Assembly of the board should be performed in accordance with Fig. 3. Note that the switch should be pressed fully home on the board and the contacts soldered to the tracks where applicable.

INSTALLATION

The board is mounted in exactly the same manner as the IC version, with the exception that a long spacer (56 mm) should be used to support the

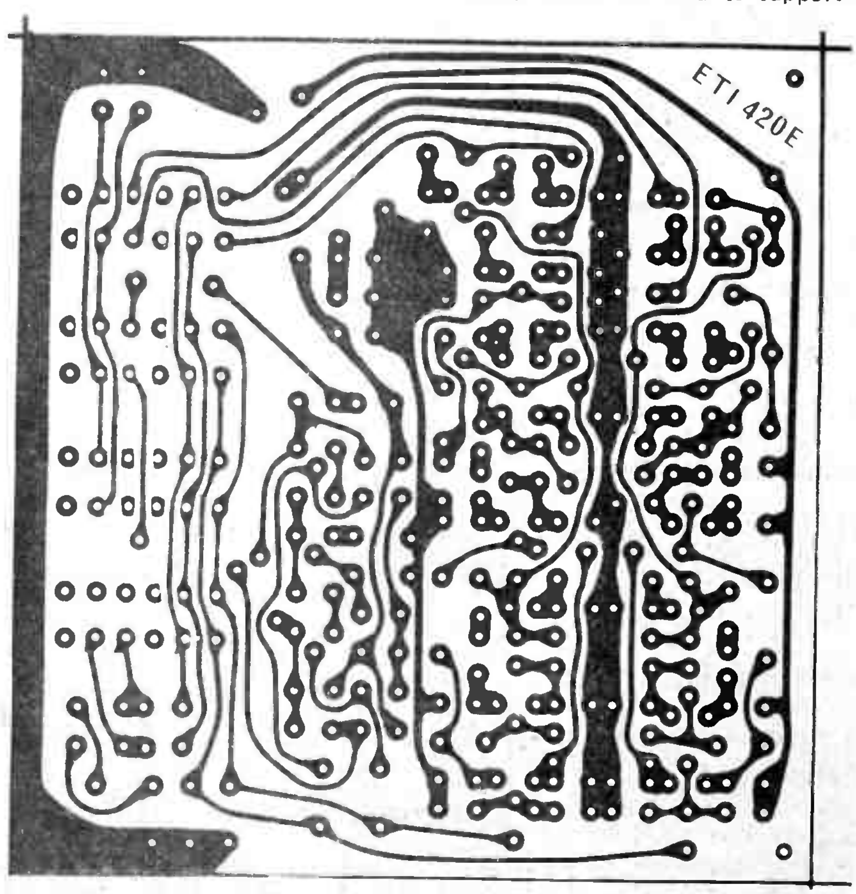


Fig. 1. Printed circuit board for the decoder (full size). ▶

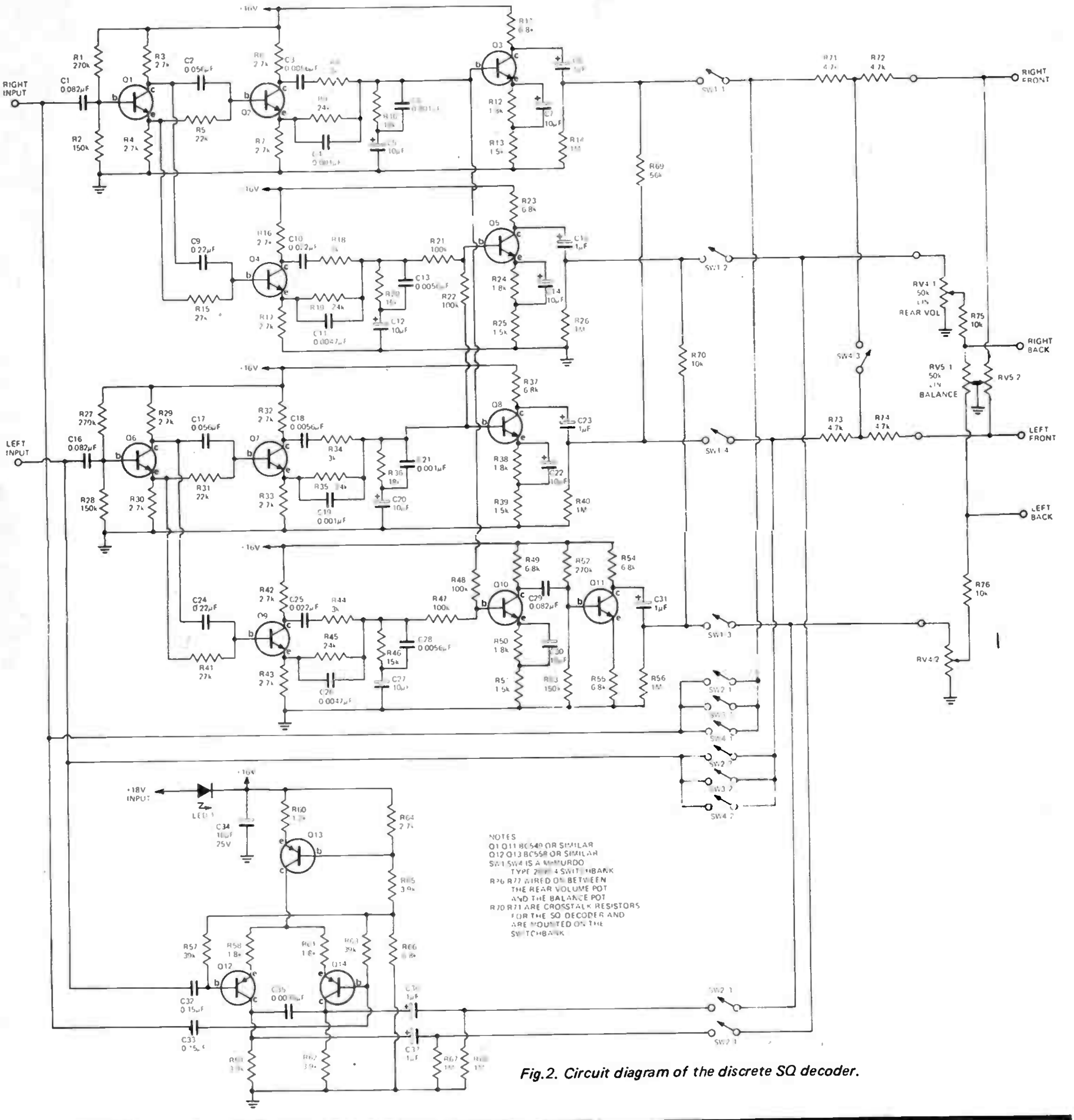


Fig.2. Circuit diagram of the discrete SQ decoder.

PARTS LIST — DECODER BOARD

R1,27,52	Resistor	270 k	½W	or ¼W	5%
R2.28.53	"	150 k	"	"	"
R3,4,6,7,16,17,29,30,32	"	2.7 k	"	"	"
R33,42,43,64	"	2.7 k	"	"	"
R5,31	"	22 k	"	"	"
R8,18,34,44	"	3 k	"	"	"
R9,19,35,45	"	24 k	"	"	"
R10,36,	"	18 k	"	"	"
R11,23,37,49,54,55,66	"	6.8 k	"	"	"
R12,24,38,50,58,61	"	1.8 k	"	"	"
R13,25,39,51	"	1.5 k	"	"	"
R14,26,40,56,67,68	"	1 M	"	"	"
R15,41	"	27 k	"	"	"
R20.46	"	15 k	"	"	"
R21,22,47,48	"	100 k	"	"	"
R57,63	"	39 k	"	"	"
R59,62,65	"	3.9 k	"	"	"
R60	"	1.2 k	"	"	"
R69	"	56 k	"	"	"
R70,75,76	"	10 k	"	"	"

R71,72,73,74	Resistor 4.7k ½W or ¼W 5%
RV4,5	Potentiometer 50k dual lin rotary
C1,16,29	Capacitor 0.082μF Polyester
C2,17	" 0.056μF "
C3,13,18,28	" 0.0056μF "
C4,6,19,21	" 0.001μF "
C5,7,12,14,	Capacitor 10μF 10V electrolytic PC mounting
C20,22,27,30	" 10μF 10V electrolytic PC mounting
C8,15,23,31,36,37	Capacitor 1μF 16V tag tantalum
C9,24	" 0.22μF Polyester
C10,25	" 0.022μF "
C11,26,	" 0.0047μF "
C32,33	" 0.15μF "
C34	Capacitor 10μF 25V electrolytic PC mounting
C35	" 0.0039μF Polyester
Q1-Q11	Transistors BC109, BC549 or similar
Q12-14	" BC178, BC558 "
LED1	light emitting diode McMurdo type 3240-01-02 or similar
PC board	ETI 420E
SW1,2,3,4	switchbank McMurdo type 2899-4

HOW IT WORKS

To decode SQ matrixed material, the left and right input signals must each be split into two signals with a constant 90° phase shift between each pair, regardless of frequency.

Although a simple integrator will provide the 90° phase shift it does not provide a constant amplitude (−6 dB/octave). A more complex network is thus required.

Basically, each input is split into two components at 180°. Each is then phase shifted by two slightly different networks in series. When referred to the input, the output phase continually changes with frequency (maximum shift 540° at 20 kHz). However when one output is referred to the other there is a constant phase shift of 90° ±10° between 30 Hz and 20 kHz.

In detail, the right channel input is buffered by Q1 which provides two, 180° out of phase, signals. A phase shift varying between 0° and 180° (90° at 130 Hz) is provided by C2 and R5 and this signal is then buffered by Q2. Additional phase shifting is provided by C3, C4, C6, R8, R9 and R10 to provide a total of 540° shift at 20 kHz. This network has a loss, constant with frequency, of 10 dB.

The values of the phase shift components are critical – stated values *must* be used. The resulting output of the above network is referred to as Rϕ°.

The output of Q1 also feeds a second network (Q4 etc), having different phase shift components, the output of which lags the first network by 90°. This output is known as the R (ϕ–90°).

The left channel input receives the same treatment and produces corresponding Lϕ° and L(ϕ–90°) outputs.

The Rϕ° and Lϕ° outputs are buffered by Q3 and Q8 respectively and amplified to compensate for the 10 dB loss in the phase shifting networks.

The right-back output is now produced by mixing equal proportions of R (ϕ–90°) and Lϕ°. These are mixed by R21 and R22 and then buffered and amplified by Q5.

Similarly the left-back output consists of equal proportions of L(ϕ-90°) and Rϕ°. These are mixed by R47 and R48 and then buffered and amplified by Q10. This output, as required by the SQ coding, is then phase shifted 180° by Q11.

Some crosstalk between left and right channels is recommended and R69 provides 10% mixing of left and right front channels, whilst R70 provides 40% mixing of the rear channels. These two resistors are mounted on the switch bank.

Except for component numbering, the ambience mode is as described last month.

rear right hand corner of the board. We used two 25 mm spacers and one 6 mm to do the job, but a piece of rod drilled and tapped both ends would be better if you have one, or can make one.

Wiring to the board differs slightly from the IC version in the following respects:—

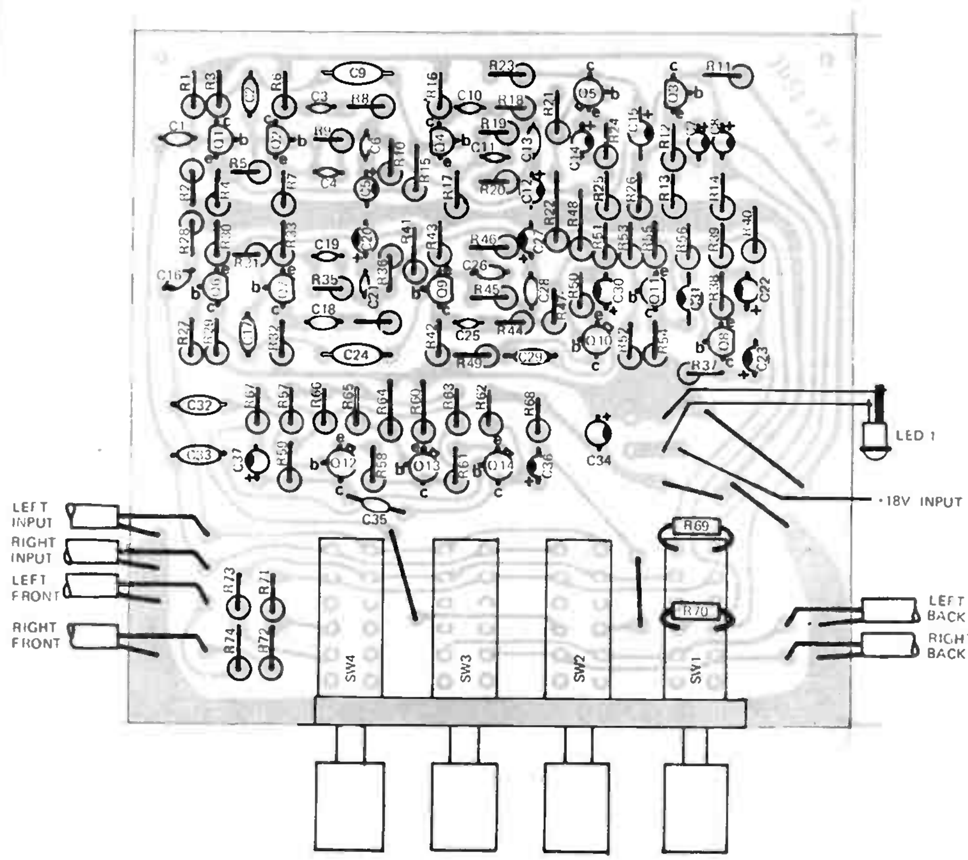
1. The positions of left and right input channel wiring is reversed to that previously used.

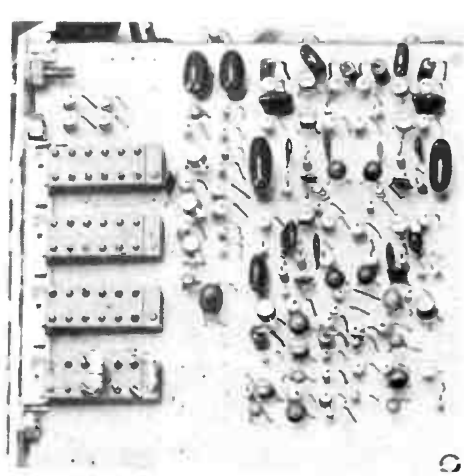
2. The position of left and right back output channels is reversed.

3. The position of left and right front outputs are reversed.

Check these points on the overlay to make sure all wiring is the correct way round.

4. Mono switching is done, on-the-board, hence wires 13 and 14 from the preamplifier are not required. ●

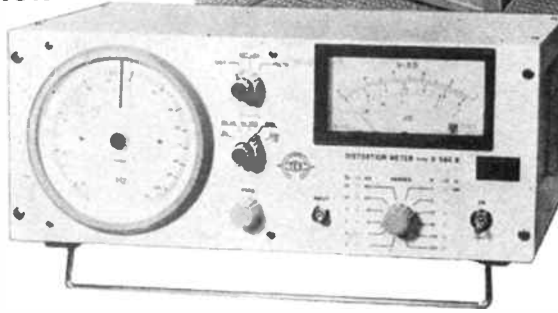




This photo shows the board with all components fitted.

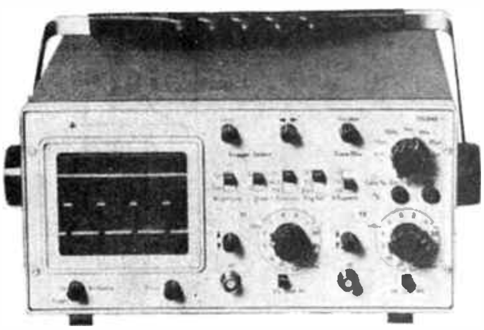
No need to get in a flap over fixing AUDIO/HI-FI gear

Not with this selection of instruments from Jacoby Mitchell to iron out your problems.



TES Model D 566 B AUDIO DISTORTION METER

Fully transistorised, easy to operate distortion meter. **Distortion Meter:** Freq. range 10 Hz to 1 MHz • Distortion factor 0.03% to 100% • Input impedance 100 KOhm; 40 pF. approx. **Millivoltmeter:** Voltage range 1 mV to 300 V f.s.d. • Level range (rel. to 0.776 V) + 52 dB to − 75 dB Freq. range 10 Hz to 2 MHz • Input impedance 2 MOhm; 50 pF approx.



ADVANCE OS240 OSCILLOSCOPE

A 10 MHz dual trace unit, with 4" flat faced CRT, using the Y1 channel for X deflection and the Y2 channel for Y deflection.

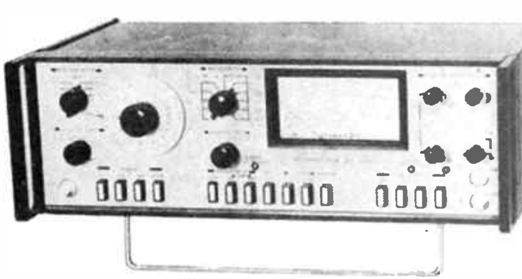
VERTICAL AXIS
Sensitivity: 5 mV/div. to 20 V/div. in 1-2-5 sequence. Input impedance 1M Ω/approx. 28pF. Input coupling DC-GND-AC. **HORIZONTAL AXIS** — Sensitivity: (Internal) ≪ 0.3 div. 40 Hz-2 MHz (External) ≪ 1.5 V 40 Hz-2 MHz. Input Impedance: 100 K Ω ≪ 10pF. Full trigger level control even when using the 'bright line' facility.

ADVANCE DMM2 DIGITAL M/METER

An inexpensive 3½ digit DVM for AC & DC voltage, current and resistance • All push button with the stability of LSI (large scale integrated circuit) • Voltage range 200 mV to 1000 V, current range 200 μA AC & DC with up to 2A with SP-2 current shunt • Resistance ranges 200 Ohm to 2000 KOhm • Accuracy 0.19 DC • Weighs only 3½ lb.

TES Model MU 472 OUTPUT POWER METER

Fully transistorised power meter calibrated in watts and dB. Built-in load resistance • Power measuring ranges 1-10-100W • Freq. range 20 Hz to 50 KHz • Accuracy within 0.5 dB • Load input resistances 4-8-16 ohms • Resistances accuracy better than 5%.

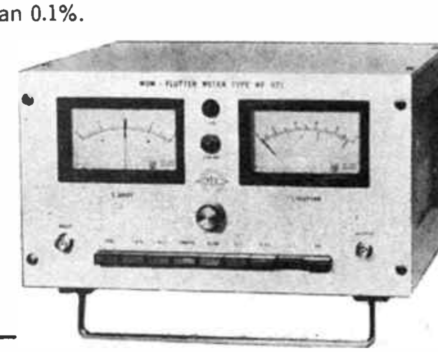


FERROGRAPH RTS-2 AUDIO TEST SET

An "all-in-one" test set to check frequency response, signal noise ratio, distortion, cross talk, wow and flutter, drift, erasure, input sensitivity, output power and gain. The facilities of the RTS-2 can be extended with the RTS Auxiliary Unit. The RTS-2 offers many advantages over conventional methods of obtaining these measurements which would require a number of separate test instruments.

TES WF 971 WOW & FLUTTER METER

For tape recorder servicing • Spec. DIN & CCIR • Input signal 20 mV rms to 20 V rms approx. • Frequencies (switchable) 3150 Hz and 3000 Hz • Ranges (flutter) ± 0.1% ± 0.3% ± 1% f.s.d. • Drift Indication ± 2% max. • Input impedance 10 KOhm max. • Built-in oscillator 3000 Hz or 3150 Hz switchable • Stability better than 0.1%.



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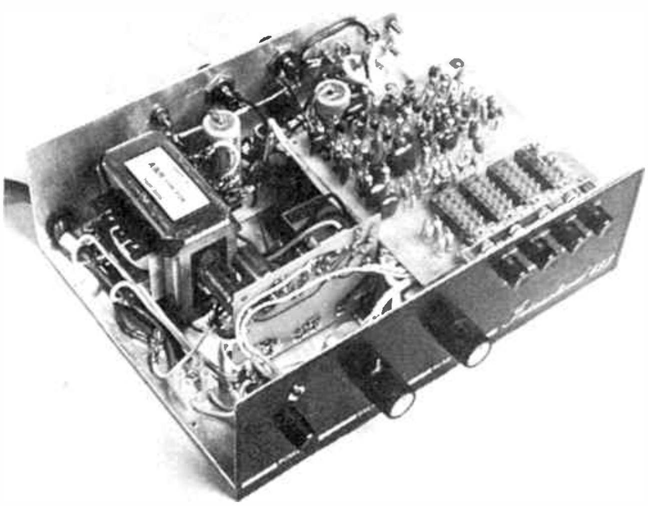
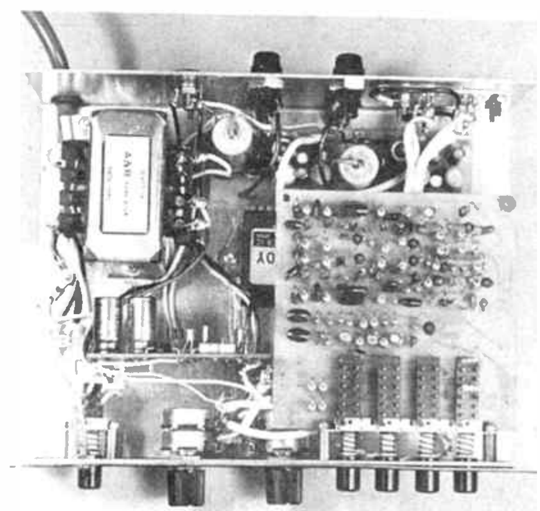
JM/149-73

PROJECT
423

PLUS TWO add~on decoder amplifier

Adapt your stereo hi-fi to full four-channel SQ operation





SPECIFICATION

Output Power (at 1% distortion)			
Both channels driven	12.5 watts per channel		
Distortion	100 Hz	1 kHz	10 kHz
At 0.1 watt output	0.15%	0.13%	0.25%
At 1 watt output	0.14%	0.11%	0.18%
At 10 watt output	0.14%	0.1%	0.15%
Maximum Input Voltage	2 V		
Gain			
Input to front output	Unity		
Damping Factor			
100 Hz	5		
1 kHz	30		
10 kHz	30		
SQ Decoder Phase Shift			
30 Hz to 20 kHz	$90^{o} \pm 10^{o}$		

MANY OF US have watched the evolution of four-channel systems with interest, but, being already possessors of a stereo system, have rejected four-channel as being too expensive to implement.

But here is a cheap and relatively simple way to convert your stereo into a full SQ, four-channel system. Apart from this unit the only extra equipment needed is two rear speakers, which need not be as high in quality as your existing front speakers.

The add-on unit is connected to your existing stereo amplifier via the pre-amplifier 'out' and main amplifier 'in' sockets. This facility — together with a 'connect/disconnect' switch is provided on most good quality amplifiers. If it is not, your existing amplifier must be modified by disconnecting the internal wiring and bringing all four points out to the rear panel via shielded cable.

Although this is a quick simple modification, it should only be attempted by those who have a good understanding of amplifier operation — if you don't know how to do it — do obtain advice.

The add-on unit's mode of operation may be readily understood by referring to Fig. 1. It will be seen that the SQ matrixed signals are amplified by the existing preamplifier tone control stages, and then passed to the add-on unit. Here they are decoded into left front, right front, left back and right back channels. The left and right back channels are amplified and passed direct to the rear speakers. The left and right front signals are passed back to the existing main amplifiers and speakers, and there you have it — inexpensive four-channel sound.

The discrete decoder board described in the previous article is used in the add-on unit. Alternatively the decoder board described in Project 420 (four-channel amplifier) may be used, although the MC1312P integrated circuit is extremely difficult (if not impossible) to obtain at present.

The power amplifier module uses the Sanken SI-1010Y modules and is

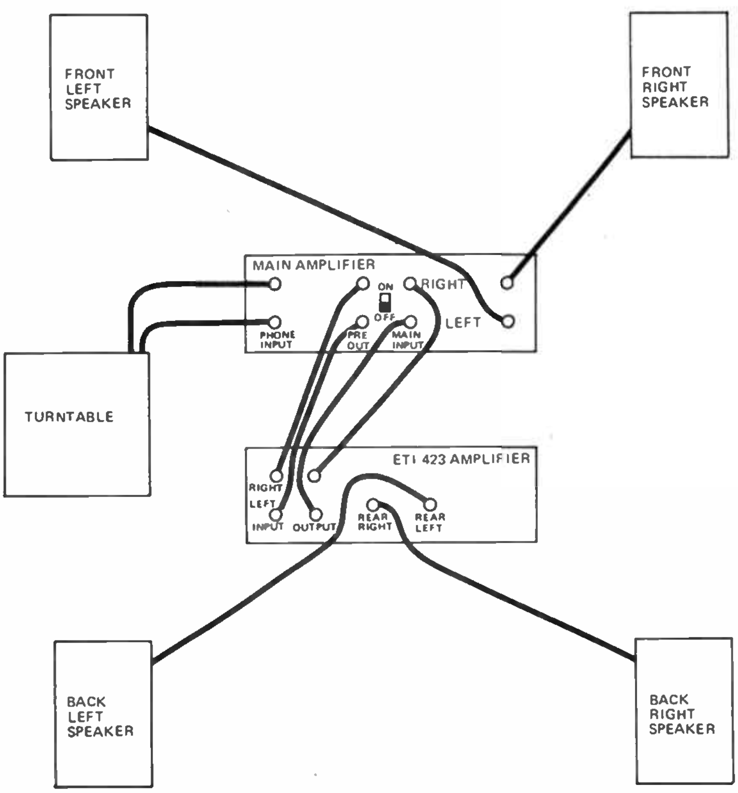


Fig. 1. This schematic drawing shows how the add-on unit is connected into the existing stereo system.

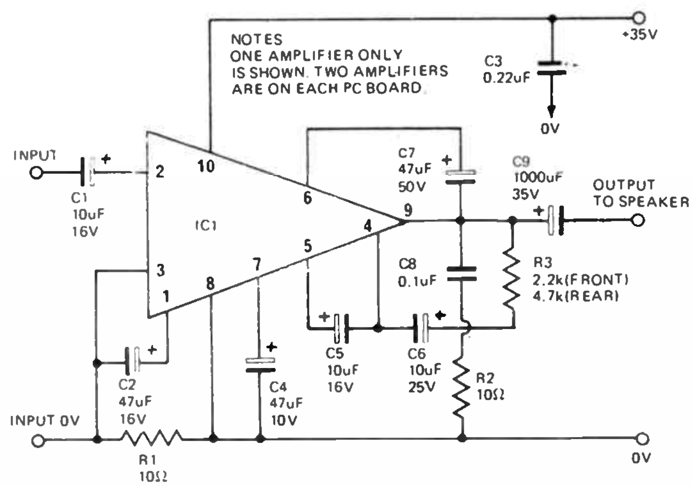


Fig. 2. Circuit diagram of one power amplifier module (two per assembly).

PLUS TWO add-on decoder amplifier

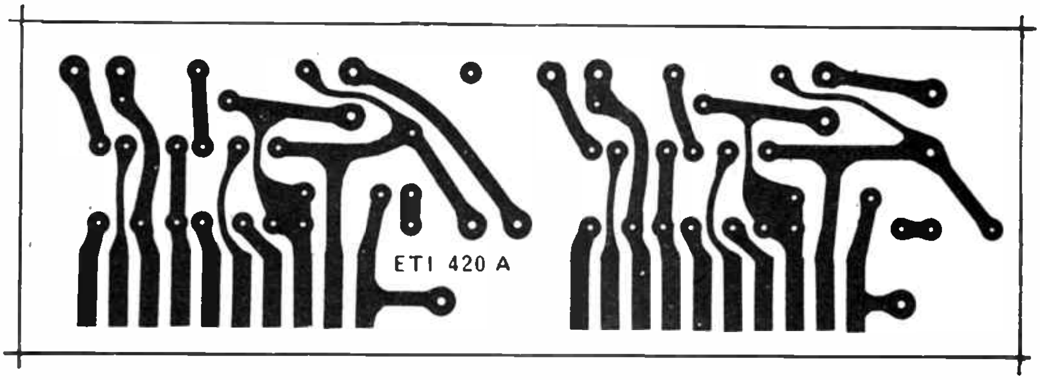


Fig. 3. Printed circuit board for the twin power amplifier assembly.

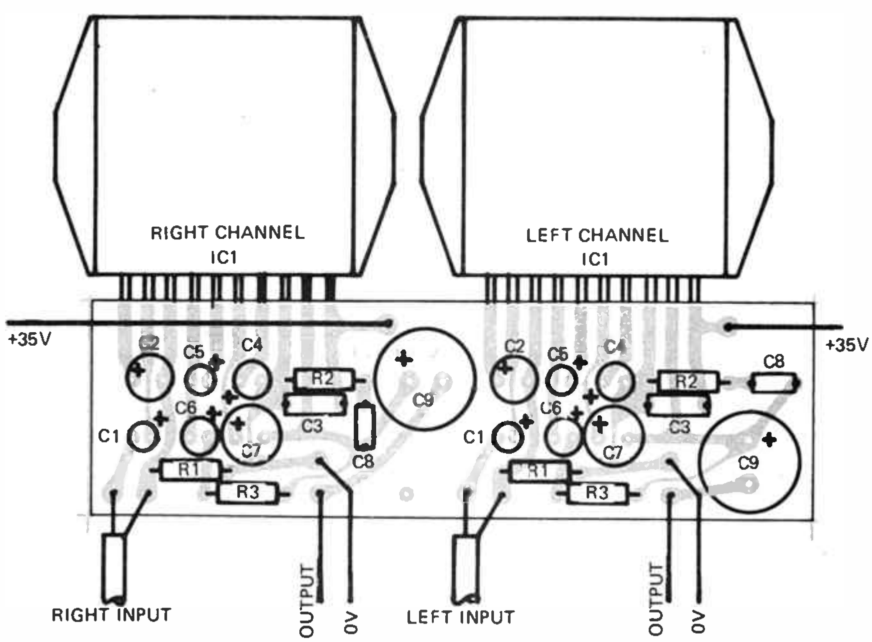


Fig. 4. Component overlay for the twin power amplifier assembly.

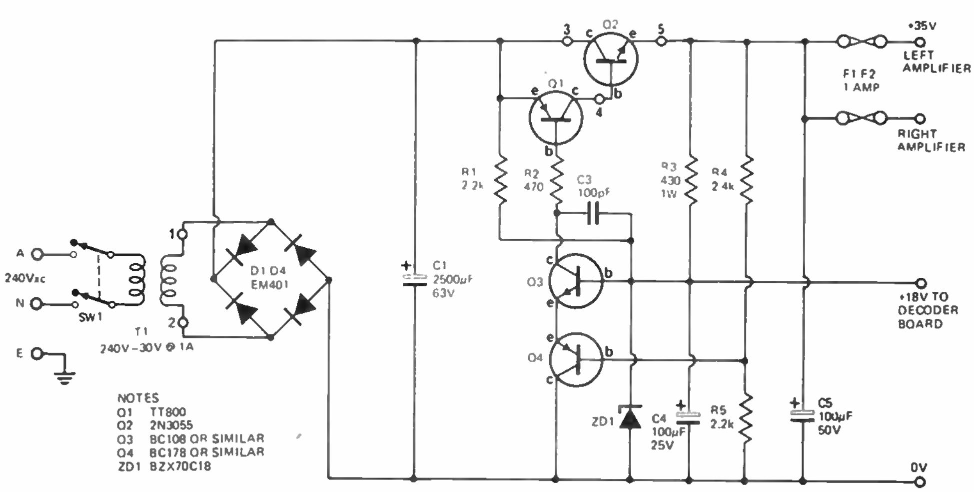


Fig. 5. Circuit diagram of power supply.

PARTS LIST

POWER SUPPLY

R1	Resistor	2.2k	½W	5%
R2	"	470	"	"
R3	"	430	1W	"
R4	"	2.4k	½W	"
R5	"	2.2k	"	"

C1 Capacitor 2500μF 63V electrolytic
C3 " 100pF ceramic
C4 " 100μF 25V electrolytic PC mounting
C5 " 100μF 50V electrolytic PC mounting

Q1 Transistor TT800 or similar
Q2 " 2N3055 or similar
Q3 " BC108 or similar
Q4 " BC178 or similar

D1-D4 Diode EM401 or similar
ZD1 Zener Diode BZX70C18

T1 Transformer 240V/30V @ 1A A & R 9129 or equivalent.
SW1 Switch McMurdo Type 2904-1
PC Board ETI423
F1-F4 1 Amp Fuse and panel mounting holders
Cover for 2N3055 transistor
Insulation kit for 2N3055

CHASSIS AND MISCELLANEOUS

Complete decoder board as published in Project 420E page 16.
1 spacer ¼" long (plain)
4 spacers ½" long (plain)
2 knobs John Carr type TK196 or similar
2 2 way RCA sockets
2 two pin DIN sockets
Mains cord, grommet and clamp
2 way terminal block
Metal chassis to Fig. 13
2 small right angle brackets to hold power supply board
Wood box to Fig. 12
23/0076 wire
Shielded cable
Front panel to Fig. 14

AMPLIFIER

R1, 2	Resistor	10Ω ½W 5%
R3	"	4.7k ½W 5%
C1,5	Capacitor	10μF 16V electrolytic*
C2	"	47μF 16V electrolytic*
C3	"	0.22μF polyester
C4	"	47μF 10V electrolytic*
C6	"	10μF 25V electrolytic*
C7	"	47μF 50V electrolytic*
C8	"	0.1μF polyester
C9	"	1000μF 35V electrolytic*

* all electrolytics should be PC mounting type

IC1 Amplifier Module Sanken SI-1010Y

PC Board ETI 420A

Note: Two amplifiers required for each unit.

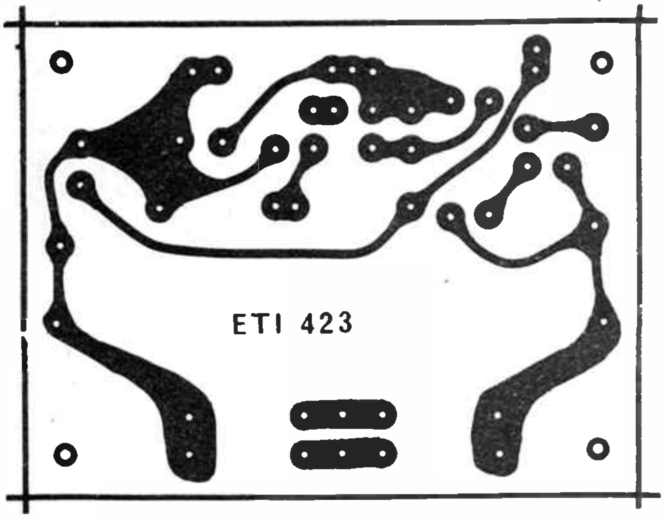


Fig. 6. Printed circuit board for the power supply.

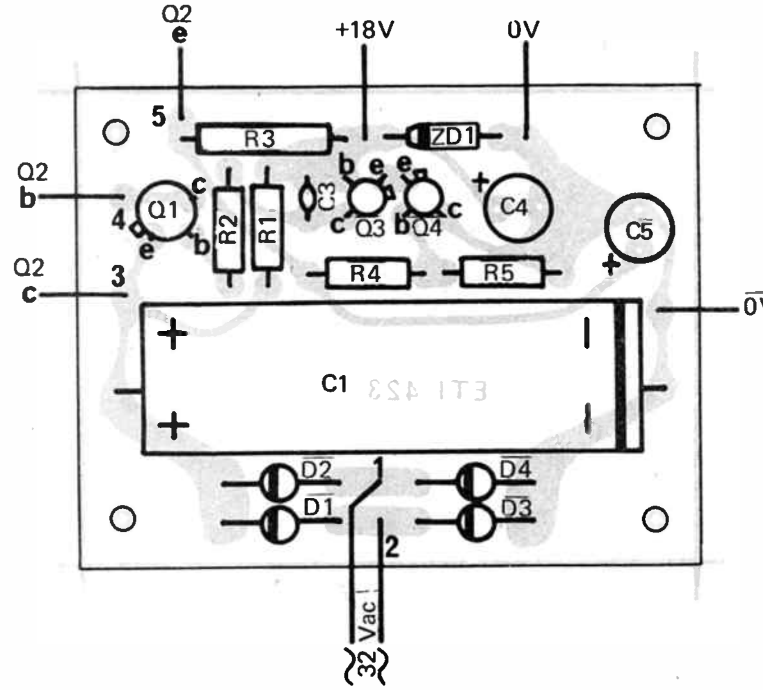


Fig. 7. Component overlay for power supply.

identical to that used in the International 420 four-channel amplifier.

CONSTRUCTION

Components should be assembled onto the printed circuit boards with reference to the appropriate component overlays. Take particular care with the orientation of polarized components such as transistors, capacitors and diodes etc.

The interconnection wiring diagrams, Fig. 8 and Fig. 9, give details of the power and signal wiring respectively. The mounting positions of the printed circuits boards, transformer and potentiometers etc may readily be seen from the metalwork drawing and from the internal photograph of the unit.

The rear-channel amplifier may be omitted if a decoder unit *alone* is required. For this, the coaxial cables, that otherwise go to the power amplifier inputs, should now be connected to two additional RCA sockets on the amplifier rear panel.

Power requirements for the decoder board are negligible (0.36 watt compared with 30 watts for the complete unit). Thus a much smaller transformer and simpler power supply circuit may be used. A transformer having a secondary of 12.6 volts at 150 mA, a bridge rectifier, D1-D4, and a single smoothing capacitor, C1, is all that is required. The complete regulator section of the power supply may be omitted.

Although the existing printed circuit board could be used, by simply leaving off the unwanted components, it would be simpler and cheaper to use a tag strip to mount the components for this simpler supply.

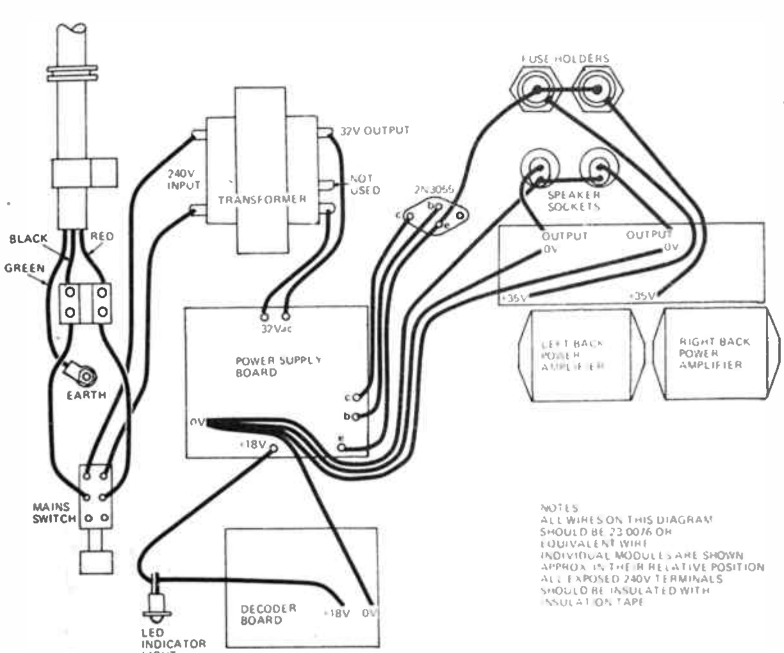


Fig. 8. Interconnections — power wiring.

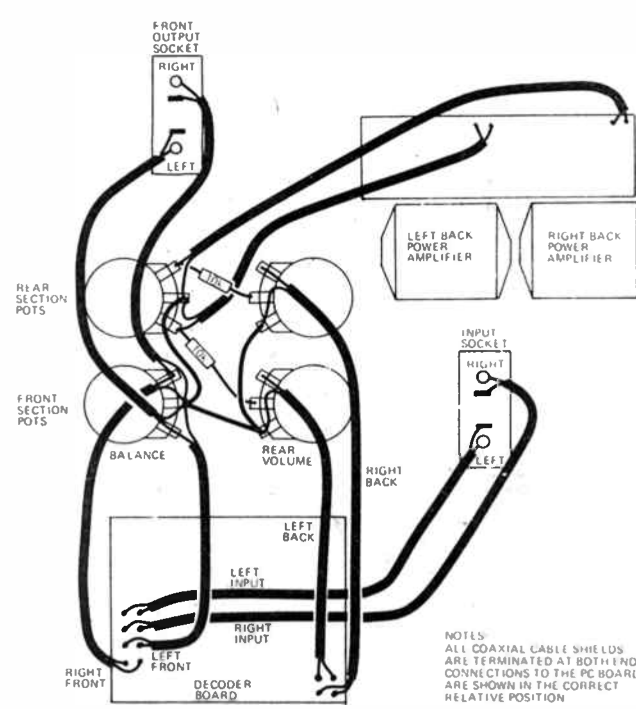


Fig. 9. Interconnections — signal wiring.

PLUS TWO add-on decoder amplifier

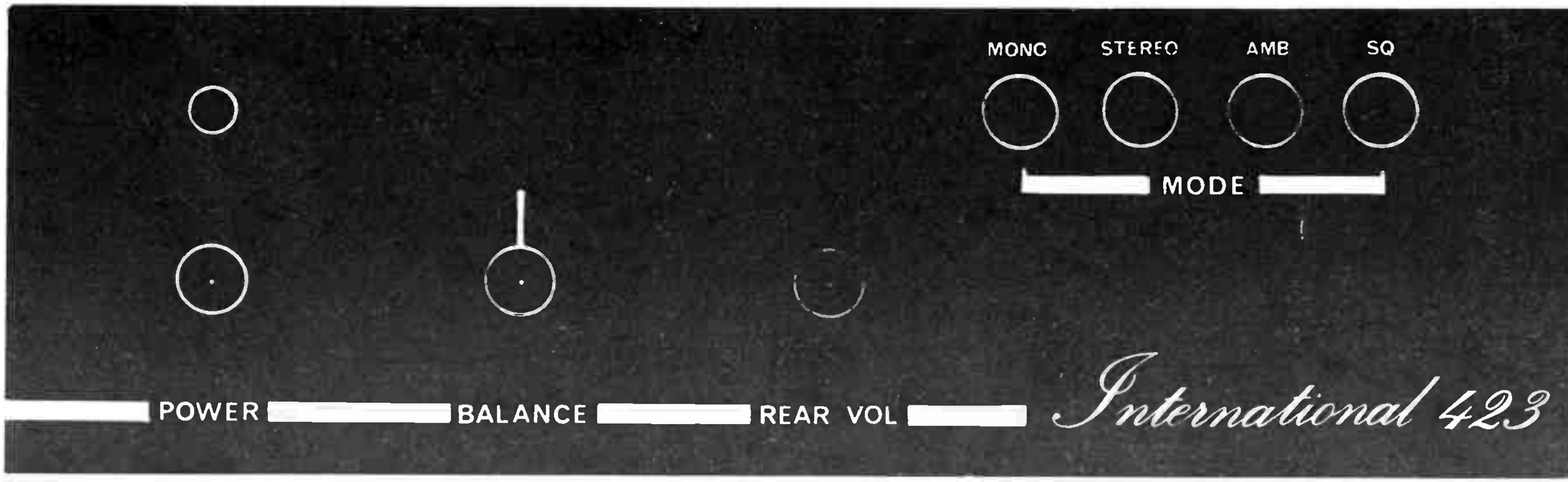


Fig. 10. Front panel artwork.

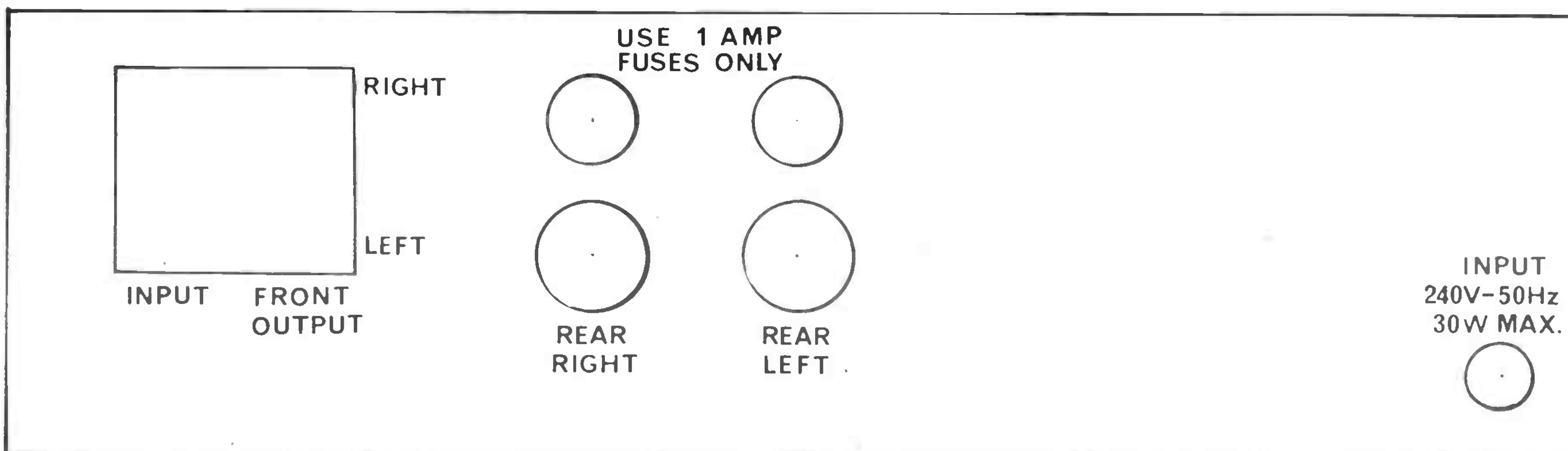


Fig. 11. Rear panel artwork.

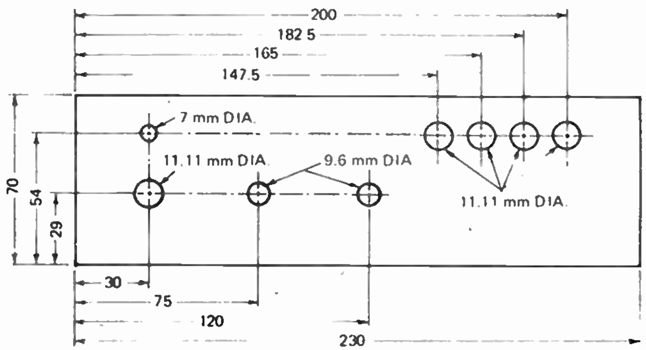


Fig. 14. Front panel escutcheon dimensions and drilling details.

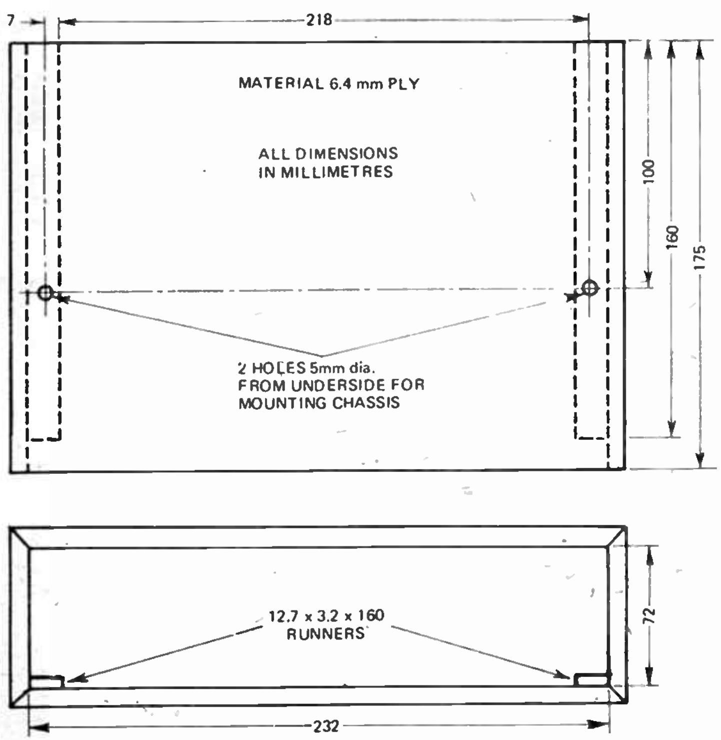


Fig. 12. Details of wooden cabinet.

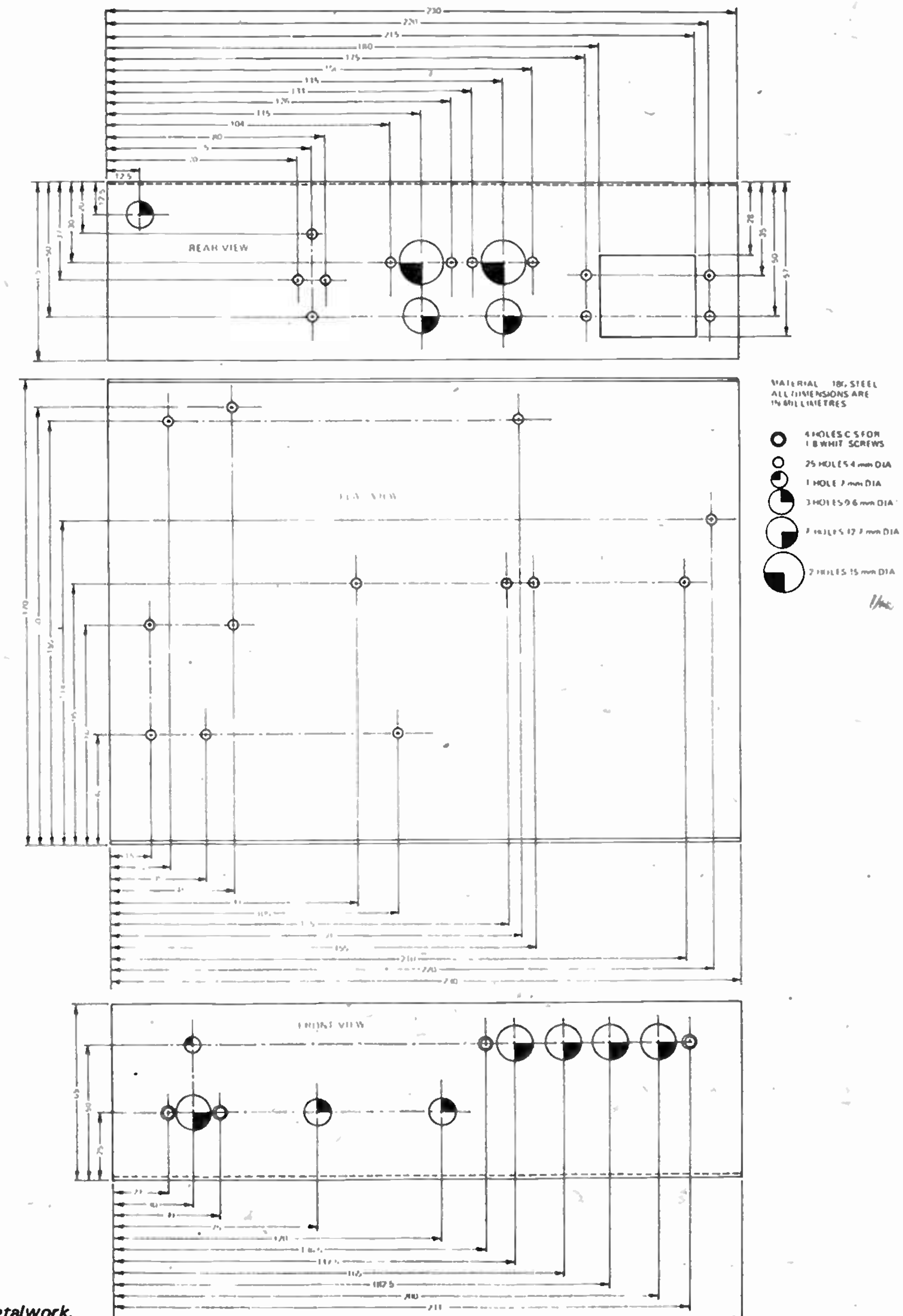


Fig. 13. Details of chassis metalwork.

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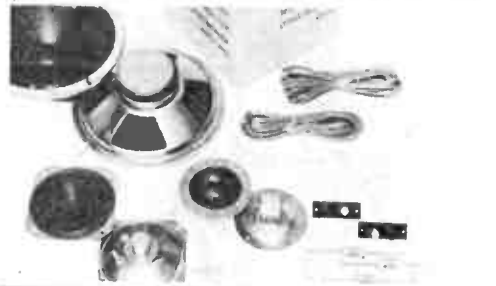
35W, 3-way 3-speaker system in kit form includes all speakers, crossovers, terminals, wire, screws, coral emblem for front of box.

The features as assembled in a designed cabinet are as follows:

Features (8SA-1)	
Type of cabinet	Closed type
Type of speaker	3-way 3-speaker
Input impedance	8 Ohms
Crossover frequency	4,000 : 9,000 Hz
Frequency response	70 ~ 20,000 Hz
Sensitivity	93 dB
Capacity	35 W

$30 PAIR



Price **$50** PAIR

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60W, 3-way, 4-speaker system in kit form includes all speakers, crossovers, terminals, wire, screws, coral emblem for front of box.

The features as assembled in a designed cabinet are as follows:

Features (12SA-1)	
Type of cabinet	Closed type
Type of speaker	3-way 4-speaker
Input impedance	8 Ohms
Crossover frequency	1,000 : 10,000 Hz
Frequency response	30 ~ 20,000 Hz
Sensitivity	95 dB
Capacity	60 W

P & P ALL CORAL KITS $2.00

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50W, 3-way, 3-speaker system in kit form includes all speakers, crossovers, terminals, wire, screws, coral emblem for front of box.

The features as assembled in a designed cabinet are as follows:

Features (10SA-1)	
Type of cabinet	Closed type
Type of speaker	3-way 3-speaker
Input impedance	8 Ohms
Crossover frequency	2,000 : 6,000 Hz
Frequency response	40 ~ 20,000 Hz
Sensitivity	93 dB
Capacity	50 W

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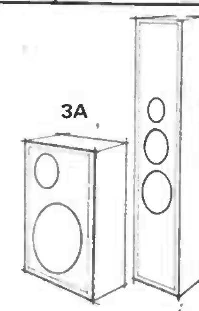
Power Handling Capacity 20W rms.

Speaker system 3A — for 8" woofer & 1" tweeter $13.50
Power Handling Capacity 20W rms.

Speaker system 4A — for 8" woofer + 5" mid range + 1" tweeter $16.50
Power Handling Capacity 20W rms.

Speaker system 5A — for 8" woofer + 5" mid range + 1" tweeter $24.00
Power Handling Capacity 10W rms. Column Type.

Speaker system 7A — for 12" woofer + 5" mid range + 1" tweeter $29.00
Power Handling Capacity 40W rms.



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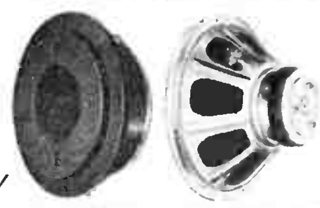

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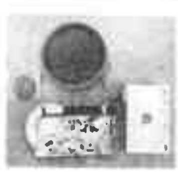

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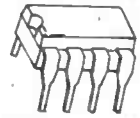

LM741 operational (OP. AMPS) amplifiers in min dip case. 80c ea or 2 for $1.50. 10 for $6.95. Post Free.

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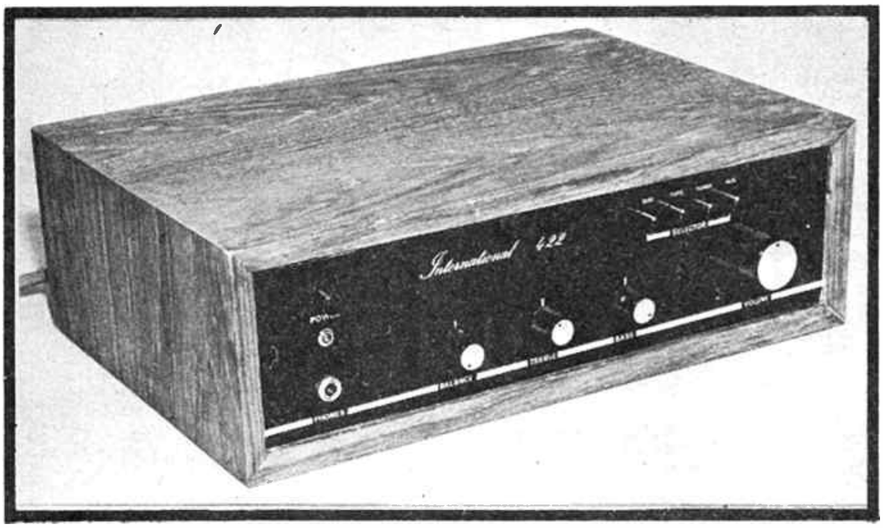
INTERNATIONAL 422 STEREO AMPLIFIER



eti PROJECT 422

This exciting new amplifier produces a full 50 watts (rms) per channel!

SINCE publication of our 100 watt guitar amplifier, in December 1972, several thousand have been built and a surprisingly large quantity of these have been for home stereo use. People have used two of these together with a separate preamplifier for stereo, and would you believe it, we know of a few people using four in a quadraphonic system!

This is not as way out as it sounds for many present-day speakers sacrifice efficiency to gain quality. Many high quality speakers need at least 50 watts ('rms') to drive them satisfactorily.

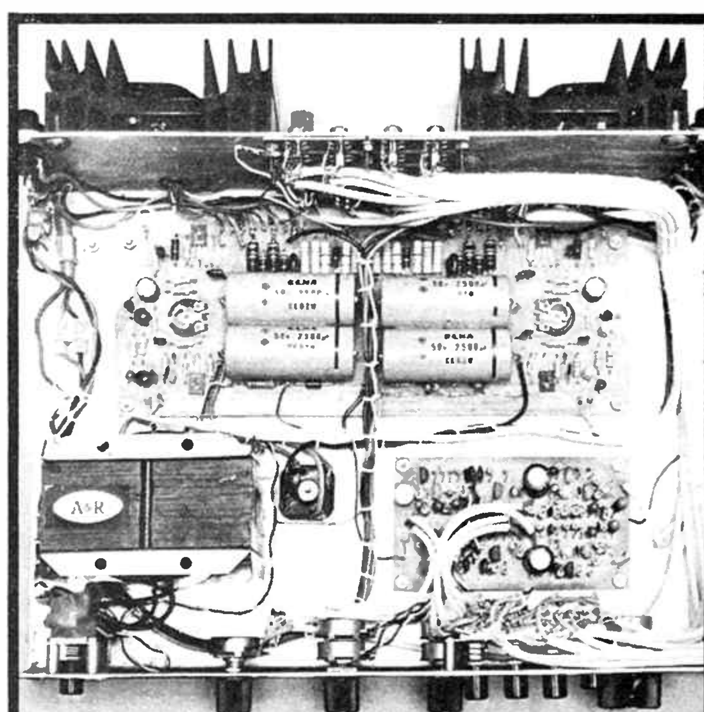
There is an obvious *need* for a high powered amplifier, and in response to many pleas, we have designed an inexpensive amplifier that will deliver a genuine 50 watts rms per channel, both channels driven, into 8 ohms.

Since most modern speakers are 8 ohms impedance, we have not designed the amplifier for 4 ohm operation. Such an amplifier would require a much larger transformer and would be considerably more expensive, so we have decided to

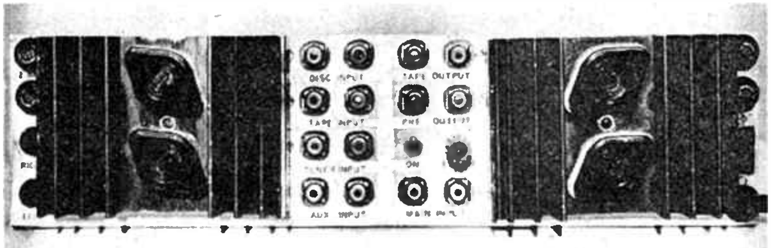
MEASURED PERFORMANCE OF PROTOTYPE UNIT

POWER OUTPUT			
Both channels driven $8\,\Omega + 8\,\Omega$ loads	50 watts rms		
FREQUENCY RESPONSE			
20 Hz-20 kHz	±0.5 dB		
CHANNEL SEPARATION			
At rated output and 1 kHz	45 dB		
HUM AND NOISE			
With respect to rated output			
Tape, Tuner and Aux. inputs	−78 dB		
Disc input (re 10 mV)	−67 dB		
INPUT SENSITIVITIES (for rated output)			
Tape, Tuner and Aux. inputs	210 mV into 47 k		
Disc at 1 kHz	2.1 mV into 47 k		
Main amplifier	500 mV into 10 k		
TOTAL HARMONIC DISTORTION			
	100 Hz	1 kHz	6.3kHz
1 watt output	0.14%	0.11%	0.12%
5 watt output	0.17%	0.13%	0.15%
10 watt output	0.16%	0.11%	0.13%
50 watt output	0.27%	0.38%	0.60%
TONE CONTROLS			
Base	±13 dB at 50 Hz		
Treble	±13 dB at 10 kHz		
DAMPING FACTOR	>70		

Internal view of the completed amplifier.

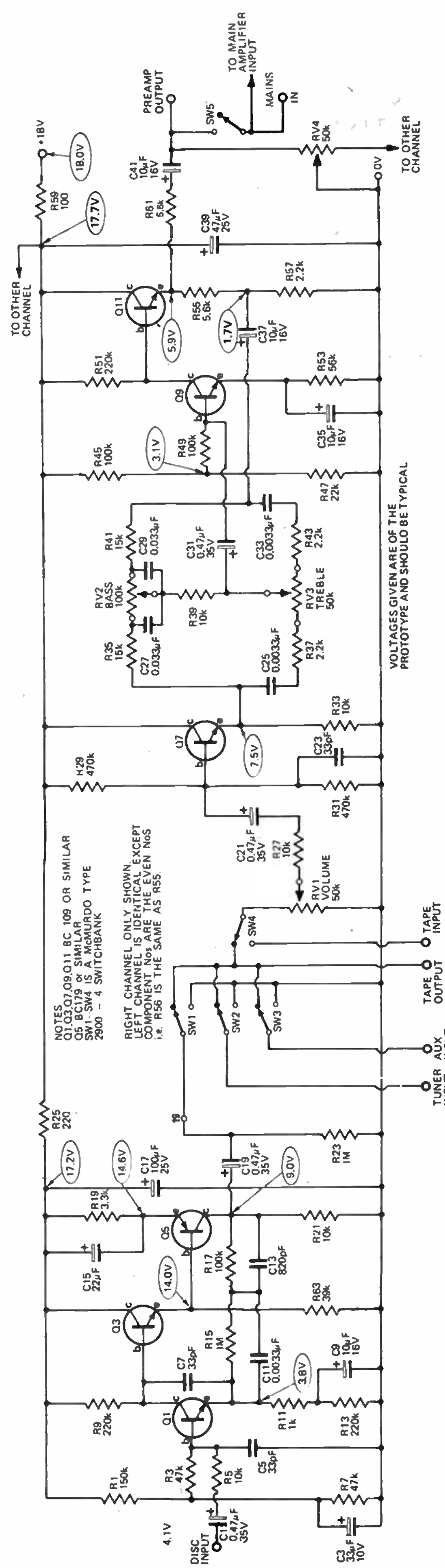


INTERNATIONAL 422 STEREO AMPLIFIER



View of the rear panel of the amplifier.

Fig.1. Circuit diagram of the preamplifier (one channel only shown).



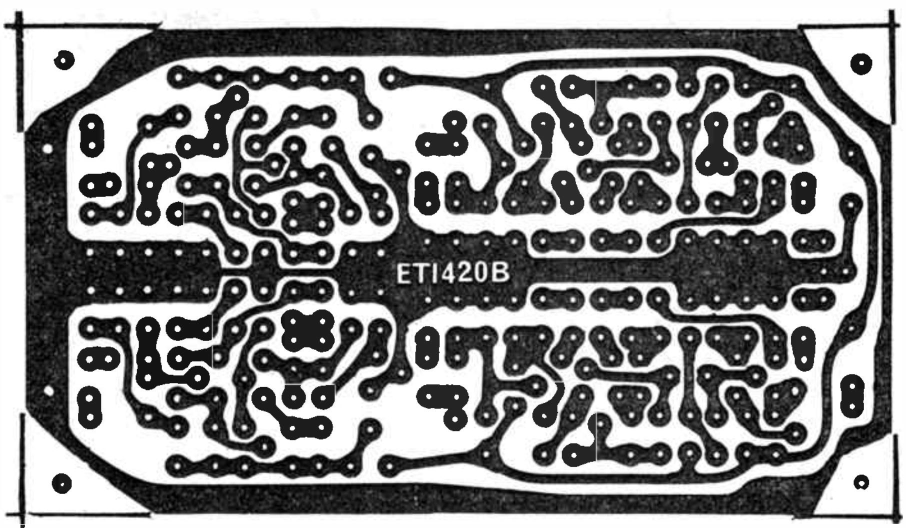
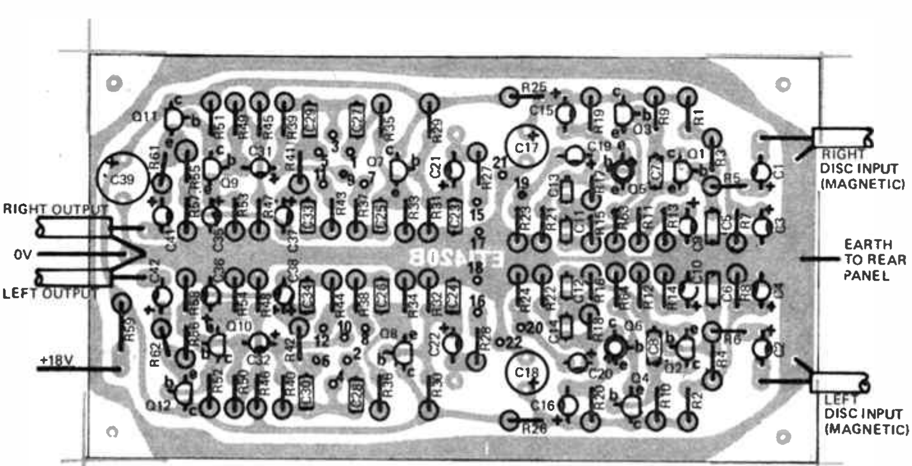


Fig.2. Printed circuit board pattern for the preamplifier.



NOTES–
PADS 1–12 ARE FOR THE TONE CONTROLS
PADS 19 & 20 ARE THE MAG. AMP. OUTPUT
PADS 21 & 22 ARE THE SHIELD OF 19 & 20
PADS 15 & 16 ARE ARE THE PREAMP. INPUTS
PADS 17 & 18 ARE THE SHIELD OF 15 & 16

Fig.3. Component overlay for the preamplifier.

satisfy the many, rather than the few.

As well as being designed to provide high power at low cost, the amplifier has been kept simple from the constructional point of view. It uses the preamplifier from our ETI420 four-channel amplifier with only a few minor changes. Tape-in, tape-out and main amp in/preamp-out facilities have been provided. Tape monitoring may be achieved by pressing, simultaneously, the tape button as well as that for the desired input.

A new main amplifier board is used. This carries the components for both main amplifiers (apart from those mounted on the heatsinks) and the power supply components. All components are mounted on a simple pan-type chassis which slides into the same wooden case as was used for the four channel amplifier.

CONSTRUCTION

The construction has been kept as simple as possible so that a person with only average electronics experience should have no problems in building the amplifier.

The printed circuit boards carry the majority of the components apart from hardware items such as switches, potentiometers and the transformer etc.

The boards should be assembled with reference to their component overlays making sure that all components are in the correct position and, that they are orientated correctly.

It is preferable that pins be used to connect all external wiring to the main amplifier board, as this will considerably facilitate wiring up the board at a later stage.

The components should be assembled onto the heatsinks with the aid of Fig. 7 and Fig. 8. Note that a mica washer should be used on both sides of the heatsink for each of the 2N3055's so that the BD139-140 transistors may also be insulated from the heatsink.

The PN3643 transistor, Q13, should be glued into the heatsink in the position shown in Fig. 7 and the wires which connect to the flying leads of the, transistors should also be secured to the heatsink with glue. The new quick-dry epoxies are ideal for this application.

The chassis hardware should be assembled in the following order:–

1. Fit RCA, DIN speaker, and power sockets. As space is short, trim ends and use only centre screw.

2, Fit the rear panel escutcheon using the fuse holders, the main-preamplifier connect toggle switch, the earth terminal and the 3-core flex and grommet, to secure it to the rear panel.

3. The heatsinks can now be fitted by passing the wiring through the rear panel holes (which should be fitted with grommets) and securing them using 12 mm long ¼" screws. The screws will screw directly into the

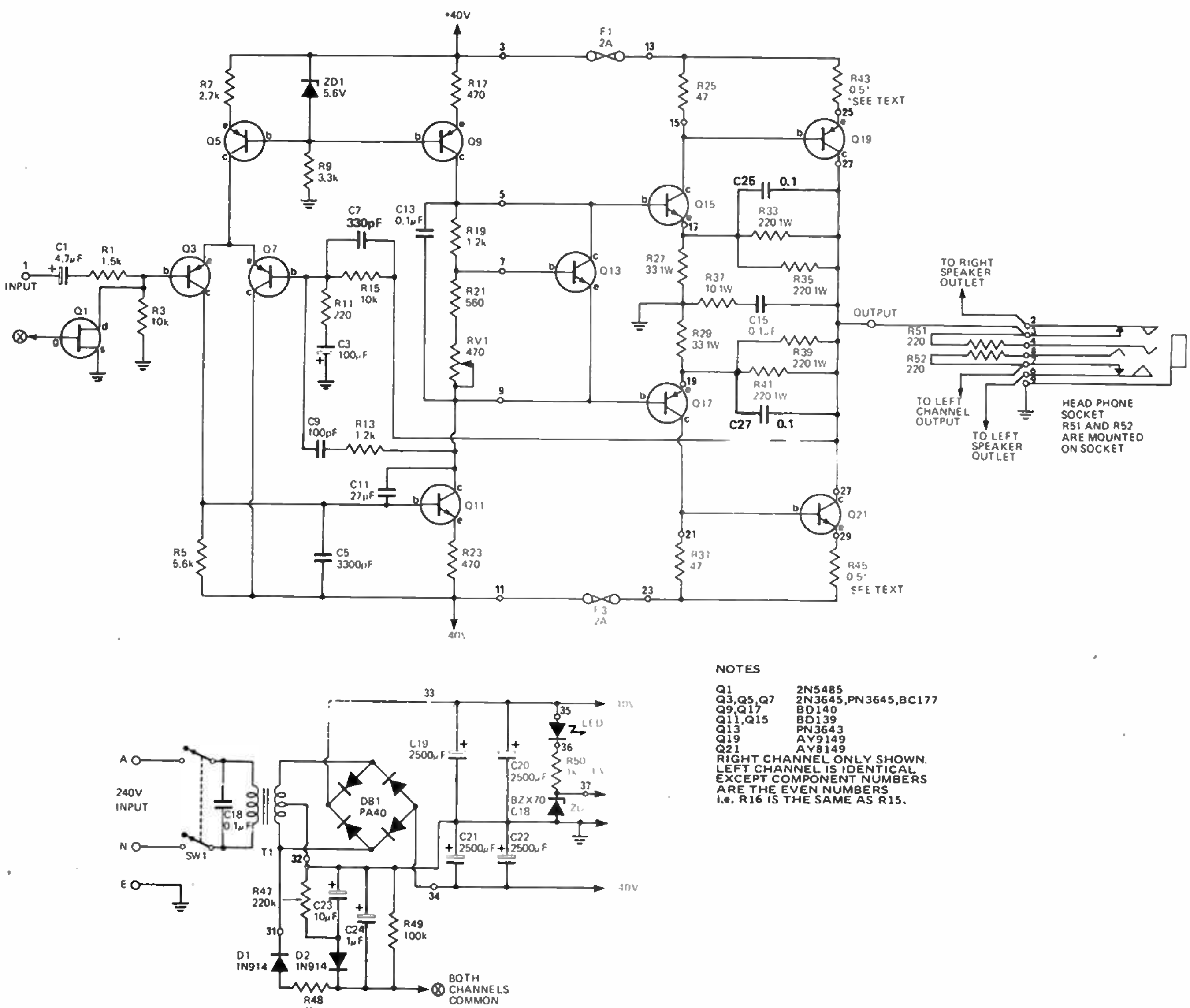


Fig. 4. Circuit diagram of the main amplifier (one channel only) and the power supply.

INTERNATIONAL 422 STEREO AMPLIFIER

heatsink-fin spacing which is designed for such a mounting technique.

4. Fit a cable clamp to the 3-core power flex and terminate the cable into a two way terminal block and a separate earth screw.

5. The power switch and the selector switch should now be mounted using 12.7 mm spacers.

6. The front panel can now be mounted. It is secured by the potentiometers, LED and the phone socket.

7. Mount the preamplifier board after connecting coax. or hookup wire where applicable. The board should be supported on 6 mm spacers.

8. Mount the power amplifier board, also on 6 mm spacers, the power transformer and the PA40 bridge rectifier.

9. The interconnection wiring should now be carried out with the aid of the schematic wiring diagrams. Note that the earth lugs of the RCA sockets for right channel PRE-OUT and MAIN-IN should be linked, and so should those for the left channel. There is no link between left and right and all other sockets have independent earths.

10. All exposed 240 volt wiring should be taped up to provide safety against personal contact. The capacitor C18 should be mounted on the power outlet socket and similarly taped up.

SETTING UP

The only setting up required is the adjustment of bias current in the output stage. For this a milliammeter having a 100 mA range is required.

Rotate trim potentiometer wipers such that they are closest to the front. This adjusts bias current to its lowest value.

Remove both fuses from the right hand channel and the top fuse of the left hand channel. Connect the milliammeter across the left channel fuseholder from which the fuse has been removed.

If a variac is available wind the ac line supply up slowly whilst monitoring the bias current. If a variac is not available the amplifier will have to be switched on, if there is any gross fault the remaining fuse will blow but no other damage should result.

The bias current should be adjusted to about 25 mA. If it is adjustable, but too high, increase the value of R21 to 820 ohms. If it is adjustable but too low, decrease the value of R21 to 330 ohms.

If it is not adjustable *at all* check for errors in the layout or wiring. In a normal amplifier the range of bias

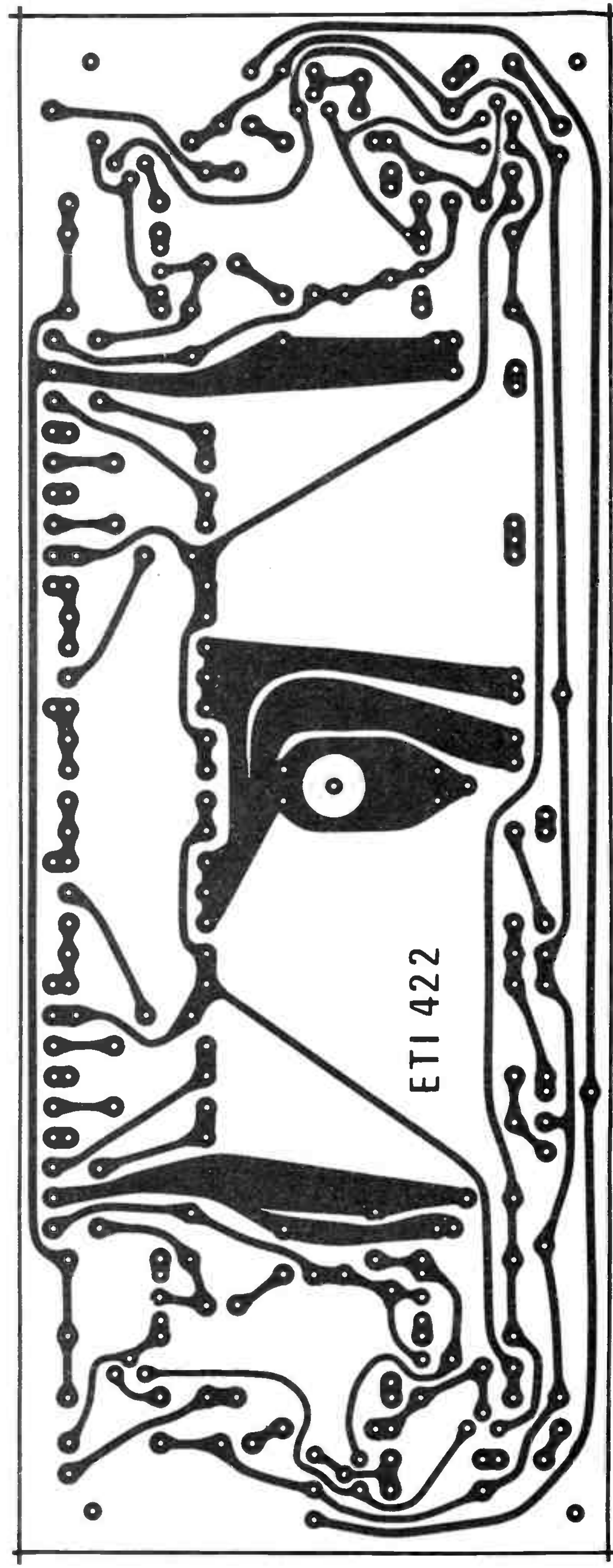


Fig. 5. Printed circuit board pattern for the main amplifier and power supply (full size).

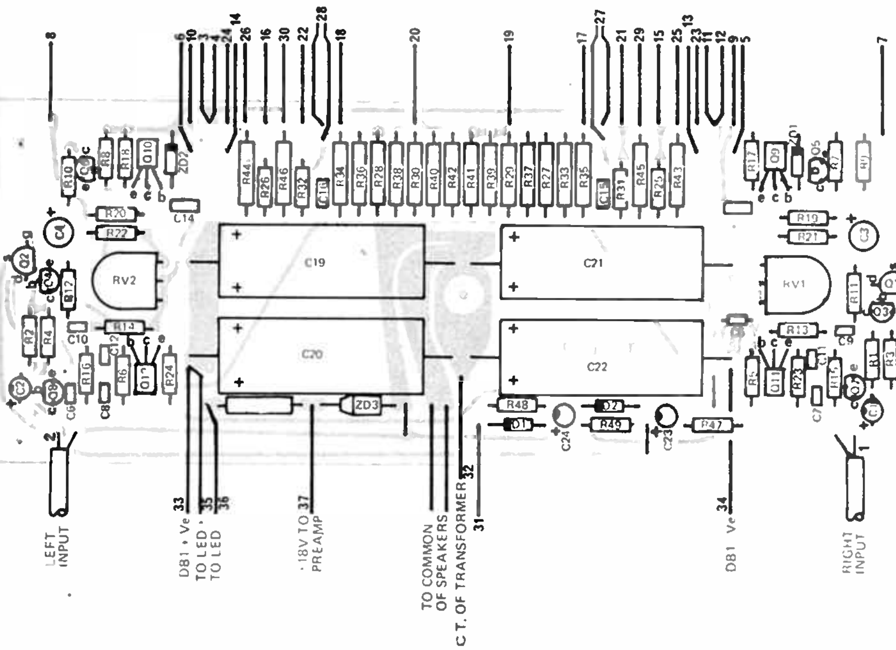


Fig. 6. Component overlay for the main amplifiers and power supply. (Note that capacitors C25, 26, 27 and 28 are not shown and should be soldered across the appropriate resistors).

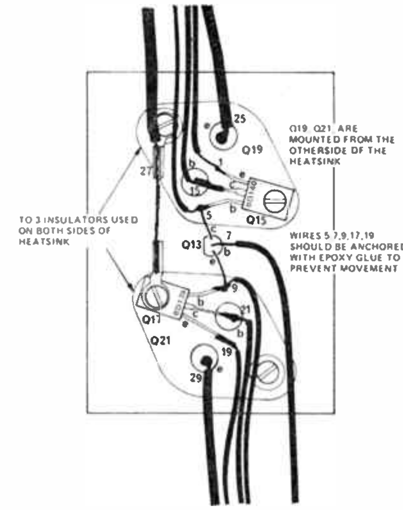


Fig. 7. Details of heatsink assembly. Note particularly the orientation of Q13.

adjustment offered by the trim potentiometer should be entirely adequate.

Switch off and replace the missing left channel fuse together with the lower right channel fuse. Using the milliammeter across the top right-channel fuse holder, adjust the right channel bias current to 20 to 25 mA as for the left channel.

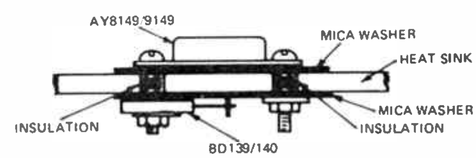


Fig. 8 Method of assembly of power and BD139-140 transistors to the heatsink.

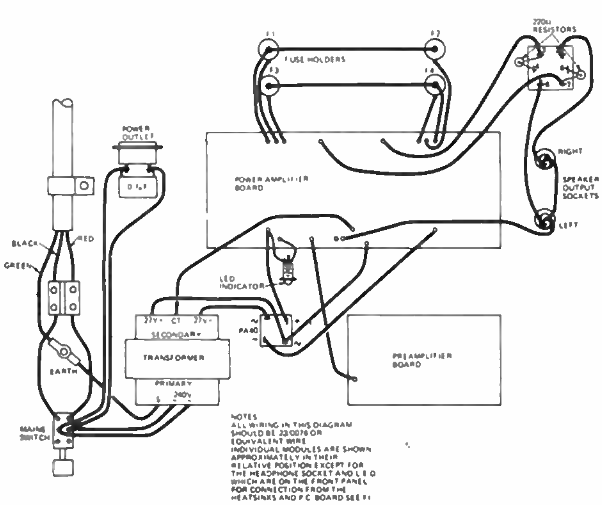


Fig. 9. Power wiring diagram of the complete amplifier.

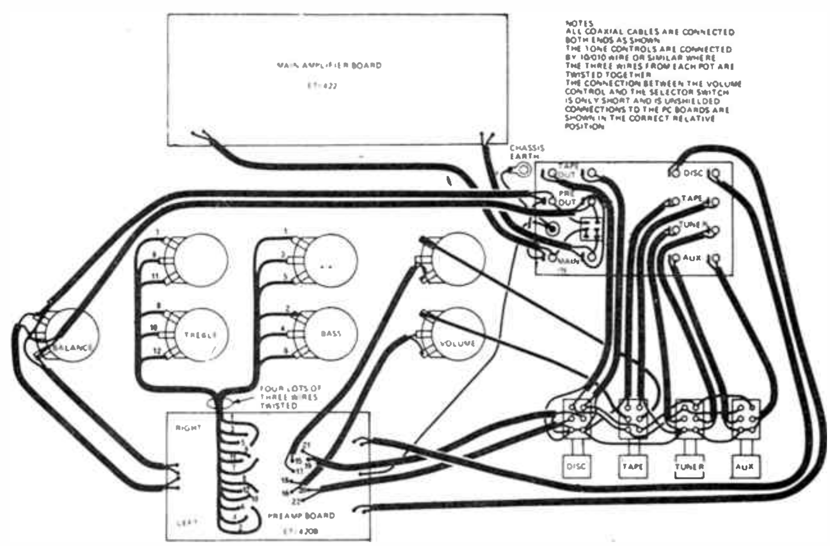


Fig. 10. Signal wiring diagram of the complete amplifier.

HOW IT WORKS – PREAMPLIFIER

The output level of a magnetic cartridge may be as low as 1 mV and this must be amplified and equalised before being applied to the tone controls.

Transistors Q1, 3 and 5 form this equalizing amplifier. The gain is controlled by R11, and the frequency response by R15, R17, C11 and C13. This complex network provides the correct RIAA equalization, the desired signal source and appropriate network being selected by SW1, 2 and 3 and 4. The signal is then passed to Q7 which buffers the output of the volume control and drives the tone control network.

Transistor Q9 and Q11 form a high gain amplifier in which the gain is determined by the relative positions of the bass and treble controls. The gain at 1 kHz is approximately 2.

HOW IT WORKS – MAIN AMPLIFIER

The input signal is fed via C1 and R1 to the base of Q3 which, with Q7, forms a differential pair. Transistor Q5 is a constant current source where the current is [5.6V (ZD1) – 0.6 (Q5)]/2700 (R7) – that is about 2 mA. This current is shared by Q3 and Q7. Transistor Q9 is also a constant current source supplying about 10 mA which, if no input signal exists, flows through Q13 and Q11. The differential pair controls Q11 and thus the voltage at its collector.

INTERNATIONAL 422 STEREO AMPLIFIER

HOW IT WORKS
MAIN AMPLIFIER (Cont'd)

The resistors R19 and R21, together with potentiometer RV1, control the voltage across Q13 and maintain it at about 1.9 volts. But as Q13 is mounted on the heatsink, this voltage will vary with heatsink temperature. Assuming that the voltage at points 5 and 9 is equally spaced about zero volts (ie ± 0.95 volts), the current will be set at about 12 mA through Q15 and Q17. The voltage drop across the 47 ohm resistors (R25 and R31) will be enough to bias the output transistors. Q19 and Q20, on slightly to give about 10 mA quiescent current. This quiescent current is adjustable by means of potentiometer RV1.

Local feedback is applied to the output stage by the network R33, R35, R39 and R41, giving the output stage a voltage gain of about four. The overall feedback resistor, R15, gives the required gain control.

Protection to the amplifier, against shorted output leads, is provided by fuses in the positive and negative supply rails to both amplifiers.

Temperature stability is obtained by mounting Q13 on the heatsink. Q13 will thus automatically adjust the bias voltage. Frequency stability is ensured by C9/R13, C5, C11 and C7.

Although the power amplifier itself does not produce a thump in the loudspeaker, when switched on, the preamplifier does. This is because the preamplifier uses a single power rail and has to stabilize. To reduce this thump to an acceptable level, Q1 is used to short the input for about 2 seconds on switch-on and immediately after switch-off.

The power supply is a conventional full-wave bridge with centre tap, providing + 40 volts and – 40 volts. Diode D1 is used to rectify a second negative supply which is used to control the FETs. Due to the resistance in series with the diode, the charge of C24 is slow. In addition, during the charge period, C23 is also being charged increasing the delay. On switch off, however, C23 cannot assist the voltage on C24 and the off-timing is much shorter than the on-timing.

The power supply for the preamplifier is derived by an 18 volt zener which is fed from the +40 volt rail via an LED power-on indicator and R50.

PARTS LIST — Preamplifier

R59	Resistor	100	¼ or ½W	5%
R25,26	"	220	"	"
R11,12	"	1k	"	"
R37,38,43	"	2.2k	"	"
R44,57,58	"	2.2k	"	"
R19,20	"	3.3k	¼ or ½W	5%
R55,56,61,62	"	5.6k	"	"
R5,6,21,22	"	10k	"	"
R27,28,33,34	"	10k	"	"
R39,40	"	10k	"	"
R35,36,41,42	"	15k	"	"
R47,48	"	22k	"	"
R63,64	"	39k	"	"
R3,4,7,8	"	47k	"	"
R53,54	"	56k	"	"
R17,18,45	"	100k	"	"
R46,49,50	"	100k	"	"
R1,2	"	150k	"	"
R9,10,13	"	220k	"	"
R14,51,52	"	220k	"	"
R29,30,31,32	"	470k	"	"
R15,16,23,24	"	1M	"	"

RV1	Potentiometer	50k log dual r'tary
RV2	"	100k lin dual r'tary
RV3	"	50k lin dual r'tary
RV4	"	50k lin rotary
C5,6,7	Capacitor	33pF ceramic
C8,23,24	"	33pF ceramic
C13,14	"	820pF ceramic
C11,12,25	"	0.0033µF polyester
C26,33,34	"	0.0033µF polyester
C27,28,29,30	"	0.033µF polyester
C1,2,19,20	"	0.47µF 35V tag tantalum
C21,22,31,32	"	0.47µF 35V tag tantalum
C9,10,35,36	"	10µF 16V electrolytic *
C37,38,41,42	"	10µF tag tantalum
C15,16	Capacitor	22µF 16V elect'ytic*
C3,4	"	33µF 10V "
C39	"	47µF 25V "
C17,18	"	100µF 25V "

* PC mounting or tag tantalum

Q1,2,3,4,7	Transistor	BC109, BC549*
Q8,9,10,11,12	"	" "
C5,6	"	BC179,BC559

* can be BC108,548 except for Q1,Q2

SW1-4 Switch assembly McMurdo 2900-4

PC board ETI 420 B

PARTS LIST — Main Amplifiers and Power Supply

R43,44,45,46	Resistor	0.5ohm	2W *	
R37,38	"	10 "	1W	5%
R27,28,29,30	"	33 "	1W	5%
R25,26,31,32	"	47 "	½W	"
R11,12,51,52	"	220 "	½W	"
R33,34,35,36	"	220 "	1W	"
R39,40,41,42	"	220 "	1W	"
R17,18,23,24	"	470 "	½W	"
R21,22	"	560 "	½W	"
R50	"	1k "	1W	"
R13,14,19,20	"	1.2k "	½W	"
R1,2	Resistor	1.5k	½W	5%
R7,8	"	2.7k	"	"
R9,10	"	3.3k	"	"
R5,6	"	5.6k	"	"
R3,4,15,16	"	10k	"	"
R48	"	47k	"	"
R49	"	100k	"	"
R47	"	220k	"	"

* If difficult to obtain, these resistors may be fabricated from a short length of electric jug element — about 90 mm is sufficient for each. Wind securely around a 1 watt resistor (100 ohms or higher) and solder into place.

RV1,2 Potentiometer 470 ohm Trim type

C11,12	Capacitor	27pF ceramic
C7,8,9,10	"	100pF "
C5,6	"	0.0033µF polyester
C13,14,15,16	"	0.1µF "
C25,26,27,28	"	0.1µF " *
C18	"	0.1µF 250V ac
C24	Capacitor	1µF 35V electrolytic *
C1,2	"	4.7µF 10V "
C23	"	10µF 25V "
C3,4	"	100µF 10V "
C19,20,21,22	Capacitor	2500µF 50V electrolytic

* should be PC mounting type

Q1,2	Transistor	2N5485
Q3,4,5,6,7,8	"	2N 3645, BC177
Q9,10,17,18	"	BD 140
Q11,12,15,16	"	BD 139
Q13,14	"	PN 3643
Q19,20	"	AY 9149, MJ2955
Q21,22	"	AY 8149, 2N 3055

D1,2 diode IN 914, EM 401
ZD1, ZD2 5.6 V 400 mV zener diode BZY88C5V6
ZD3 zener diode BZX70C18**
** 20V or 16V will do if 18V unobtainable

DB1 diode bridge PA40

T1 Transformer 56V CT 1.5A or Ferguson PF 3577.

PC Board ETI 422
F1-F4 miniature 2 AMP Fuse and holders John Carr AH750 or equiv.
SW1 McMurdo type 2904-1
4 covers for TO3 transistors
4 insulation kits for TD3 transistors
4 extra mica washers for TO3

PARTS LIST — General

Chassis to Fig. 15.
Front panel to Fig. 13.
Rear panel to Fig. 14.
Wooden box as per Fig. 16.
12.7 mm spacers 4 req.
6.4 mm spacers 9 req.
3 small knobs John Carr type TK196 or similar
1 large knob John Carr type TK195 or similar
1 stereo phone jack McMurdo 1291-04-02 or similar
1 LED indicator McMurdo 3240-02-02
2 heatsinks Warburton Franki WBF-003/3
Miniature DPDT toggle switch
Earth binding post
3 core flex and cable clamp
3 rubber grummets
6 ¼" Whit ½" long screws
17 1/8" Whit 3/8" long c/s screws
4 1/8" whit 3/8" long c/s screws
17 1/8" whit 3/4" long r/h screws
coax cable
23/0076 cable
10/0076 cable

7 dual RCA sockets, John Carr type AT-620 or similar.

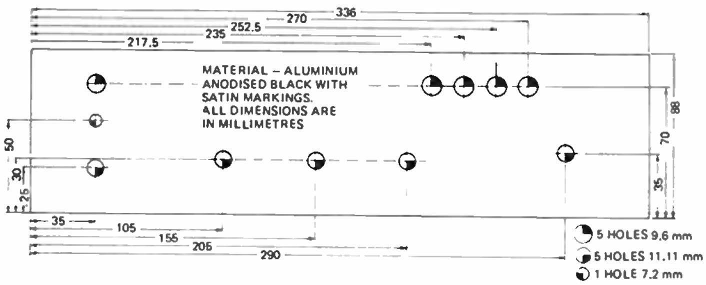


Fig. 13. Drilling details of the front panel escutcheon.

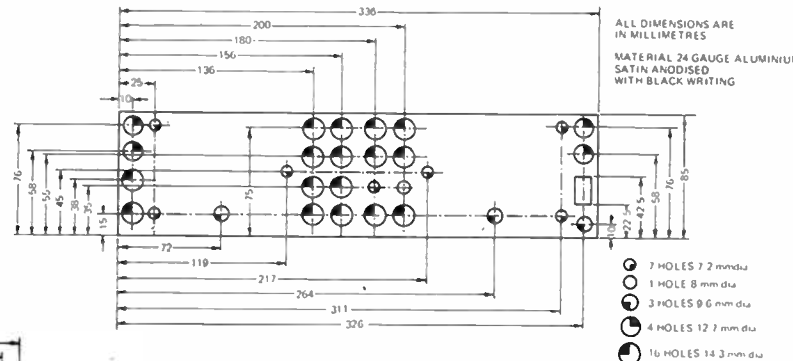


Fig. 14. Drilling details for rear-panel escutcheon.

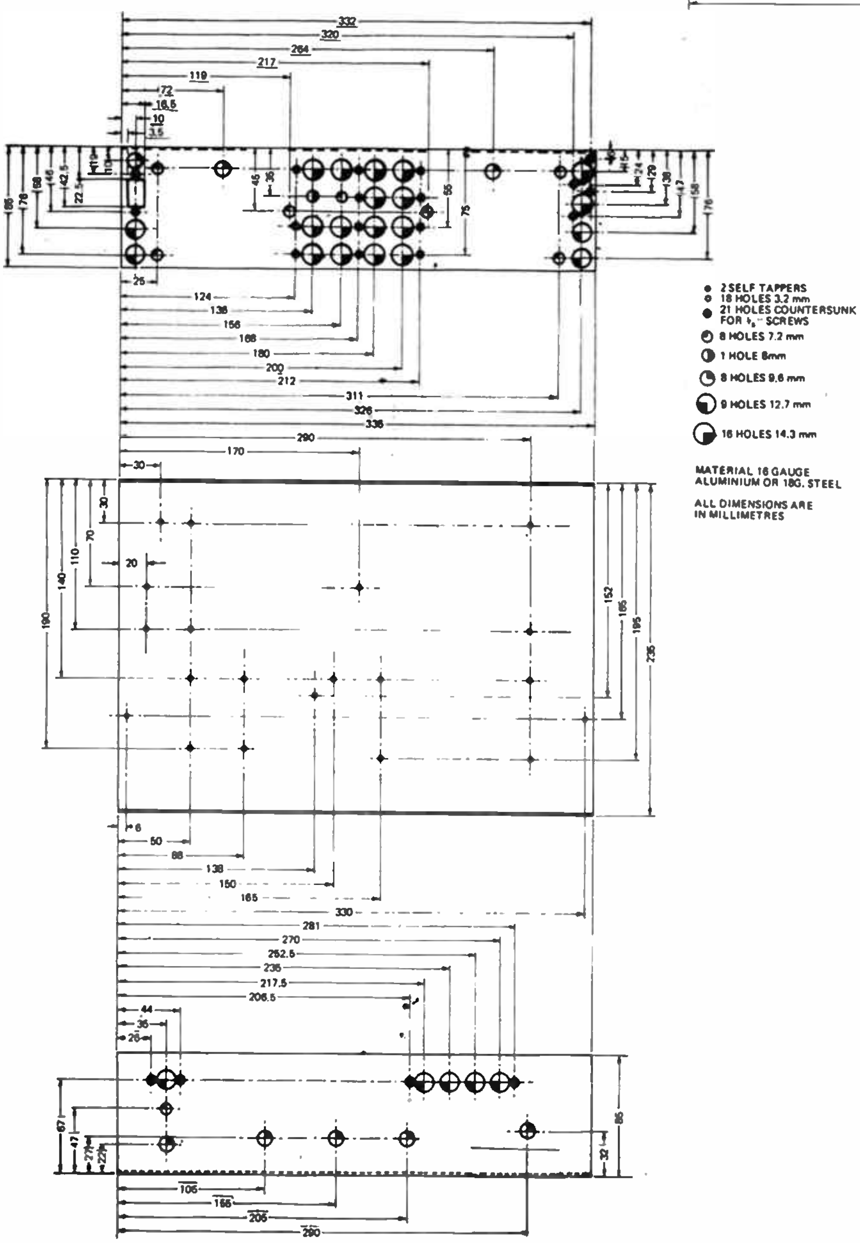


Fig. 15. Drilling details and dimensions of the chassis.

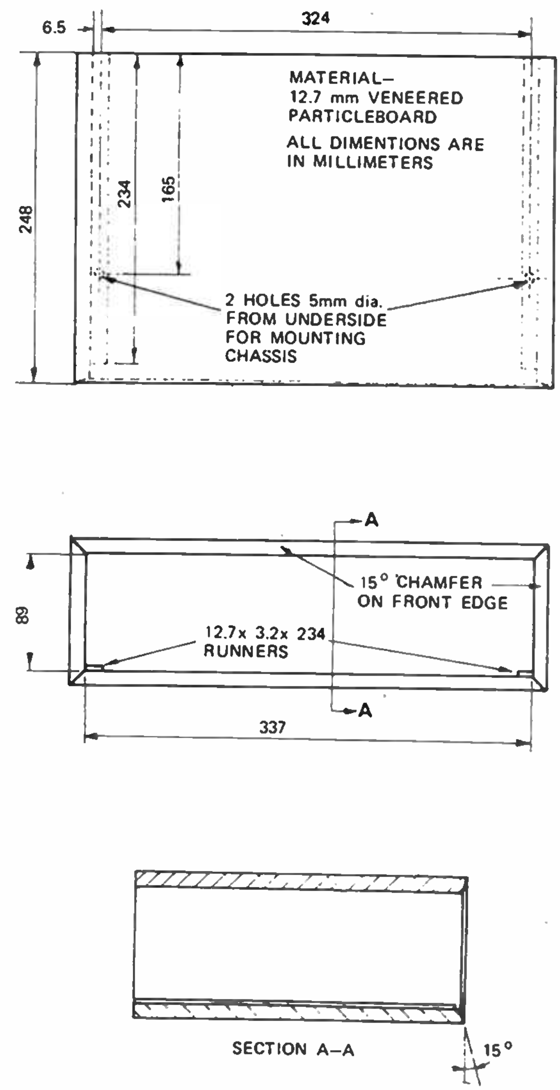


Fig. 16. Constructional details of the cabinet.

SUBSTITUTE ZENER

The 18 volt zener, ZD3, which supplies the stabilized 18 volts for the preamplifier may be difficult to obtain. If so, a 16 volt unit may be used without further modification. A 20 volt zener may also be used, but in this case R53 and R54 should be increased in value from 56 k to 68 k.

The amplifier, because of its wide frequency response, will reproduce pops and clicks, etc, introduced into the mains by equipment (such as refrigerators) switching on and off. Some protection against this is given by C18 (across the primary of the power transformer). If this is insufficient, 0.0047 microfarad 250 Vac capacitors may be fitted between the active end of C18 and earth, and also between the neutral end of C18 and earth.

Note that values higher than 0.0047 microfarad *should not be used.* The use of higher values is illegal, and is a safety hazard. Take care to insulate the leads of these capacitors to avoid accidental personal contact. ●

Fig. 11. Artwork for front-panel escutcheon (half-size).

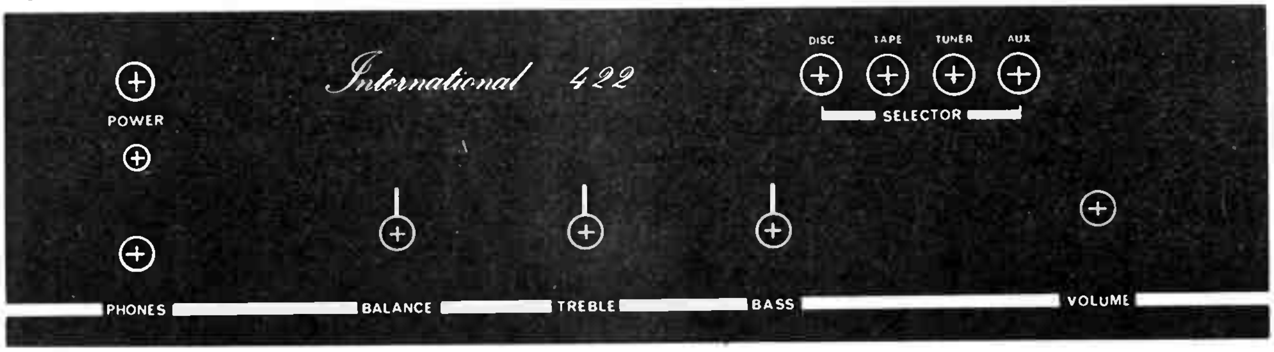
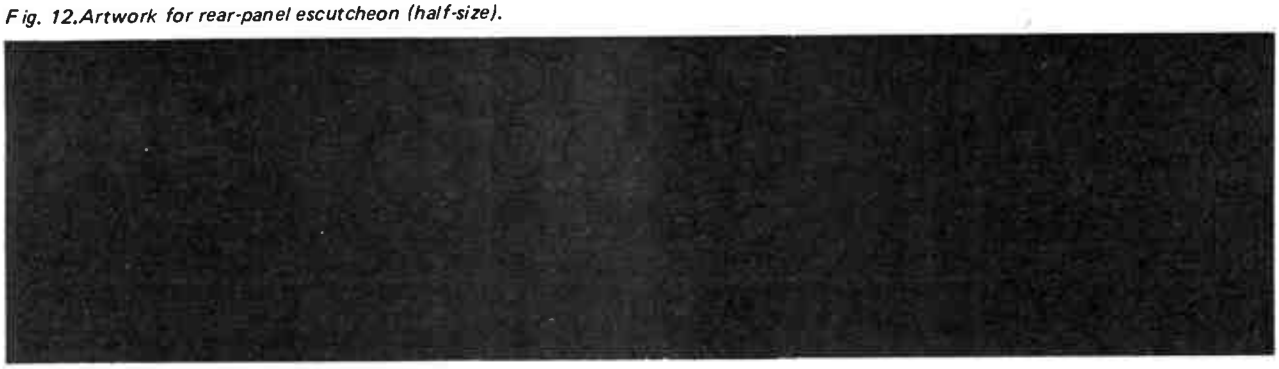
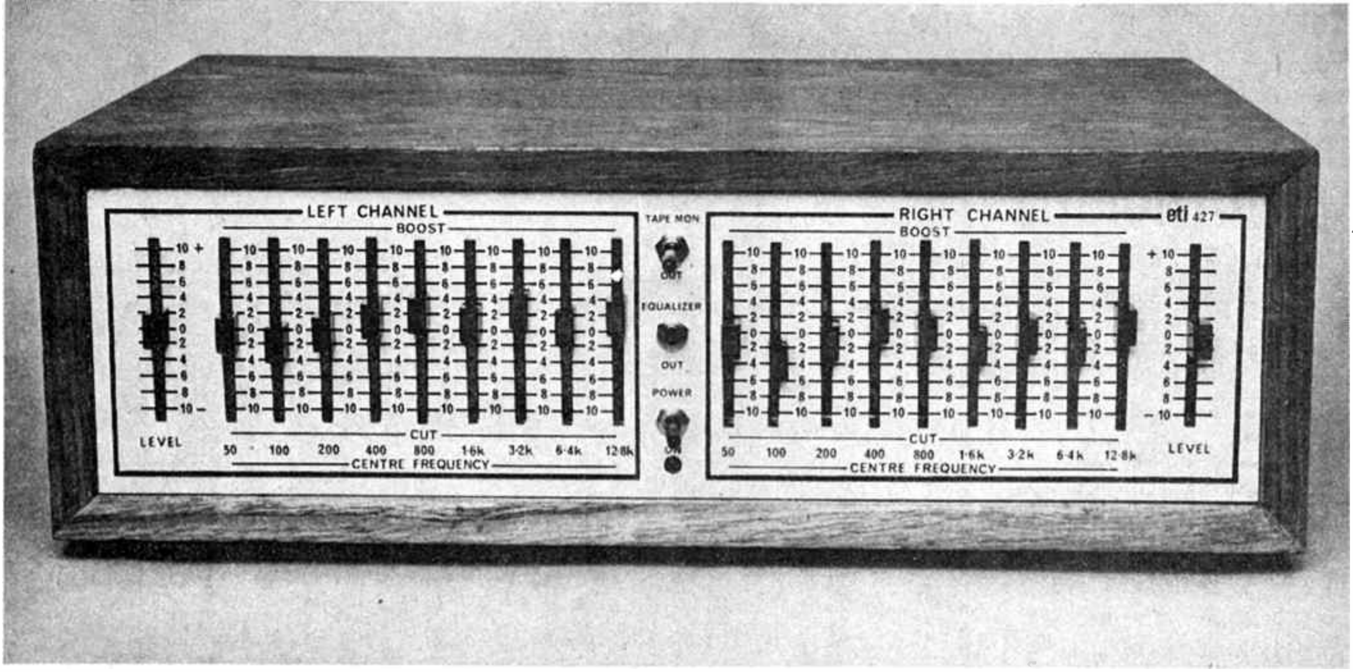


Fig. 12. Artwork for rear-panel escutcheon (half-size).





PROJECT 427 GRAPHIC EQUALIZER

Inexpensive unit compensates for speaker and room deficiencies.

MANY audiophiles are discovering the advantages of graphic equalizers in domestic as well as professional sound systems. Unfortunately the costs of such units have prevented them becoming as popular as warranted by the many advantages they offer.

The advantages of an equalizer are not generally well known but are as follows.

Firstly an equalizer allows the listener to correct deficiencies in the linearity of either his speaker system alone, or the combination of his speaker system and his living room.

As we have pointed out many times in the past, even the best speakers available cannot give correct reproduction in an inadequate room. It is a sad fact that very few rooms are ideal, and most of us put up with resonances and dips, sadly convinced that this is something we have to live with.

Whilst the octave equalizer will not completely overcome such problems, it is possible to minimize some non-linearities of the combined speaker/room system.

In a concert hall it is also possible to use the unit to put a notch at the frequency where microphone feedback occurs, thus allowing higher power levels to be used.

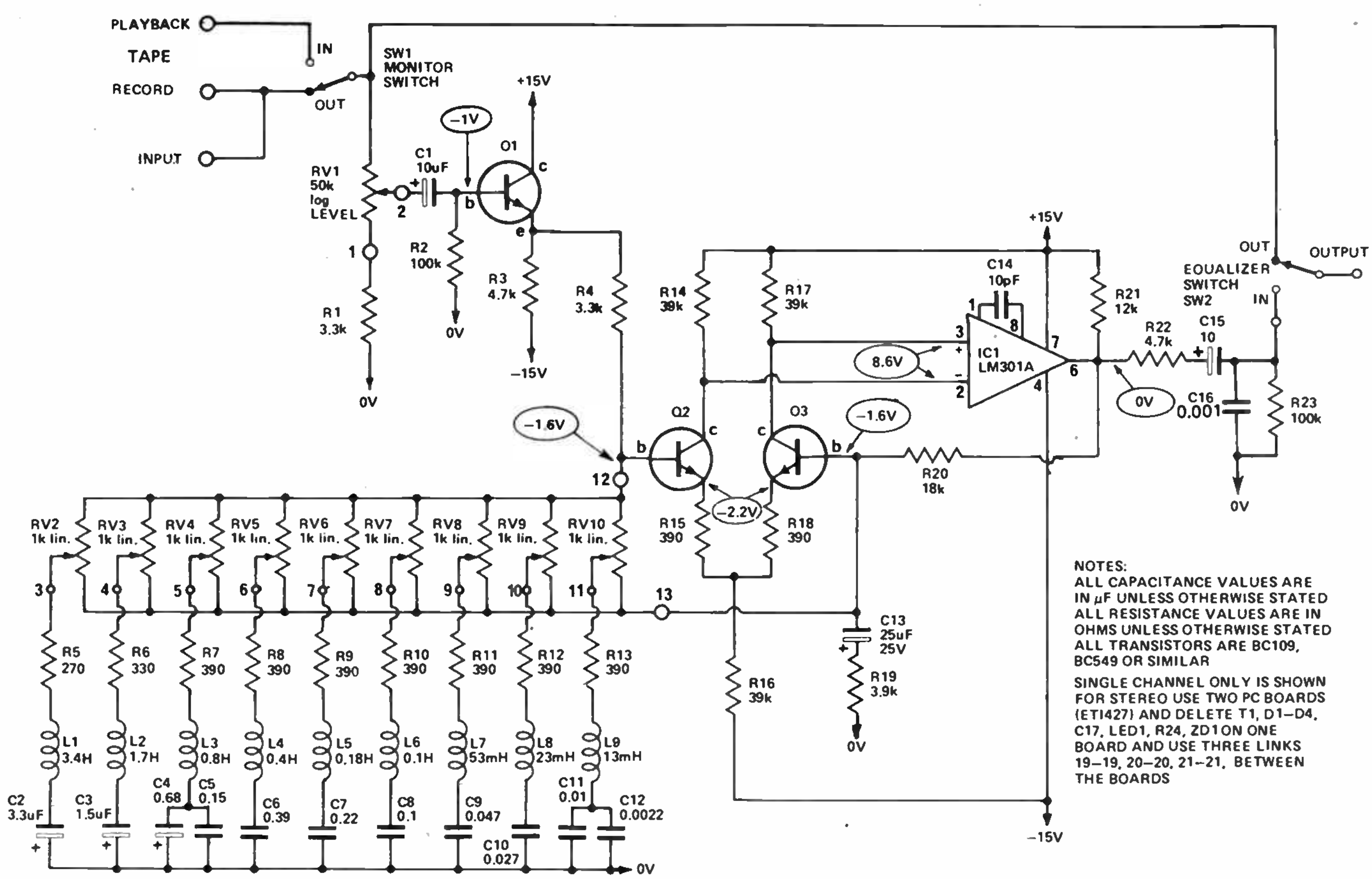
Thirdly, for the serious audiophile, an equalizer is an exceedingly-valuable tool in evaluating the deficiences in a particular system. One adjusts the equalizer to provide a uniform response, the settings of the potentiometer knobs then graphically display the areas where the speaker etc is deficient.

There is a snag, however, one must have an educated ear in order to properly equalize a system to a flat response. It is not much use equalizing to your own preference of peaky bass etc in order to evaluate a speaker.

Ideally, a graphic equalizer should

MEASURED PERFORMANCE (of Prototype)

Frequency Response			
Equalizer out		Flat	
Equalizer in	10 Hz – 10 kHz	±½ dB	
and all controls at zero	1.5 Hz – 30 kHz	+ ½ – 3 dB	
Range Of Control			
Individual filters		±13 dB	
Level control		+ 14 – 9 dB	
Maximum Output Signal at < 1% distortion		> 6 volts	
Maximum Input Voltage		3 volts	
Distortion			
at 2 volts out, controls flat	100 Hz <0.1%	1 kHz <0.1%	6.3 kHz <0.1%
Signal to Noise Ratio at 2 volts out (unweighted)		69 dB	
Input Impedance		50 k	
Output Impedance		4.7 k	



▲ *Fig. 1. Circuit diagram of one channel of the equalizer.*

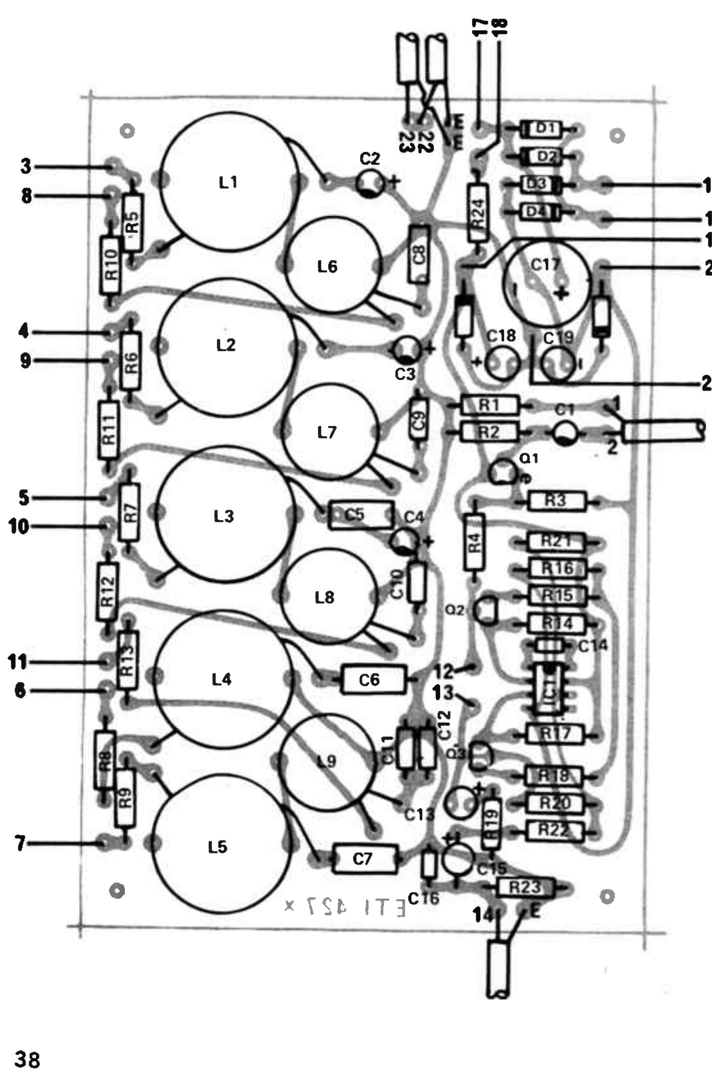


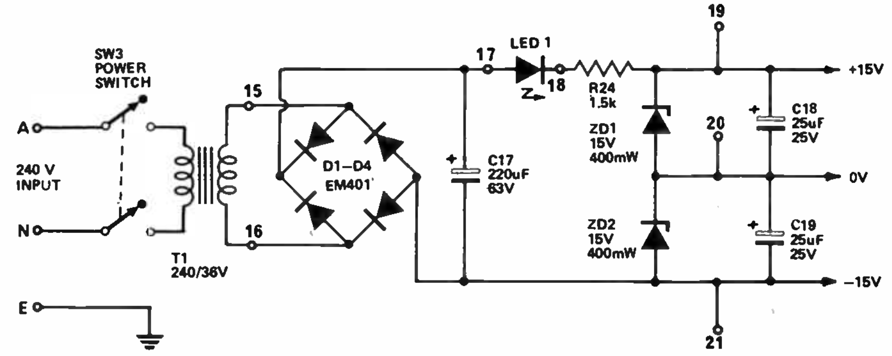
Fig. 2. Component overlay of the equalizer (one channel only)

GRAPHIC EQUALIZER

have filter at 1/3 octave intervals, but except for sound studios and wealthy pop groups, the expense and size of such units are too much for most people.

Recently some excellent commercial units have become available with filters spaced at octave intervals. These are relatively inexpensive and cater for the needs of most professionals and domestic users. Such a unit is the Soundcraftsmen 2012 reviewed in the July 74 edition of Electronics Today International.

The Electronics Today Equalizer has been designed to provide nine filters spaced at octave intervals in each of two channels. It is simple to construct and should be available inexpensively in kit form.



Circuit diagram of the equalizer power supply.

HOW IT WORKS ETI 427

This equalizer is basically similar to those used in the ETI Synthesizer and master mixer projects with the exception that it has nine filter sections per channel.

The equalizer stage is a little unusual in that the filter networks are arranged to vary the negative feedback path around the amplifier. If we consider one filter section alone, with all others disconnected, the impedance of the LCR network will be 390 ohms at the resonant frequency of the network. At either side of resonance the impedance will rise (with a slope dependant on the Q of the network which is 2.5) due to the uncancelled reactance. This will be inductive above resonance and capacitive below resonance. We can therefore represent the equalizer stage by the equivalent circuit below.

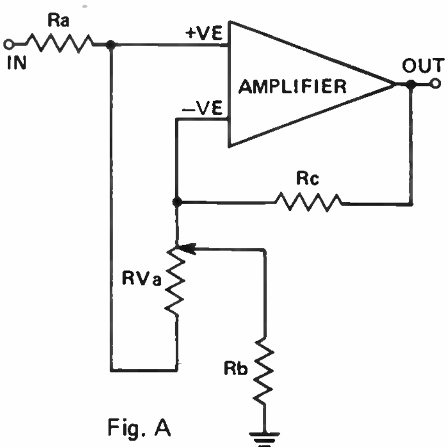


Fig. A

It must be emphasized that this equivalent circuit represents the condition with one filter only, at its resonant frequency.

Additionally letters have been used to designate resistors to avoid confusion with components in the actual circuit.

With the slider of the potentiometer at the top end (Fig. A) we have 390 ohms to the OV line from the negative input of the amplifier, and 1 k between the two inputs of the amplifier. The amplifier, due to the feedback applied, will keep the potential between the two inputs at zero. Thus there is no current through RVA. The voltage on the positive input to the amplifier is therefore the same as the input voltage since there is no current through, or voltage drop across resistor RA.

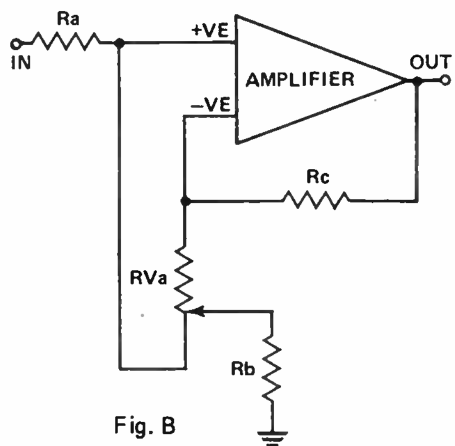


Fig. B

The output of the amplifier in this case is approximately the input signal times (3300 + 390)/390 giving a gain of 19 dB. If the slider is at the other end of the potentiometer, (Fig. B), the signal appearing at the positive input, and thus also the negative input, is about 0.11 (390/(3300 + 390) of the input. There will still be no current in the potentiometer and in RC, thus the output will be 0.11 of the input. That is, the gain will be -19 dB.

If the wiper is midway, both the input signal and the feedback signal are attenuated equally, and the stage will have unity gain.

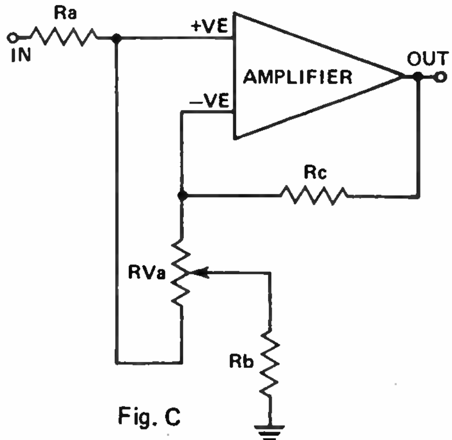


Fig. C

With all filter sections in circuit the maximum cut and boost available is reduced, but ±14 dB is still available.

Reverting back now to the actual circuit, the amplifier consists of IC1, Q2 and Q3. The transistors help to reduce the effect of the noise in the IC and add gain at the high-frequency end. This additional gain is required because the negative feedback, due to the potentiometer between the two inputs, causes high-frequency roll off. This does not affect operation of the unit provided the open-loop gain is above 60 dB over the entire audio range. An overall closed-loop gain of about 15 dB is maintained by R20/R19 with the filter potentiometer at mid position.

The output of the amplifier is decoupled to the output of the unit via C15, and C16/R22 provide a cutoff above 30 kHz.

The input signal is buffered by Q1 because the equalizer stage requires a low impedance signal source for correct operation. Potentiometer RV1 provides level control with 0 to -23 dB range which, combined with the equalizer characteristic, results in an overall level range of + 14 to -9 dB.

The power supply used is a simple, full-wave bridge filtered by C17. Plus and minus supplies are derived by means of two 15 volt zeners in series fed via R24. The front-panel power indicator is an LED connected in series with the dropping resistor R24.

CONSTRUCTION

All components, with the exception of the transformer and the slide potentiometers, are mounted on two printed circuit boards — one for each channel. Whilst the layout is not critical, any alternative construction method could be used, we strongly recommend the use of printed-circuit boards to ease construction and eliminate a possible source of faults.

The components should be assembled to the boards with the aid of the overlay Fig. 2 Carefully check polarities of ICs, capacitors and transistors, etc, before soldering in place. Check particularly the BC549 transistor as there are two lead configurations manufactured for the same type number. The Philips type is the one shown on the overlay. Attach lengths of wire and Coax of adequate length to the board before mounting in position by means of 13 mm spacers.

Due to the close spacing used for the slide potentiometers it is necessary to mount the 9.6 mm spacers, to the potentiometer support-bars, before mounting the potentiometers. Use 6.4 mm long countersunk screws for this purpose.

The potentiometer assembly, and all other external components, (switches etc) can now be assembled to the chassis and the unit wired as shown in the interconnection diagram.

The circuits used have very high gains and it is necessary to take precautions

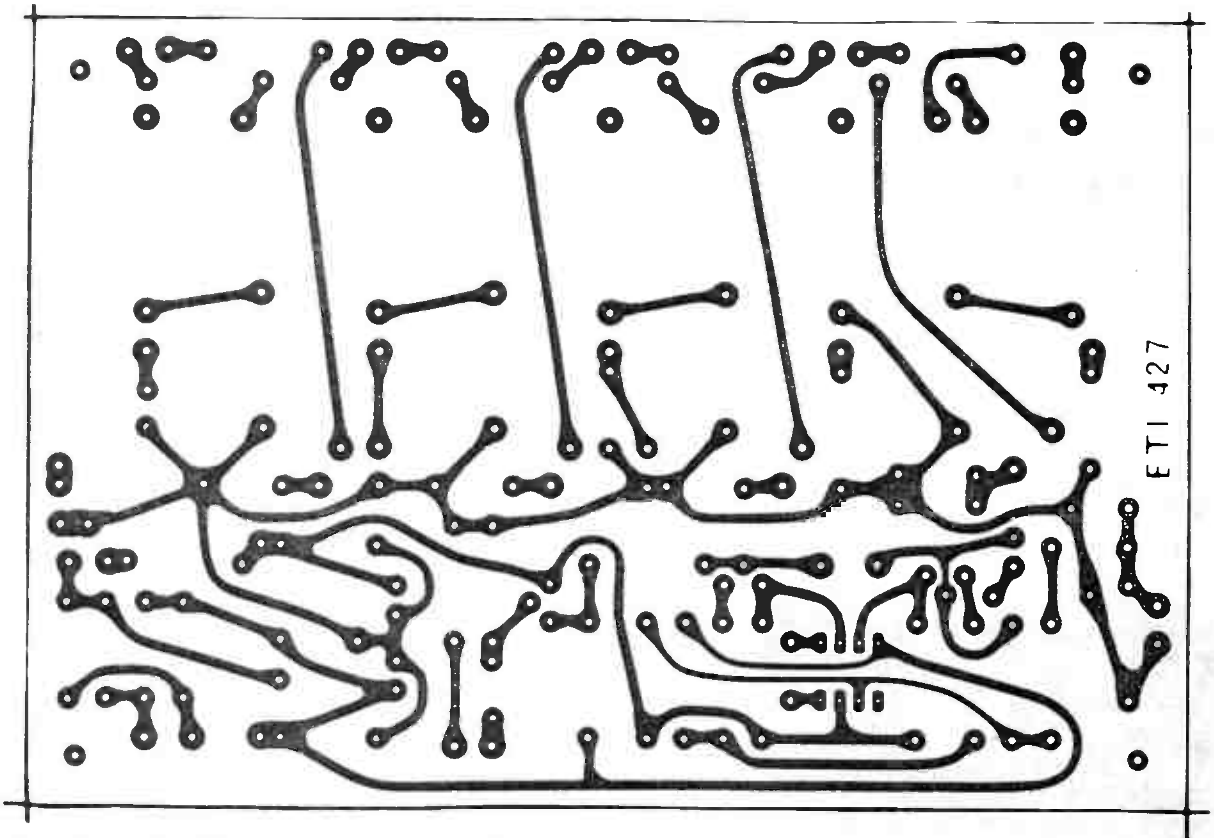


Fig. 4. Printed circuit board for the equalizer. Full size 152 x 103 mm.

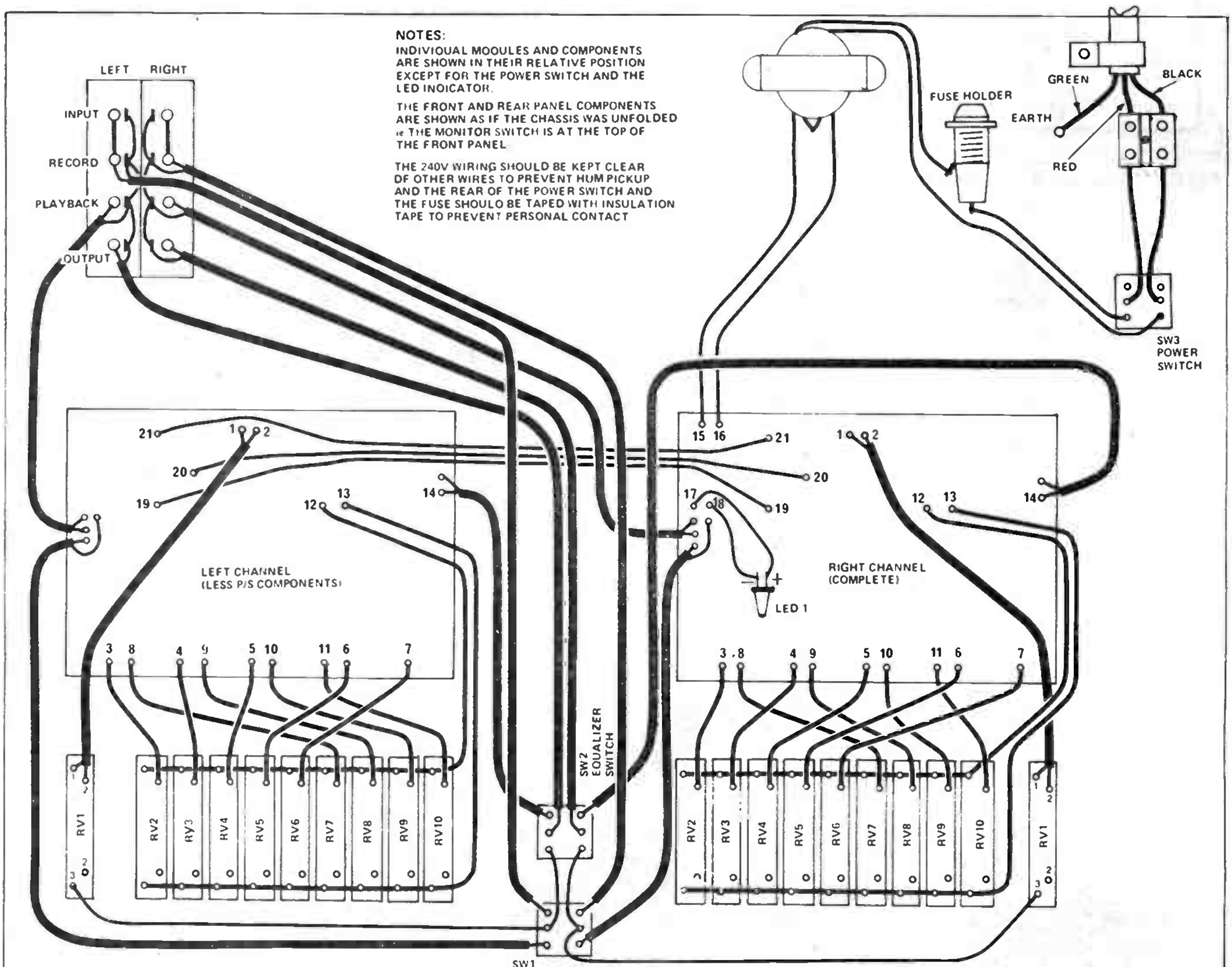


Fig. 3. Interconnection diagram for the unit.

against mains hum-pickup. The transformer should be mounted in the position shown, and the 240 volt wiring, to the front power switch, should be run down the right-hand side of the chassis and along the front, in front of the potentiometer support brackets. If a different transformer is used, or if hum pickup does occur, it may be necessary to mount the transformer inside a metal box to shield it.

Due to tolerances of resistors variations in V_{be} of Q2 and Q3 etc, the steady-state output of IC11 may be anywhere within plus or minus one volt of zero.

Hence it is desirable to determine the polarity of the steady state voltage at pin 6 of IC1 in order to determine which way round C15 should be inserted. If the output is positive insert as shown in Fig. 1. Alternatively C15 should be a non-polarized type such as the Sonar RBP series.

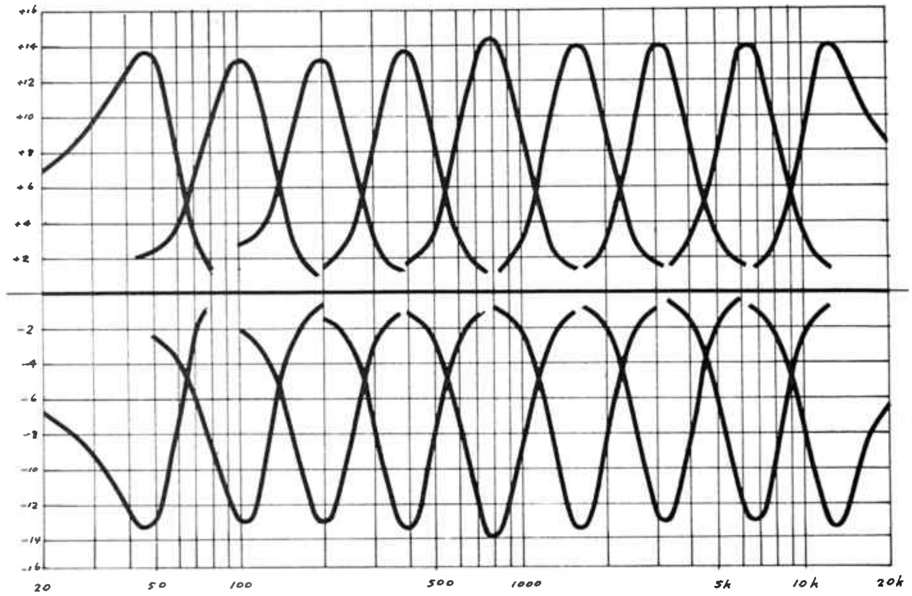


Fig. 5. Individual filter responses for the unit. Boost at top and cut at bottom.

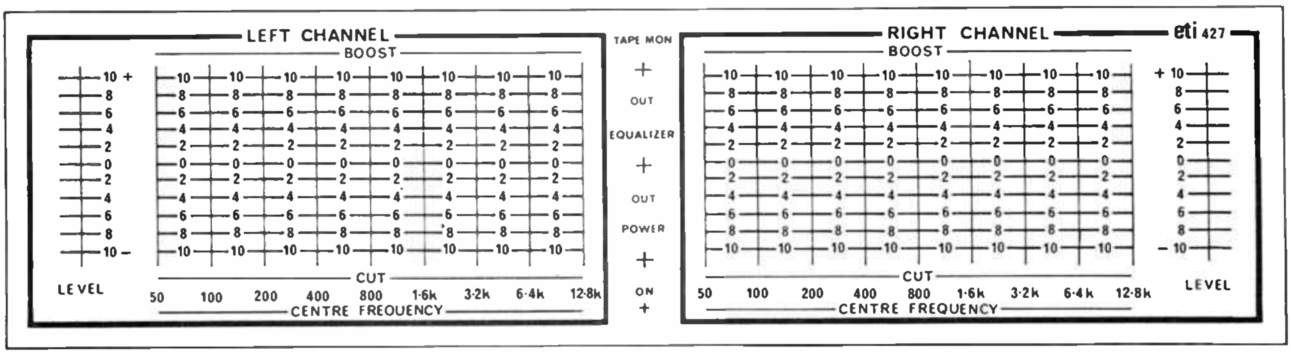
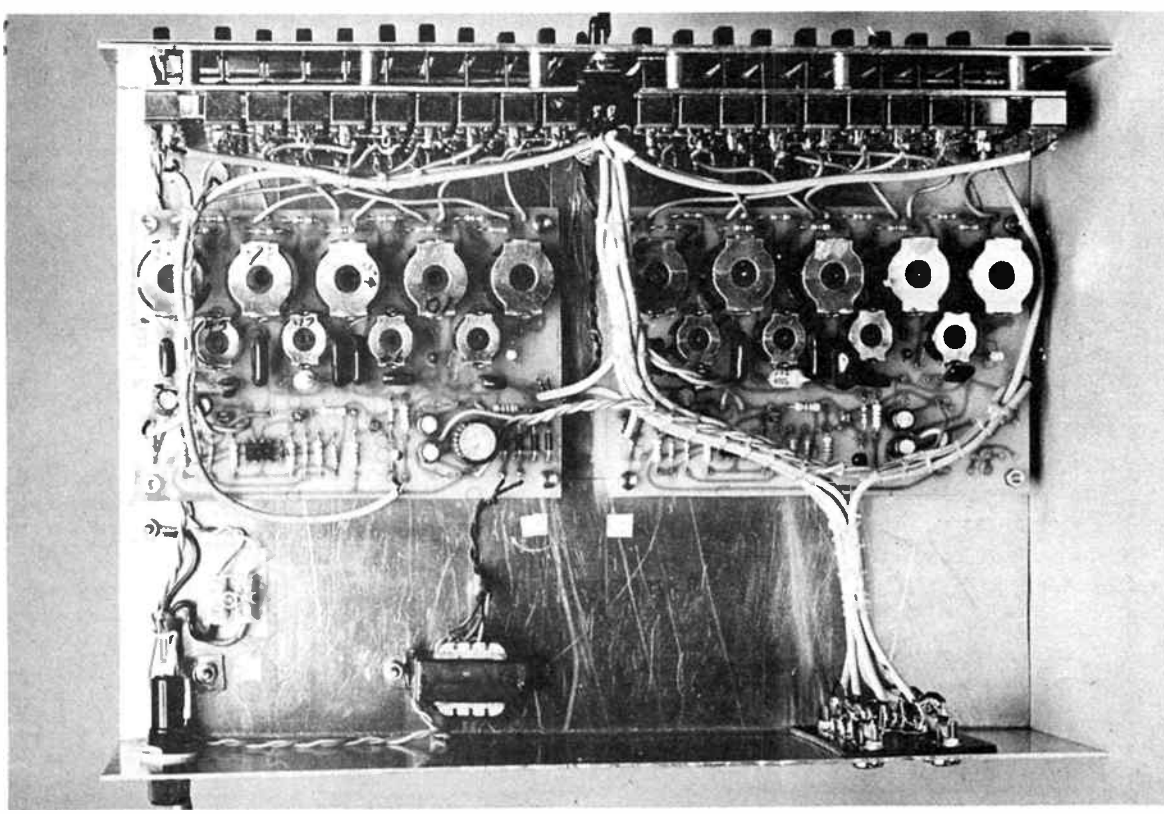


Fig. 6. Front panel artwork for the equalizer. Full size 336 x 88 mm.



Internal layout of the equalizer.

GRAPHIC EQUALIZER

TABLE I. CHOKE WINDING DATA

The types of potcore available at any given time is variable. We have therefore provided winding details for several different cores.

Note that R5 and R6 may alter with certain cores. Part numbers quoted are for Elcoma Ferrite potcores.

Part Nos listed are Philips

Core 4322.022.29120 or 4322.022.29320 (AL = 1600)
Bobbin 4322.021.30330. Clip 4302.021.20020

	Inductance	Turns	Wire Size	Resistance	Notes
L1	3.4 H	1450	0.125 mm	104 ohms	R5 = 270 ohms
L2	1.7 H	1030	0.125 mm	70 ohms	R6 = 330 ohms
L3	0.8 H	700	0.16 mm	30 ohms	
L4	0.4 H	500	0.16 mm	20 ohms	
L5	0.18 H	340	0.16 mm	13 ohms	

Core 4322.022.28090 or 4322.022.28290 (μe = 330)
Core 4322.022.29110 or 4322.022.29310 (AL = 1000)
Bobbin 4322.021.30330 Clip 4302.021.20020

	Inductance	Turns	Wire Size	Resistance	Notes
L1	3.4 H	1800	0.10 mm	180 ohms	R5 = 180 ohms
L2	1.7 H	1280	0.125 mm	90 ohms	R6 = 300 ohms
L3	0.8 H	860	0.16 mm	40 ohms	
L4	0.4 H	630	0.16 mm	28 ohms	
L5	0.18 H	420	0.16 mm	16 ohms	

Core 4322.022.24080 or 4322.022.24280 (μe = 220)
Bobbin 4322.021.30270 Clip 4307.021.20000

	Inductance	Turns	Wire Size	Resistance
L6	100 mH	465	0.125 mm	23 ohms
L7	53 mH	340	0.16 mm	10 ohms
L8	23 mH	225	0.16 mm	6.2 ohms
L9	13 mH	170	0.16 mm	4.5 ohms

Core 4322.022.25100 or 4322.022.25300 (AL = 630)
Bobbin 4322.021.30270 Clip 4307.021.20000

	Inductance	Turns	Wire Size	Resistance
L6	100 mH	400	0.16 mm	12 ohms
L7	53 mH	290	0.16 mm	8 ohms
L8	23 mH	190	0.16 mm	5 ohms
L9	13 mH	145	0.25 mm	1.6 ohms

PARTS LIST — ETI 427

R5	Resistor	270	½W	5%
R6	"	330	½W	5%
R7,8,9	"	390	½W	5%
R10,11,12	"	390	½W	5%
R13,15,18	"	390	½W	5%
R24	"	1.5k	½W	5%
R1,R4	"	3.3k	½W	5%
R19	"	3.9k	½W	5%
R3,22	"	4.7k	½W	5%
R21	"	12k	½W	5%
R20	"	18k	½W	5%
R14,16,17	"	39k	½W	5%
R2,23	"	100k	½W	5%

RV1 Potentiometer 50k log 45 mm slide
RV2-10 Potentiometer 1 k lin 45 mm slide
C17 Capacitor 220 µF 63V electrolytic
C13,18,19 Capacitor 25 µF 25V electrolytic
C1,15 Capacitor 10 µF 16V electrolytic
C2 " 3.3 µF 10V tag tant.
C3 " 1.5 µF 25V " "
C4 " 0.68 µF 25V tag tantalum
C6 " 0.39 µF polyester
C7 " 0.22 µF "
C5 " 0.15 µF "
C8 " 0.1 µF "

C9 " 0.047 µF "
C10 " 0.027 µF "
C11 " 0.01 µF "
C12 " 0.0022 µF "
C16 " 0.001 µF "
C14 " 10 pF ceramic
L1-L9 Chokes See table I
Q1,2,3 Transistor BC109, BC549 or similar
D1,2,3,4 Diodes EM401 or similar
ZD1,2 Zener Diode BZX70C15
LED 1 light emitting Diode
IC1 Integrated Circuit LM301A
PC Board ETI 427
For stereo operation double the above components except R24, C17, LED 1, ZD1, ZD2, D1-D4 where only one is required.
Transformer 240V — 36V @ 30 mA PF3787 or similar
SW1,2,3 switch DPDT miniature toggle
4-way RCA socket, 2 off
Chassis to Fig. 6.
Front panel to Fig. 7 and 8.
20 off knobs for slide pots McMurdo P/N 4093 — Black (Isostat)
4 pot support rails (Fig. 9)
12 threaded spacers 9.6 mm long
8 plain spacers 12.7 mm long
24 screws, countersunk head, 6.5 mm long to suit spacers
3 core flex & plug
Cable clamp, grommet, terminal block

GRAPHIC EQUALIZER

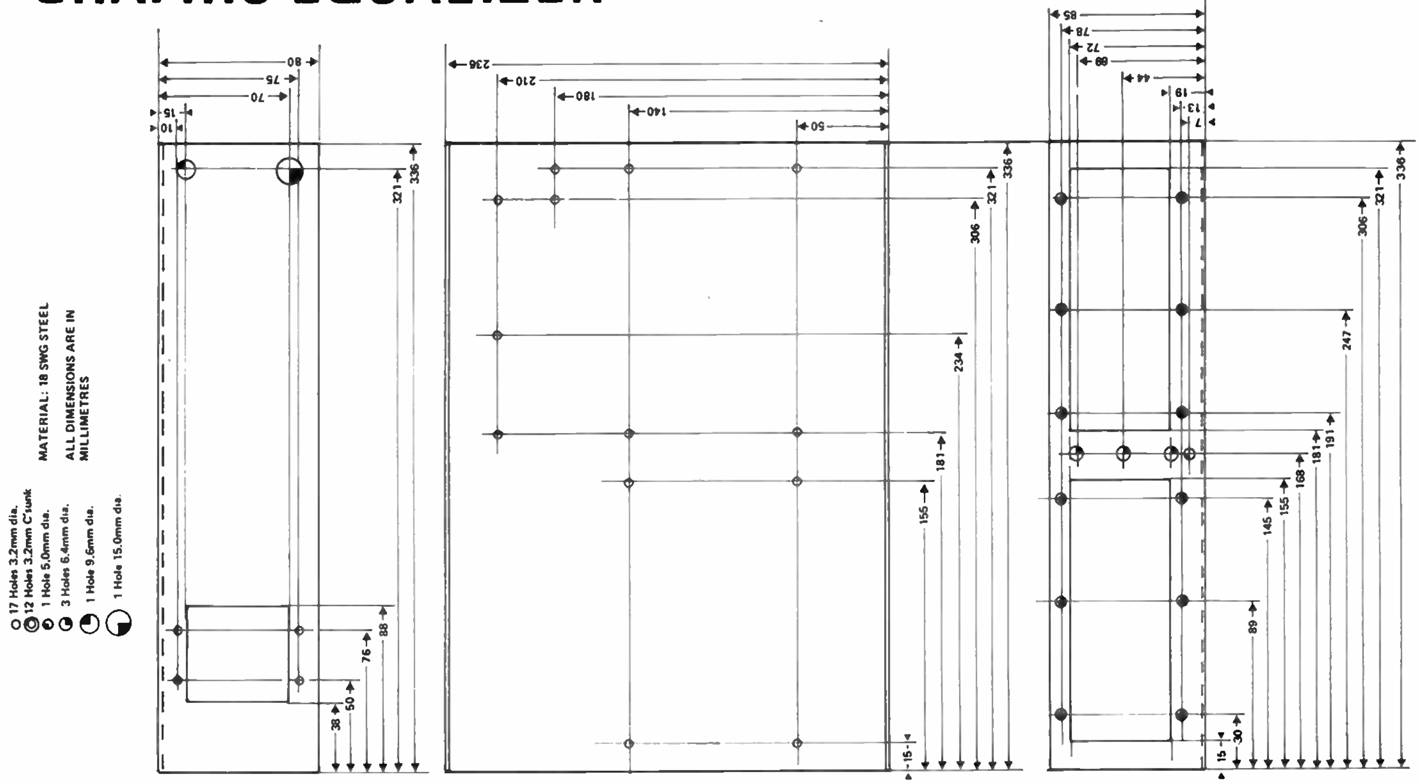


Fig. 7 Detail of the chassis.

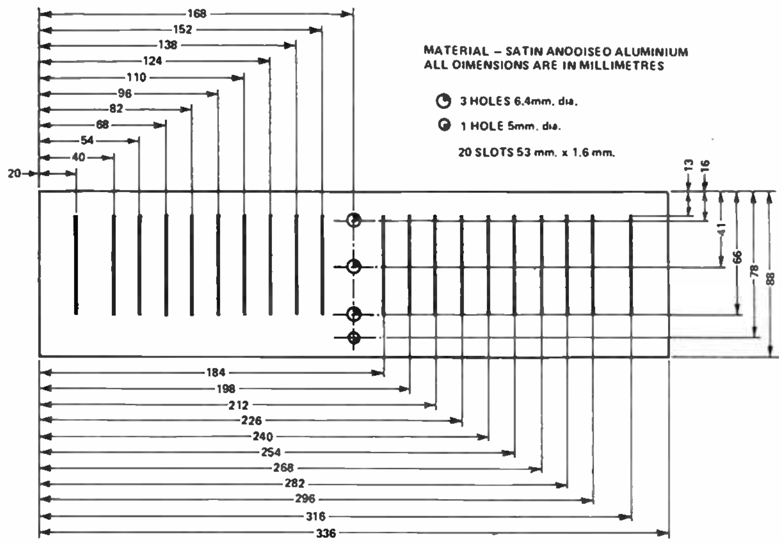


Fig. 8 Metalwork details of the front panel.

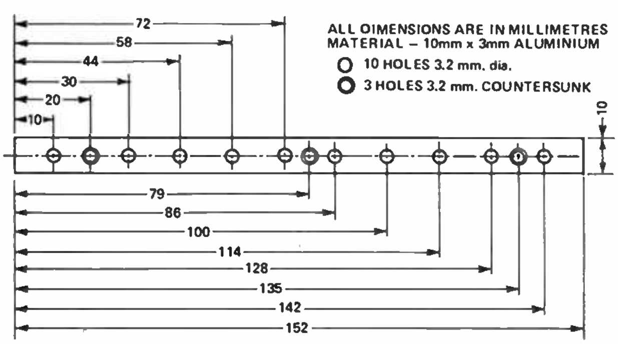


Fig. 9 Drilling details for potentiometer support brackets.

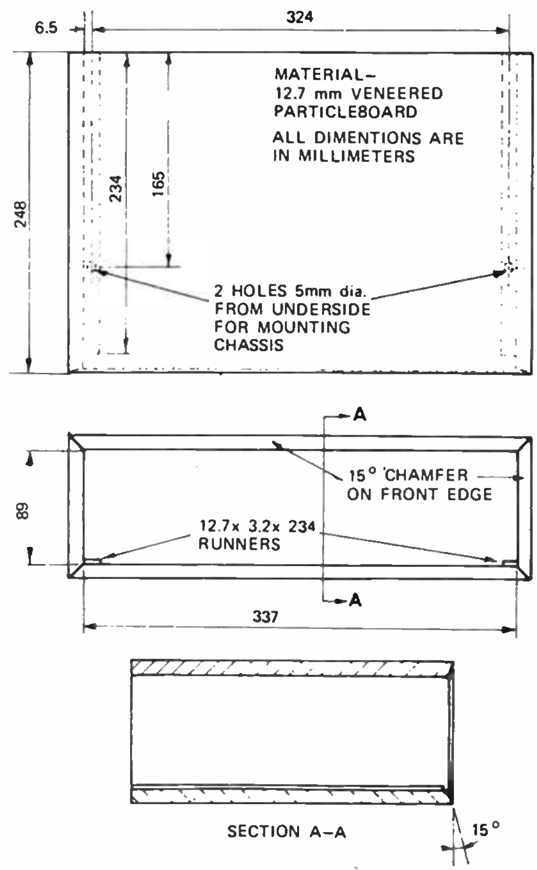
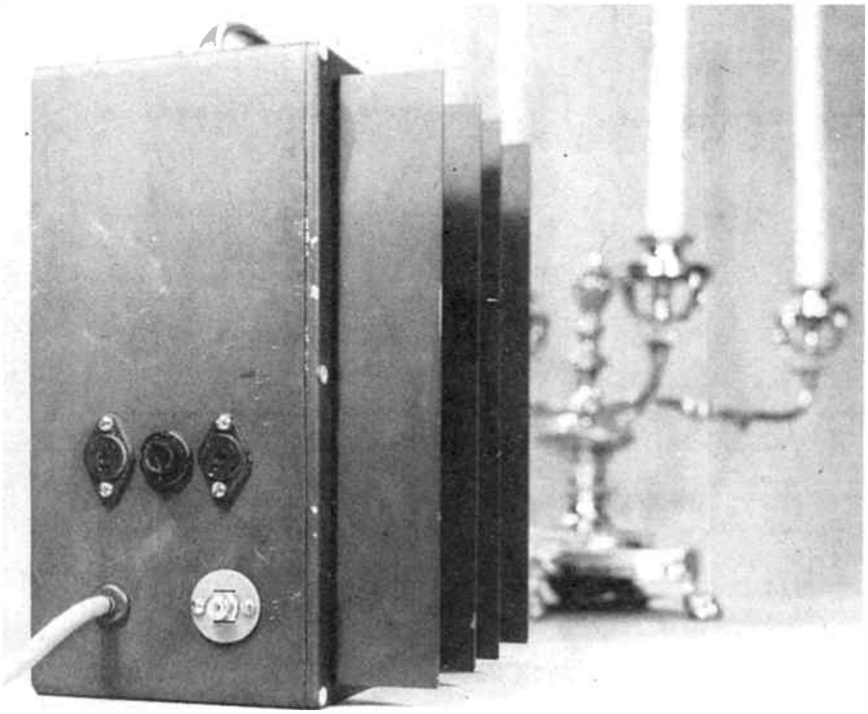


Fig. 10. Constructional details of the cabinet.

100W GUITAR AMPLIFIER



In the early days of radio one of the standard acceptance tests for shipborne radio apparatus was its ability to withstand a 13 stone radio operator climbing up the equipment rack wearing heavy boots.

Electronic equipment used by pop groups and for public address systems – whilst often built to substantially lower standards – often receives similarly rugged treatment.

For this type of use, the ability to operate reliably despite having spent the previous six hours rolling around in the boot of a car will be of far more importance than a stainless steel facia with a lot of coloured indicator lights and VU meters.

The amplifier described in this project has been specifically designed for just such applications.

It is intended primarily as a guitar amplifier and for public address systems. In the interests of ruggedness it has been put together entirely without frills. It has no tone or volume controls and must be used with a suitable preamplifier.

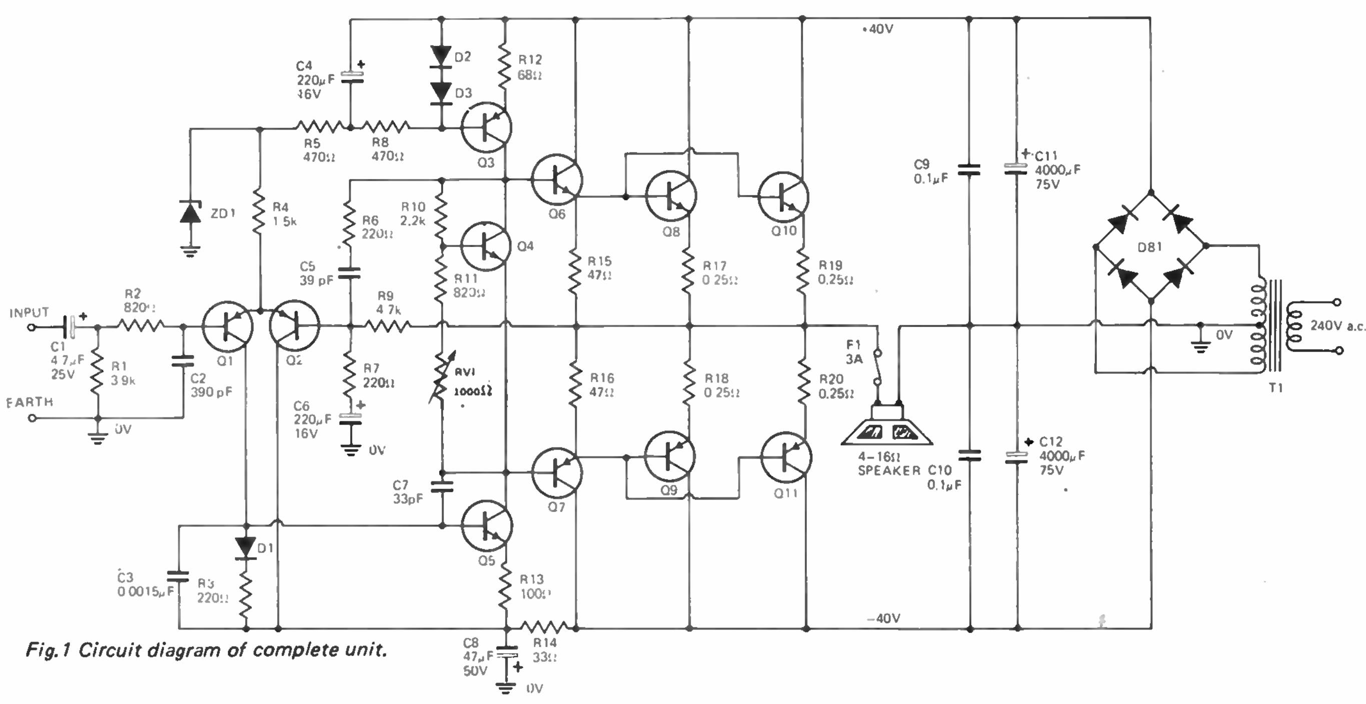


Fig. 1 Circuit diagram of complete unit.

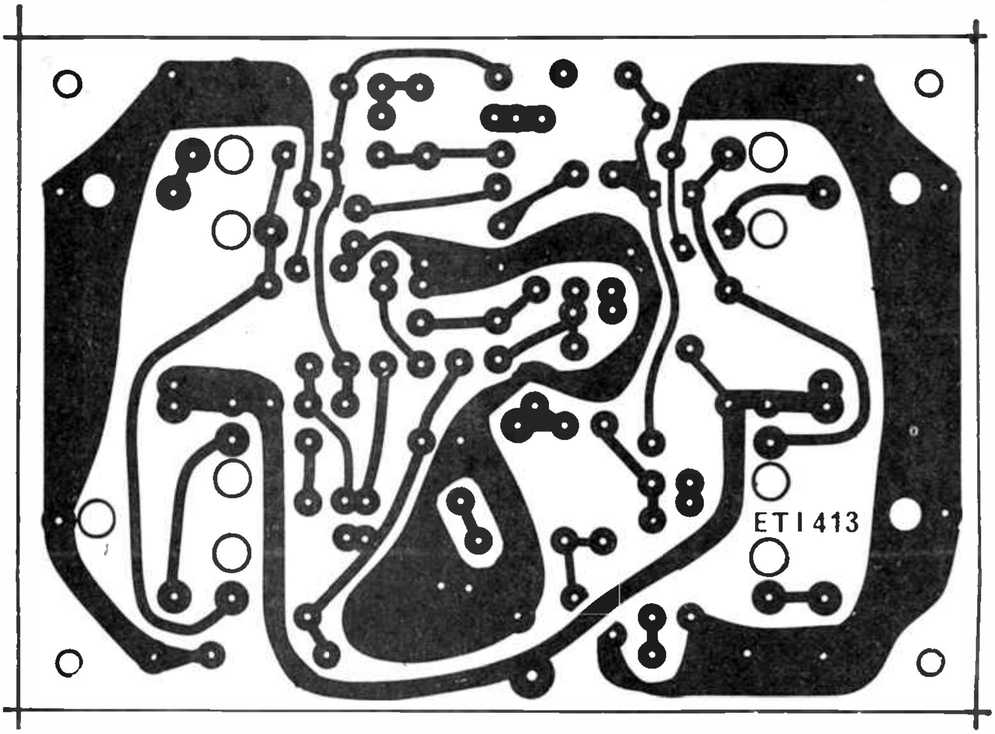
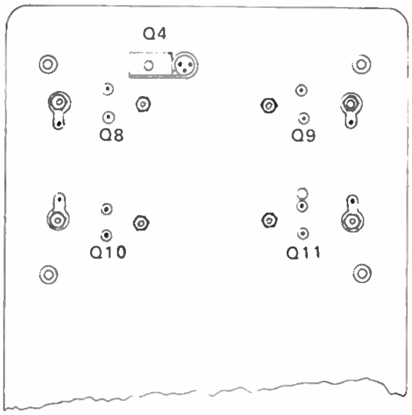


Fig. 2 Foil pattern for printed circuit board (full size).

It is not only rugged mechanically, for it will handle over a hundred watts continuously with a sine-wave input.

Despite the design criteria of ruggedness, the performance specifications put the unit well into the hi-fi area. Frequency response — as the accompanying table shows — is virtually flat from 50 Hz to 20 kHz and total harmonic distortion is less than 0.5% from 0.1W to 80W.

Any number of speakers may be driven from this amplifier providing their combined impedance is equal to or exceeds four ohms.



Inside view of lid showing position of Q4, Q8, Q9, Q10 and Q11 (See also Fig. 6.)

CONSTRUCTION

Construction is quite straightforward as most components are mounted on the printed circuit board.

Start by soldering the components on to the printed circuit board according to the layout shown in Fig. 3. Make sure that all capacitors, diodes and transistors are put in the right way round. Metal 'fan' type heatsinks are used on Q3 and Q5. Make sure that these are well away from any other component.

A heatsink is fitted between Q6 and Q7 (Fig. 4) and is insulated by mica washers. Note that the heatsink will be slightly skewed and the transistor slightly twisted so that the heatsink can be bolted on to the 'metal side' of the transistors. Remember that insulating washers must be used.

The printed circuit board will be mounted onto the lid of the die-cast metal box and short connecting leads will be used to connect the board to the output transistors which are mounted on the outside face of this lid.

HOW IT WORKS

The amplifier is of conventional design using a quasi-complementary symmetry, output stage and a differential input stage.

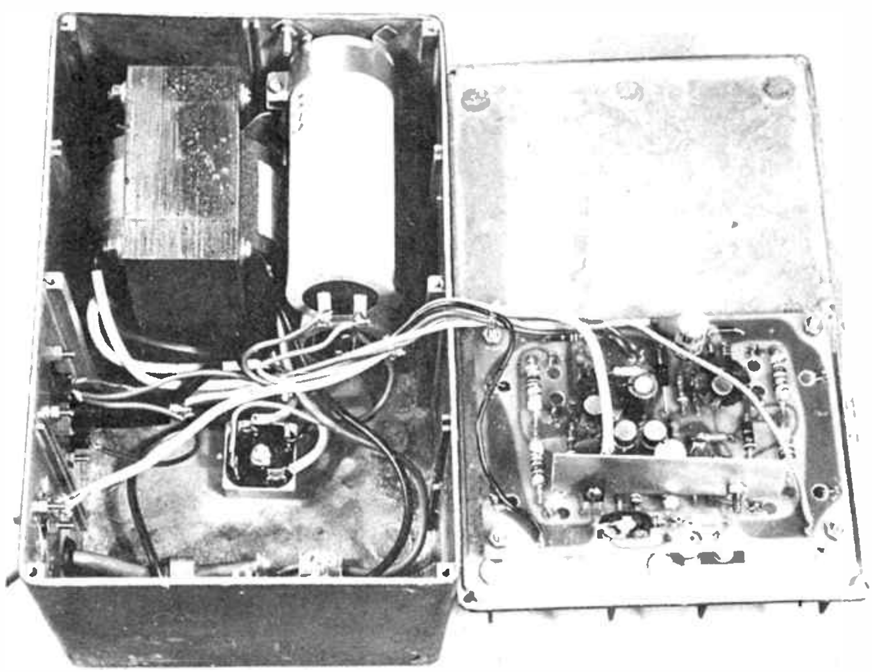
Output transistors are paralleled for greater output capacity – and transistors Q6 and Q7 connected in a Darlington configuration provide current gain.

Q3 is a current regulator supplying approximately 10 mA. This controlled current passes through Q4, thus setting the bias for the output stage, and Q5. The voltage at the collector of Q5 is set by its own base-emitter voltage. Since this transistor is working with an almost constant current in its collector it has a very high voltage gain. This gain is attenuated at high frequencies by C7.

Transistor Q5 is controlled by the differential pair Q1 and Q2. Due to the negative feedback via R7 & R9, the action of Q1 and Q2 is that of an error amplifier. Thus it tries to keep the voltage at its two inputs (the bases of Q1 and Q2) constant. Because of this action, the output voltage is held equal to the input voltage multiplied by (R9+R7)/R7. This gives the amplifier a voltage gain of approximately 22. This gain may be changed by varying the value of R7. An appropriate change must then also be made to C6 as R7/C6 determine the lower –3dB point. The value of R9 should not be altered.

The output bias current – which is necessary to prevent cross-over distortion – is set by RV1.

PROJECT 413



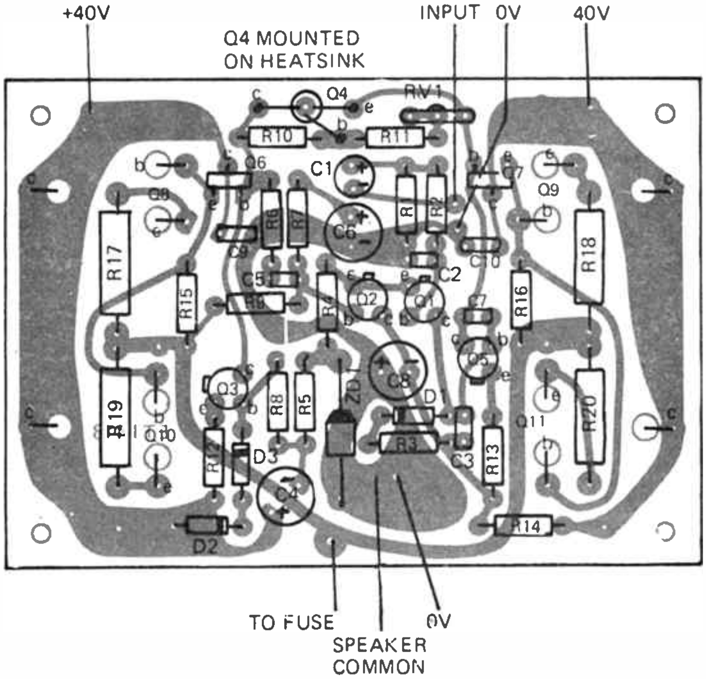


Fig. 3. How the components are mounted on the printed circuit board.

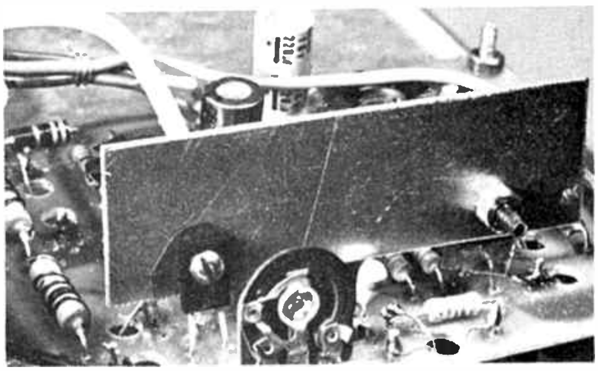


Fig. 4 A heat sink (detailed in Fig. 5) locates Q6 and Q7. In this illustration, Q7 can be seen just to left of the potentiometer.

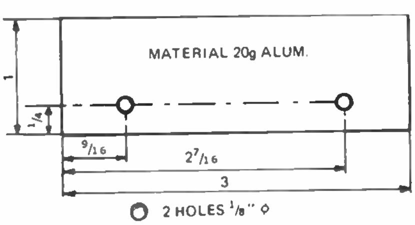
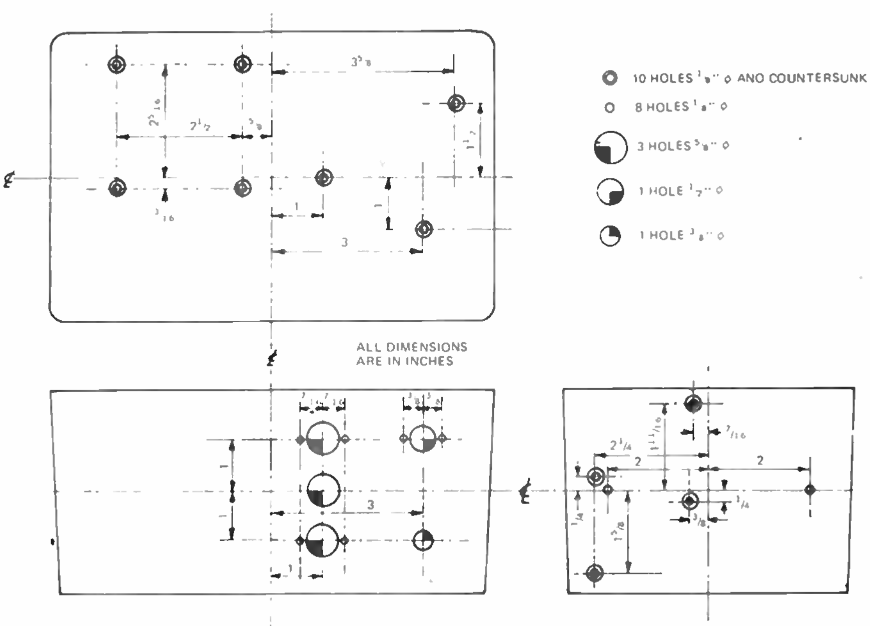


Fig. 5 Details of heat sink for Q6 and Q7.

Drilling details for STC die-cast box 8¾" x 5¾" x 4"



Countersunk screws and spacers are used to ensure that the printed circuit board stands well clear of inner face of the lid. These should be installed at this stage – but do not yet attach the board itself.

The heatsink for Q4 should be attached to the lid using a countersunk screw and insulating washers. The heatsinks for the output transistors, and the output transistors, should now be installed. Make quite sure that the correct transistors are in the right places. Insulating washers must again be used.

Connect short leads to the emitter, base and collector of the output transistors (the connection to the collectors is made via the transistor mounting screw)'

Press transistor Q4 into its heatsink.

Install metal connecting pins in the printed circuit board for terminating connections to the output transistors Q8, Q9, Q10, and Q11. Pins are also required for Q4. The pin positions are clearly marked on the printed circuit board overlay.

Now connect all leads from the power supply etc, to the printed circuit board and then fit the board over the leads from the output transistors and screw firmly into place.

Solder the leads from the various external connections to the appropriate pins on the board. Do not wrap the wire around the pins by more than half a turn as it will otherwise be very difficult to remove later (if necessary).

Install and connect all remaining components.

Ensure that the mains earth lead is securely attached to the case as must also be the transformer shield. The input shield should be earthed to the case at the input socket.

Fig. 6. Lid of the die-cast box showing heat sinks (and output transistor leads). Transistor Q4 and its associated heat sink is clearly visible.

100W GUITAR AMPLIFIER

Now carefully check out all connections — ensure that there are no loose ends of wire laying around inside the case.

The unit is now ready for testing.

TESTING PROCEDURE

A multimeter capable of measuring 100mA d.c. is required. Insert the meter in series with the +40V supply and rotate trimpot RV1 so that the wiper is nearest Q4 (i.e. maximum resistance). Switch the unit on and adjust RV1 until a reading of 65 mA is obtained. Allow the amplifier to warm up for about five minutes and then readjust the output current to 70–80 mA. (Note — the current will increase as the unit warms up). Switch the unit off and reconnect the positive power lead to the pc board.

Switch the multimeter to the volts range and check the voltage between the outputs and OV. It should be within 200 mV of zero (either polarity).

If both measurements are correct the amplifier is ready for use. Switch off and disconnect the multimeter.

Connect a loudspeaker to the output and again switch on — no sound should be heard from the speaker.

PREAMPLIFIER

The preamplifier used to drive this unit must be capable of producing approximately 1 volt into 3.9k. ●

If the amplifier is not made using the construction shown the following should be observed.

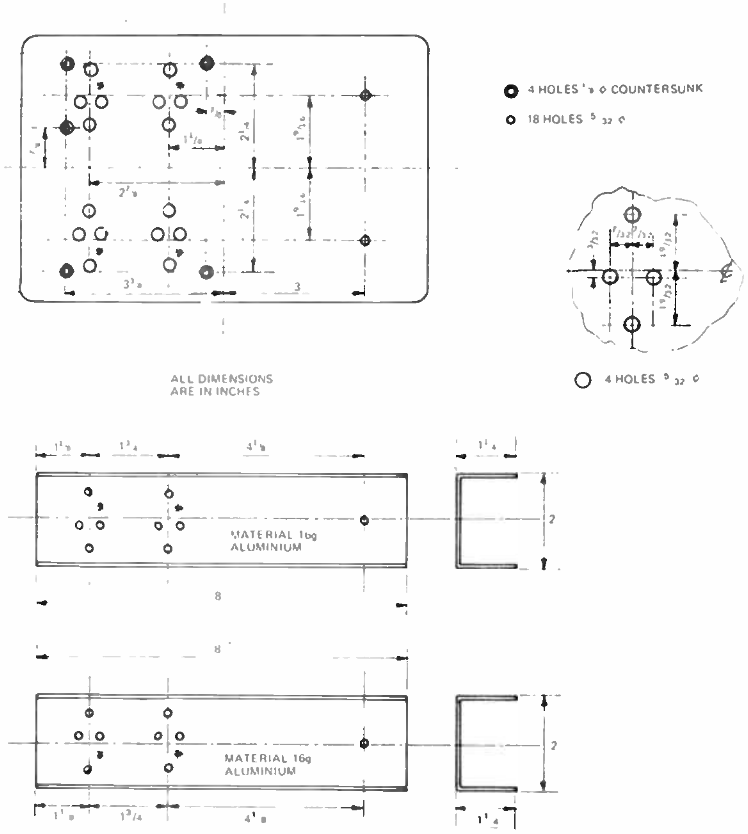
Q4 is used for temperature stabilization and MUST be mounted on, and insulated from, the main heatsink.

Failure to do this can destroy the output transistors and the transformer.

The leads to the output transistor should be as short as practicable.

SPECIFICATIONS

OUTPUT POWER (at 1% distortion)	
4 ohm load	100 watts
8 ohm load	60 watts
FREQUENCY RESPONSE	
50 Hz – 20 kHz	±1 dB
20 Hz – 150 kHz	±3 dB
TOTAL HARMONIC DISTORTION	
0.1 watts – 80 watts into 4 ohms	<0.5% typically 0.2%
DAMPING FACTOR	
4 ohm load	80
INPUT IMPEDANCE	≏3.9 k



Drilling details of STC box lid and associated heat sinks

PARTS LIST: ETI 413

R1	Resistor	3.9k	½W	5%	
R2	"	820 ohm	½W	5%	
R3	"	220 "	½W	5%	
R4	"	1.5k	½W	5%	
R5	"	470 ohm	½W	5%	
R6	"	220 "	½W	5%	
R7	"	220 "	½W	5%	
R8	"	470 "	½W	5%	
R9	"	4.7k	½W	5%	
R10	"	2.2k	½W	5%	
R11	"	820 ohm	½W	5%	
R12	"	68 ohm	½W	5%	
R13	"	100 "	½W	5%	
R14	"	33 "	½W	5%	
R15	"	47 "	½W	5%	
R16	"	47 "	½W	5%	
R17	"	0.25 "	2W	5%	ASW2
R18	"	0.25 "	2W	5%	"
R19	"	0.25 "	2W	5%	"
R20	"	0.25 "	2W	5%	"
RV1	Variable Resistor	1000 ohm Large type Trimpot			
C1	Capacitor	4.7µF 25V TAG tantalum			
C2	"	390pF ceramic			
C3	"	0.0015 µF polyester			
C4	"	220 µF 16V electrolytic (PC type)			
C5	"	39pF ceramic			
C6	"	220 µF 16V electrolytic (PC type)			
C7	"	33 pF ceramic			
C8	"	47 µF 50V electrolytic (PC type)			
C9	"	0.1 µF polyester			
C10	"	0.1 µF "			
C11	"	4000 µF 75 V electrolytic			
C12	"	4000 µF 75 V electrolytic			
D1	Diode	EM401, IN 4005 or similar			
D2	"	" " "			
D3	"	" " "			
ZD1	Zener diode	BZX70 C15			
DB1	Bridge Rectifier	PA 40			
Q1	Transistor	TT 800			
Q2	"	TT 800			
Q3	"	TT 800			
Q4	"	TT 801			
Q5	"	TT 801			
Q6	"	BD 139			
Q7	"	BD 140			
Q8	"	AY 8149, 2N 3055			
Q9	"	AY 9149, MJ 2955			
Q10	"	AY 8149, 2N 3055			
Q11	"	AY 9149, MJ 2955			
T1	Transformer	56V CT @ 1.5A (or Ferguson type PF3577)			

PC Board ETI 413
3A Fuse
Fuse holder
Die cast box 5¾ x 8¾ x 4"
2 Heat Sinks
1 " "
1 RCA socket
Output socket (2 pin din etc.)
4 spacers, ½" long for P.C. board.
1 three core flex and plug
2 Fan top heat sink for Q3 and Q5
Heat sink for Q4 (chassis mounting type, transistor vertical to chassis)
Screws etc.

HOW IT WORKS

The input signals are attenuated as required by RV1-RV4 before being summed by amplifier IC1. The gain of IC1 is determined by the value of the resistor in series with each input. The value of this resistor must be at least five times the value of the input potentiometer otherwise input impedance will change with variations in the potentiometer setting.

For dynamic or electret microphones (250 or 600 ohm) a 1 k potentiometer and a 22 k series resistor will provide full output with an input of 2 millivolts. For guitar inputs a 47 k potentiometer and a 220 k series resistor will provide full output with 20 millivolts input and an input impedance of 47 k.

Amplifier IC1 is followed by conventional tone controls and a further amplifier IC2 in which RV7 alters the gain and hence is used as the master volume control. This stage is configured as a positive, or in phase, amplifier in which a gain of less than one cannot be obtained. However the gain variation of 37 db will be found adequate for most purposes.

Two different power supplies are described. The first is a separate mains supply (where the unit is to be built into a separate box) which is simply a 12.6 volt CT transformer rectified to provide ±9 volts for the ICs. The second is used where the unit is built into the 100 watt amplifier and derives its supply by resistively dividing down the power amplifier supplies.

MIXER PREAMPLIFIER

Simple yet effective unit is specifically intended for use with our 100 W guitar amplifier.

OUR 100 watt guitar amplifier, ETI 413, has proven to be extraordinarily successful. A very large number have been built, and are in use in conjunction with the 8 channel master mixer for which it was specifically designed.

There has also proven to be a large demand, as evidenced by letters to kitset suppliers and to ourselves, for a simple pre-amplifier to be used with the guitar amplifier. This project describes such a preamplifier, which may be built as a separate unit, or within the 100 watt amplifier as desired.

The basic preamplifier may have up to four inputs, each with separate volume control, and the sensitivity and input impedance of each can be tailored to suit individual requirements. The inputs are mixed in a summing operational amplifier and the combined signal is then operated on by a common set of bass and treble controls.

A master volume control is provided so that the level of the combined signal may be varied. Although specifically designed for the 100 watt amplifier, this unit is very flexible and may be used as a separate general purpose mixer/preamplifier.

PARTS LIST

R1	resistor)		
R2	")See text		
R3	")and Table I		
R4	")		
R5	"	470 k	½ Watt 5%
R6	"	2.7 k	" " "
R7	"	12 k	" " "
R8	"	1.8 k	" " "
R9	"	3.3 k	" " "
R10	"	470	" " "
R11 *	"	2.7 k	" " "
R12 *	"	1 k	" " "
R13 *	"	1 k	" " "
R14 *	"	2.7 k	" " "

* dc powered version only

RV1	potentiometer)	See text
RV2	")	and Table
RV3	")	1
RV4	")	
RV5	"	100k log
RV6	"	100k log
RV7	"	250k log
C1	capacitor)	
C2	") See text	
C3	") and Table I	
C5	33 pF ceramic	
C6	0.0022µF polyester	
C7	0.022µF "	
C8	0.022µF "	
C9	0.22µF "	
C10	33 pF ceramic	
C11	1µF 25V tag tantalum	
C12	47µF 16V electrolytic PC mount	
C13	47µF 16V " " "	
IC1	integrated circuit LM301A mini-dip or T03	
IC2	" " LM301A mini-dip or T03	

PC board ETI 419
* Transformer A & R 6474 or similar
* D1-D4 diode EM401 or similar
* 240 volt ac version only
Plugs, sockets, knobs to suit individual requirements.

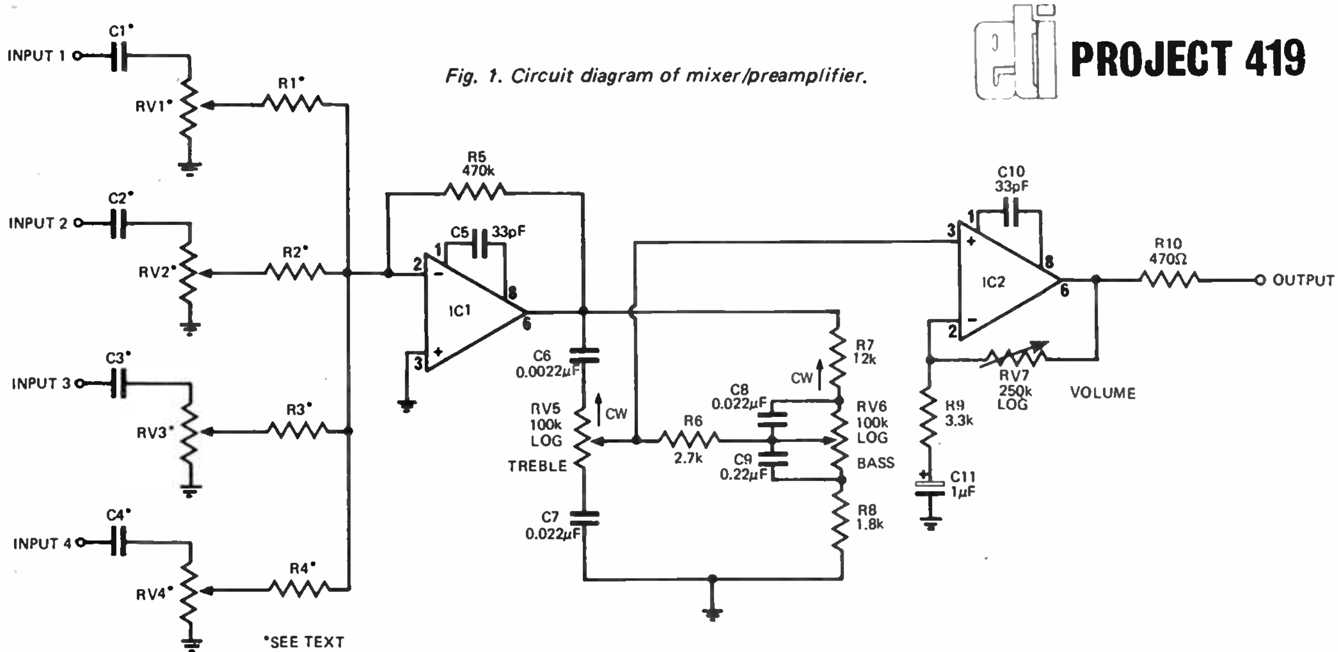


Fig. 1. Circuit diagram of mixer/preamplifier.

CONSTRUCTION

We mounted our prototype unit within the existing 100 watt amplifier: This is a simple and neat method of housing the unit, but has a limitation in that there is only room for two sets of input channel controls – so if this arrangement is used, R3 and R4 should be omitted from the printed circuit board.

Apart from limitation of space, due to the proximity of the power transformer, mounting the unit within the main amplifier increases the possibility of hum pickup. This may be minimized by using twisted leads or shielded cables for wiring to the controls and using a single earth only for the preamplifier, namely the shield of the input cable to the main amplifier. In addition the input sockets should be of the insulated type to prevent earth loops. If the insulated types are unavailable, then standard types may be mounted on a piece of bakelite or fibreglass board. These precautions will enable a hum level of −65 dB to be obtained.

If the unit is constructed in a separate box with its own power supply a much lower hum level should be realised. In this case the resistors shown in Fig. 3 are omitted from the board and the power supply of Fig. 2 should be used. In the latter case diodes D1 – D4 are mounted on the board.

Mount the components to the PC board in accordance with the overlay applicable to the ac or dc version as required. Ensure that ICs and capacitors are correctly orientated.

Select values for the input components from Table I depending on your individual requirements. We used a 1 k microphone input and a 47 k guitar input on our prototype.

SPECIFICATION
(when used with ETI 413 amp)

Number of inputs	4
Input level	
Input impedance 1 k	2 mV
" " 47 k	20 mV
Distortion at 80 watts	< 0.5%
Tone control range	
bass	± 10 dB at 100 Hz
treble	± 10 dB at 10 kHz
Master volume control range	37 dB
Hum level referred to 100 watt output	−65 dB
Noise level (excluding hum) referred to 100 watt output	−75 dB

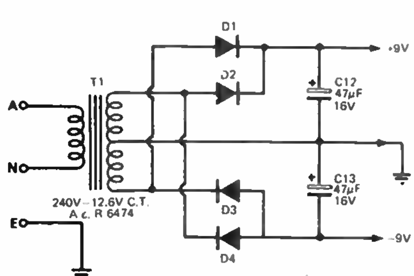


Fig. 2. Power supply for preamplifier as a separate unit.

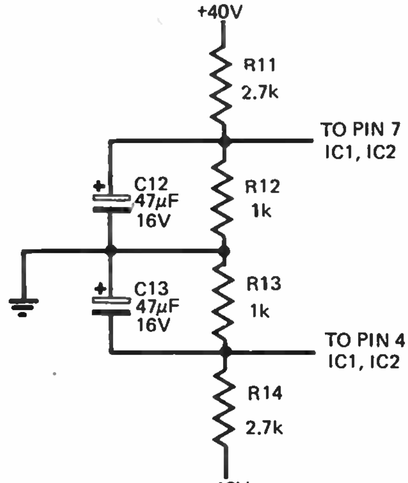
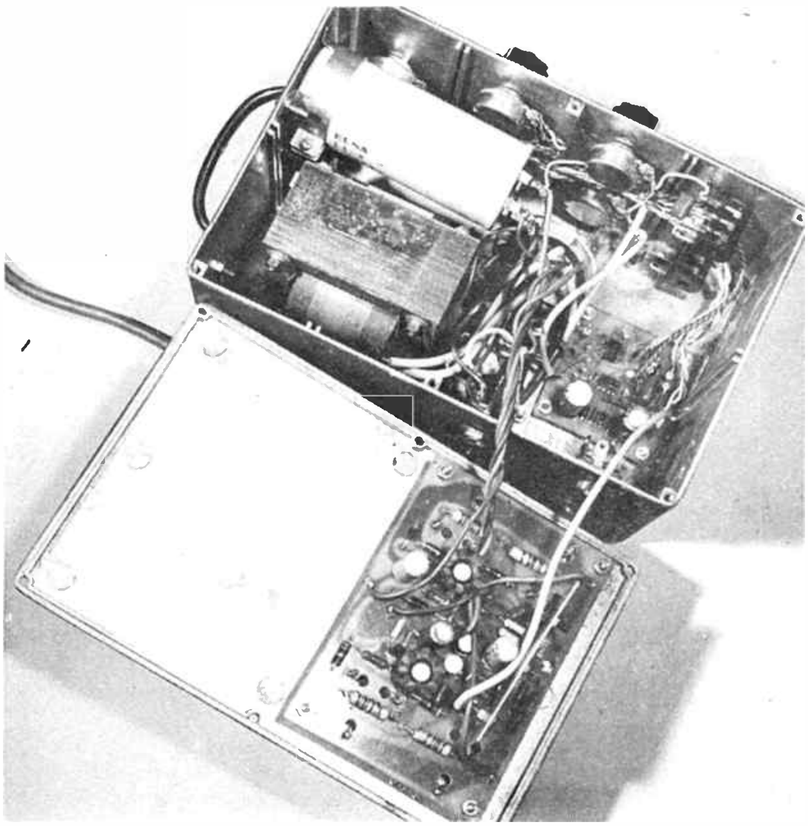


Fig. 3. Divider network; resistors as shown are fitted to PC board if 100 watt amplifier power supply is used.

MIXER/PREAMPLIFIER

TABLE I				
APPLICATION	RV1	R1	C1	SENSITIVITY
Microphones (600 or 250 ohms)	1 k log	22 k	4.7μF	2 mV
Microphones or guitars (47 k)	47 k log	220 k	0.1μF	20 mV
Crystal microphones, line inputs, ceramic pickup etc	1 M log	2.2 M	0.1μF	200 mV



◀ *The preamplifier is mounted in the bottom of the existing power amplifier and the controls and input sockets on the right-hand side.*

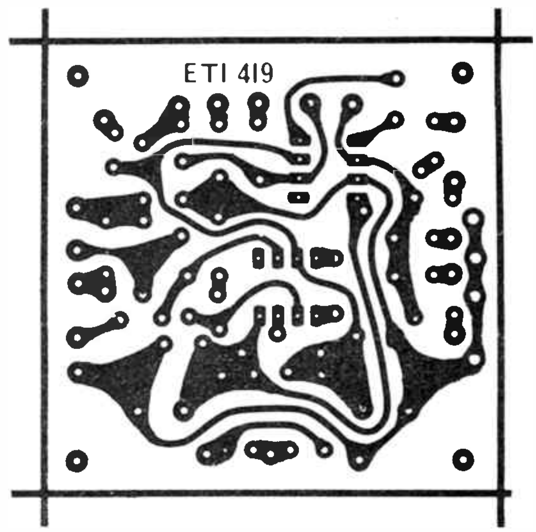
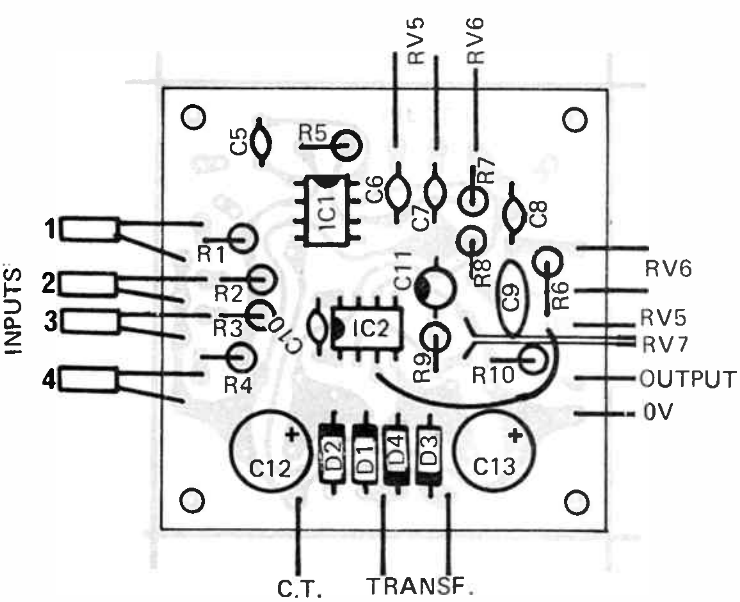


Fig. 4. PC board layout (both versions).



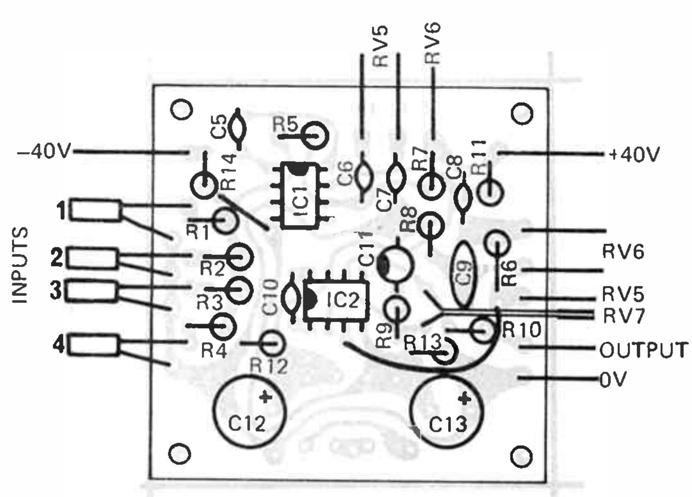
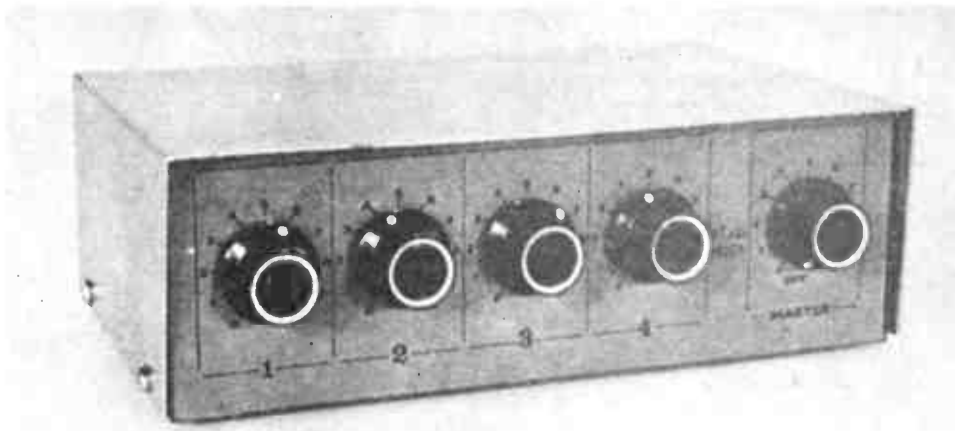


Fig. 5, Bottom left: Component overlay for ac powered preamplifier

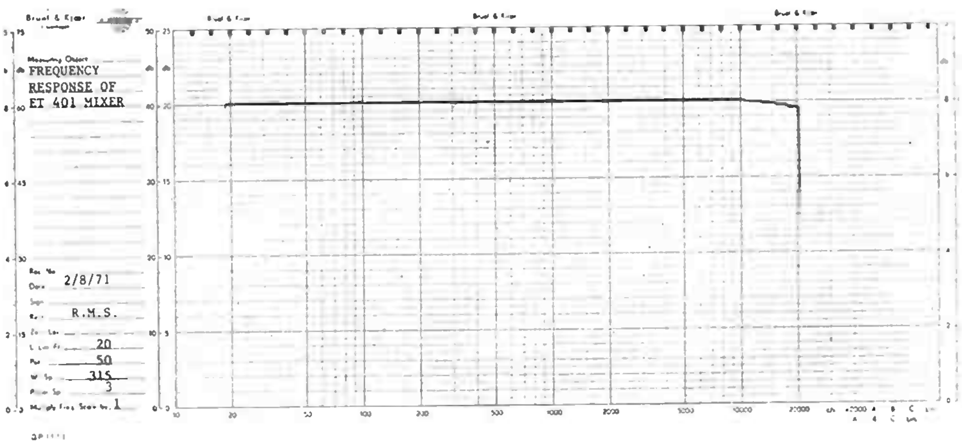
Fig. 6, Bottom right: Component overlay for preamplifier powered from 100 watt amplifier supply.

ETI PROJECT 401



FET FOUR-INPUT MIXER

Mix any combination of four audio signals with this easily constructed unit.



THREE guitars and a microphone. One record player, two tape decks and an electronic organ. Or any combination you like of two, three or four separate audio signals can be smoothly blended together by using this simply constructed Input Mixer.

The unit can handle input levels from 2mV to 2V, input impedances from 4 ohms to 1 megohm. It provides a maximum gain of 20 dB, and has a frequency response that is absolutely flat from 20 Hz to 10 kHz, and is still within 1 dB at 20 kHz. The response curve is shown in Fig. 1.

Battery operation has ensured that internally generated noise is kept to a very low level, and the unit is suitable for all types of inputs with the possible exception of very high performance dynamic microphones with outputs of less than 2mV. Life expectancy of the batteries specified in the parts list is at least 100 hours of continuous use.

CONSTRUCTION

The circuit diagram of the complete unit is shown in Fig. 2, the circuit board pattern is shown in Fig. 3, and the component layout in Fig. 4.

Make sure that the electrolytics are connected the right way round, and do not use excessive heat when soldering

Interior of completed unit.

The mixer consists of four identical input stages and one summing amplifier.

Each input stage consists of a variable attenuator and a FET source follower. The attenuators are one megohm potentiometers, and since the input impedance of a FET is very high, the input impedance of the mixer is that of the attenuator — i.e. 1M.

Input signals are coupled to the mixer via standard jack plugs and sockets or RCA connectors. Both types of input connector were fitted to each input of the prototype unit — each pair being wired in parallel. Provision for both types has been made on the sheet metalwork drawings and included in the parts list. Either type may be omitted if required.

The output impedance of the FET stage is approximately 1k, and the internal gain of the FET is unity.

The summing amplifier is a LM 301A operational amplifier. This has an open loop gain of around 100,000 and a cut off frequency of approximately 10 MHz. Gain control is varied by the feedback potentiometer RV5.

Output impedance of the mixer is less than 100 ohms. It is desirable that the load impedance should exceed 2k. The mixer is however, short circuit proof, and the only effect of excess load is distortion. The output is dc coupled, and the offset voltage (dc component) is typically 2mV.

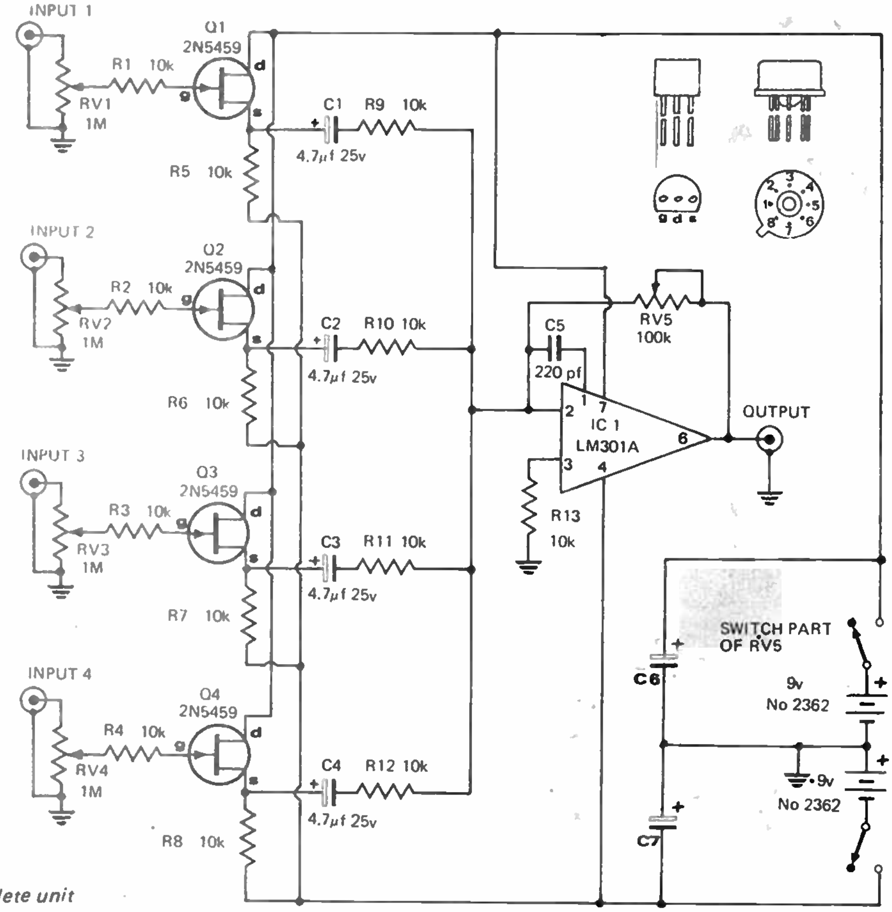
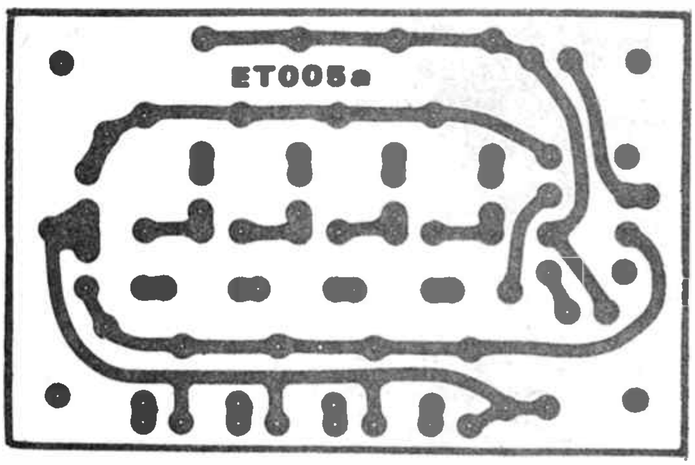


Fig. 2, Circuit diagram of complete unit



PARTS LIST

R1-R13	—	resistor 10k, ½W, 5%
RV1-RV4	—	potentiometer, 1 Megohm, logarithmic.
RV5	—	potentiometer, 100k, logarithmic, with double pole switch.
Q1-Q4	—	field effect transistors, 2N5459
C1-C4	—	Tag tantalum capacitors, 4.7μF.
C5	—	capacitor, 220μF
C6-C7	—	capacitor, pc board mounting, 100μF electrolytic.
IC1	—	integrated circuit LM 301A (National Semiconductor) metal can type, or μA 301A Fairchild metal can type.

Plus — 4 closed circuit jack sockets, McMurdo type 1291-06-01 or similar.
2 double type RCA sockets.
1 single type RCA socket
1 printed circuit board — ET 005a
2 nine volt batteries, Eveready type 2362 or equivalent.
1 metal case and cover.
4 spacers for pc board.
1 perspex front panel.
5 control knobs.
Coax cable, screws etc.

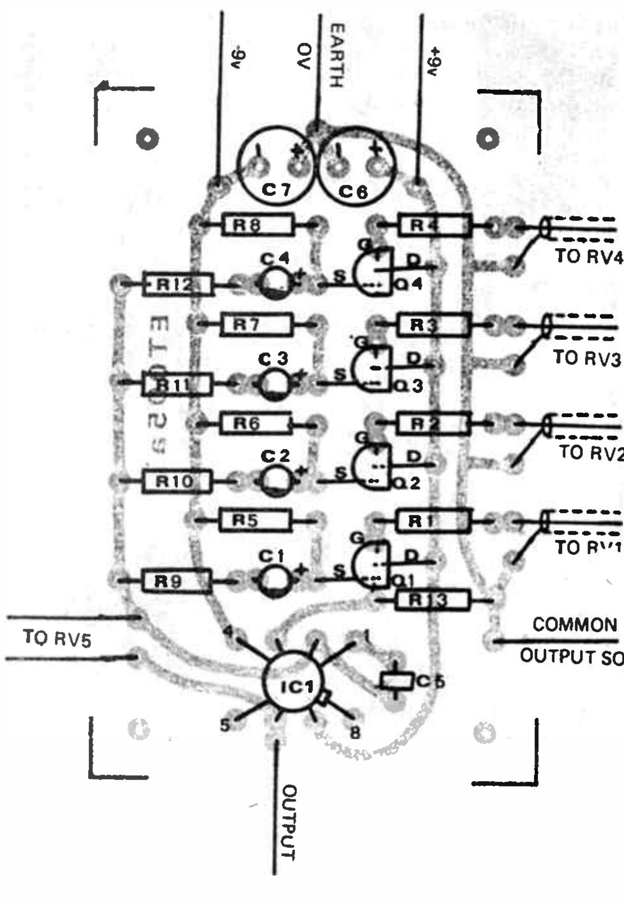


Fig. 4. How components are mounted on the printed circuit board — seen from components side.

FET FOUR-INPUT MIXER

the connections of the FETs and the I/C. Screened wire must be used for all leads from the input sockets to the potentiometers, and from the potentiometers to the printed circuit board.

Sheet metal drawings for the chassis and cover are shown in Figs. 5 and 6.

ARTWORK

Finished artwork for the front panel is reproduced on page 62. Cut the artwork around the outer edges, punch out the holes for the potentiometer bushes, and then clamp the artwork between the front metal panel of the chassis and a sheet of perspex (smoked perspex looks superb – if you can locate it). The complete assembly of chassis, artwork and perspex is sandwiched together by the potentiometer mounting nuts.

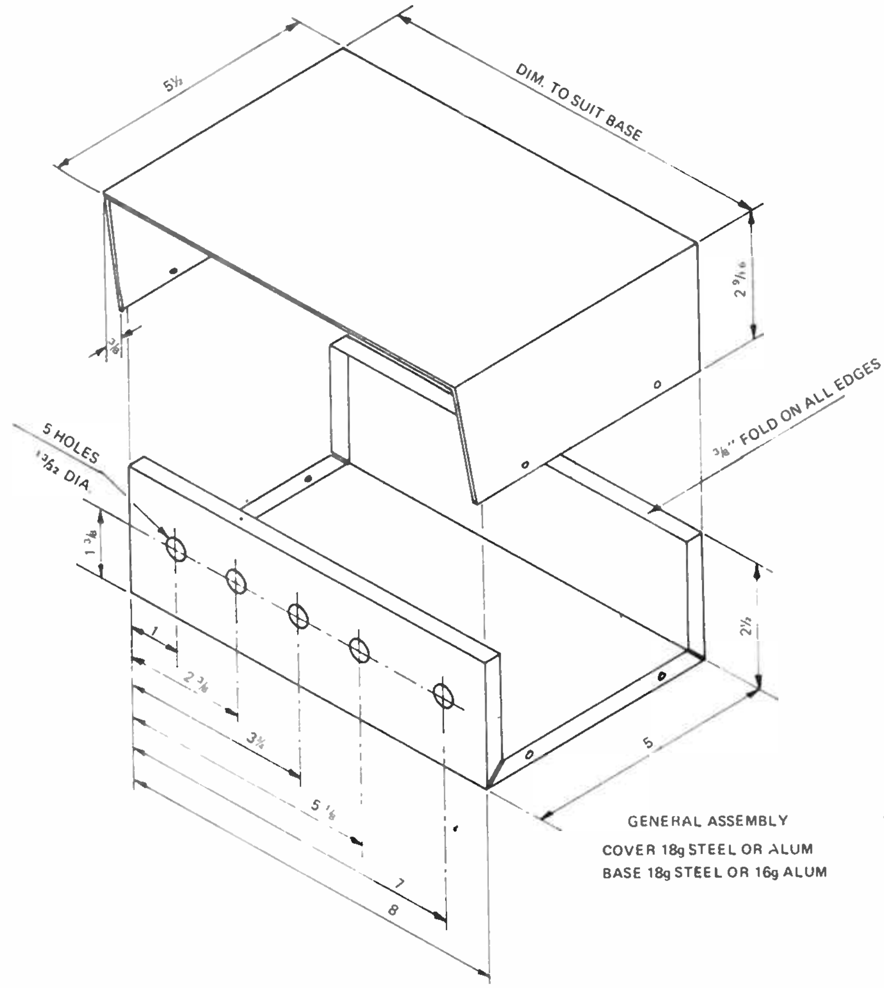
This will result in a very professional looking unit.

THE UNIT IN USE

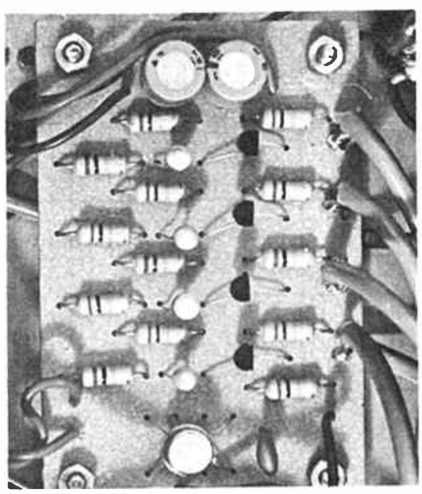
The mixer may be connected, via screened cable, to any tape recorder or amplifier.

Connect the required audio inputs – again via screened lead – and set all four channel input controls to zero. Set the master gain control to just beyond halfway. Adjust the level of your amplifier (or if you are using a tape recorder, to the normal recording level) and bring up the level of each signal input as required. Leave the level of all unused inputs at zero.

Adjust the overall sound level by using the master gain control on the mixer. ●



GENERAL ASSEMBLY
COVER 18g STEEL OR ALUM
BASE 18g STEEL OR 16g ALUM

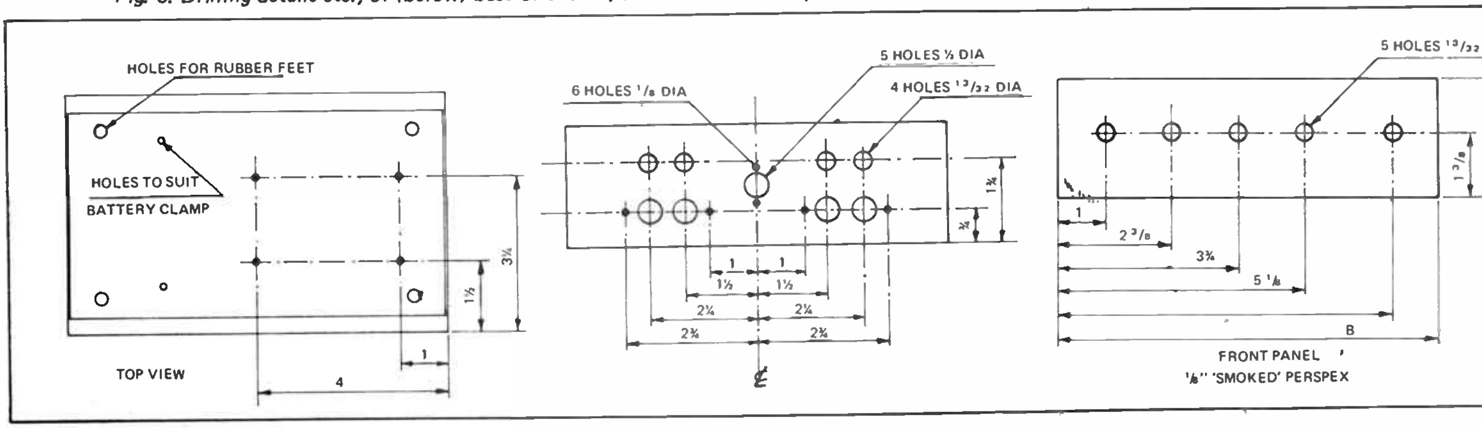


Compare this photograph of finished board with Fig. 4.

ET 401 MIXER

No. of inputs	–	4
Input impedance	–	1 Megohm
Input voltages	–	2mV - 2V
Gain - max.	–	20 dB
Max. output V.	–	3V rms
Output impedance		<100 ohms
Output load	–	>2000 ohms
Freq. response	–	within 1dB (20Hz to 20kHz)
Power supply	–	18V at 15ma
Batteries	–	2 x 2362
Battery life	–	100 hours approx.

Fig. 6. Drilling details etc., of (below) base of chassis, (below centre) rear panel of chassis, (below right) perspex cover.



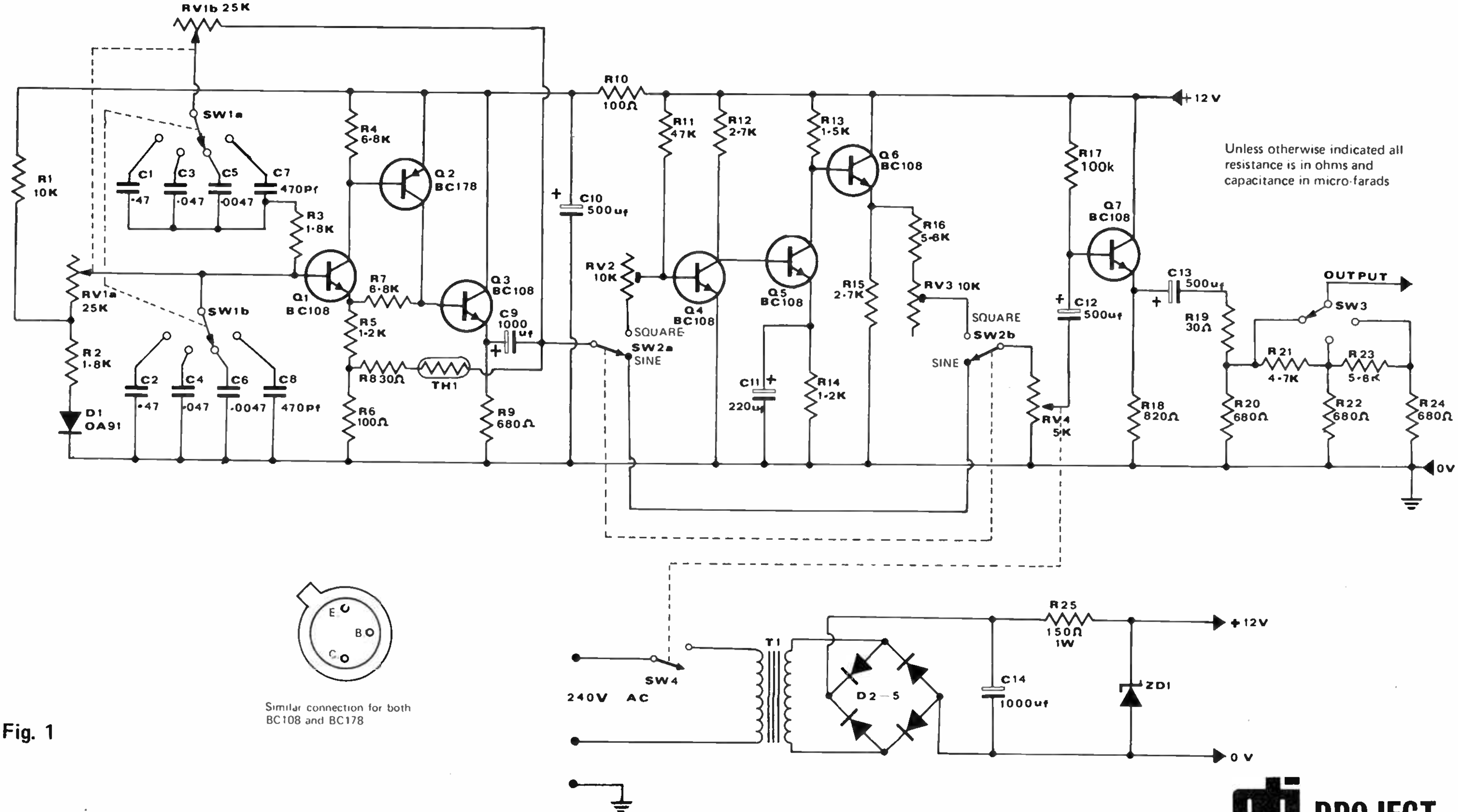


Fig. 1

ETI PROJECT 102

Audio Signal Generator

THIS ARTICLE gives all necessary data for constructing a reliable, stable audio signal generator which covers frequencies from 15Hz to 150kHz in four switch-selected ranges.

Both sine and square wave outputs are provided. Sine wave amplitude is one volt rms, square wave amplitude is one volt peak, adjustable by both fine and coarse attenuators in the emitter follower output stage.

The generator uses a total of seven silicon transistors, six of which are npn types and one a pnp type. The Wien bridge oscillator (Q1, Q2, Q3) is a slightly modified version of the well-known Mullard circuit which uses fixed capacitive elements and variable resistance elements in the bridge and includes a thermistor to ensure constant amplitude output. The modifications to the original Mullard Wien bridge circuit have been made to accommodate transistors readily available in Australia.

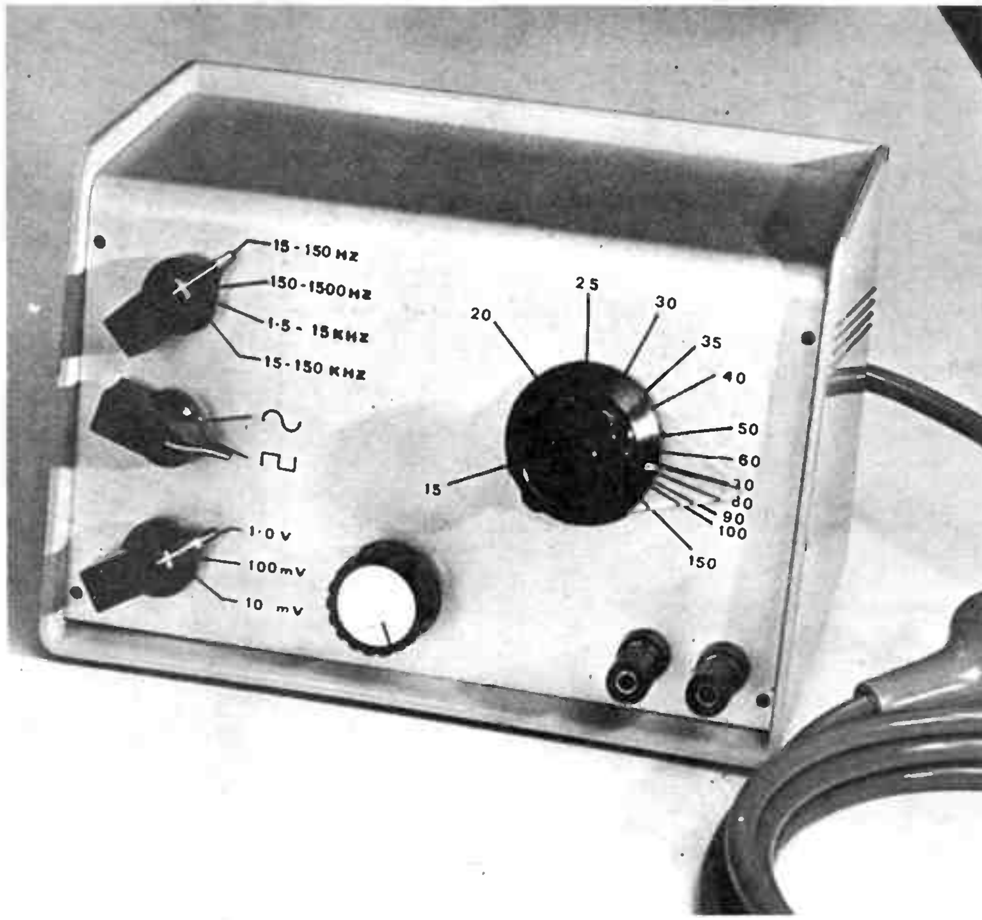
The full circuit is shown in Fig. 1.

Frequency is varied by the ganged potentiometers RV1a and RV1b; these form the resistive elements of the bridge. Constant amplitude output is ensured by the thermistor TH1 in the feedback loop to the emitter circuit of Q1.

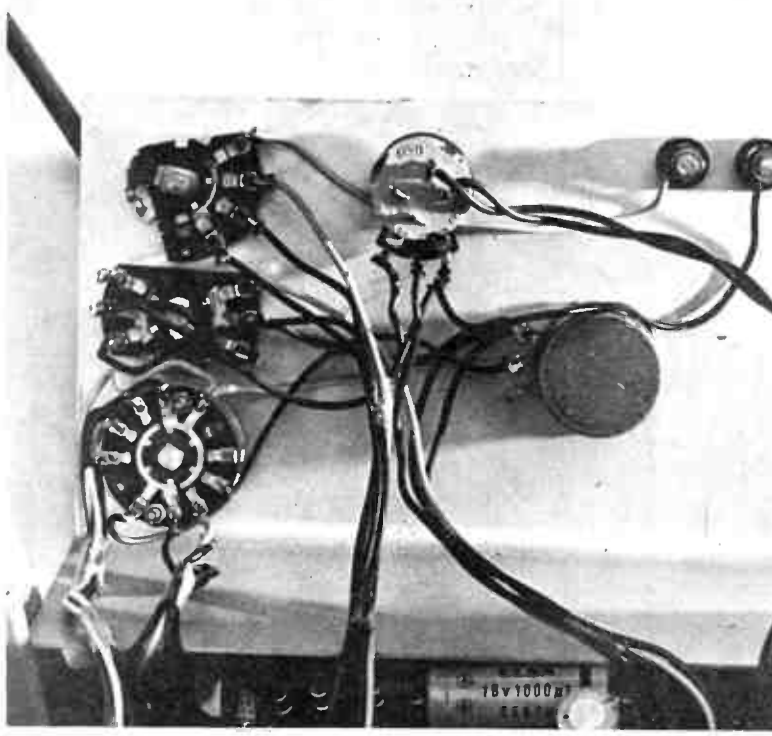
The npn/pnp pair Q1 and Q2 form a high gain amplifier which is coupled to the npn emitter follower Q3.

For sine-wave output the signals are taken via SW2a and SW2b, and the fine attenuator control RV4, to the base of emitter follower Q7. The output of Q7 then goes via the switched attenuation circuit (SW3, R21, R22, R23, R24) which can be used to adjust the sine-wave output to a maximum of 1 volt, 100 mV, or 10 mV rms.

For square-wave signals, the sine-wave output from Q3 is taken to Q4 and Q5, which together form a

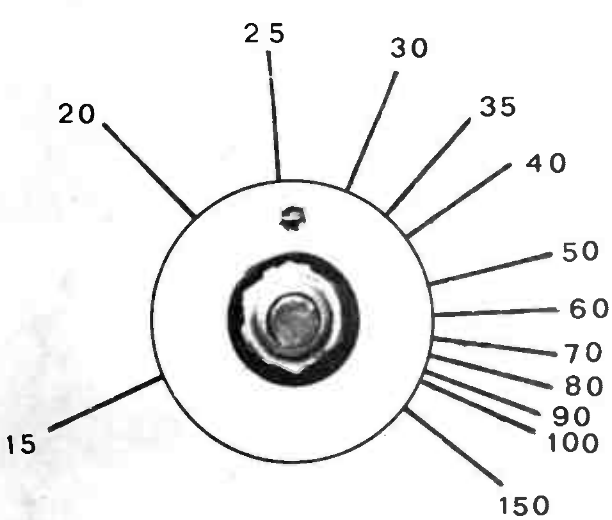


Our completed unit



Front panel wiring details.

Fig. 5. Calibration scale (full size)



Simply constructed audio signal generator provides adequate performance for home and shop use.

Audio Signal Generator

PARTS LIST

R1	–	Resistor 10k ½ watt 5%
R2	–	" 1.8k " " "
R3	–	" 1.8k " " "
R4	–	" 6.8k " " "
R5	–	" 1.2k " " "
R6	–	" 100Ω " " "
R7	–	" 6.8k " " "
R8	–	" 30Ω " " "
R9	–	" 680Ω " " "
R10	–	" 100Ω " " "
R11	–	" 47k " " "
R12	–	" 2.7k " " "
R13	–	" 1.5k " " "
R14	–	" 1.2k " " "
R15	–	" 2.7k " " "
R16	–	" 5.6k " " "
R17	–	" 100k " " "
R18	–	" 820Ω " " "
R19	–	" 30Ω " " "
R20	–	" 680Ω " " "
R21	–	" 4.7k " " "
R22	–	" 680Ω " " "
R23	–	" 5.6k " " "
R24	–	" 680Ω " " "
R25	–	" 150Ω 1 watt 10%
RV1 (a/b)	–	Dual gang potentiometer 25k linear
RV2	–	Potentiometer preset 10k linear
RV3	–	" " " "
RV4	–	Switch potentiometer 5k linear
C1	–	Capacitor – metallized 0.47 µf, 15 volt
C2	–	" – " 0.47 µf, " "
C3	–	" – " 0.047µf, " "
C4	–	" – " 0.047µf, " "
C5	–	" – " 0.0047µf, " "
C6	–	" – " 0.0047µf, " "
C7	–	Capacitor – metallized 470 pf, 15 volt
C8	–	" – " 470pf, " " (silver mica capacitors may be used for C7 and C8).
C9	–	Capacitor – electrolytic 1000 µf, 16 volt
C10	–	" – " 500 µf, " "
C11	–	" – " 220 µf, " "
C12	–	" – " 500 µf, " "
C13	–	" – " 500 µf, " "
(Note C10–C13 are single ended type capacitors)		
C14	–	" – " 1000 µf, " "
D1	–	diode – OA91 (or equivalent)
D2-5	–	diodes EM401 (or equivalent)
ZD-1	–	zener diode BZY96, 12v., 1 watt (or similar)
Q1	–	Transistor BC108
Q2	–	" BC178
Q3–Q7	–	Transistors BC108
TH1	–	Thermistor type STC type R53
T1	–	Transformer 240 volt to 12 volt 150 ma (Ferguson type PF 2851 or similar)
SW1 (a/b)	–	Switch rotary, two pole, four way
SW2 (a/b)	–	Switch rotary, two pole, two way
SW3	–	Switch rotary, single pole, three way
SW4	–	(rear of switch pot. RV4)
PC ET-006	–	Printed circuit board
Case	–	ATC type plastic case
Various	–	Output terminals, connecting block, mains cable and plug, control knobs.

Errata: Capacitors C1,C2,C3,C4,C5,C6,C7 and C8 should be 100 volts not 15 volts as shown above.

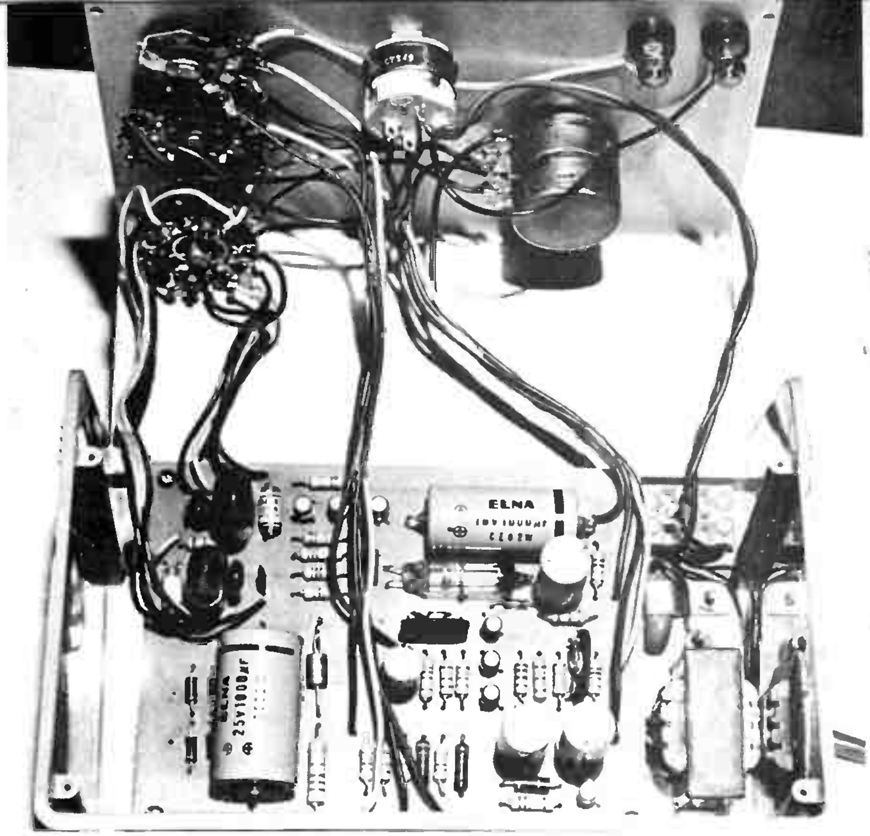
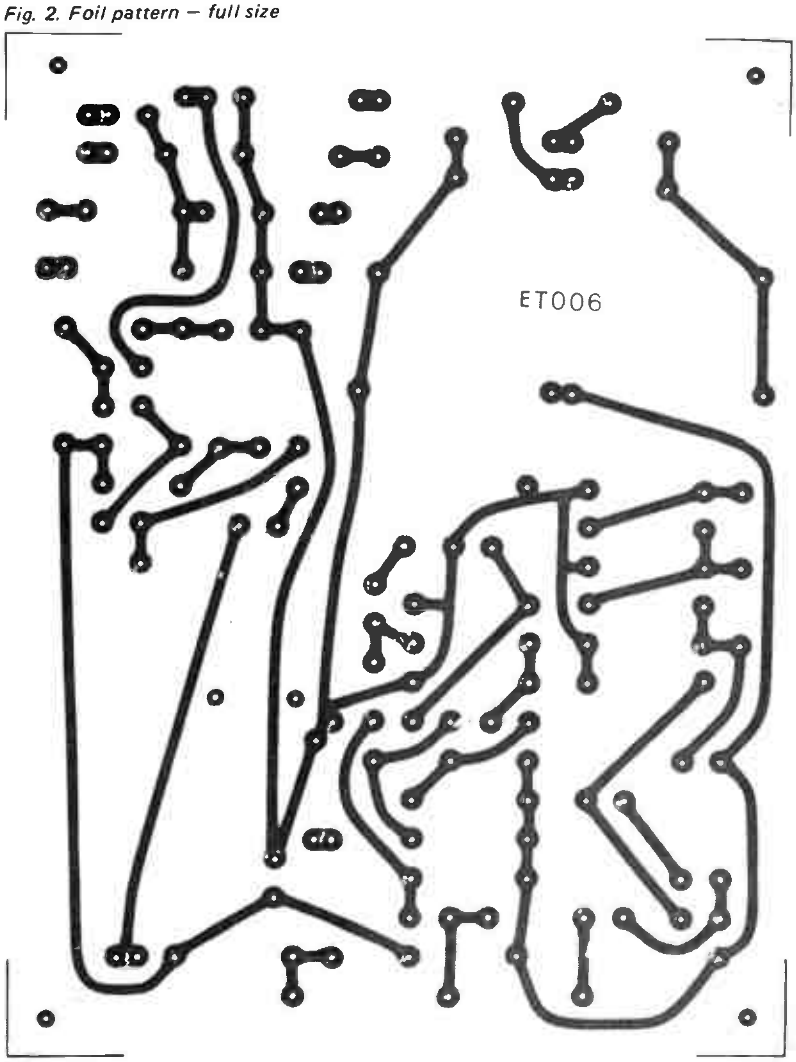


Fig. 4. General layout of components, in our final version we found it was desirable to use screened signal leads between the circuit board and the panel mounted switches.

Fig. 2. Foil pattern – full size



Audio Signal Generator

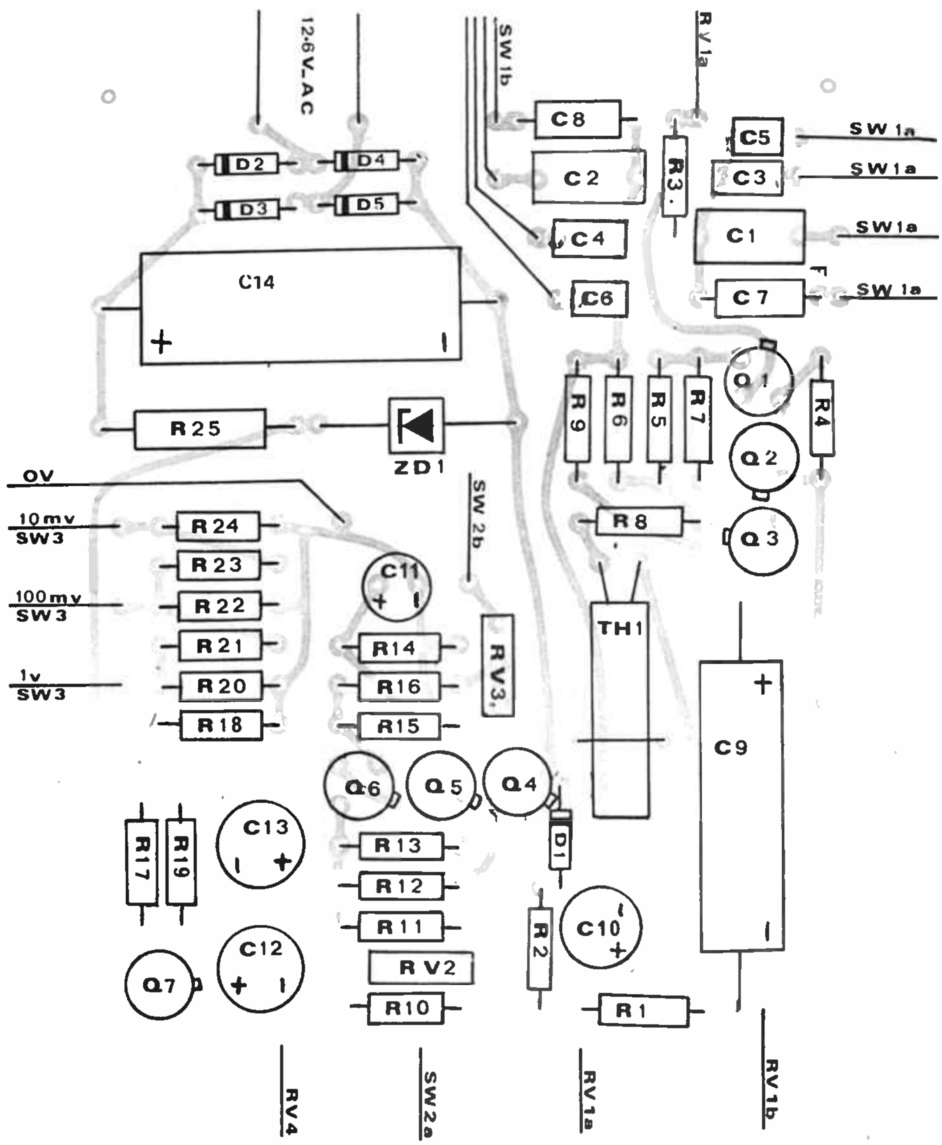
Frequency range 1:	15Hz – 150Hz
Frequency range 2:	150Hz – 1,500Hz
Frequency range 3:	1,500 Hz – 15,000Hz
Frequency range 4:	15,000Hz – 150kHz
(range 4 will in fact extend beyond 150kHz)	
Output variation:	Less than ±1dB from 15Hz – 150kHz.
Distortion:	<1%
Output impedance:	600 ohms.
Sine-wave output:	0-1 volt (rms) 0-100mV (rms) 0-10mV (rms)
Square-wave output:	0-2 volts (peak-to-peak) 0-200mV (peak-to-peak) 0-20mV (peak-to-peak)
Square-wave mark/space ratio:	nominally 1:1
Square-wave rise time:	less than 1 μsec.

squaring amplifier. The pre-set potentiometer RV2 is used to set the mark-space ratio to 1:1. The squared signal is then taken from the emitter follower Q6, through pre-set attenuator RV3, and then to the output transistor Q7.

The preset potentiometer RV3 is adjusted to produce a maximum square-wave output of 2 volts peak-to-peak. The coarse output attenuator will reduce this to either 200mV or 20mV peak-to-peak. Each of three output levels (both sine-wave and square-wave) is then steplessly variable from zero to maximum by means of RV4.

A circuit for a 12 volt power supply has been included in Fig. 1. However, if the generator is used infrequently or for short periods, it can be operated from a 12 volt battery. Current drain is less than 30mA.

Our prototype unit was built on a printed circuit board the foil pattern of which is reproduced in Fig. 2. The layout of components is shown in Fig. 3; compare this with Fig. 4, which is a photograph of the completed unit.



If desired, the unit can be assembled on Veroboard. The layout of components is not too critical, although we have found that it is necessary to use screened leads from the oscillator board to the output level switch, and from the output level switch to the output terminals.

It is essential that, metallized capacitors be used for C1, C2, C3, C4, C5 and C6. Silver mica capacitors may be used for C7 and C8 (470pF). DO NOT use ceramic capacitors.

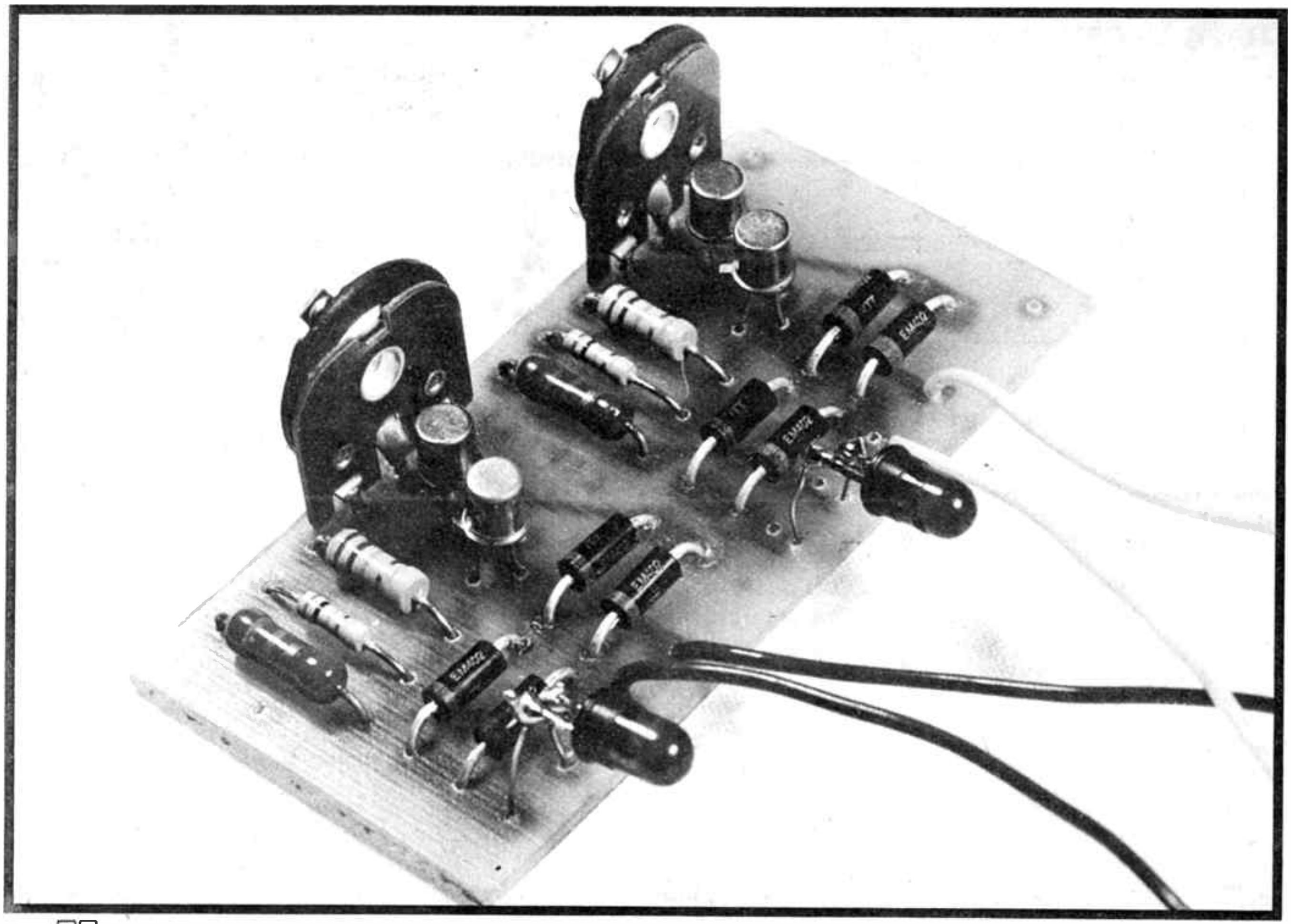
The frequency range scale illustrated in Fig. 5 will prove accurate enough for most audio work. If greater accuracy is required, the unit must be checked against another oscillator of known accuracy.

The output waveform should, if possible, be checked with an oscilloscope to verify that a good sine-wave is being obtained and that the square-wave is uniform. The mark-space ratio of the square-wave output should be set to 1:1 by the pre-set potentiometer RV2 and the peak-to-peak output level by RV3. If no oscilloscope is available, RV2 and RV3 should be set to mid-range.

If an audio frequency voltmeter is available, the sine-wave output should be checked from 15Hz to about 100kHz to ensure that reasonably constant output is maintained.

The output signal should remain constant within ±1dB up to 100kHz or so, but there may be changes in level whilst changing frequency. The envelope stability (or the time taken for the amplitude to stabilize after a change in frequency) is mainly a function of the quality of the dual ganged potentiometer.

Performance of the prototype unit is detailed above. ●



PROJECT 417

THE OVER-LED

Is your power amplifier clipping? This simple monitor lets you know.

TABLE 1

RMS watts per channel	SPEAKER IMPEDANCE 4Ω R1	4Ω R3	8Ω R1	8Ω R3	16Ω R1	16Ω R3
5	68	5.6k	82	8.2k	120	12k
10	82	8.2k	120	10k	180	18k
15	100	10k	150	15k	220	22k
20	120	12k	180	18k	240	24k
25	150	15k	220	22k	270	27k
35	180	18k	240	24k	330	33k
50	220	22k	270	27k	390	39k
75	240	24k	330	33k	470	47k
100	270	27k	390	39k	560	56k

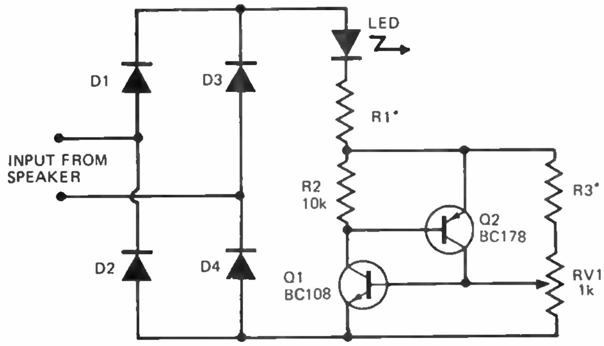
MANY people are aware of distortion when they turn up the volume control on their hi-fi equipment — but are usually unaware of the cause.

Nine times out of ten this distortion is caused by 'clipping'. That is, the amplifier does not have enough reserve power to handle the peak music transients at the required volume.

During such peaks, the amplifier is driven into an overload condition and as a result the music peaks are 'clipped'. This results in harsh sounding reproduction.

This simple device, which may be built into your existing amplifier, or separately located, flashes a warning light if the power level at which clipping occurs is exceeded.

Two completely independent circuits are provided so that each channel of a stereo system may be monitored separately.



*SEE TABLE 1 FOR VALUES

ONE CHANNEL ONLY SHOWN

Fig. 1. Circuit diagram of overload detector. One channel only shown.

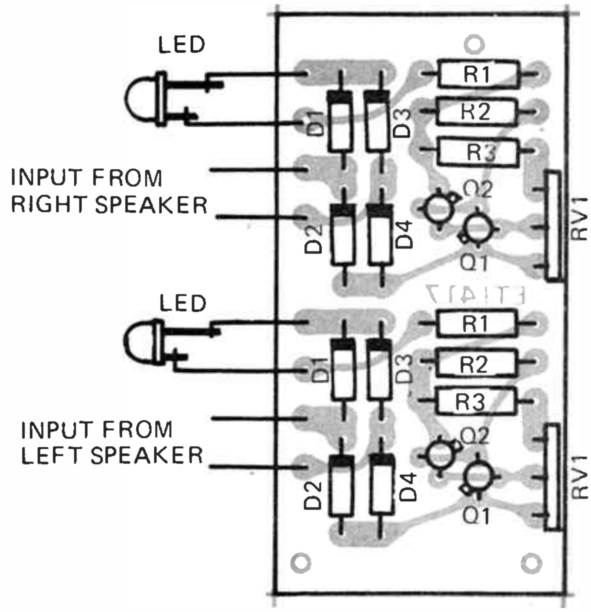


Fig. 2. Component overlay.

HOW IT WORKS

The output of each power-amplifier channel is monitored at the speaker terminals. The output is bridge rectified by D1-D4 so that both positive and negative transients may be detected.

Transistors Q1 and Q2 (together) are equivalent to a sensitive gate SCR (silicon controlled rectifier). If the voltage at the base of Q2 is more than about 0.6 volts above its emitter, Q1 and Q2 will each turn hard on and latch on, until the current through them drops to zero.

When transistors, Q1 and Q2 are on, the current flowing through them also flows through the LED causing it to illuminate. Resistor R1 limits the peak current through the LED to about 100 mA. The range of calibration potentiometer RV1 is set by resistor R3. The values of R1 and R3 are provided in Table 1 for various amplifier power ratings and speaker impedances. These values are not critical. If your amplifier has a power rating other than that specified, the nearest values will do.

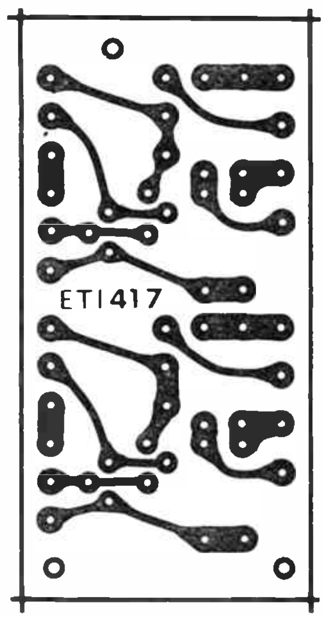


Fig. 3. Printed circuit board (full size).

PARTS LIST

One set required for each channel.

R1	resistor	as Table 1
R2	"	10k 5% ½ watt
R3	"	as Table 1
Q1	transistor	BC108 or similar
Q2	"	BC178 or similar
D1 to D4	Diode	EM401 or similar

LED light emitting diode FLV110, MV5025 or HP4403 etc.

PCB ETI 417 (1 only required — each board has provision for two channels)

CONSTRUCTION

Mount all components on to the printed circuit board in accordance with the component overlay. Make sure that all diodes are correctly orientated, in particular the LED's. The LED's will not be damaged by reverse polarity but will not operate in that mode.

Whether the unit is mounted inside the amplifier or external to it in a small box will be a matter for the individual constructor. The printed circuit board may be mounted in any suitable position within the amplifier and leads extended to front-panel mounted LEDs if required.

Polarity of the leads to the amplifier output terminals is immaterial but make sure that the leads of separate channels are not mixed. This is best avoided by twisting each pair of leads to each channel.

CALIBRATION

There are several ways of calibrating the unit.

By far the best way is to connect an audio oscillator to the input of the amplifier (both channels driven at the same time), then, with the amplifier volume control at a low setting, adjust the oscillator to provide a 1 kHz sine-wave.

Set both trim potentiometers (RV1) so that their wipers are nearest R3.

Now increase the amplifier volume until clipping occurs. This is very easily identified as a sudden harshness of tone. Do not leave the volume control at this setting for more than a second or two, as apart from the pounding you are giving to your ears, some amplifiers will not tolerate a sine-wave input at clipping level for extended periods without damage.

Once the clipping point has been established, turn the volume down again, and then quickly turn up to the clipping point momentarily, meanwhile adjusting the trimming potentiometers RV1 until a point is reached where the light emitting diodes just come on.

Repeat the procedure a few times — finally arriving at a setting at which the LED's come on just before the clipping point.

If you do not have access to an oscillator, the device can be set by playing a test record that contains a sine-wave tone — or failing this — by playing a record of a solo instrument such as a flute. A recording of the human voice is also very effective. In such cases the same calibration procedure described above should be followed. ●

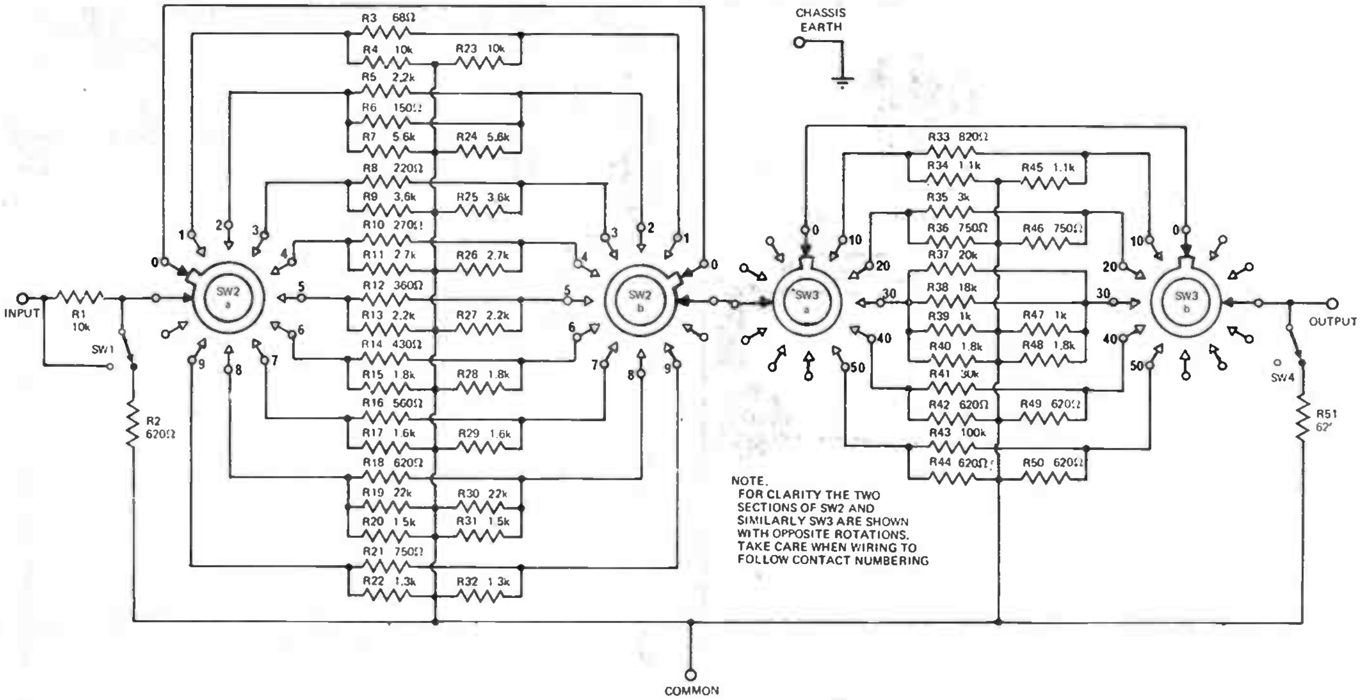
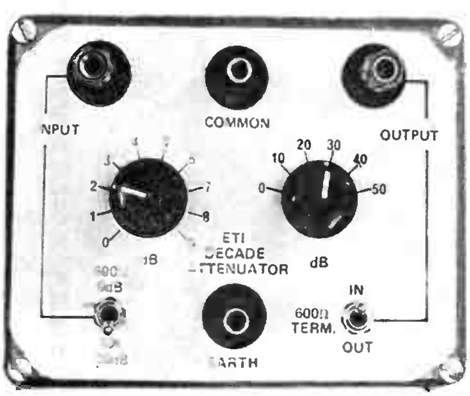


Fig. 1. Circuit diagram of the attenuator.

This useful audio attenuator project for the experimenter provides 0-59dB attenuation in one dB steps.

AUDIO ATTENUATOR



ACCURATE attenuators are required in a multitude of design, service, testing and measuring situations. These units are designed with varying degrees of accuracy and as many steps of attenuation as the designer feels necessary. They may be balanced or unbalanced and have whatever input and output impedances the designer requires.

There are three common types of attenuator configuration, Pi, T or L. The latter is mainly employed where the output impedance is not required to be constant.

PROJECT 112

We have chosen Pi type sections for our unit. We could have connected the various sections in tandem to form a ladder attenuator, but this would have made more complex rotary switches necessary. Instead, we chose to employ a separate section for each step of attenuation, making only simple rotary switches necessary.

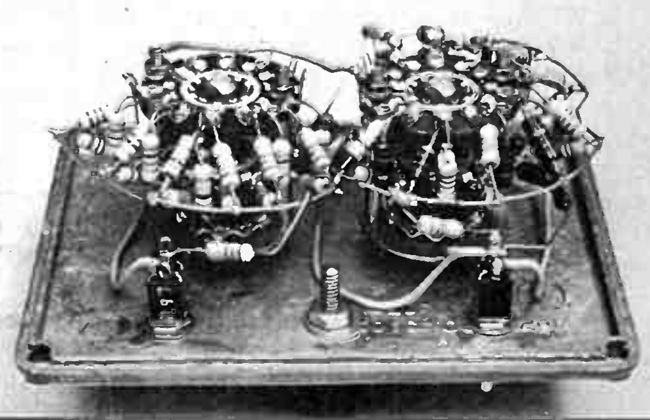
The input and output resistances of the unit remain relatively constant at 600 ohms over the full attenuation range. The input impedance can be changed to 10k by SW1 but an additional 30dB of attenuation is added. The output can also be terminated internally by SW4 when using a high impedance load such as a meter.

The maximum attenuation when the input and output resistances are set at 600 ohms is 59dB. There are ten 1dB steps from 0dB to 9dB, via a 10 position rotary switch, and a further six 10dB steps from 0dB to 50dB via a six position rotary switch, giving a

SPECIFICATION

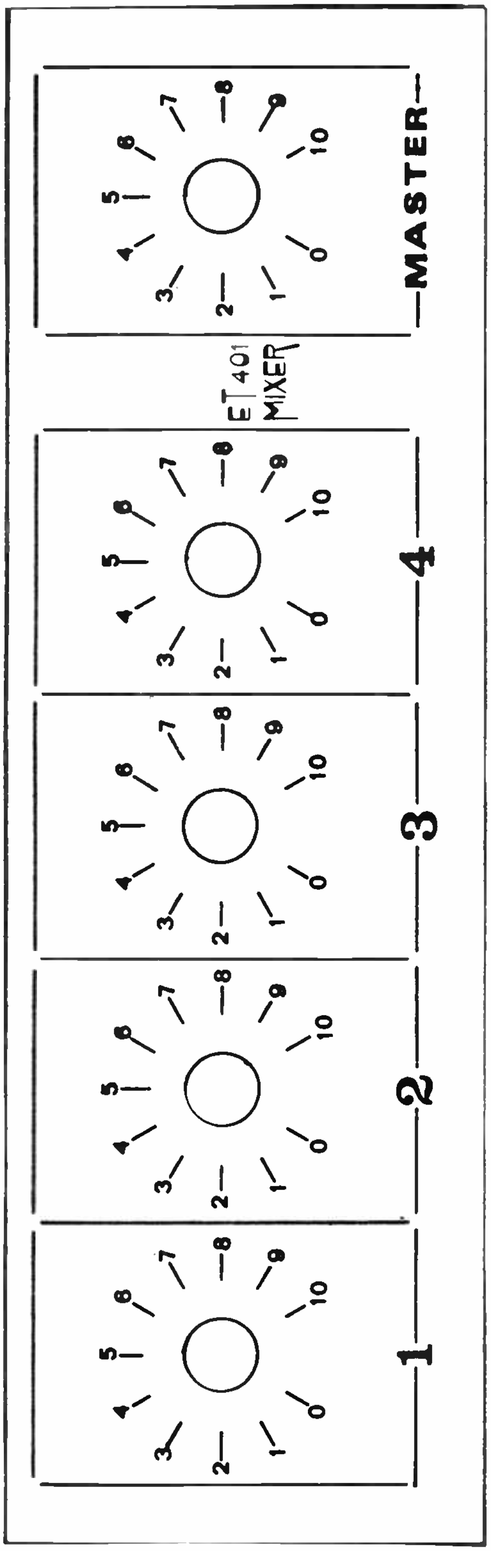
Max attenuation	59dB
Resolution	1dB
Accuracy	±0.3dB
Frequency range	dc to 100kHz
Input impedance	600 Ω nominal 10k switched (+30dB attenuation)
Output impedance	600 Ω nominal
Max input voltage	15 volt

Internal switched termination resistor for use with high impedance loads.



FET-FOUR INPUT MIXER

(Continued from page 53)



This is the front panel layout reproduced exactly full size. Carefully cut around the outer edge, cut out the centre holes and mount it on the front panel behind a thin sheet of perspex.

PARTS LIST ETI 112

R1	Resistor	10k	2% ½W
R2	"	620Ω	" "
R3	"	68Ω	" "
R4	"	10k	" "
R5	"	2.2k	" "
R6	"	150Ω	" "
R7	"	5.6k	" "
R8	"	220Ω	" "
R9	"	3.6k	" "
R10	"	270Ω	" "
R11	"	2.7k	" "
R12	"	360Ω	" "
R13	"	2.2k	" "
R14	"	430Ω	" "
R15	"	1.8k	" "
R16	"	560Ω	" "
R17	"	1.6k	" "
R18	"	620Ω	" "
R19	"	22k	" "
R20	"	1.5k	" "
R21	"	750Ω	" "
R22	"	1.3k	" "
R23	"	10k	" "
R24	"	5.6k	" "
R25	"	3.6k	" "
R26	"	2.7k	" "
R27	"	2.2k	" "
R28	"	1.8k	" "
R29	"	1.6k	" "
R30	"	22k	" "
R31	"	1.5k	" "
R32	"	1.3k	" "
R33	"	820Ω	" "
R34	"	1.1k	" "
R35	"	3k	" "
R36	"	750Ω	" "
R37	"	20k	" "
R38	"	18k	" "
R39	"	1k	" "
R40	"	1.8k	" "
R41	"	30k	" "
R42	"	620Ω	" "
R43	"	100k	" "
R44	"	620Ω	" "
R45	"	1.1k	" "
R46	"	750Ω	" "
R47	"	1k	" "
R48	"	1.8k	" "
R49	"	620Ω	" "
R50	"	620Ω	" "
R51	"	620Ω	" "

SW1 Single pole change over minature toggle switch
SW2 2 pole 11 position rotary switch
SW3 2 pole 11 position rotary switch
SW4 Single pole change over minature toggle switch
Diecast box 4¾ x 3¾ x 2
4 Terminals type L1568/1S or similar
2 Knobs

total of 60 steps from 0dB to 59dB. This range of attenuation is adequate for most purposes. Although further sections could be added, noise becomes a limiting factor in a simple attenuator such as this.

CONSTRUCTION

It is advisable to employ separate wafers for each switch pole. If the type of switch that has two poles on one wafer is employed, there may be problems at the high frequency end due to stray capacitance. This would be evident as spikes on the leading edges of high frequency square waves.

The common rail for each switch is a length of 18 gauge tinned copper wire formed into a ring to allow termination of the shunt resistors (R4, R23, R7 and so on). The series resistors are connected directly between the relevant switch contacts. Layout of the unit may be seen by the accompanying photographs. ●

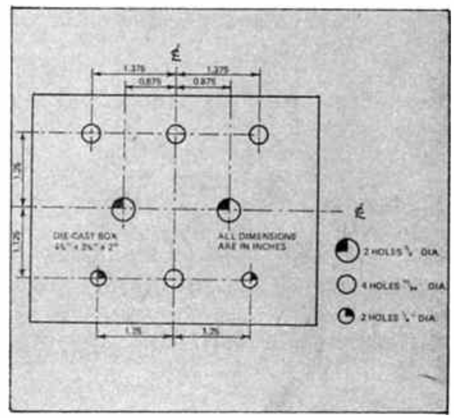


Fig. 2. Drilling details for the die cast box.

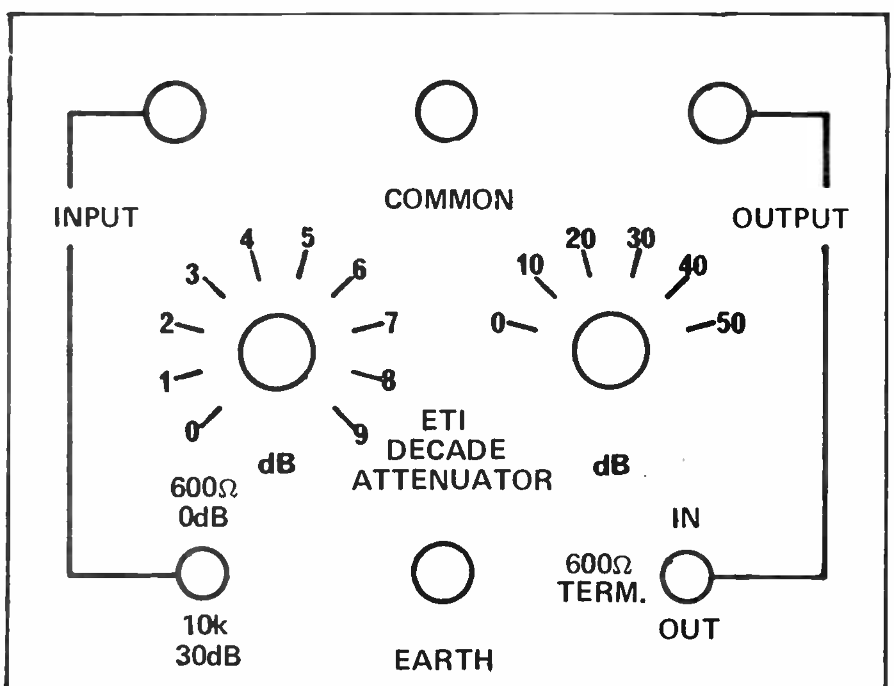


Fig. 3. Lettering and front panel artwork — full size.

FOUR-CHANNEL SOUND
A SIMPLER WAY

Four-channel sound can be an expensive hobby, especially if you have already a satisfactory stereo set-up. This project explains how to adapt your present system to produce simulated four-channel.

IS FOUR-CHANNEL sound *really* worth while, or is it just a technically interesting gimmick?

Here's a simple and relatively inexpensive way of finding out. It won't give the same results as a really good matrix or discrete system – but on some recordings it comes surprisingly close.

When a normal stereo record is made, a great deal of sound – especially reverberation – is masked by the direct sound from the performers. Nevertheless the sound that has been recorded is actually right there on the record. All we have to do is to obtain it.

This circuit does just that. It works by extracting signals which are the difference between the right hand and left hand programme channels.

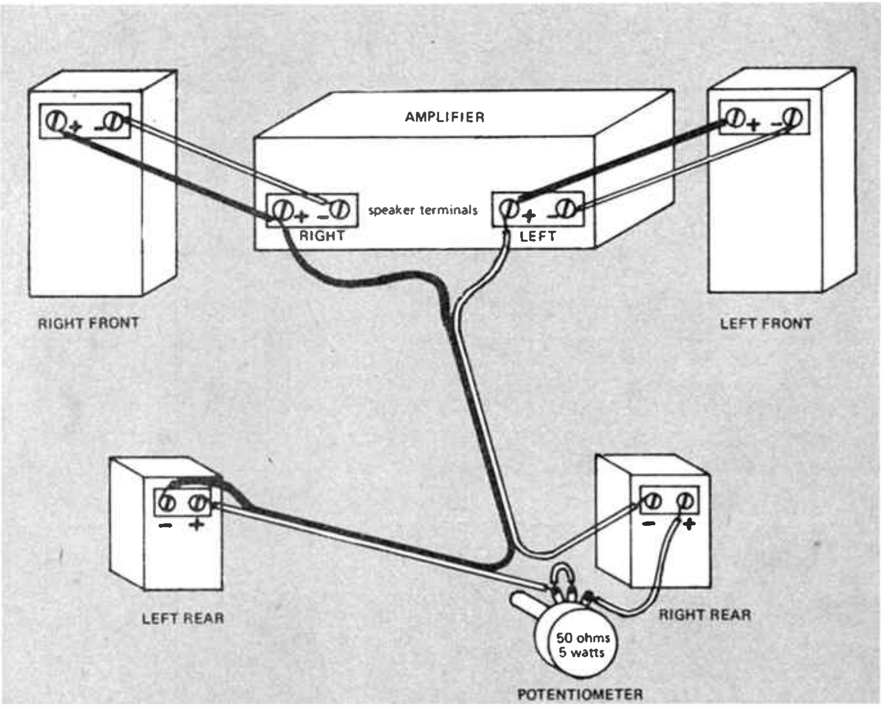
It works best on classical music that has been recorded 'live'. However, good results will be obtained from almost any 'live' recording.

The rear speakers don't need to be anything special, in fact best results will be obtained from units having restricted treble response. Wiring may be twin lighting flex or even bell wire – the rear speakers don't draw a great deal of current.

The potentiometer should preferably be a 50 ohm, 5 watt unit – at a pinch 2 watts will do.

For best results, play a short piece of classical music and adjust the potentiometer so that you are only barely aware of sound from behind you.

ETI PROJECT 402



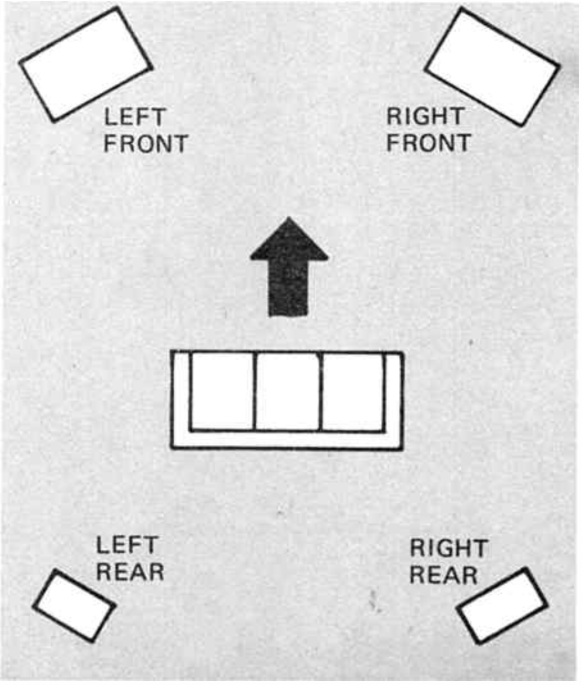


Fig. 1. How the connections are made. Note that all the speakers and the amplifier are shown as seen from the rear. Just connect them exactly as shown in this drawing and then position the speakers as shown in Fig.2.

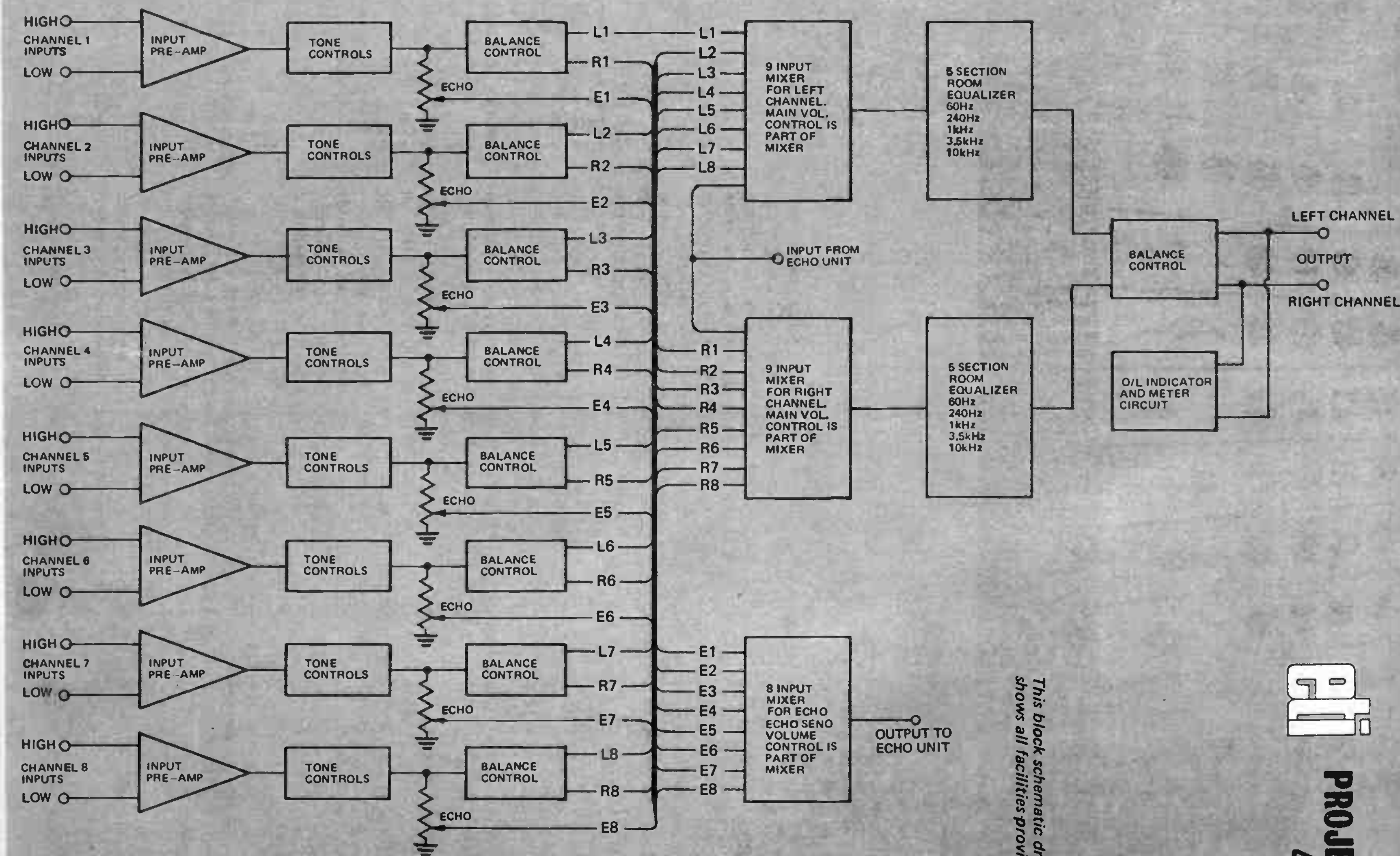
Some amplifiers may have the terminals arranged vertically but if in doubt be guided by the 'L' and 'R' markings that will appear somewhere near the terminals.

Terminal markings may also vary. The terminals that we have indicated as '+' may be marked 'hot' or 'high', or just with a red dot. Some may be unmarked but a red wire will be connected to the equivalent terminal that we have marked '+'. Again, the terminals that we have marked with a '–' may be labelled 'common', 'ground', 'earth' or 'low'.

Fig. 2. How the speakers are positioned.

ETI MASTER-

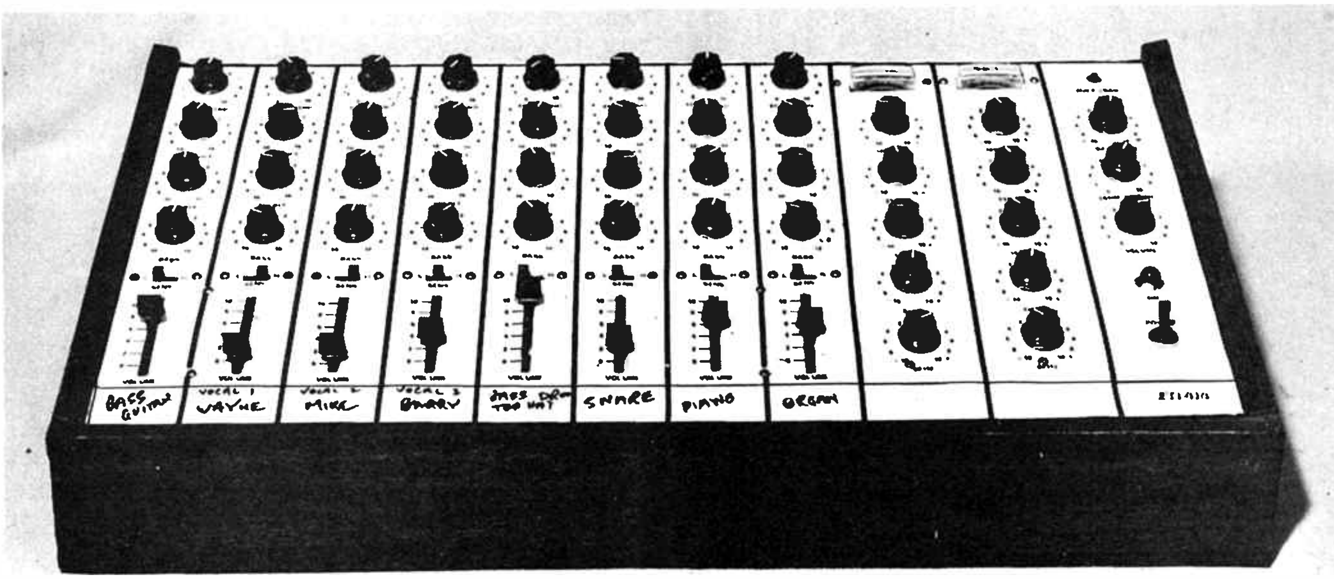
PROJECT 414



This block schematic drawing shows all facilities provided.

MIXER

Versatile multi-input mixer/preamplifier has all the facilities required for professional PA use.



* **Multiple inputs**
* **Low noise**
* **Stereo outputs**
* **Inbuilt equalizers**
* **Echo facilities**
* **Professional design**
* **Overload immunity**
* **Stage monitor facility**

Anyone who is associated with a pop group or band, will be familiar with the steps one must take to ensure optimum sound in varied localities and halls.

Outdoors, each amplifier and/or public address system must be adjusted separately to ensure sufficient sound and optimum overall mix.

Indoors, one must also cope with the acoustics of the particular building.

Many of the smaller groups merely adjust their sound on stage with one member at the back of the hall giving a subjective indication of the sound he is hearing. Larger groups often employ a person whose main function is to ensure that the final sound is exactly as it should be (as regards volume, mix, quality, etc).

The 8-channel mixer described in this article will allow the total sound to be adjusted at the one point – perhaps the rear of the hall, while at the same time, eliminating several expensive amplifiers, and still ensuring an optimum overall sound. (This is only part of the story as the reader will realize from the full description of the unit).

INPUTS

As the name of the unit indicates, there are eight separate input channels. Each of these input channels has two input sockets, one of 47k impedance, and the other adjustable by changing one resistor, (maximum 4.7k). In our case we have a 200 ohm resistor in circuit so we shall refer to the 200 ohm input from here on.

Each input channel has a slide control potentiometer for volume. This potentiometer is in series with a sensitivity network that is adjusted by a three position slide switch.

The remaining input channel controls are rotary potentiometers facilitating balance, bass, treble and echo-send volume. We shall discuss these controls in detail later in this article.

Each input, after passing through the preamplifier and tone control stages, is divided to provide identical signals. The relative level of these signals can be varied by the input channel balance control. The outputs from the balance controls drive the output mixers. This creates a stereo effect, allowing the performers to be audibly "positioned" on stage.

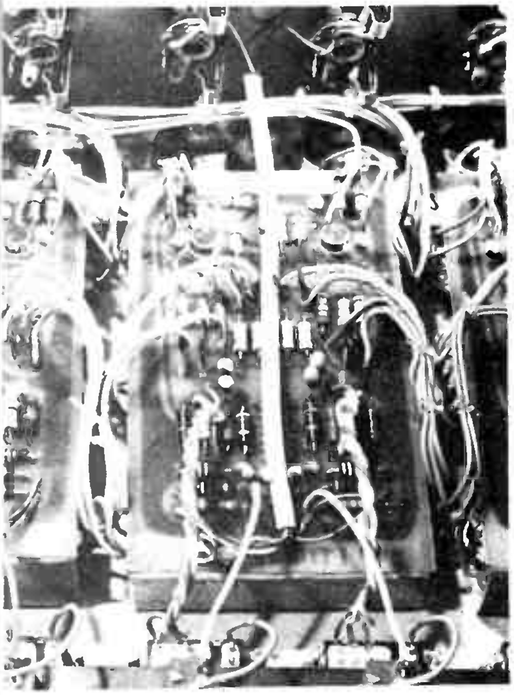
OUTPUTS

The unit has two output channels. The unit can of course be modified simply to provide one main output and an onstage monitor output. These receive signals from the input channel balance controls and external echo unit, or similar, if one is employed. Rotating any of the input channel balance or echo-send controls will affect the output for that particular input only.

There are also controls provided for overall volume, balance and echo volume. Finally five more rotary controls per output channel have been provided for frequency equalisation. These allow compensation for hall acoustics etc. These controls operate at 60Hz, 240Hz, 1000Hz, 3500Hz and and 10kHz and provide approximately 10dB boost or cut.

Two VU meters also feature on the front panel, together with an overload indicator light which becomes

ETI MASTER-MIXER



Input pre-amps are built in modular form — two to each board.

illuminated should either output exceed one volt — which is the overload point of the 100 watt amplifier on page 44.

Having briefly covered the various controls and facilities provided, we shall now describe the operation and specifications of each section more extensively before we commence constructional details.

Each input channel is identical, so we only need concern ourselves with one.

The two input jacks for each channel are situated at the back of the unit, directly behind their respective control panels.

The 47k input is typical for electric guitar pickups, microphones and such, but if long leads are used, problems could arise due to hum pickup or other radiated interference. If this is the case, a matching stage or transformer may have to be inserted between the input source and the low impedance input socket.

Some microphones have an impedance of 50k, and in this case the same would apply if long leads are to be used. The optimum situation is a low impedance source (microphone etc) into the low impedance socket, but if there is a mismatch, a low impedance source and a high impedance input is preferable.

There may be situations where one wishes to feed two or three microphones to the same input channel. In this case a separate low cost mixer would be needed.

The situation above could occur for example with a drummer or with organs that have more than one output.

Each input employs an operational amplifier. The gain of this amplifier is varied by changing the negative feedback, as is customary with this type of device. Maximum gains of 20dB, 40dB and 55dB are available via the volume control and the switched sensitivity network.

The output from each input op-amp, feeds a second op-amp which acts as a tone control stage. The output from each tone control stage is then fed via a potentiometer to one of eight inputs of an echo send mixer I.C., and is also split by the input channel balance control network before being diverted to the output channel mixers.

The output from the echo-send mixer is brought out at the rear of the unit. This output is intended to drive a complete echo or reverberation unit etc. The output from the external unit is then fed back into the unit via another socket to a resistive splitter, which provides two identical signals for the output mixers. It is important to realize that all signals are "echoed" if their particular echo send controls are turned up, and that both output channels amplify the result equally, as indicated above. The overall echo gain control varies the feedback of the echo-send mixer.

SPECIFICATIONS

Inputs	eight (but may be expanded or reduced — in multiples of two — as desired)
Input impedance	
(high)	47k
(low)	nominally 200 ohms, but may be any preset value under 4.7k
Sensitivity (high impedance input)	10mV
(low impedance input)	1mV
Tone controls (on each input)	bass ±10dB at 100Hz treble ±10dB at 10kHz
Outputs	two, left and right
Output level	maximum 5 V rms
Output impedance	approx 4000 ohms
Output tone control	each channel has its own equalizer providing ±10dB boost or cut at following frequencies — 60Hz; 240Hz; 1kHz; 3.5kHz; 10kHz

A nine-input mixer is employed at the input of each output channel. One of the nine inputs is in both cases used for echo input, while the others take the outputs from the eight input channels. The negative feedback of these op-amps is varied for overall volume control.

The outputs from the main mixers pass through the graphic equalizers and then to an overall balance control. The two VU meters and the overload indicator are connected at the output of the unit.

The IC employed in the preamplifier stages is a National type LM381 dual low noise preamplifier IC. We have used one IC per every two input channels — a total of four, if all input channels are required.

The total equivalent input noise is specified as maximum 1μV rms with a 600 ohm source impedance, over a frequency range from 10 Hz to 10 kHz.

The open loop gain of each amplifier is typically 112 dB, the supply range 9 to 40 volts, and power supply rejection better than 120 dB.

Supply current is typically 10 mA over the voltage range quoted above. Channel separation measured at 1 kHz is typically 60 dB. Total harmonic distortion measured at 1 kHz with the gain set at 75 dB is typically 0.1%.

The maximum recommended input voltage is 300 mV, and the typical available peak-to-peak output voltage swing is Vcc minus two volts. This IC is short circuit protected.

CONSTRUCTION — Preamplifier

There are four preamplifier boards each with two channels. Assemble the components to each board in accordance with the circuit diagram and component overlay provided. Take care not to damage the ICs with excessive heat (use a lightweight iron and solder quickly) and pay particular attention to the orientation of the TAG tantalum capacitors.

Printed circuit boards will be available from kitset suppliers. However for those who prefer to etch their own boards, a full-size pattern is provided. Details of the connections between the preamplifier boards and their associated controls are given in Fig. 1. It is suggested that leads of adequate length should be connected to the boards first. The boards may then be fixed in position and the leads routed to their respective controls.

The 200 ohm resistors across the low impedance inputs should preferably not be fitted across the input sockets. Rather, they should be fitted across the output of the low impedance device (eg within the jack plug) itself.

This procedure will prevent excessive noise when low impedance input is not being used.

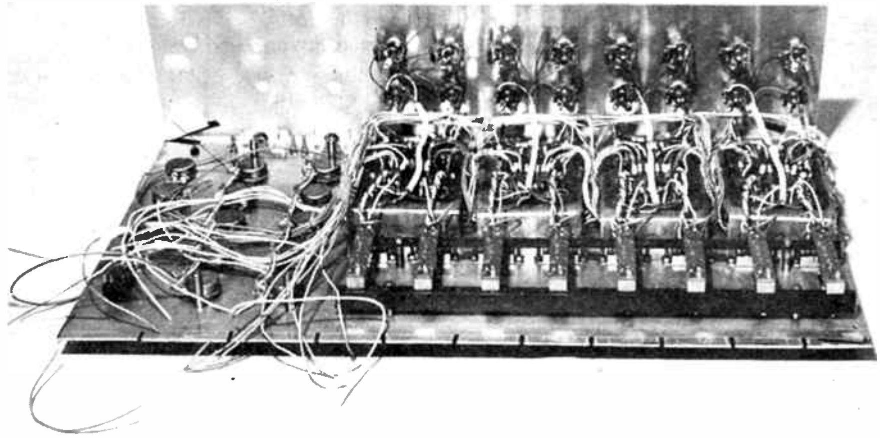
Equalizer

After the preamplifier boards are assembled, we can assemble the main mixer/equalizer boards of which there are two. The winding data for the inductors associated with this section is given in Table 1.

The coils must be layer wound with care. Jumble winding will almost certainly prevent the full number of turns fitting on the bobbin.

The metal panel of our unit is folded from one piece of 18 gauge steel. Eleven aluminium escutcheons are used, although of only three different types. These should be available from kit suppliers, however should the reader wish to make his own panels and cabinet, diagrams of both metal work and woodwork are published later in this article.

The only remaining printed circuit board accommodates the power supply – echo mixer, overload and meter circuitry.



This photograph was taken during construction of our prototype unit. The four input pre-amps may be seen on the right. Directly in front of the pre-amps are the slide potentiometers used for individual volume controls.

HOW IT WORKS MAIN MIXERS – EQUALIZERS

There are nine inputs to each main mixer IC. This IC is connected in an inverting amplifier configuration, with the gain controlled by varying the negative feedback. This gives a control range from zero output to about 30 dB gain.

The output from the main mixer is direct coupled to the input of the equalizer stage. This stage is a little unusual, since the equalizing networks are arranged to vary the negative feedback. If we consider one section with the others disconnected, at the resonant frequency of the series LCR combination the impedance of the entire network will be equal to 680 ohms. Either side of resonance the impedance of the network will increase (with a slope dependent on the Q of the network), due to uncancelled inductive reactance above resonance and uncancelled capacitive reactance below resonance. We can therefore represent the equalizer stage with equivalent circuits as reproduced below. These circuits consider only one network is in circuit, the input signal frequency is the resonant frequency of the network, and the resistance of the inductor is negligible.

With the slider of the potentiometer at the top end (Fig. 2a) we have 680 ohms to the zero volt line from pin 2 of IC2, and a 1k ohm between pin 3 and pin 2. The IC will act due to the feedback to keep the potential between pins 2 and 3 virtually zero, thus there is zero current through

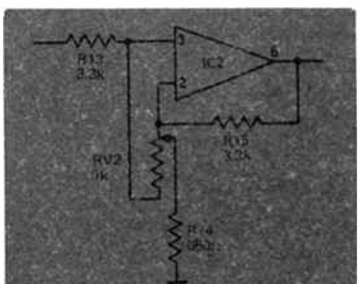


Fig. 2a – Equivalent circuit of the equalizer with potentiometer set for maximum boost at the resonant frequency of the network.

RV2. The voltage on pin 3 (IC2) is therefore equal to the output of the mixer since there is virtually no current through and no voltage drop across R13.

The output of IC2 in this case is approximately the input signal times (R15 + 680)/680 ohms, indicating a

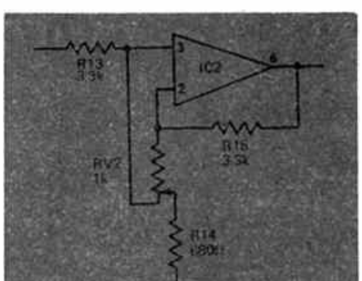


Fig. 2b – Equivalent circuit of the equalizer with the potentiometer set for maximum cut at the resonant frequency of the network.

gain of about 15dB. If the slider is at the other end of the potentiometer (Fig 2b) the signal appearing at pin 3 and thus also at pin 2 is about 0.2 of the output of the previous stage due to the voltage division of R13 and the 680Ω. There is still zero current through RV2 and also zero current through R15 since there is no path. The output voltage is therefore the same as that at pin 2, which happens to be about 0.2 times the output of the previous stage. The gain is therefore 0.2 or –13dB.

With all networks in circuit, the maximum boost and cut will be

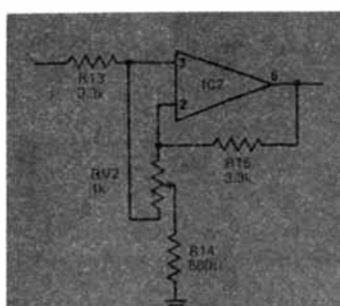
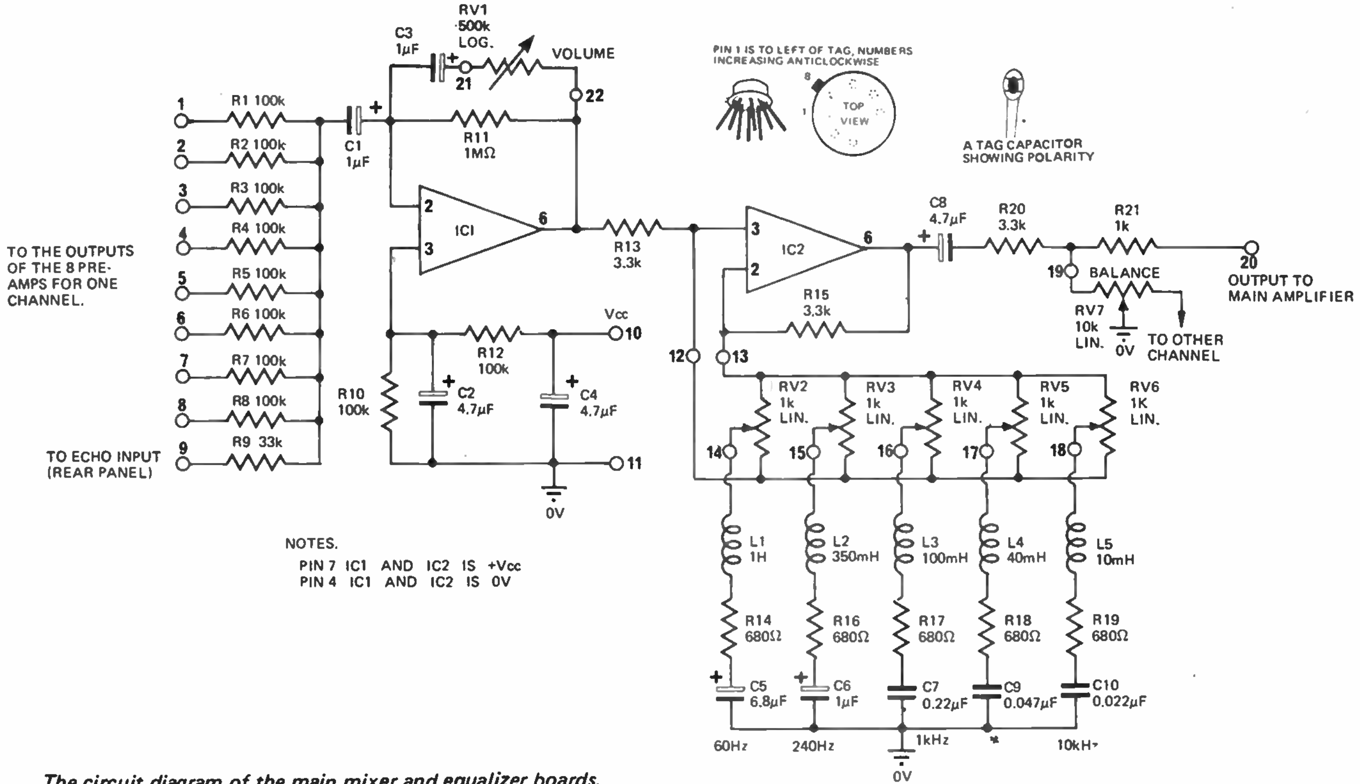
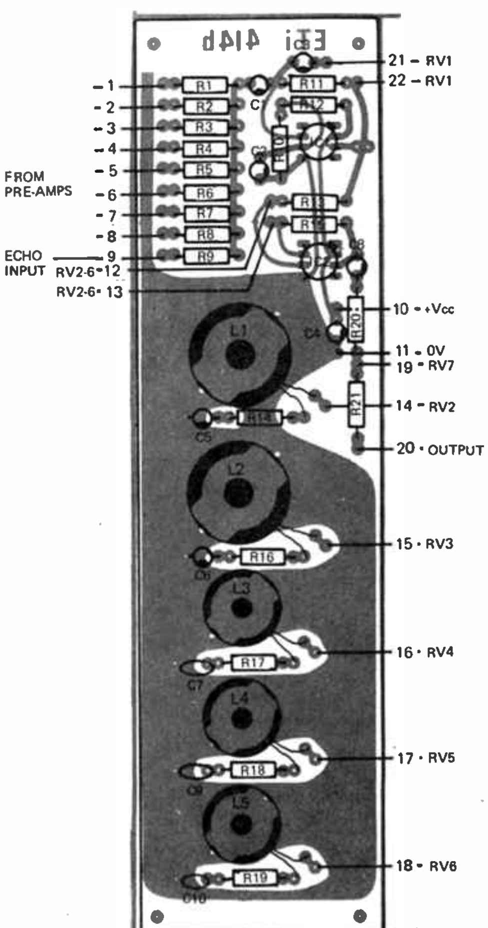


Fig 2c – Equivalent circuit of the equalizer with the potentiometer set for unity gain regardless of frequency.

reduced, but a range of ±10dB is still available. With the wiper of the potentiometers set midway – Fig 2c, the gain will be unity regardless of frequency, due to the symmetry of the entire network.



The circuit diagram of the main mixer and equalizer boards.



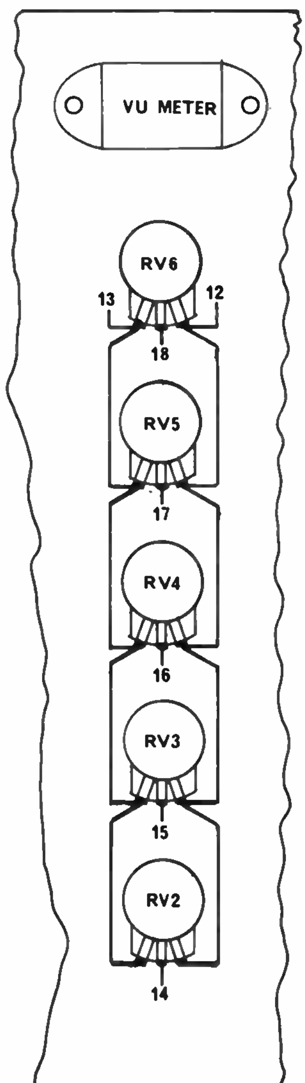
Component overlay for main mixer and equalizer.

THIS LIST CONTAINS ALL THE PARTS FOR ONE MIXER-EQUALIZER. (TWO SETS REQUIRED)

R1	resistor	100 k	5%	½ watt
R2	"	100 k	"	"
R3	"	100 k	"	"
R4	"	100 k	"	"
R5	"	100 k	"	"
R6	"	100 k	"	"
R7	"	100 k	"	"
R8	"	100 k	"	"
R9	"	33 k	"	"
R10	"	100 k	"	"
R11	"	1 M	"	"
R12		100 k	"	"
R13	"	3.3 k	"	"
R14	"	680 ohm	"	"
R15	"	3.3 k	"	"
R16	"	680 ohm	"	"
R17	"	680 ohm	"	"
R18	"	680 ohm	"	"
R19	"	680 ohm	"	"
R20	"	3.3 k	"	"
R21	"	1 k	"	"
C1	capacitor	1μF	35V TAG tantalum	
C2	"	4.7μF	35V " "	
C3	"	1μF	35V " "	
C4	"	4.7μF	35V " "	
C5	"	6.8μF	25V " "	
C6	"	1μF	35V " "	
C7	"	0.22μF	polyester	
C8	"	4.7μF	35V TAG tantalum	
C9	"	0.047μF	polyester	
C1	"	0.022μF	polyester	
L1	audio choke	1H	(see winding data table 1)	
L2	" "	350 mH	"	
L3	" "	100 mH	"	
L4	" "	40 mH	"	
L5	" "	10 mH	"	
IC1	integrated circuit	uA741,LM307 (metal can or mini dip only)		
IC2	" "	uA741,LM307 (metal can or mini dip only)		
*RV1	potentiometer	500 k ohm LOG dual		
RV2	"	1 k LIN		
RV3	"	1 k LIN		
RV4	"	1 k LIN		
RV5	"	1 k LIN		
RV6	"	1 k LIN		
*RV7	"	10 k LIN		

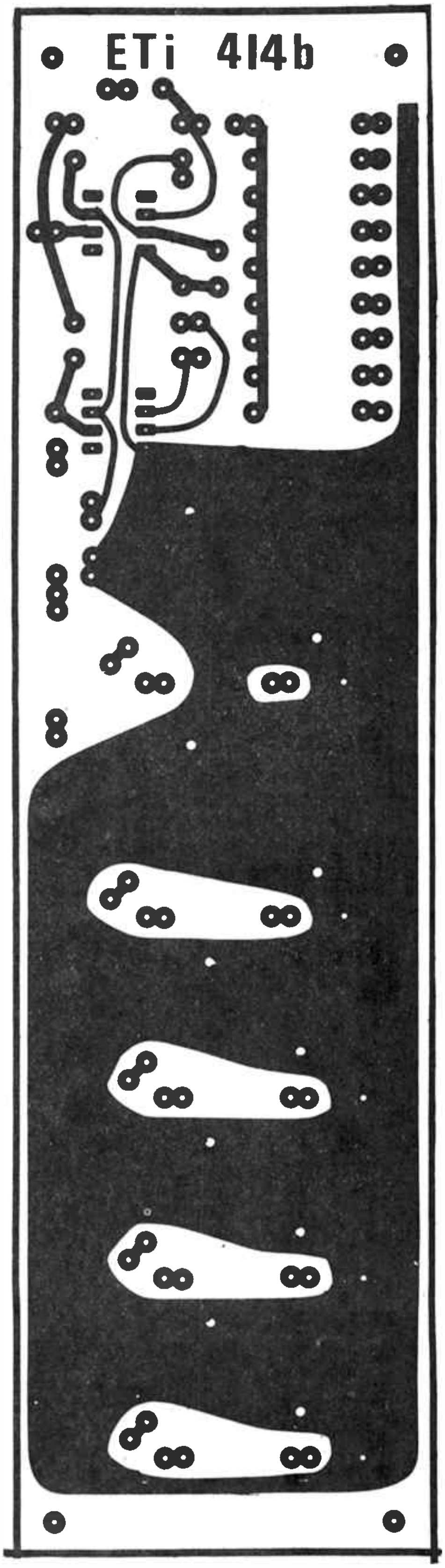
*ONE ONLY REQUIRED FOR COMPLETE UNIT

PC Board ETI 414B
5 knobs
4 1" spacers



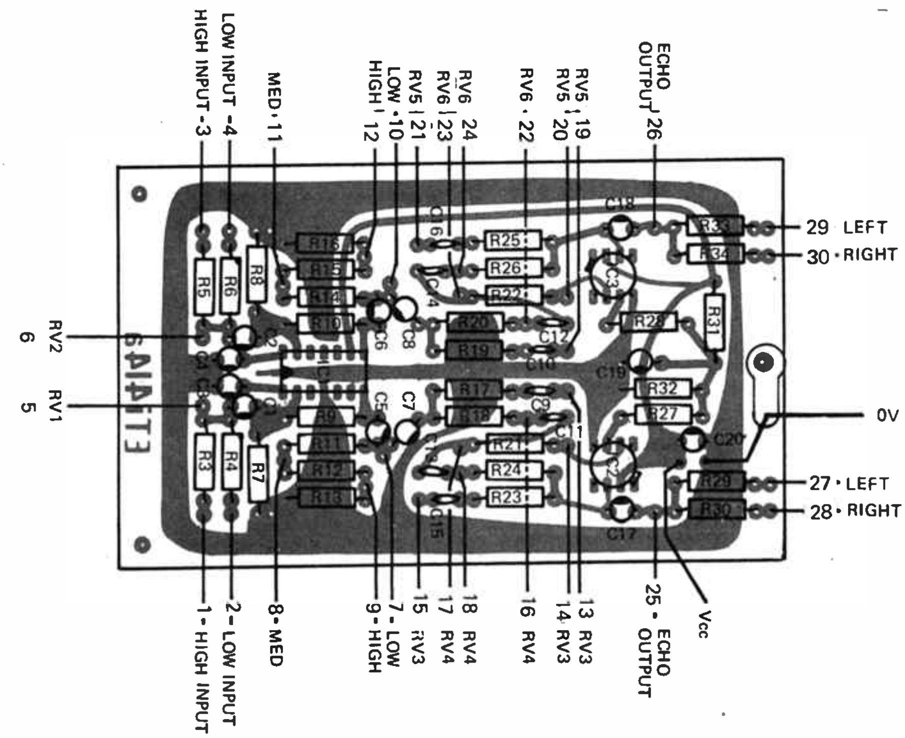
This diagram shows the connections to the potentiometers associated with the equalizers. The numbers correspond one-to-one, to those on the main mixer – equalizer circuit and overlay diagrams.

ETI MASTER MIXER



The main mixer and equalizer printed circuit board pattern – shown full size.

TABLE 1:– WINDING DETAILS EQUALIZER COILS	
L1	1000 Turns 34 B&S wire Core Philips 4322 022 28290 Former Philips 4322 021 30330 Clip Philips 4302 021 20020
L2	585 Turns 32 B&S wire Core, former and clip same as L1
L3	460 Turns 34 B&S wire Core Philips 4322 022 24280 Former Philips 4322 021 30270 Clip Philips 4302 021 20000
L4	300 Turns 34 B&S wire Core, former and clip same as L3
L5	150 Turns 32 B&S wire Core, former and clip same as L3



Preamplifier component overlay

HOW IT WORKS – PREAMPLIFIERS

Considering channel 1 of the board only, ICI is wired as an inverting amplifier. The gain of this amplifier is varied by RV1 – the volume control, and set at high, medium or low by SW1 – the sensitivity switch. These controls vary the gain of the amplifier by adjusting the negative feedback. More feedback, less gain, and vice-versa.

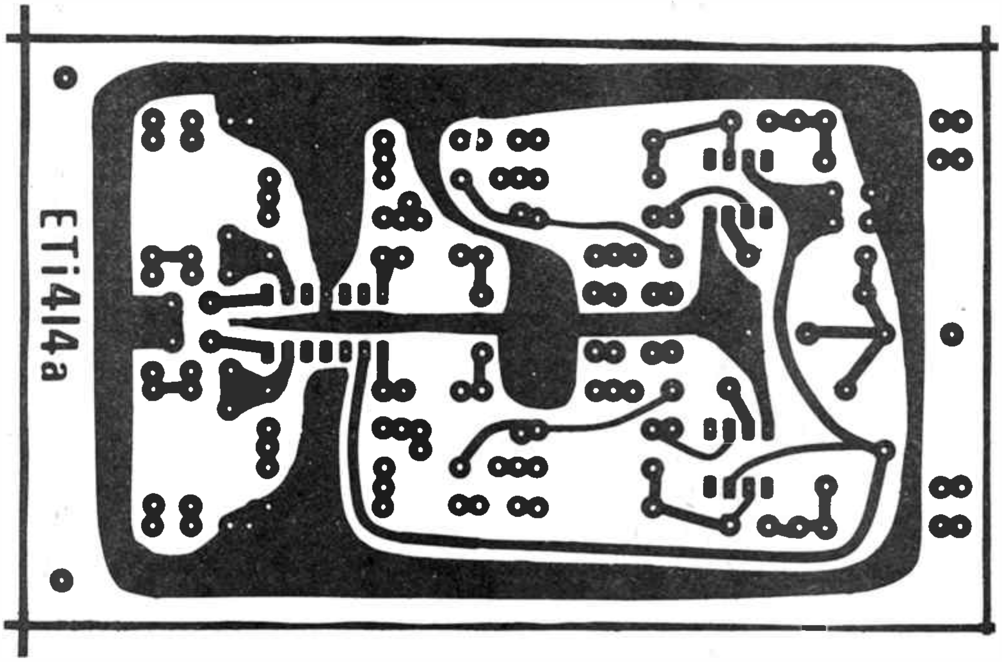
SW1 changes the range of RV1 for maximum gains of 20dB, 40dB and 55dB when the low impedance input is employed. With the sensitivity switch at low the minimum output of this stage is virtually zero, while a minimum gain of 6dB is realised when the sensitivity is set at either medium or high. Gains when the high impedance input is employed are all 20dB lower than those given above.

The input impedance to the IC is virtually zero, when used as an inverting amplifier. Therefore the input impedance to the preamplifier is determined by R3 for the high impedance input, and by R1 in parallel with R4 for the low impedance input. R9 and R7 set the bias of the IC. The tone control stage is a conventional feedback type.

Note that where different input impedances from those specified are required, the values of R1 (or R2) required may be calculated by the following formula

R = (4700 x Zin)/(4700 – Zin)
where Zin is the desired input impedance.

ETI MASTER MIXER



The printed circuit board pattern for the pre-amplifier sections — shown full size.

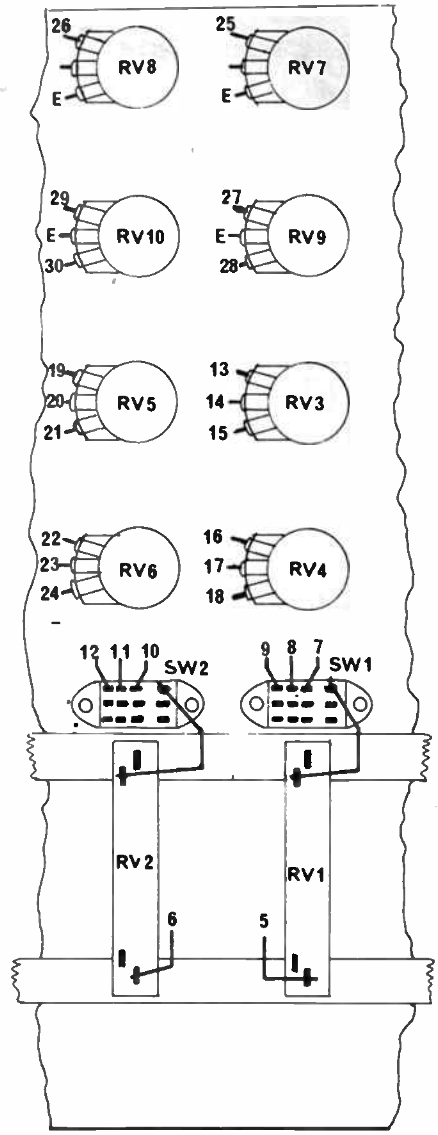


Fig. 1. This diagram shows connections to the potentiometers and sensitivity switches associated with the preamplifier boards. The numbers corresponds to those on the pre-amplifier circuit and overlay diagrams.

PARTS LIST-PREAMPLIFIER

This list contains all parts (except metal work) for a complete preamplifier and tone controls. For an eight-channel mixer, four sets of components are required.

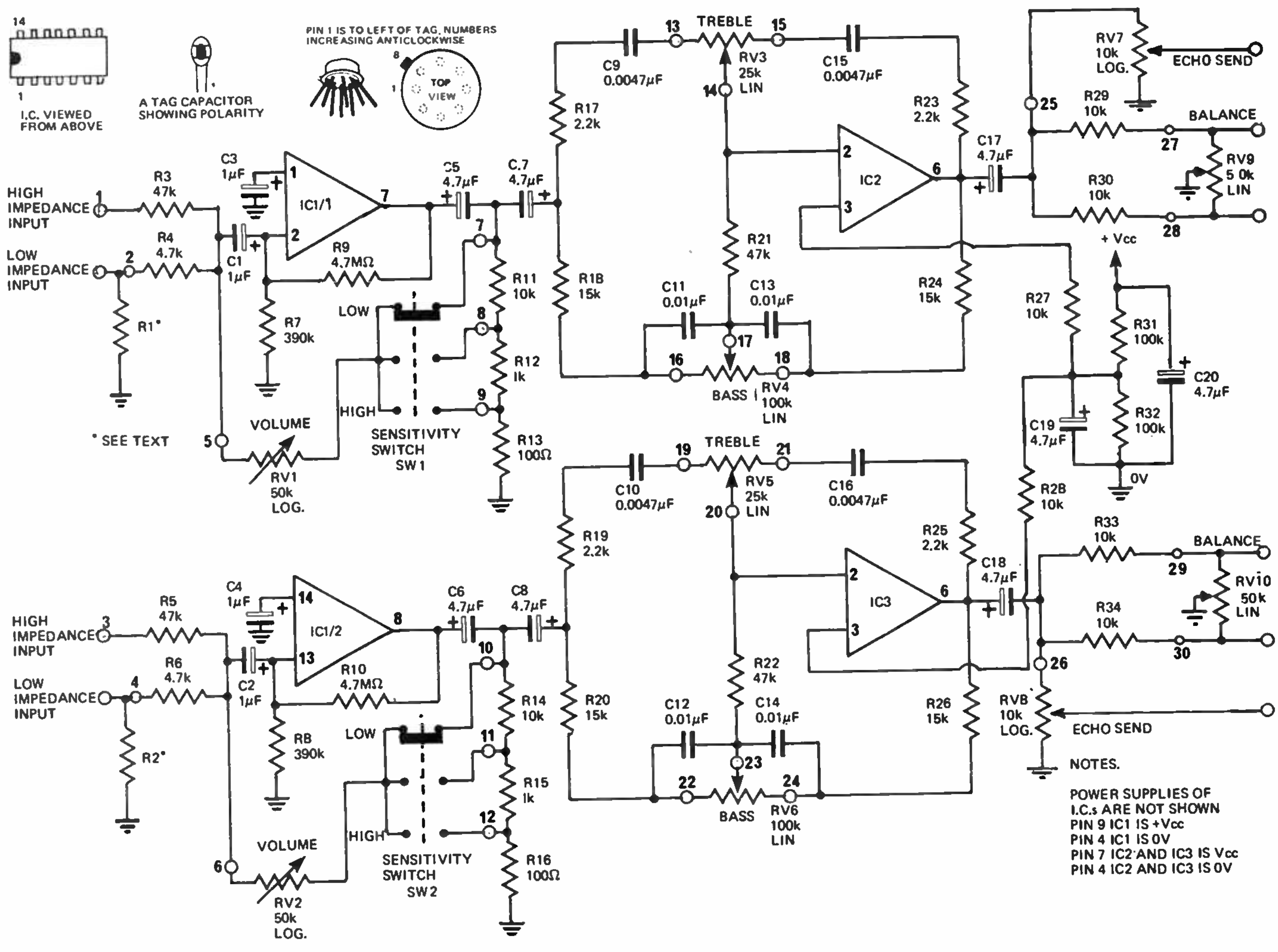
R1*	Resistor	200ohm	5%	½W
R2*	"	200ohm	"	"
R3	"	47k	"	"
R4	"	4.7k	"	"
R5	"	47k	"	"
R6	"	4.7k	"	"
R7	"	390k	"	"
R8	"	390k	"	"
R9	"	4.7M	"	"
R10	"	4.7M	"	"
R11	"	10k	"	"
R12	"	1k	"	"
R13	"	100ohm	"	"
R14	"	10k	"	"
R15	"	1k	"	"
R16	"	100ohm	"	"
R17	"	2.2k	"	"
R18	"	15k	"	"
R19	"	2.2k	"	"
R20	"	15k	"	"
R21	"	47k	"	"
R22	"	47k	"	"
R23	"	2.2k	"	"
R24	"	15k	"	"
R25	"	2.2k	"	"
R26	"	15k	"	"
R27	"	10k	"	"
R28	"	10k	"	"
R29	"	10k	"	"
R30	"	10k	"	"
R31	"	100k	"	"
R32	"	100k	"	"
R33	"	10k	"	"
R34	"	10k	"	"

C1	Capacitor	1µF	35V	TAG tant.
C2	"	1µF	35V	" "
C3	"	1µF	35V	" "
C4	"	1µF	35V	" "
C5	"	4.7µF	35V	" "
C6	"	4.7µF	35V	" "
C7	"	4.7µF	35V	" "
C8	"	4.7µF	35V	" "
C9	"	0.0047µF	35V	polyester
C10	"	0.0047µF		"
C11	"	0.01µF		"
C12	"	0.01µF		"
C13	"	0.01µF		"
C14	"	0.01µF		"
C15	"	0.0047µF		"
C16	"	0.0047µF		"
C17	"	4.7µF	35V	TAG tantalum
C18	"	4.7µF	35V	" "
C19	"	4.7µF	35V	" "
C20	"	4.7µF	35V	" "

RV1	potentiometer	50 k	LOG 45mm slide (Soanar)
RV2	"	50 k	LOG 45mm slide (Soanar)
RV3	"	25 k	LIN
RV4	"	100 k	LIN
RV5	"	25 k	LIN
RV6	"	100 k	LIN
RV7	"	10 k	LOG
RV8	"	10 k	LOG
RV9	"	50 k	LIN
RV10	"	50 k	LIN

IC1 Integrated circuit LM381
IC2 " " uA741, LM307 (metal can or mini dip only)
IC3 " " uA741, LM307 (metal can or mini dip only)

PC board ETI 414A
8 knobs for rotary potentiometers
2 knobs for slide potentiometers
2 3P-3T slide switches (John Carr or McMurdo)
3 1" spacers
4 6.5 mm phone sockets



The circuit diagram of one of four identical preamplifier boards. I.C. 1/1 and I.C. 1/2 are in the same package.

MASTER

Assembly and operation of the power supply — overload indicator and metering circuits, and final assembly details are provided in this third article in the series.

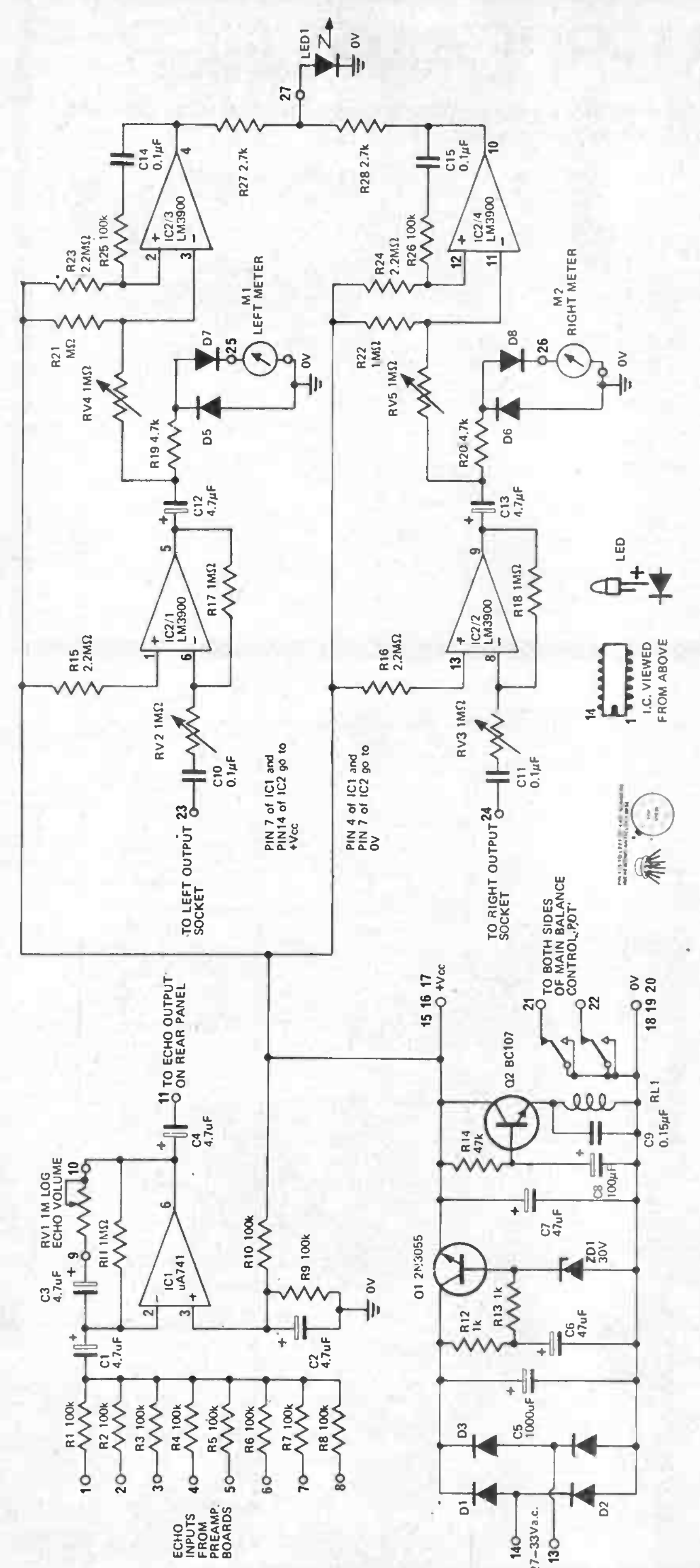


Fig. 1. Circuit diagram of power supply and metering board

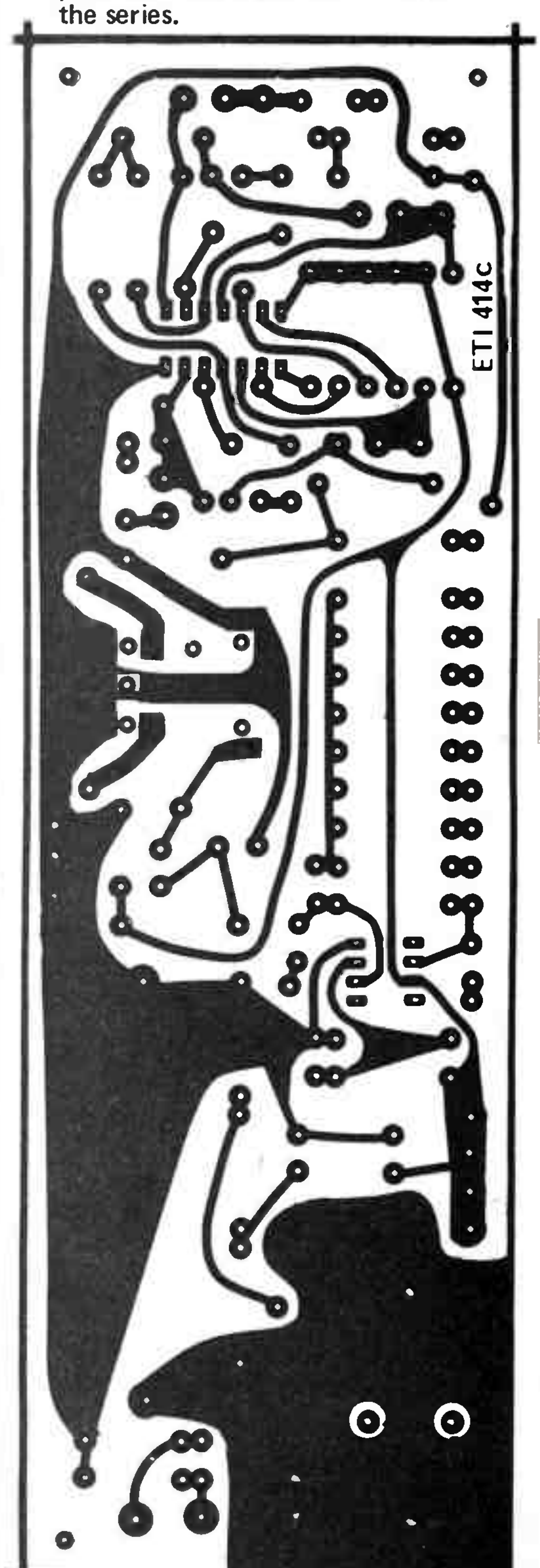
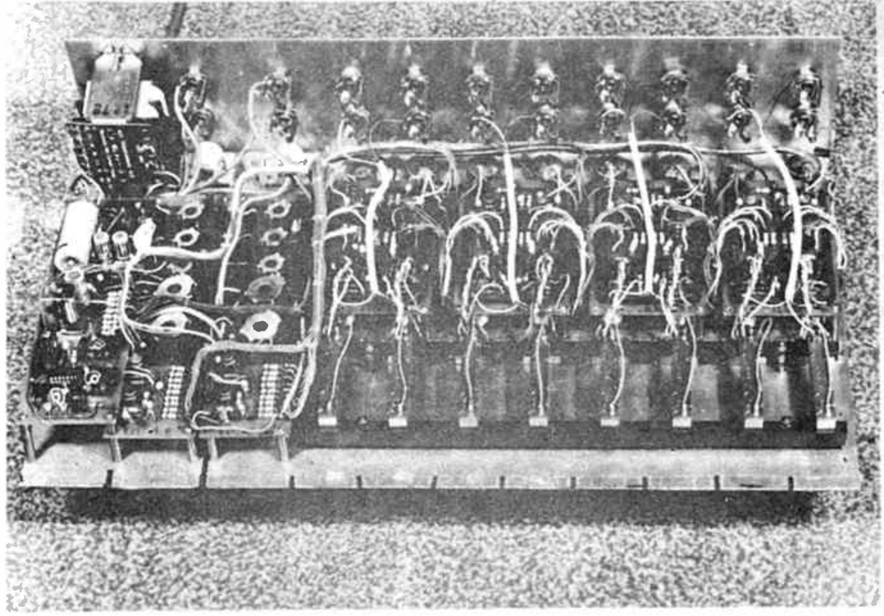


Fig. 2. Foil pattern for power supply board

MIXER



ASSEMBLING COMPONENTS

Construction should be commenced by assembling all relevant components on to the power supply printed circuit board. This should be done following the component overlay (Fig. 3) shown below.

Make sure that the integrated circuits and tantalum capacitors are fitted to the printed circuit boards the correct way round (refer to the small component sketches inset on the circuit diagram shown on page 78).

Use care when soldering to avoid damaging the components by excess heat – especially integrated circuits. Use a light-weight, low-wattage soldering iron and work quickly.

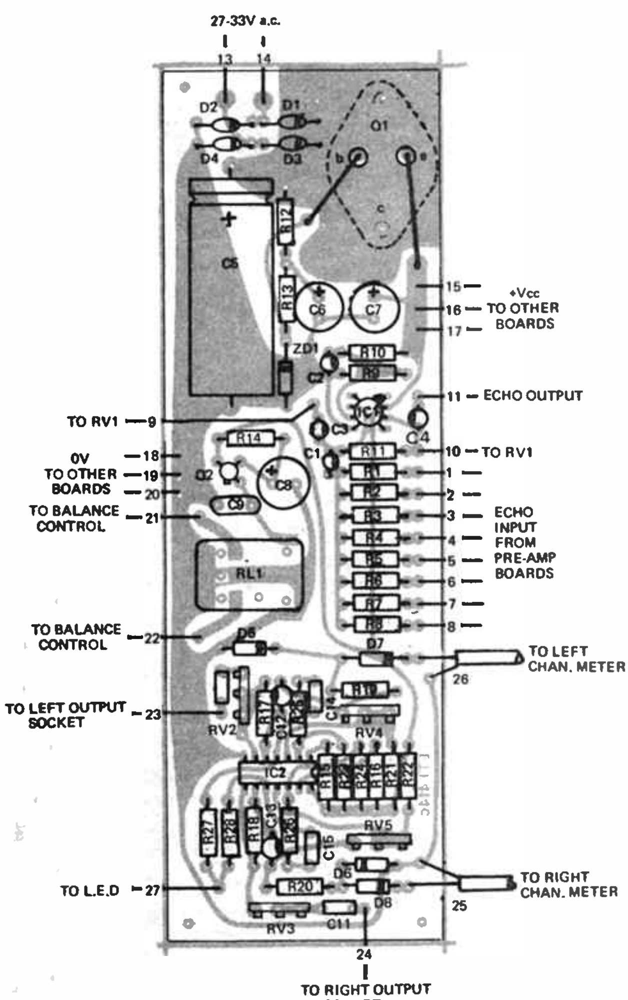


Fig. 3. Component overlay of power supply board.

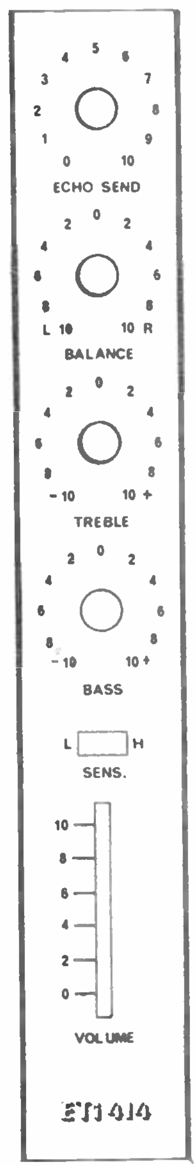


Fig. 4. Escutcheon for preamplifier (actual size 12" x 1¾").

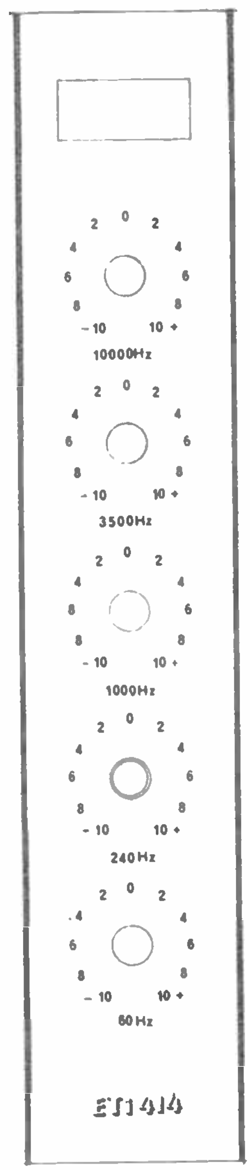


Fig. 5. Equalizer panel escutcheon (actual size 12" x 2¼")

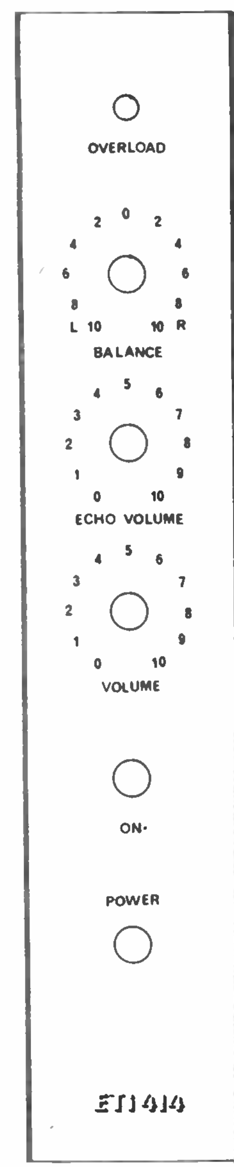


Fig. 6. Main control panel escutcheon. (actual size to be 12" x 2¼").

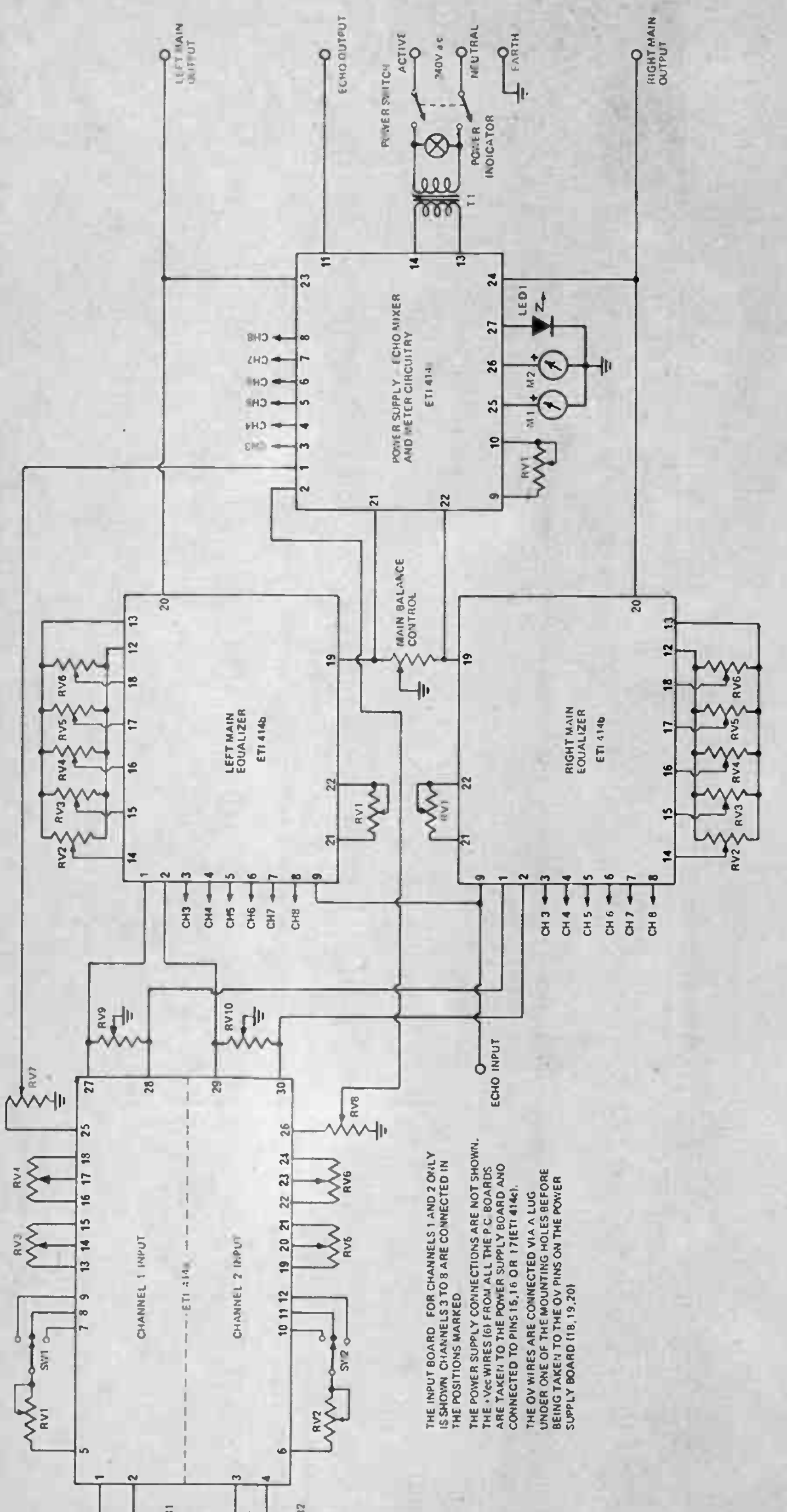


Fig. 8. Main interconnection diagram.

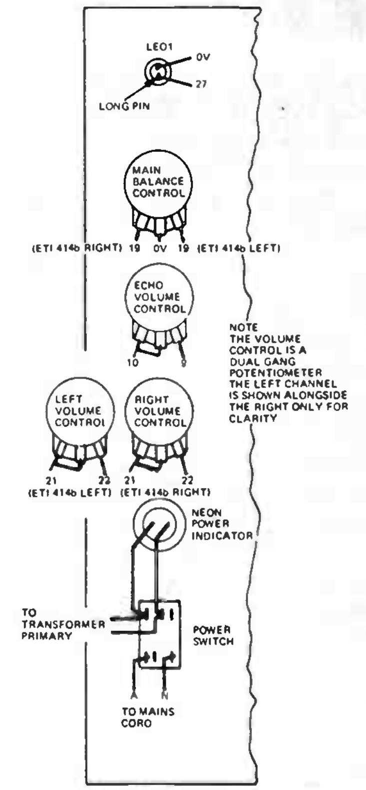


Fig. 7. Wiring to rear of main control panel.

Printed circuit boards purchased from kitset suppliers may be varnished with resin or similar. Clean off the varnish where the 2N3055 regulator transistor is mounted, to allow electrical contact. Silicon grease should be used between the copper pattern and the transistor to aid heat transfer.

The pins of the relay are inserted into the holes provided in the board and bent to make contact with the copper tracks before soldering. We inserted pins to allow connection of the positive and negative supply leads which have to be routed to the various other boards.

There are three pins for positive leads and three for negative leads. Two leads connect to each positive pin (six leads total). The common leads from the four preamplifier boards and the two main mixer boards, are soldered to lugs secured between each respective board and one of its mounting pillars. Two of these leads are terminated at each negative pin on the power supply board.

By referring to the metalwork drawings and the photograph of the unit, boards and other components

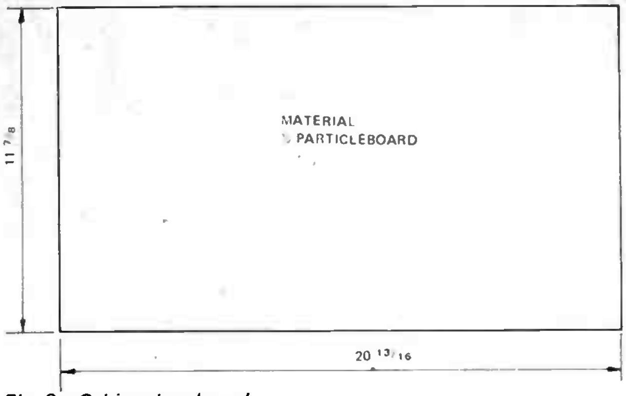


Fig. 9a. Cabinet baseboard.

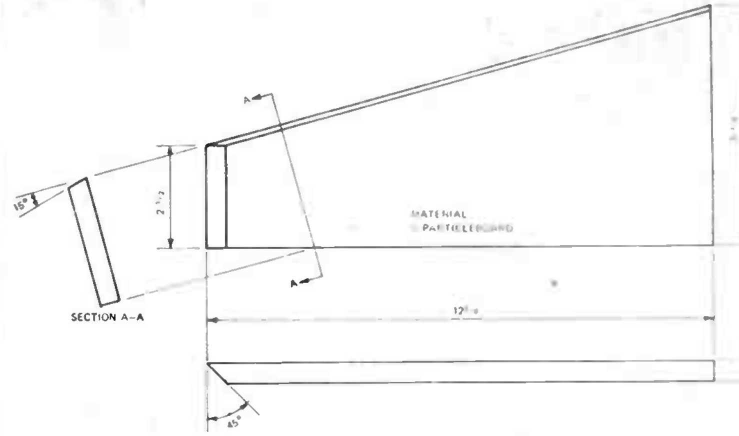


Fig. 9d. Cabinet sides — two required. Note that boards should be mirror image of each other, that is, chamfers should have opposite slope.

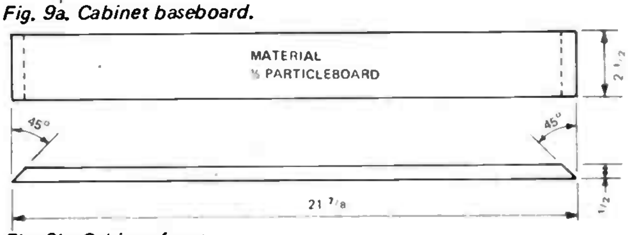


Fig. 9b. Cabinet front

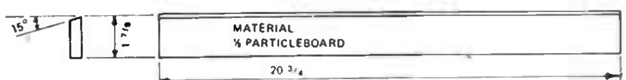


Fig. 9c. Front panel support

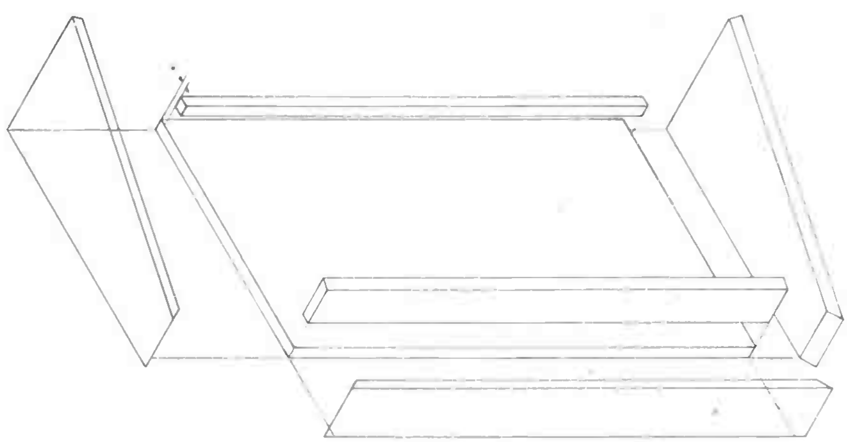


Fig. 10. Cabinet assembly details.

may be mounted to the panel in the following order.

1: Each preamplifier board is mounted on three 1" long threaded pillars. The main mixer – equaliser boards each employ four of these pillars which should be secured to the front panel with countersunk screws.
2: Mount the VU meters, with countersunk screws.
3: Mount the sensitivity switches.
4: The slide potentiometers are mounted on two rails, each of which is spaced from the chassis by four, ¾" long threaded pillars – eight in all. Ensure that pin 1 of each potentiometer is orientated towards the front of the panel.
5: Glue on the escutchions with contact cement and mount the rotary potentiometers, switches and indicator lights.

Note: Two of the escutchions will have to be drilled to allow the front panel to be secured (see the metalwork diagram).

6: Mount the input jacks on the rear of the panel.
7: Mount the transformer and the printed circuit boards.

This completes the front panel assembly and we can now make the interconnections.

WIRING THE UNIT

The interboard wiring should be carried out with reference to the underchassis photograph and to the interconnection diagram, Fig. 8.

All wiring should preferably be colour coded and should be routed down one side only of each board so that the board may be swung-up, sideways, if servicing is required at some later date.

Use one mil plastic tubing, or lacing twine, to tie the wiring into looms. This, as well as improving the appearance of the unit, also facilitates servicing.

Leads to the VU meters, output sockets, echo input and output sockets, and the main balance control must be in shielded cable. These and, as far as possible, all other wiring should be kept well clear of the mains transformer to prevent hum pickup.

WOODWORK

Cut the five pieces shown in Fig. 9 from ½ inch particle board, note that the two pieces cut as per Fig. 9d are mirror images of each other. Veneer the inside surfaces of the two sides (Fig.9d) and the front strip (Fig. 9b).

Assemble the box as per Fig. 10. Screws or nails should be used to hold the panels together while the glue sets. Take care to ensure that the sides are square to the base, otherwise the metal panel may not fit in place. In fact it is a good idea to use the panel as an assembly guide. The support piece (Fig. 9c) is assembled with the short side to the front. The rear panel support is merely a half inch square piece of timber, positioned 3/8 inches from the rear edge of the base (Fig. 10).

When the glue is set, the box can be sanded and all visible outside surfaces veneered, before final sanding and finishing operations are carried out. The inside of the box should be lined with "Alfoil", and this earthed to the metal chassis. If the Alfoil goes over the rear panel support, the metal panel will make contact with it and no other connection need be used.

TESTS AND ADJUSTMENTS

Before initially switching on, remove from the power supply board the +Vcc

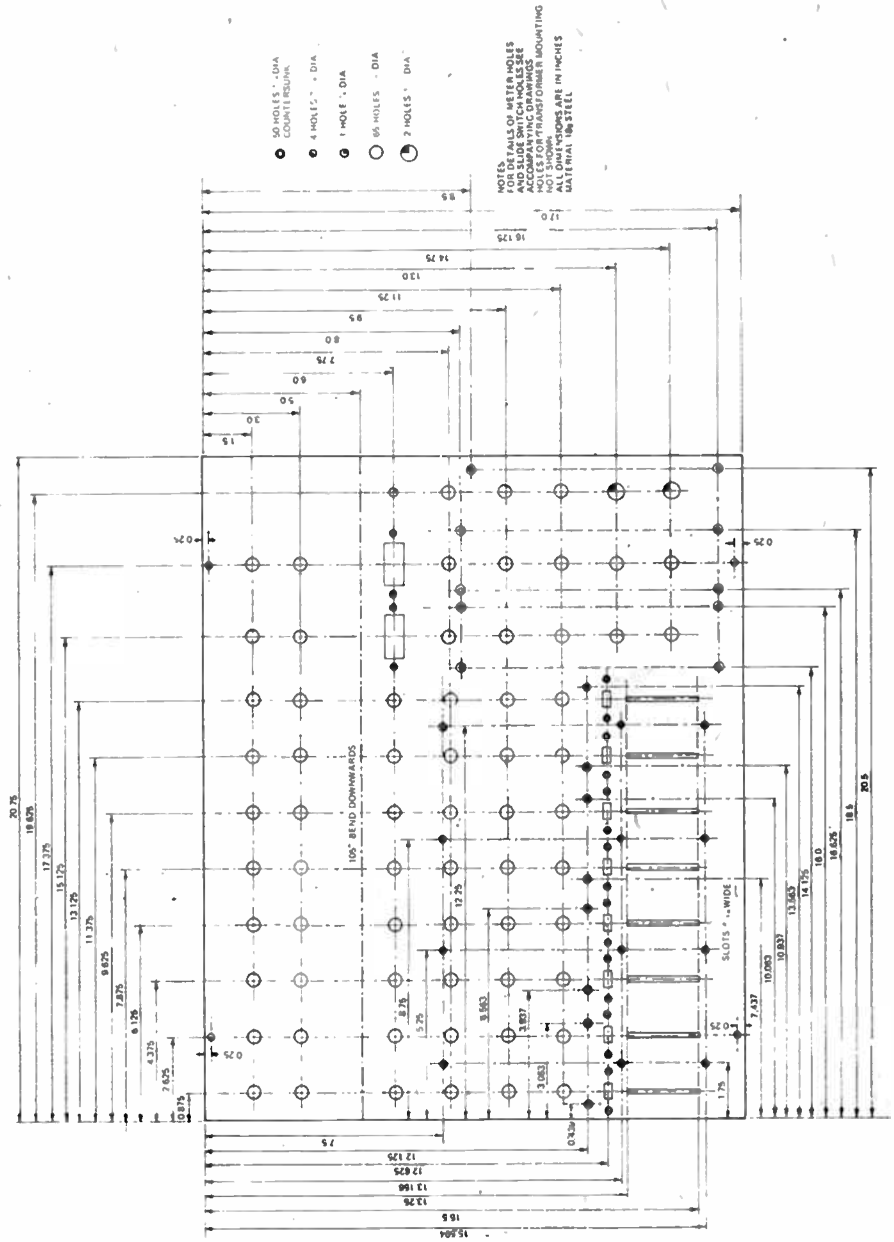


Fig. 11a. Drilling details of front panel.

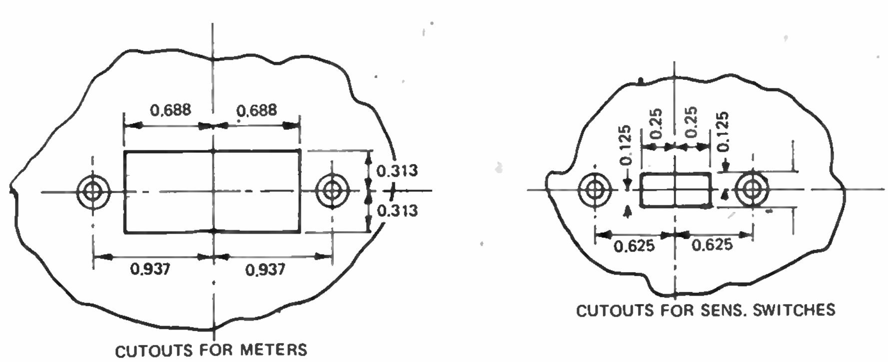
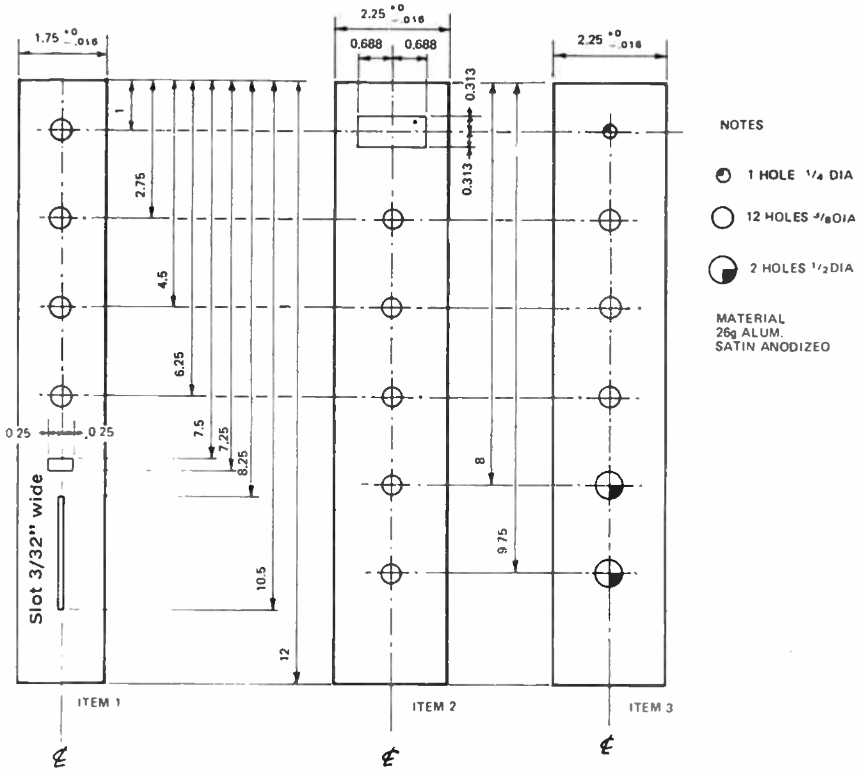


Fig. 11b. Cutouts for meters and sensitivity switches.

Fig. 12. Drilling details – Item 1: preamplifier panel. Item 2: equalizer panel. Item 3: main control panel.



PARTS LIST FOR POWER SUPPLY BOARD ETI 414

R1	Resistor	100k	½W	5%
R2	"	100k	"	"
R3	"	100k	"	"
R4	"	100k	"	"
R5	"	100k	"	"
R6	"	100k	"	"
R7	"	100k	"	"
R8	"	100k	"	"
R9	"	100k	"	"
R10	"	100k	"	"
R11	"	1M	"	"
R12	"	1k	"	"
R13	"	1k	"	"
R14	"	47k	"	"
R15	"	2.2M	"	"
R16	"	2.2M	"	"
R17	"	1M	"	"
R18	"	1M	"	"
R19	"	4.7k	"	"
R20	"	4.7k	"	"
R21	"	1M	"	"
R22	"	1M	"	"
R23	"	2.2M	"	"
R24	"	2.2M	"	"
R25	"	100k	"	"
R26	"	100k	"	"
R27	"	2.7k	"	"
R28	"	2.7k	"	"
RV1	Potentiometer	1 Meg Log		
RV2	"	1 Meg Trim type		
RV3	"	1 Meg " "		
RV4	"	1 Meg " "		
RV5	"	1 Meg " "		
C1	Capacitor	4.7µF 35V "TAG" Tantalum		
C2	"	4.7µF 35V "TAG" Tantalum		
C3	"	4.7µF 35V "TAG" Tantalum		
C4	"	4.7µF 35V "TAG" Tantalum		

C5	"	1000µF 50V Electrolytic Pigtail leads
C6	"	47µF 50V Electrolytic PCB mounting
C7	"	47µF 50V Electrolytic PCB mounting
C8	"	100µF 50V Electrolytic PCB mounting
C9	"	0.15µF 100V Polyester
C10	"	0.1µF 100V "
C11	"	0.1µF 100V "
C12	"	4.7µF 35V "TAG" Tantalum
C13	"	4.7µF 35V "TAG" Tantalum
C14	"	0.1µF 100V Polyester
C15	"	0.1µF 100V "
D1-D4		Diodes PA2121, EM401, or similar
D5-D8		" 1N914
LED1		NSL5023
ZD1		Zener Diode BZY88C30
IC1		Integrated Circuit LM307, µA741 (metal can or minidip)
IC2		Integrated Circuit LM3900 (National Semiconductor)
Q1		Transistor 2N3055
Q2		" BC107
PC board		ETI 414c

Level indicator edge meters 400µA 410ohms, type PV31 or similar (two required).
Brass spacers, ½" by 1/8" clearance hole (three required).
Brass spacers, 1", tapped 1/8" (two required).
Phone jacks 6.5mm (two required).

Transformer 240V primary 27-33V secondary —200mA.
Power switch type MSP625 dpdt (or similar).
Neon indicator 240V (chassis mounting).
Three-core flex and plug, nuts, bolts, etc.
Relay 1250 ohm, miniature type VP2, two change-over contacts.

wires leading to the preamplifier and mixer boards making sure they cannot touch other circuitry. Rotate the trim potentiometers to their mid position and switch on. Check the voltage between the Vcc and OV terminals. This should be between 27 and 32 volts. If not, there is a fault in the supply which should be located before proceeding further.

Using an oscillator, feed a signal into the output socket of the left channel. An indication should be visible on the left hand meter. Set the input level to that required to drive the power amplifier to full output (1V for the ETI 413 amp.), and adjust RV2 to give full scale deflection. Now adjust RV4 to the point where the LED just stops flashing. Now repeat the process for the right channel, adjusting RV3 for full scale deflection and RV5 for LED indication. This completes the metering circuit calibration.

Now connect the equalizer boards and one of the preamplifier boards. This preamplifier can be checked either with an oscillator or a microphone. Check that the gain increases when the sensitivity switch is moved to the right, also that the tone controls give maximum boost when moved clockwise. Make sure that the balance control operates correctly and the wires going to the mixers have not been crossed.

Add the other preamplifiers one at a time testing each as above. When all the above procedure is complete the unit is ready for operation.

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All advertisers and advertising agents are advised to study those provisions very carefully as heavy penalties are imposed.

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FALSELY represent that goods or services are of a particular standard, quality or grade, or that goods are of a particular style or model.

FALSELY represents that goods are new.

REPRESENT that goods or services have sponsorship, approval, performance characteristics, accessories, uses or benefits they do not have.

REPRESENT that he or it has a sponsorship, approval or affiliation he or it does not have.

MAKE false or misleading statements concerning the existence of, or amounts of, price reductions.

MAKE false or misleading statements concerning the need for any goods, services, replacements or repairs.

MAKE false or misleading statements concerning the existence or effect of any warranty or guarantee.

Penalty for an individual is $10,000, or 6 months imprisonment; for a corporation, $50,000.

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In cases of doubt, consult your lawyer.

ELECTRONICS TODAY INTERNATIONAL

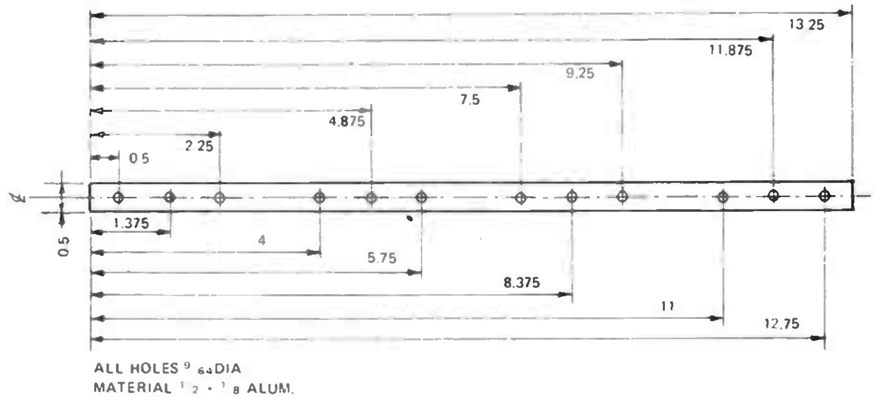


Fig. 13. Slide potentiometer support bars (two required)

HOW IT WORKS

MASTER MIXER POWER SUPPLY

The power supply is of conventional design. Any transformer which will supply 27 to 33 volts at 200mA will suffice. The regulator employs a 2N3055 as a series regulator, and by virtue of the 30V zener diode between the transistor base and the negative rail, maintains the output voltage at approximately 29.5 volts.

At switch-on Vcc rises immediately but the output of the unit is shorted out by relay RL1 for approximately four seconds while C8 charges exponentially via R14. Transistor Q2 is simply an emitter follower driving relay RL1. The voltage at its emitter is approximately 0.5 volts less than that on capacitor C7. After approximately four seconds the voltage across the relay rises sufficiently to activate it, removing the short from the output.

This prevents accidental damage to power amplifiers due to switching transients or other warm-up anomalies.

ECHO MIXER

The echo mixer is straight forward. As indicated earlier there are eight separate inputs. These receive signals from the input channel echo-send controls. The gain of the echo amplifier is controlled by RV1 which varies the negative feedback. The output goes to the echo output socket on the rear panel. From here it is intended to pass through an echo tape, reverberation unit, or similar type of device before returning to the unit and being split equally to provide an input to each main mixer stage.

METERING CIRCUITS

The metering and overload indicator circuits employ a quad-amplifier IC type LM3900 from National Semiconductor. This package accommodates four independent, internally compensated amplifiers which are designed to operate from a single power supply voltage and to provide a large output voltage swing. Each amplifier makes use of a "current mirror" to provide the non-inverting input.

Unlike a normal operational amplifier, the two inputs are current driven, not voltage. This means that when used as an amplifier the output tries to balance the current in the two inputs. Therefore an initial bias is required. This is provided by R15. For the amplifier to be balanced, an equal current must flow in R17. This sets the quiescent output voltage to approx. 15V.

The ac voltage gain is equal to R17/RV2 where RV2 is the preset value of RV2. The meter is driven by R19 and rectified by D5 and D7.

The second stage (IC2/3) is a comparator-monostable. Both inputs of this amplifier are biased from the supply rail although the current is higher into the negative input. Since this is outside the linear region the output is almost at 0V. When in use current is being added and subtracted to the current into the negative input.

If enough current is subtracted, such that it is less than the current into the positive input, the output of the IC will go high. Due to the positive feedback of R25 and C14 the IC will stay in the high state for approximately 0.1sec, even if the initiating signal has ceased. The overload light LED1 is on while either monostable (IC2/3 or IC2/4) is high.

If the output is continuously high the light will flash rapidly.

Two of these amplifiers are employed in each of the metering indicator circuits. A variable resistor in series with the input to the first amplifier allows zero VU to be adjusted for outputs in the range of 100mV to 3V.

If a single transient exceeds a preset level the indicator light will flash for approx 100 ms. This will allow the "transient" to be seen and thus act as a warning. On a continuous overload the light will flash rapidly. With the ET1413 amplifier this level should be approx IV rms.

How to use the master-mixer in the most effective way — and how to modify it to suit individual requirements.

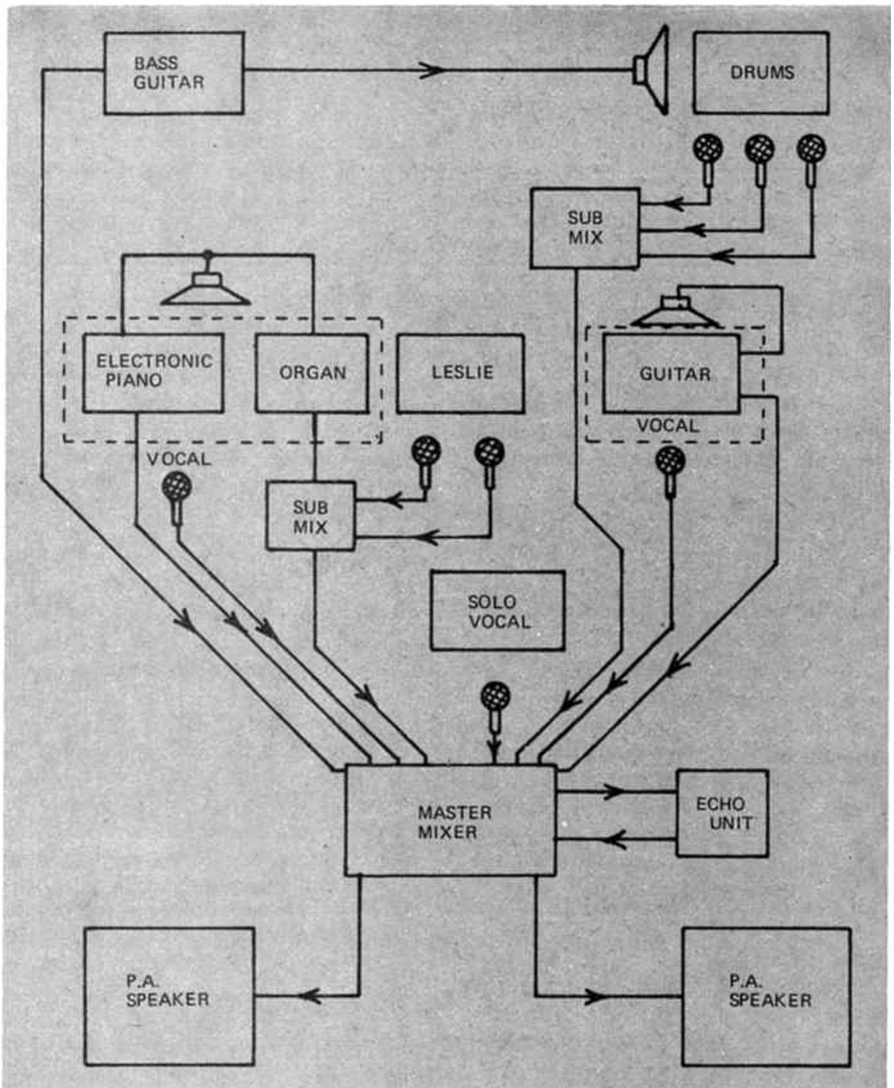


Fig. 1. Arrangement of average-sized group

USING THE UNIT

When you have built the ETI Master-Mixer, you will wish to use it in the most effective way, and perhaps modify its performance to suit individual requirements. We cannot possibly cover all eventualities, but here are some details of a typical installation and some commonly-needed alternative configurations.

BASIC PHILOSOPHY

The unit has been designed to provide master-mixing for the average sized group (which is usually similar to that shown in Fig. 1). It provides a stereo output which may be used to drive the main amplifiers for an auditorium, or may be used for recording purposes. We have taped major performances using our own prototype master-mixer and have achieved very pleasing results indeed. Remember however, that a system configuration suitable for recording is not necessarily suitable for auditorium use and vice versa.

Basically the unit should be located in the auditorium so that the operator may judge acoustic quality as the audience hears it — and to make appropriate adjustments as necessary.

Most groups nowadays use half acoustic and half electronic instruments. Instruments such as drums may not need 'miking' at all except in a very large auditorium or out-of-doors. Naturally when making recordings, all instruments have to be 'miked'. In such cases four microphones are usually needed adequately to cover the drums and these are best combined in a sub-mixer. Similarly, an electronic organ with Leslie is perhaps best handled by a sub-mixer. All other inputs will of course go direct to the master mixer.

One of the main problems within the group is that of monitoring. Each player of an electronic instrument needs to be able to hear himself and the drummer particularly needs to hear the bass guitar but there is so much noise on stage that this is usually not possible. As each player usually has his own amplifier/speaker for use in practice, these may be used on stage to provide the necessary monitor facilities. To split the instrument output for both monitor amplifier and master mixer a simple plug to twin socket adapter may be used. Another method is to use a separate monitor box, as shown in Fig. 2, or monitor outputs may be fitted to the mixer unit itself as explained later.

It is of course posssible to 'mike' the output of monitor speakers but this usually results in loss of fidelity. On the other hand such a procedure, together with deliberate overloading, is often used to provide special effects by distorting the output.

RECOMMENDED MICROPHONE
Low output (magnetic) microphones are not recommended as noise problems will be encountered. It is strongly recommended that the 'Electret' type microphone be used.

SETTING UP THE MIXER

Before connecting any inputs, set each input channel sensitivity switch to low, volume controls to zero, tone controls to centre position.

Switch on, connect the instruments one at a time and perform the following adjustments. Adjust both master-volume and channel volume to position 7 and then switch channel sensitivity for maximum desired level at these settings.

Then adjust the tone controls for the

MASTER MIXER

nicest sound for each instrument — without destroying its natural sound. Bear in mind that to increase the response in mid-range it is necessary to turn down bass and treble and turn up the volume.

If echo is to be used, connect the 'Echo Send' input and output to an echo unit such as the 'Echolette', or alternatively, to a suitable reverberation unit such as the ETI project 424 "Spring Reverberation Unit". The echo effect may be increased or decreased by using the echo-send control.

Audibly position each member of the group left or right, by adjusting his channel balance control. Note that a balance control at centre will make the instrument appear audibly centred as well. These controls may need some readjustment when the full group is playing. The master balance control is then adjusted to achieve overall uniformity.

The equalizers may now be used to obtain a level overall frequency response by subjective listening and appropriate adjustments. Note that a five-section equalizer cannot correct major defects in auditorium acoustics, but can compensate for minor problems and for poor quality speakers.

As said before, the unit may be used for recording on stereo tape or disc and this is done by taking direct line outputs from the mixer to the recording equipment. Again, as said before, all instruments need to be 'miked'. Remember that the quality of the acoustics, particularly when recording, is affected very much by the choice of microphone. Most dynamic microphones drop off at the high end and we suggest that, providing sufficient funds are available, a good Electret microphone (such as the Sony ECM 22P) be used. It is essential that microphones should be as directional as possible to avoid problems with acoustic-feedback.

MODIFYING THE SYSTEM

Innumerable individual variations may be required — a few of those most commonly requested are dealt with here.

Some of these modifications can be performed without changing the basic wood and metal-work, others cannot. Because of the variety of combinations that may be used, details of wood and metal-work must be left to the individual constructor.

These modifications are therefore of necessity presented in a general way and should only be undertaken after careful consideration of exactly what is needed, and only if what needs to be done is fully understood. We regret that we cannot assist in individual design requirements, however do tell us about your requirements and problems, and, if sufficient people ask for the same thing, we may be able to publish details of a modification at some later date.

Before dealing with specific modifications we will expand on the general theory previously given so that limitations may be more readily understood.

PREAMPLIFIERS

With reference to the circuit diagram on page 71, we see that the input amplfiier ICl has three selectable gains, the maximum gain being 500. This means that a one millivolt signal will become 500 millivolts at the output. A higher gain may be obtained by reducing the value of R4/R6 but to maintain input impedance R1/R2 will have to be increased (see How it Works — Preamplifier page 69 for gain formula). Note however that the tone-control stage is a standard feedback-type providing a maximum boost of 15 dB which corresponds to a voltage gain of approximately 6. The maximum output voltage of IC2 is 6 volts RMS and the maximum output of the preamplifier must therefore not exceed IV RMS if clipping under maximum boost conditions is to be avoided. In addition an overload margin of 20 dB should be allowed, and this implies a maximum nominal output of only 100 mV from the preamplifier.

MIXER AND EQUALIZERS

The mixer is simply a summing amplifier, the output voltage being the vector sum of the input voltages multiplied by the resistance of RV2 divided by 100,000. The maximum gain, one channel only driven, is 3 1/3 and although the individual gain remains constant the power level is greater with all channels driven. Overall gain is controlled by RV1, the master volume control.

Each section of the equalizer is a series LCR filter whose sharpness is determined by the circuit Q and with the coils given, the reactance at resonance is approximately 700 ohms. If more than five sections are required the filter must be made sharper and hence the reactance of the capacitor and inductor must be increased. Note however that phase shift problems limit the number of sections to seven in this type of circuit.

POWER SUPPLY

The current consumption is approximately 10 mA per channel and the power supply has adequate reserve for up to 20 channels, however if more than 10 channels are used a heatsink of about four square inches should be added to Q1.

If meter and overload indicators are required for each channel then a printed circuit board with this section only wired up should be made for each channel. If each channel is required to have a separate LED overload indicator, separate R27 and R28 (Fig. 1 page 72) and use each resistor to drive an LED.

CHANGING THE NUMBER OF CHANNELS

If less channels are required it is simply a matter of deleting the appropriate number of preamplifier/tone control boards and fitting blank panels to the cabinet in

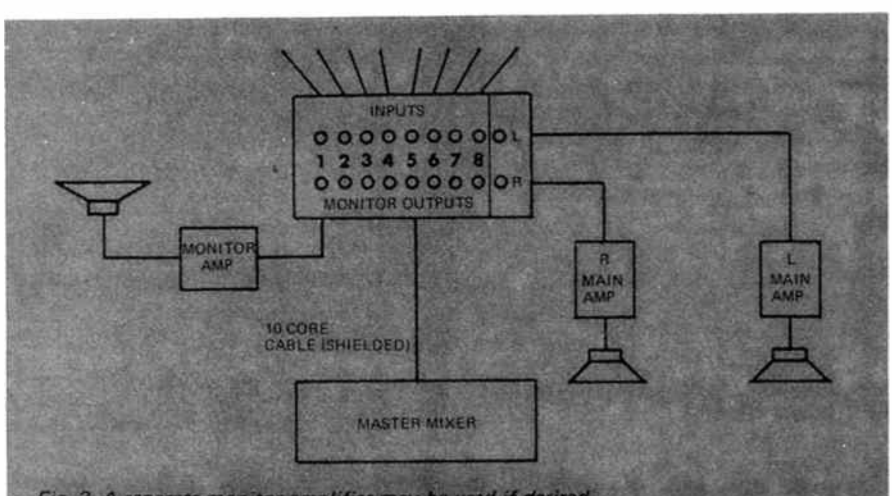


Fig. 2. A separate monitor amplifier may be used if desired.

their place.

If more channels are required, the existing metalwork and woodwork will have to be extended to accommodate the extra preamplifiers.

One 100k resistor must be added to the main mixer summing network for each additional channel. These may be mounted by glueing them to the existing resistors with epoxy cement and making flying lead connections. Alternatively a small sub-board may be constructed for them.

In an exactly similar manner the echo mixer may be modified to accommodate the required extra channels. Extra input sockets must also be provided and the appropriate interwiring carried out.

SUB-MIXERS

As discussed earlier, sub-mixers may be required to implement a complete system. A simple sub-mixer may be constructed using the circuit shown in Fig. 3. This circuit is quite simple, is based on the echo mixer, and may be built on veroboard. Alternatively the echo-mixer PC board could possibly be adapted fairly readily.

As the instruments associated with each sub-mixer are usually grouped left-and-right, splitting may be performed after the sub-mixer as shown in Fig. 3. If balance is required before mixing it will be necessary to use two sub-mixers controlled by a ganged potentiometer, and to use balance circuitry similar to that in the circuit on page 68. The outputs of the sub-mixers are taken to the normal inputs of the main mixer.

MONITOR OUTPUTS

The need for monitoring has been explained previously, and if only one monitor channel is required, and echo is not required, the echo channel may be used to provide a monitor output. However two or more monitor outputs are often required and they may need to each have an equalizer for the elimination of microphone feedback.

This may be achieved by wiring additional potentiometers in parallel with the echo potentiometers as monitor level controls. The output from these potentiometers may then be fed directly or via additional equalizer/main-mixer boards to the monitor amplifiers. A balance control is not required on monitor, hence R21 and RV7 (page 68) may be omitted and the output taken from terminal 19. Again, if equalization is not required, a mixer similar to that of Fig. 3 may be used.

CUEING OUTPUTS

When recording it is sometimes necessary to suppress the main output of the mixer while still monitoring the final mixed sound.

This may be done quite simply by taking an output from the junction of R20 and C8 (page 68.) of the final mixer to a cue-monitor outlet, and using a good-quality key switch to short terminal 19 to ground.

This allows monitoring of equalizer output whilst inhibiting output to the main amplifier.

That completes our project. We trust that this versatile unit helps you become a good mixer! ●

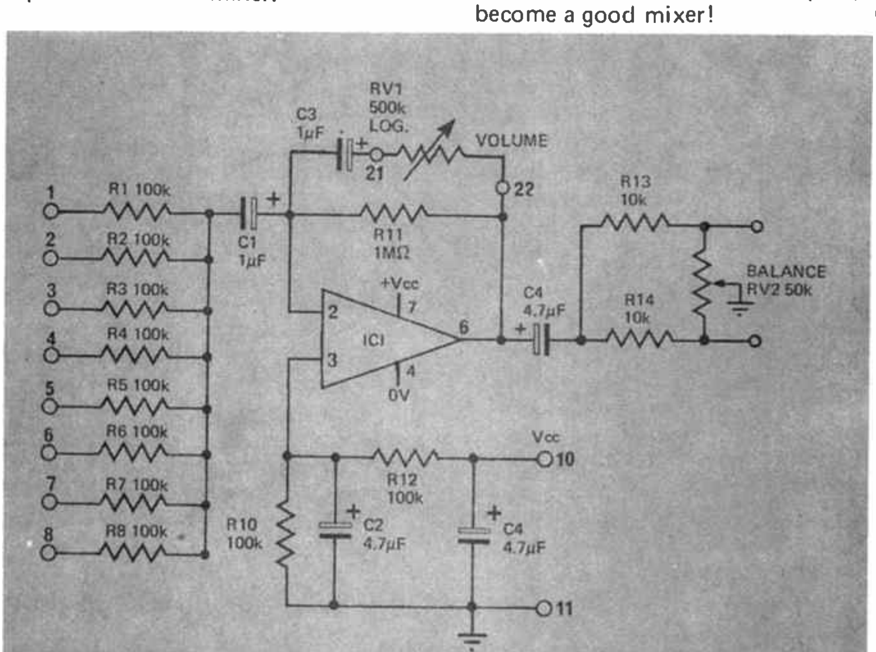
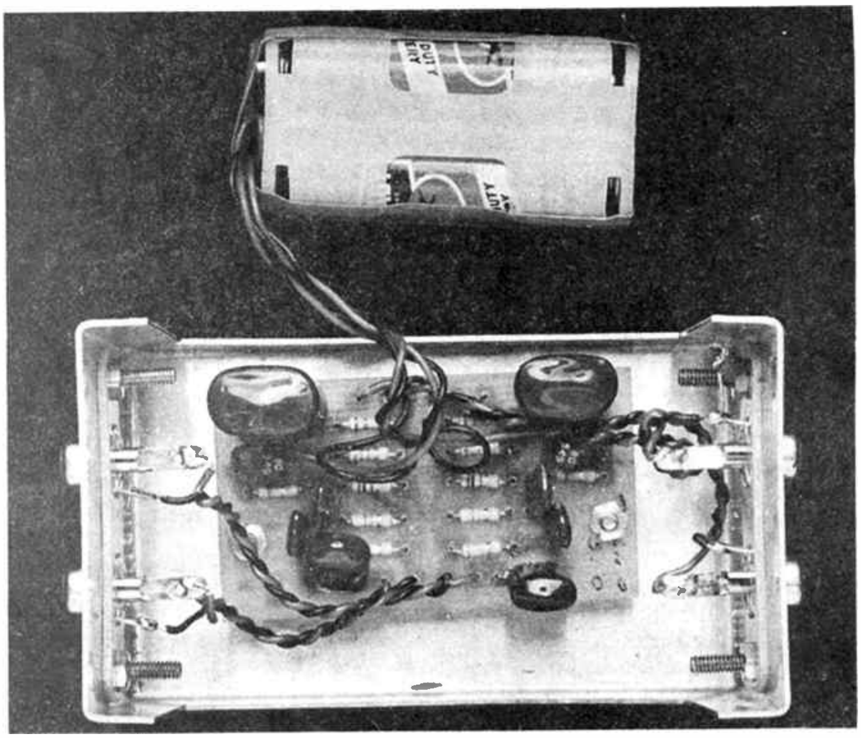


Fig. 3. Simple sub-mixer

STEREO RUMBLE FILTER

PROJECT 426



This internal view shows how the rumble filter is assembled.

Active filter design improves clarity of bass reproduction.

IN BYGONE DAYS rumble filters were very popular because even the best of turntables, used then, generated considerable vibration due to bearing and motor deficiences. These vibrations, mechanically transmitted to the pickup cartridge, resulted in an audible output. Hence high-pass filters were often incorporated in amplifiers to reduce this objectionable rumbling sound to an acceptable level, and as bass response seldom extended below 50 Hz, a simple RC filter with 6 dB per octave roll-off below 50 Hz was considered adequate.

Modern turntables have far smoother bearing and drive arrangements than their early counterparts – and for this reason many amplifier manufacturers no longer include a rumble filter facility.

Those that do are rarely satisfactory. Their slope is generally inadequate and the main effect of switching them in is to roll off the low-frequency response to the detriment of programme content.

At first sight it would seem better to exclude the rumble filter altogether and just make sure that our turntables do not generate any appreciable rumble.

Surprisingly perhaps, a rumble filter is still very much required and if designed correctly can make an appreciable improvement to reproduction – even when used with turntables that generate no rumble at all!

The reason why will be clearly apparent if you take the front grille

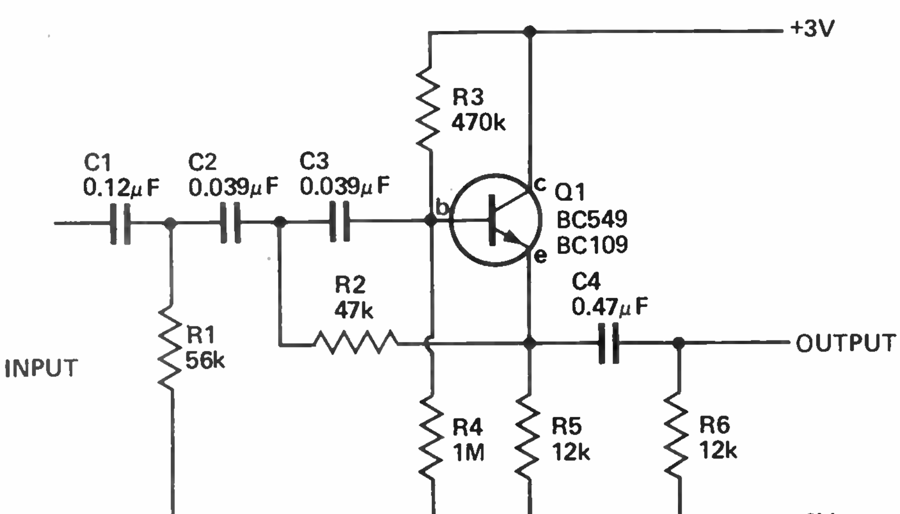


Fig. 1. Circuit diagram of the rumble filter. Two required for stereo.

HOW IT WORKS

The filter consists of three separate sections:–

1. A passive RC filter consisting of R1 and C1.
2. An active filter comprising C2, 3, R2, 3, 4 & 5 and Q1.
3. A passive filter comprising C4 and R6.

The active filter (from input of C2 to output to C4) is a standard design with the exception that values have been selected to give a peak in the response at the cut-off frequency. The maximum lift is about 2 dB and this characteristic, combined with those of the two RC filters, gives a sharp knee to the roll-off. The composite filter has a lift of 0.2 dB before turning over sharply.

Thus low frequency response is maintained substantially flat down to 50 Hz and is only 2 dB down at 40 Hz. Thereafter the response drops very rapidly and is in excess of 30 dB off one of your speakers and — with the phono-cartridge tracing a section of record that has no recorded content (or very low level content) — turn the volume control up fairly high. You will almost certainly find that the cone of the bass driver is making wild excursions to and fro, probably at frequencies between 5 Hz and 15 Hz.

So it's sub-audible — why then does it matter?

Well it really does — and we'll explain just why later in this article — but first let us consider just where this 5 Hz — 15 Hz content comes from.

Firstly, modern turntables and arms have mechanical resonances lying within the 5-15 Hz region. Secondly, stereo cartridges are sensitive in the vertical as well as horizontal planes and will respond to uneveness in record or turntable surfaces. They will also respond to a defect in the record surface known as pressing rumble.

In addition the noise finds its way onto the record during the actual recording process. This recorded noise is due to LF noise and rumble sometimes being induced in the recording lathe by seismic disturbances, and by vibration in drive gears and cutting head carriage rails.

Lastly vibration of a low frequency nature, due to people walking past the turntable or vehicles passing by outside, may well excite the turntable and arm resonances even though the turntable is reasonably well sprung.

WHY SUB-AUDIBLE NOISE MATTERS

This very low-frequency noise is responsible for a remarkable amount of intermodulation distortion which generally makes the bass sound muddy. In extreme cases it may cause the reproduction to sound as if speaker cone break-up is occurring. The reasons for this are as follows.

Preamplifier stages usually have two or three transistors around which large negative feedback is applied for equalization and/or tone control. At sub-audio frequencies these feedback networks are not generally effective. Thus the LF signals may well receive considerably more amplification in the preamplifier than would normally be expected. Secondly although the magnitude of the LF signal may not itself be sufficient to overload the preamplifier, the combined LF and music signals may well cause the preamplifier to clip. Even if clipping does not occur the LF signal will cause intermodulation distortion despite the fact that the LF signal is inaudible!

Most modern power amplifiers are quite capable of amplifying this noise signal, presenting it to the loudspeaker at a surprisingly high power level. The speaker itself has very little acoustic loading at these low frequencies and the cone will thus move considerably and may even be driven beyond its linear excursion region. Even if not actually overdriven, the presence of such large cone excursions will produce a high level of intermodulation distortion.

Whilst elimination of factors causing the noise is by far the best procedure, a lot of these factors are completely beyond the control of the average hi-fi owner. Hence a rumble filter would seem to be the obvious answer. But, we do not want to sacrifice any low frequency response and we want signals in the offending 5-15 Hz region to be attenuated as far as possible — two apparently conflicting requirements. In addition, as LF noise cannot be allowed to enter the equalization stages of the preamplifier, down below 15 Hz where most LF noise occurs.

Current drain of the two filters is only 100 μA and the batteries will last their normal shelf life of about 12 months, thus no power switch is required. Batteries should be replaced annually.

SPECIFICATION

Input Impedance (rises below 50 Hz)	47k
Output Impedance	< 5k
Input voltage (maximum)	250mV
Cut-off Frequency (−3dB)	36 Hz
Cut-off Slope (maximum)	24dB/octave
Attenuation at 10 Hz	37 dB
Gain at 1 KHz	−0.2 dB.

PARTS LIST ETI 426

R1	Resistor	56k	¼W	5%
R2	"	47k	"	"
R3	"	470k	"	"
R4	"	.1M	"	"
R5,6	"	12k	"	"
C1	Capacitor	0.12μF	polyester	
C2,3	"	0.039μF	"	
C4	"	0.47μF	"	
Q1	Transistor	BC109,BC549 or similar		

* for stereo 2 off each of the above parts are needed.

PC board ETI426
2 dual RCA sockets (or whatever is on your table)
2 dual AA size battery holders or one 4 way holder.
4 AA size batteries.
2 1/8" x 1/2" bolts and nuts
2 8mm long spaces
1 mini box type AMB7 or similar (58 x 58 x 100mm).

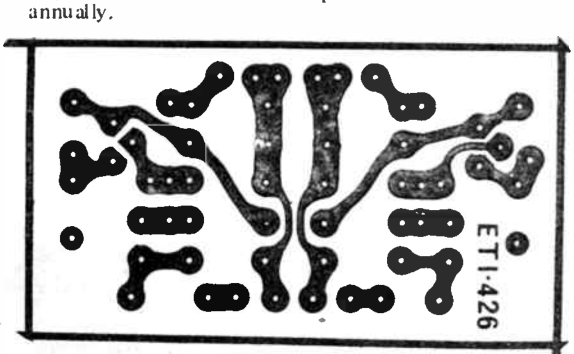
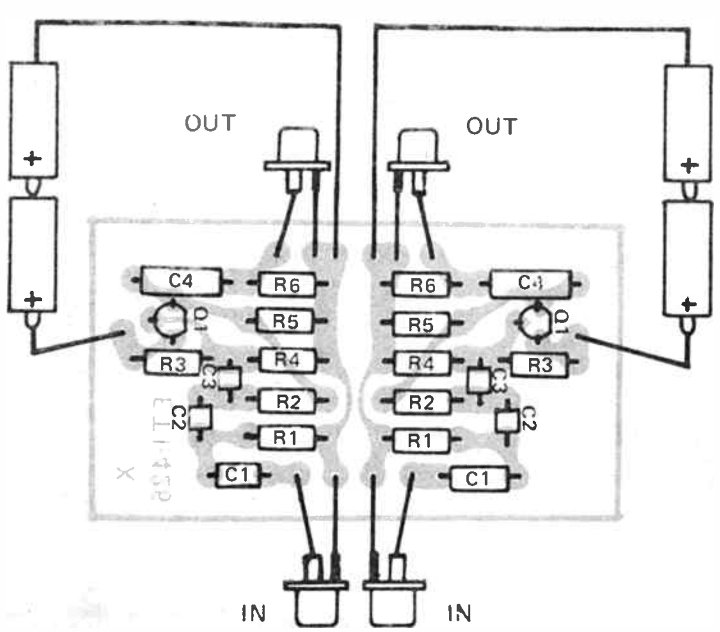
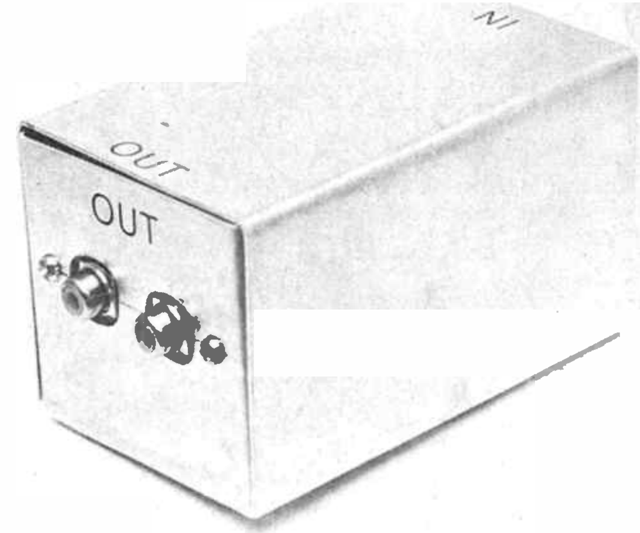


Fig. 2. Printed circuit board layout for the rumble filter 40mm x 70mm.



STEREO RUMBLE FILTER



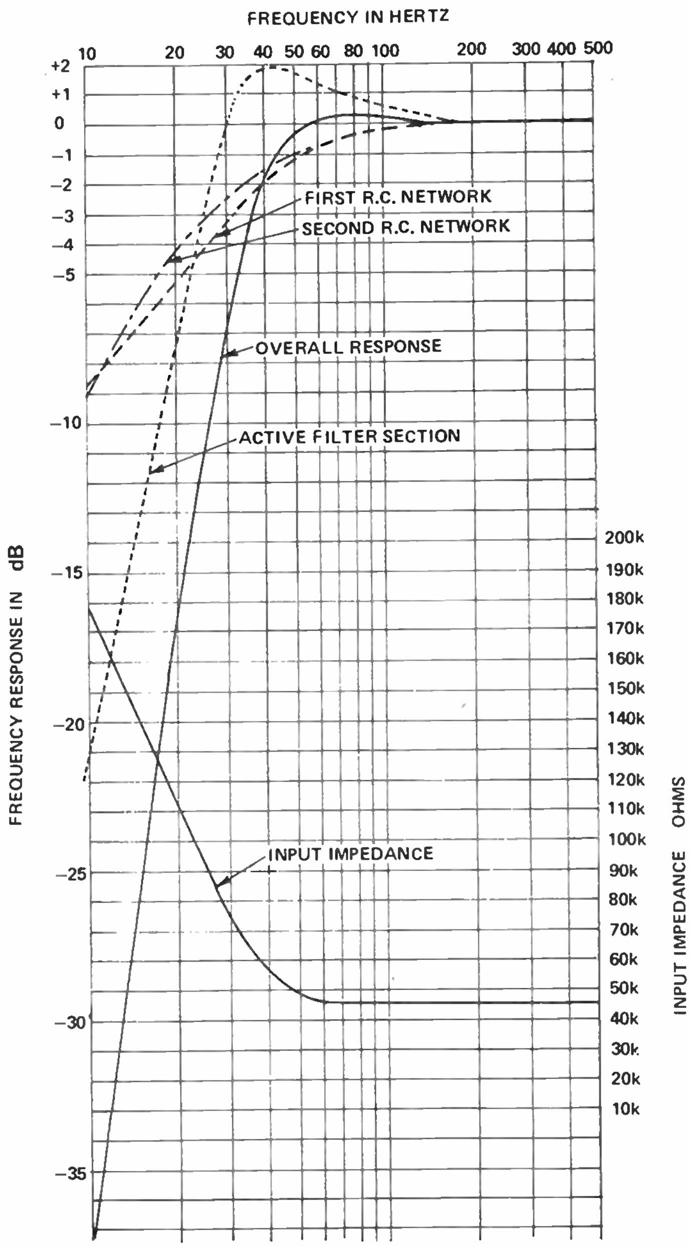


Fig. 4. Characteristics of the rumble filter.

the filter must be situated before the preamplifier. This also poses problems as the signals at this point are very low-level, and there is a danger of introducing hum which would be merely replacing one fault by another.

THE SOLUTION

To maintain response down to at least 50 Hz, whilst obtaining 30 dB or more attenuation to LF noise, we must use a filter which has a sharp knee and an ultimate attenuation slope of 24 dB per octave. The most satisfactory (and cheapest) method of doing this is to use an active high-pass filter — and this is the approach we have used. To obviate the possibility of hum-pickup, the unit uses a battery power supply, one each for left and right channel filters. The use of separate batteries prevents earth loops and ensures that channel separation is maintained. As current drain is very low the batteries may be expected to last their shelf life (12 months or so) and for that reason an on/off switch has not been included.

The unit fits between the turntable and the amplifier, cuts any frequency below 35 Hz and has a total attenuation of 37 dB at 10 Hz increasing at 24 dB/octave below that.

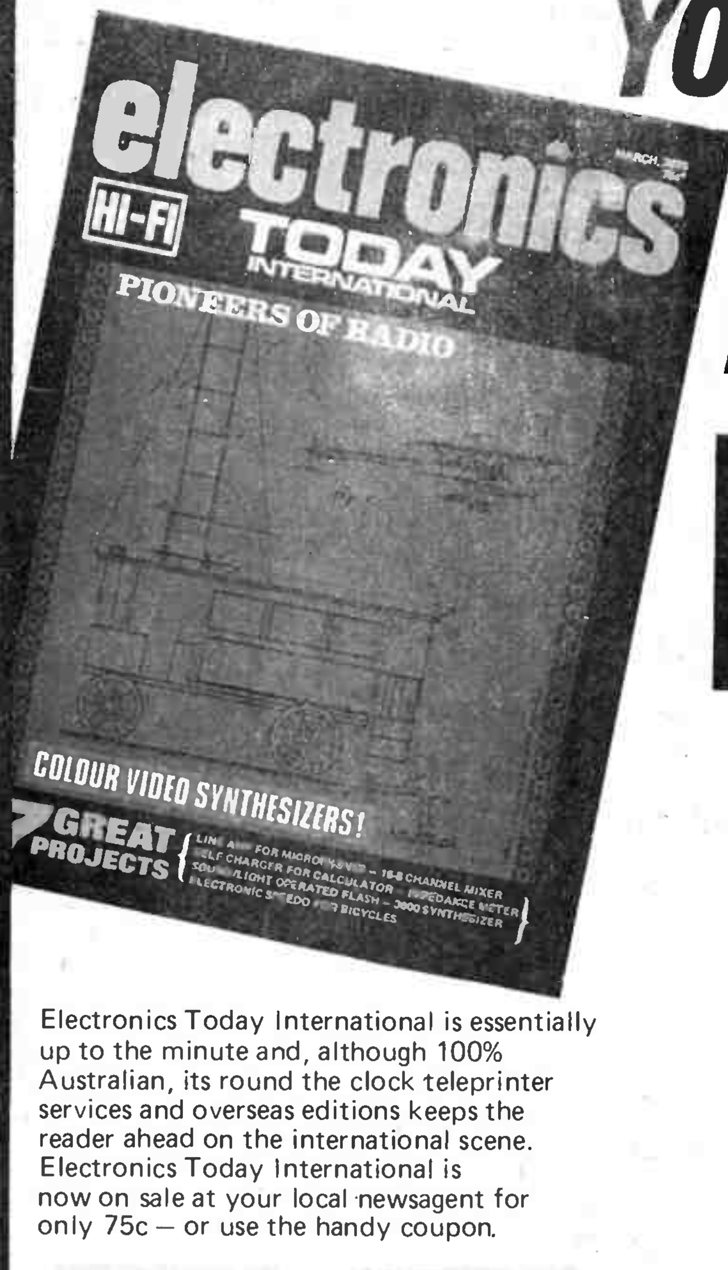
CONSTRUCTION

We built our unit onto a small printed circuit board, but layout is not critical and other alternative methods, such as matrix or Veroboard, may be used successfully. Be careful with the orientation of the transistors especially as there are two different pin configurations in use for the BC549 transistors.

The signal levels involved are extremely small (about 100 μV at 50 Hz) and for this reason a metal box is a must if hum pickup is to be minimized. And, as said before, two separate battery supplies should be used in order to avoid earth loops. We used a conventional four-way, AA battery holder to hold the two sets of batteries. These holders normally connect all four batteries in series. However it is a simple matter to snip the connection between the two sets of two cells.

The RCA sockets for both input and output should be insulated from the metal case. When connecting the unit we found minimum hum was introduced by earthing the turntable to the metal box and then, by taking a separate earth from the metal box to the amplifier. However experimentation in the positioning of earths may well show that some other configuration is best for your particular setup. ●

CONSTRUCTIONAL PROJECTS



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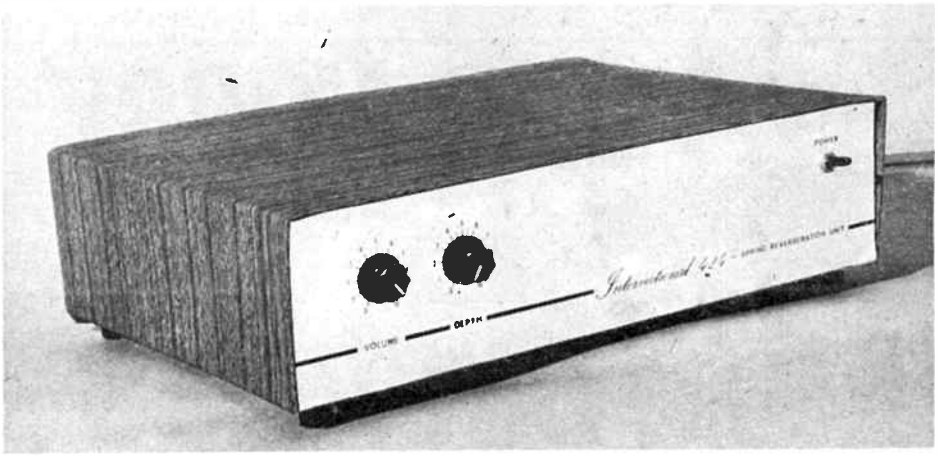
SPRING REVERBERATION UNIT



Built-in mixing facilities and stereo operation are provided in this versatile unit.

THE SOUND of many musical instruments may be "enhanced" by the addition of reverberation. Particular examples of instruments, to which reverberation is commonly applied, are the electronic organ and the guitar.

Reverberation is defined as the persistance of sound within an enclosure after the original sound has ceased. It may also be defined as a series of multiple echoes, decreasing in intensity, so closely spaced in time as to merge into a single continuous sound eventually dying away to nothing.

Reverberation, added with discretion, gives life and brilliance to the music from individual instruments which otherwise appear dull and flat. It is less commonly known that, when reproducing recorded material, the addition of reverberation can considerably enhance the liveliness of the material and its apparent spatial depth.

Artificial reverberation can be achieved in several ways. One system employs echo chambers to achieve the delay. A second system employs magnetic tape-loop techniques, whilst a third, the one used in this project, uses an amplifier that drives springs to provide the delay. It is also possible to achieve delay by fully electronic means but, for normal instrumental or home use, the circuitry is prohibitively complex and expensive.

The unit described is based on a readily available reverberation spring assembly and is suitable for incorporation into existing amplifier instrumental setups, or for adding reverberation to the reproduction from stereo Hi-Fi systems.

In March of 1972 a simpler unit was described in Electronics Today. This was very popular but, required a separate mixer in order that the generated echo and original signal could be combined in controllable proportions.

This unit has the required mixing facilities built-in, the proportion of echo to original signal being adjustable by a control called DEPTH. In addition, we decided to make the unit capable of adding reverberation to stereo systems. This involves very few extra components since both channels are mixed into the reverb spring and the combined echo then separately mixed with the original left and right channels. This extra expense is only that of an extra transistor stage and is well justified, even if the unit is mainly intended for monophonic work.

As the unit is completely functional within itself, and fitted into a strong but attractive metal cabinet it will be equally suitable for use by professionals or high-fidelity audio enthusiasts.

SPECIFICATION

INPUT VOLTAGE	
Maximum	1 volt
Range	100 mV – 1 volt
FREQUENCY RESPONSE	
Direct	–3 dB at 20 Hz, 50 kHz
Delayed	50 Hz – 4 kHz
IMPEDANCE	
Input	approx. 47 k
Output	< 5 k
CROSS TALK	
With 10 k source impedance	– 40 dB
GAIN	
Maximum	unity
SIGNAL TO NOISE RATIO	
Direct	> –60 dB ref IV
Reverberation	> –50 dB ref IV

CONSTRUCTION

We housed our unit in a simple pan-shaped chassis with metal cover.

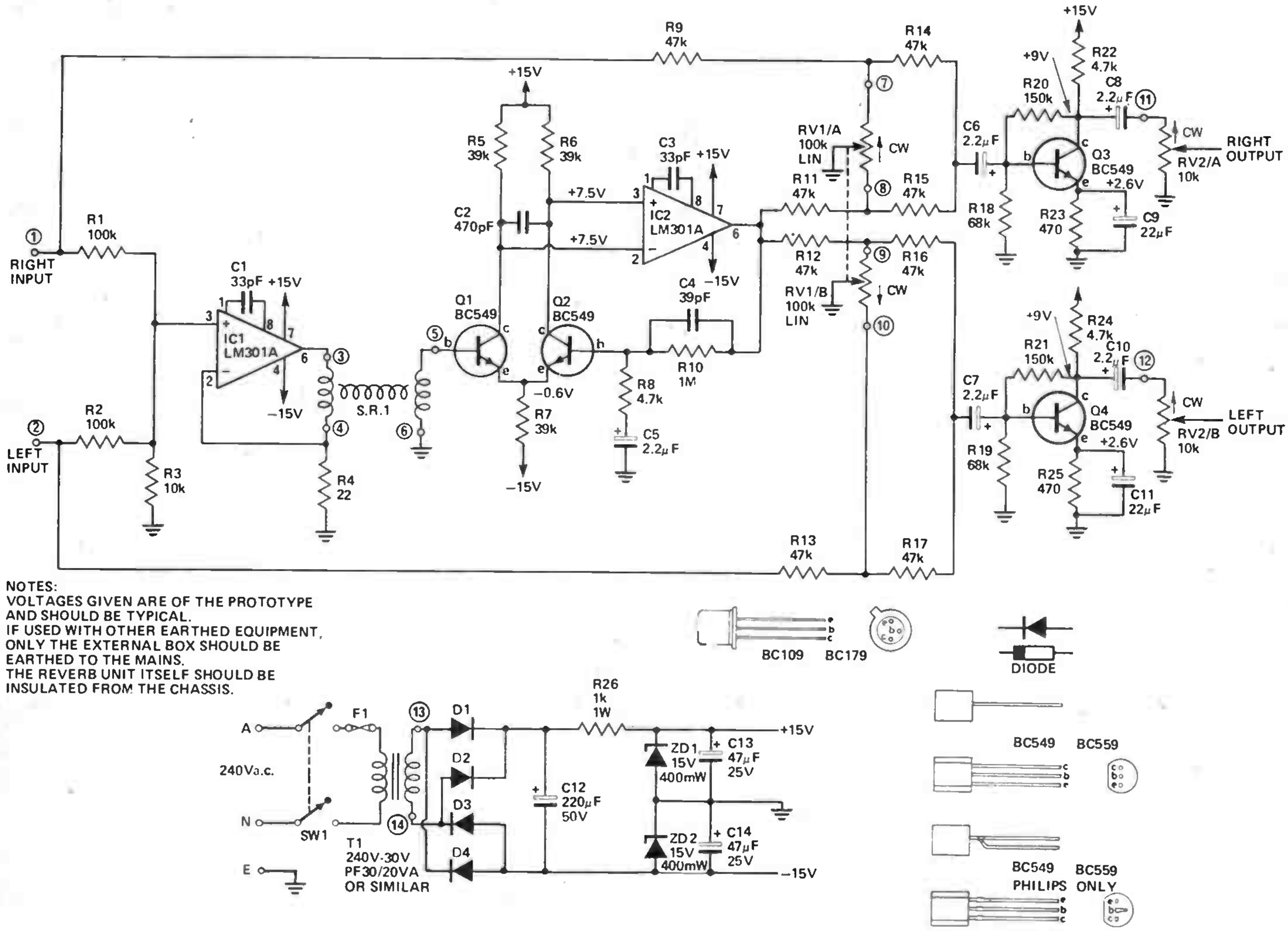


Fig. 1. Circuit diagram of the spring reverberation unit.

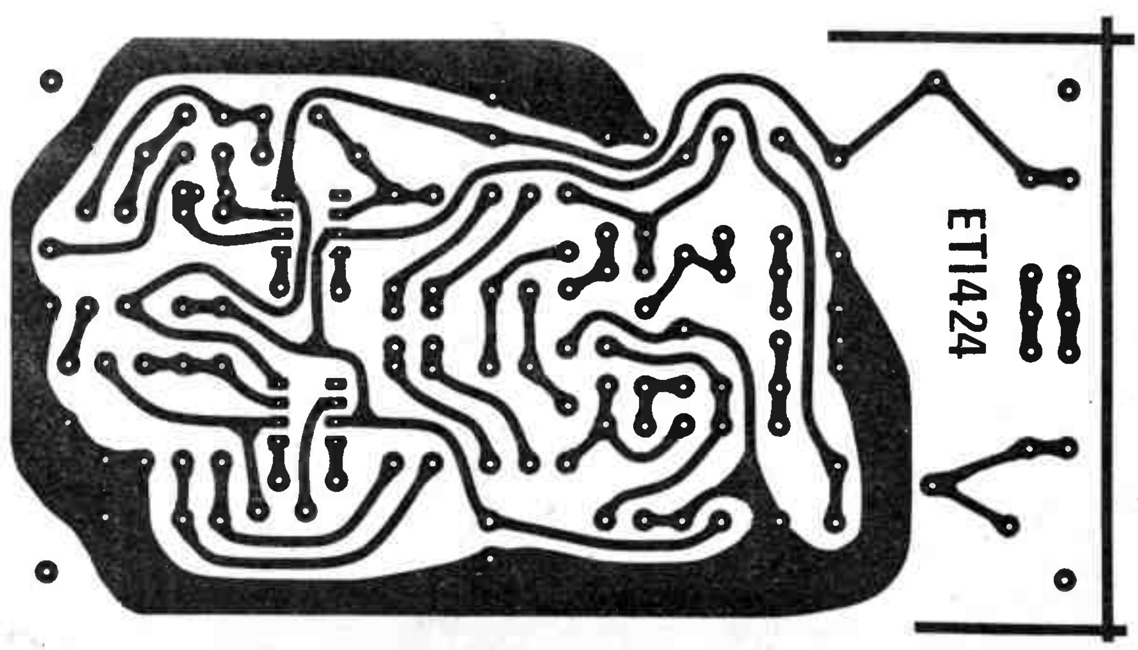


Fig. 2. Full size printed circuit board layout.

SPRING REVERBERATION UNIT

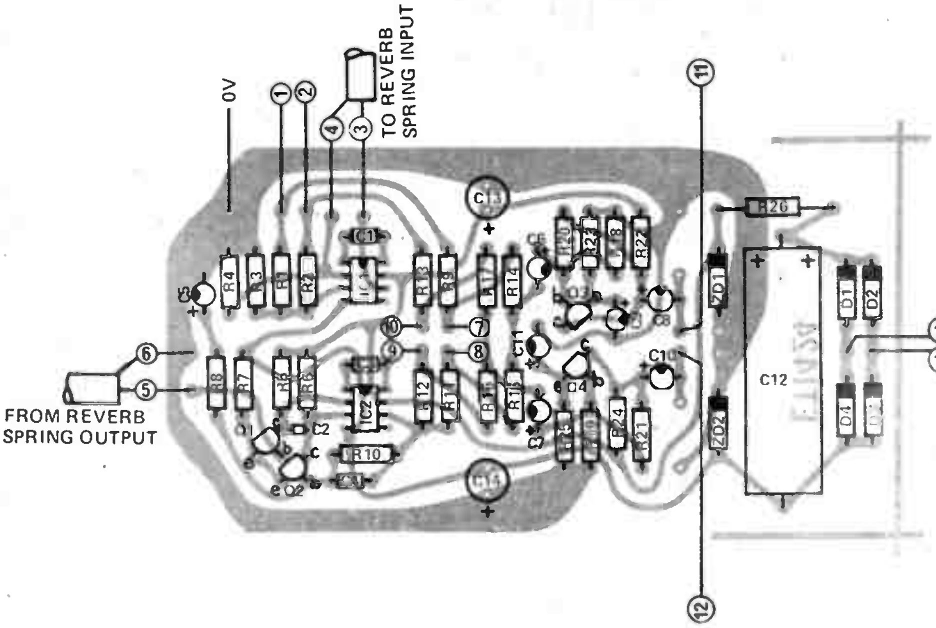


Fig. 3. Component overlay.

This enables the unit to be used as a flexible system component, but, if desired, the electronics may easily be incorporated within an existing system-box if room permits.

The majority of the components are mounted upon one single printed-circuit board, although matrix or veroboard can quite easily be used if preferred.

Whichever constructional method is used, it is essential to check polarized components, for correct orientation, before soldering. Note especially that two different pin configurations for the BC549 are available and that it is the Philips type which is shown on the overlay.

The input socket of the reverberation spring must be removed and replaced with an insulated type. To do this it is

HOW IT WORKS

The reverberation spring is an electro-mechanical device for delaying and producing echo on audio signals – it operates in the following manner. A relay-like transducer vibrates one end of a spring in response to an input audio signal. The spring continues to vibrate after the excitation has been removed and thereby produces a decaying 'echo' as well as delaying the propagation of the signal to the transducer at the other end.

The mechanical system naturally has many resonances and the frequency response therefore cannot be flat over a small frequency range, but is substantially flat over the broad frequency range of 50 Hz to 4 kHz.

Integrated circuit IC1 is connected so as to provide current drive to the input transducer of the spring. The transducer is inductive and hence, the voltage across it will increase with frequency. However, since the current remains constant, the power in the transducer also remains constant. The stereo input is summed into R3 by resistors R1 and R2 (with a loss of 20 dB) to provide a composite signal at pin 3 of IC1. As the amplifier always tries to keep pin 2 at the same potential as pin 3, the voltage across R4, and the current through it, is therefore proportional to the input voltage. As very little current flows into pin 2 of the IC, all this current flows through the transducer.

The output signal from the transducer at the other end of the spring is very small (about -50dB referred to the input and is therefore amplified back to a reasonable level by Q1, Q2 and IC2. Transistors Q1 and Q2 are low noise types and are arranged as a differential pair to add gain before the inherently noisy IC. The gain is set by (R10+R8)/R8 to about 46 dB. The low frequency cutoff is set by C5 and R8, and the high frequency cutoff by R10 and C4. Note that these last figures refer only to the receiving transducer amplifier and not to the whole system.

The direct inputs, left and right, are now both mixed with the common reverberation signal in mixers Q3 (right) and Q4 (left). The proportion of direct and reverberation signals is adjustable by means of depth control RV1. The gain of the output stage is set by R20, R21 and the bias by R18, 19, the overall gain of the complete system being approximately unity.

If single channel operation only is required, simply delete the second mixer transistor and its associated components. If reverberation only, without the mixing facility, is required the output may be taken direct from pin 6 of IC2.

In the event that a volume control is not required resistors may be fitted to the board (holes provided on board) to set the volume to any desired level. These resistors may have any value between 10 k and 1M.

ETI 424 PARTS LIST

R4	Resistor	22	½W	5%
R23,25	"	470	"	"
R26	"	1 k	1W	"
R8,22,24	"	4.7 k	½W	"
R3,	"	10 k	"	"
R5,6,7	"	39 k	"	"
R9,11,12,13	"	47 k	"	"
R14,15,16,17	"	47 k	"	"
R18,19	"	68 k	"	"
R1,2,	"	100 k	"	"
R20,21	"	150 k	"	"
R10	"	1 M	"	"

RV1	Potentiometer	100k dual lin rotary
RV2	"	10 k dual log rotary
C1,3	capacitor	33pF ceramic
C4	"	39pF ceramic
C2	"	470pF ceramic
C5,6,7,8,10	"	2.2μF 10V electrolytic
C9,11	"	22μF 10V electrolytic
C12	"	220μF 50V electrolytic
C13,14	"	47μF 25V electrolytic

Note. all electolytics except C12 are pc mounting

D1-D4	diodes	EM401 or equivalent
ZD1-ZD2	Zener diodes	BZX 79 C15 or equivalent
Q1-Q4	transistor	BC549, BC109 or equivalent
IC1,2	operational amplifier	LM301A

PC board ETI 424
SW1 switch 2 pole on-off 240V rated
F1 fuse and fuse holder 500 ma chassis mounting
Spring reverb unit Plessey type 51 or equivalent
T1 transformer PF30/20VA, PF 3600 or equivalent
3 core flex and plug
2-way RCA sockets — 2 off
12mm long spacers 4 off
chassis to Fig. 7
metal cover to Fig. 8
front panel to Fig. 6
rubber grommet for power cord and insulating reverb unit.
insulated RCA socket for reverb.

necessary to enlarge the mounting hole to provide adequate clearance. Take care not to damage the spring mechanism whilst doing this.

The reverberation spring must also be insulated from the chassis by means of rubber grommets or similar. This is necessary to prevent earth loops, when used in conjunction with other equipment, which would cause high hum levels.

The unit should be wired, as shown in Fig. 1, taking care to keep all 240 volt ac wiring well clear of the electronics and especially clear of the receive end of the reverberation spring. The metal case itself should be earthed even though the electronics itself is not earthed.

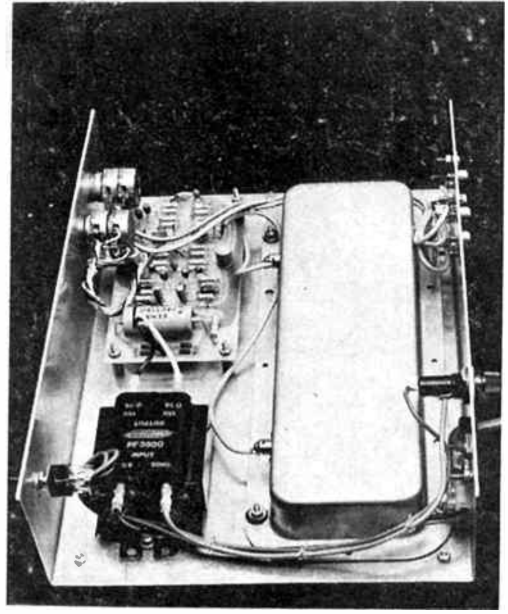


Fig. 4. Method of mounting the hardware and printed circuit board into the chassis is illustrated in this internal view.

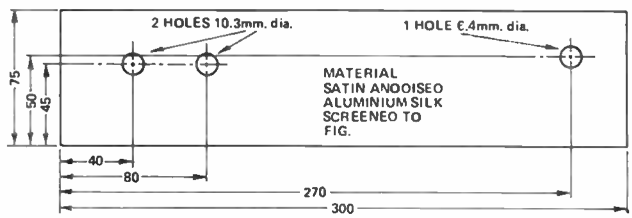


Fig.5 Front panel drilling details.

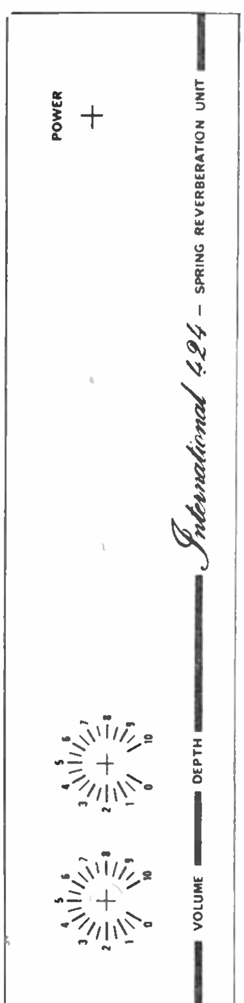


Fig. 6. Front panel artwork for the spring reverberation unit (half size)

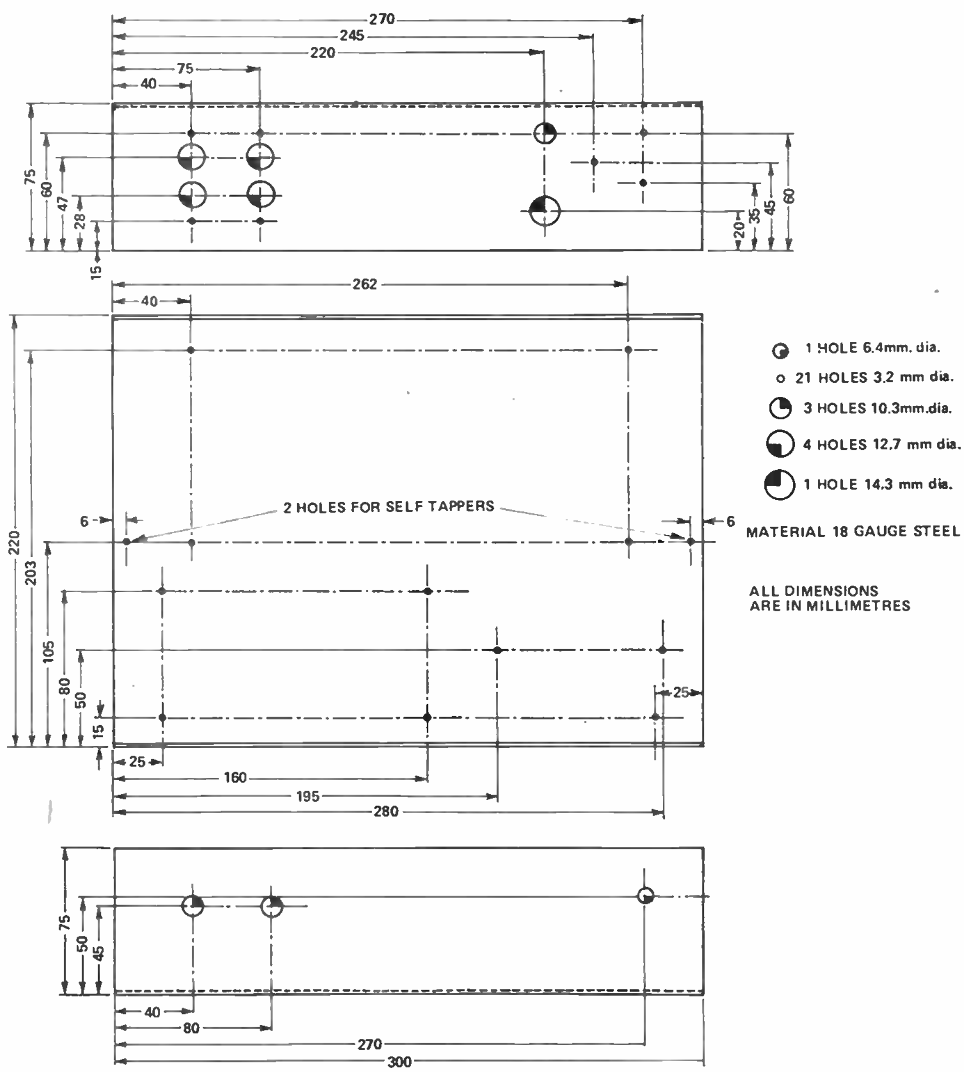


Fig. 7. Dimensions and drilling details of the chassis.

SPRING REVERBERATION UNIT

SETTING UP

As the reverberation spring is a mechanical device, vibration will produce unwanted outputs. Hence it is an inherently noisy device and should be used at a point in the system where the signal level is high.

Two typical points at which the unit may be inserted in the system are:—

1. Between the preamplifier and the main amplifier.
2. After the disc preamplifier, or high level input and the preamplifier.

If inserted between pre and main-amplifiers, i.e. after the volume control, turn the reverb volume control to maximum and adjust the preamplifier volume control such that the main amplifier is just below clipping level. The reverb volume control can then be used to set the level required.

If the reverberation unit is inserted before the system volume control, the volume control on the reverberation unit should be set to maximum (or deleted altogether if desired) and the preamplifier volume control used to set the required level. ●

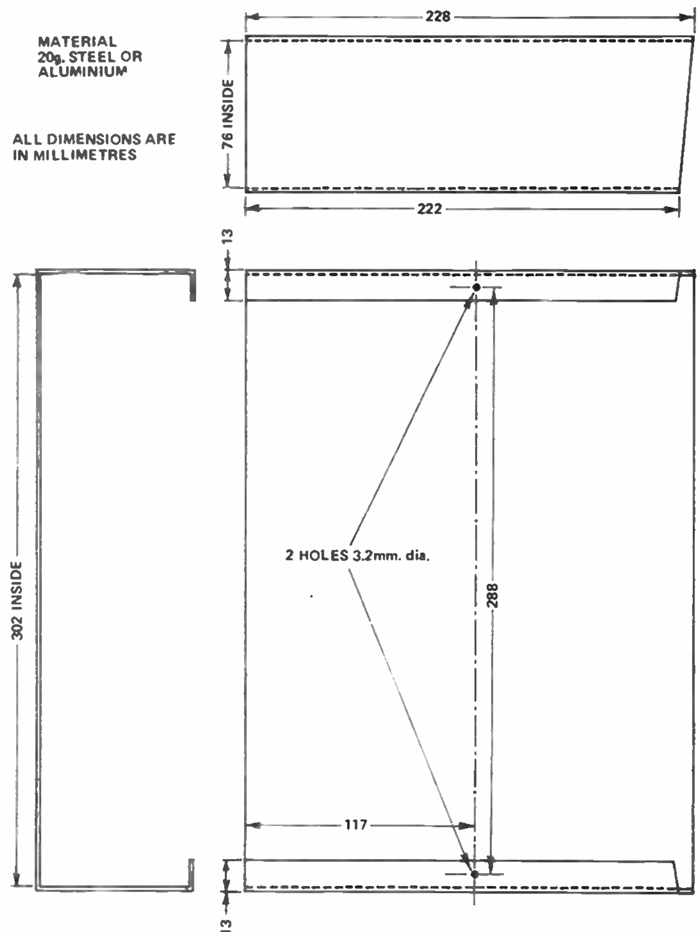


Fig. 8. Detail of the cover.

SIMPLE BASS-REFLEX CABINET

Loudspeaker enclosure design is normally a very complex procedure. But the simplified approach presented here will provide surprisingly effective results.

WHERE space is at a premium the only really satisfactory way of producing good high-fidelity sound is to combine a totally enclosed 'infinite baffle' type of speaker with a high powered amplifier.

There is no way in the world that a bass reflex enclosure can compete because bass reflex enclosures have certain critical dimensions below which they are ineffective.

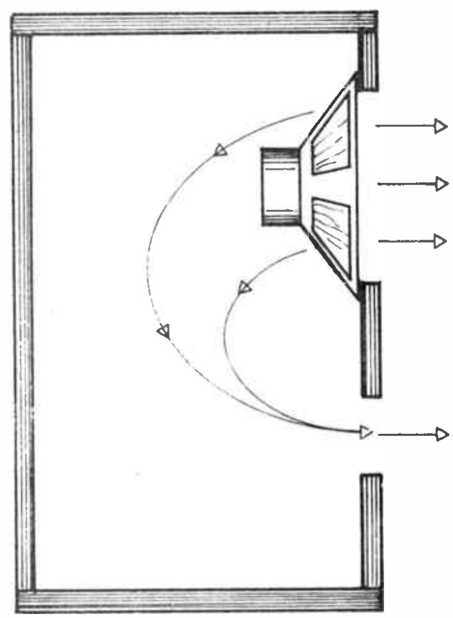
But not everyone lives in rooms 10ft. square, nor has to fit speaker enclosures in the space normally considered adequate for a pair of china cats.

For those whose choice of speakers is not completely dictated by considerations of space, the bass reflex design has still a lot going for it.

This type of enclosure was first used commercially by the USA's Jensen company back in 1936, and was based on research by a number of workers including Voight, Olsen and Thuras, all of whom in turn based their studies on the Helmholtz resonator discovered in the 1800s (Fig. 1).

The basic bass reflex enclosure consists of a box, airtight except for a loudspeaker drive unit mounted on the front panel, and a vent (or tuned port) generally located on the front panel below the speaker.

The actual *location* of the vent is not too critical because the wavelength of the frequencies at which the vent operates is far longer than the overall dimensions of the speaker enclosure.



The *size* of the vent is important, for it is a combination of this and the physical dimensions of the enclosure, that determines the behaviour of the system and provides the smooth, extended low frequency response for which this type of enclosure is renowned.

The *purpose* of the vent is to allow out-of-phase radiation from the back of the cone to be 'reflexed' so as to bring it in phase with the front radiation at low frequencies.

Simple bass-reflex cabinets may be designed, either by calculating the enclosure dimensions from the speaker diameter – or more satisfactorily – by determining the speaker's free air resonance and then designing the enclosure and vent to suit.

No matter which design method is used, the method of construction will be the same. Primarily, the aim is to produce a rigid, non-resonant enclosure, airtight except for the loudspeaker cutout and vent.

Various materials may be used – from concrete, to plywood or pineboard. The thickness of material will depend upon the size of the enclosure. Generally, 1/2" or 5/8" plywood will suffice for the smaller enclosures, increasing to 1" to 1½" for the largest.

It is literally impossible to make the enclosures *too* rigid; if space allows, use the most massive material that you have available, or can afford.

Unless really heavy material is used, reinforce all diagonals (except the front panel) with 3" by 1½" bracing and use wooden blocks to reinforce all joins and corners. All joints should be securely glued and screwed.

Rubber or cork gaskets should be used to seal any removable panels.

The completed cabinet should then be checked for airtightness and if satisfactory, then lined on at least

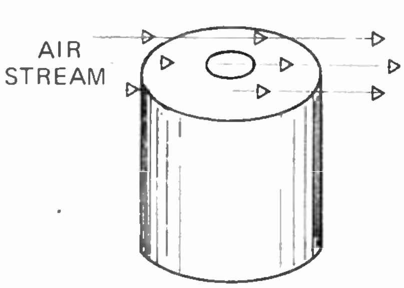


Fig. 1.
A Helmholtz resonator consists of a cavity with a single hole open to the outside air. Air blowing across the hole will cause a sound to be generated at a frequency dependent on the volume of the cavity. The bass-reflex enclosure is in reality a Helmholtz resonator in which the acoustical capacitance of the enclosed air resonates with a mass of air enclosed within the confines of the port opening.

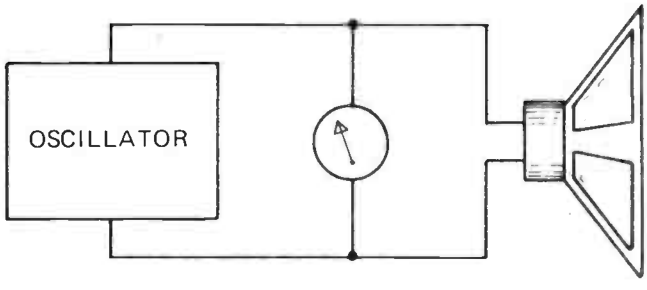


Fig. 2. The free-air resonance of a speaker is determined by measuring the voltage across the speaker's voice coil whilst it is energised over a swept frequency range. The speaker must be suspended away from walls or other reflecting surfaces.

DO YOU THINK YOU COULD GIVE US SOME IDEA OF HOW YOU CAME TO INVENT STEREO

three facing surfaces with two-inch thick Fibreglass or Innerbond. The lining material should be glued in place using a contact adhesive.

If mid-range and tweeter drive units are to be incorporated, these should be boxed in with separate airtight enclosures. These secondary enclosures should be as small as possible and their cubic capacity taken into account when calculating the total enclosure volume.

The positioning of the auxiliary drive units is not critical — but keep them at least 3" from other speakers and the walls of the enclosure.

The front panel of the speaker enclosure should be painted matt black and then covered with an open weave grille cloth (this can be obtained from many specialist hi-fi dealers).

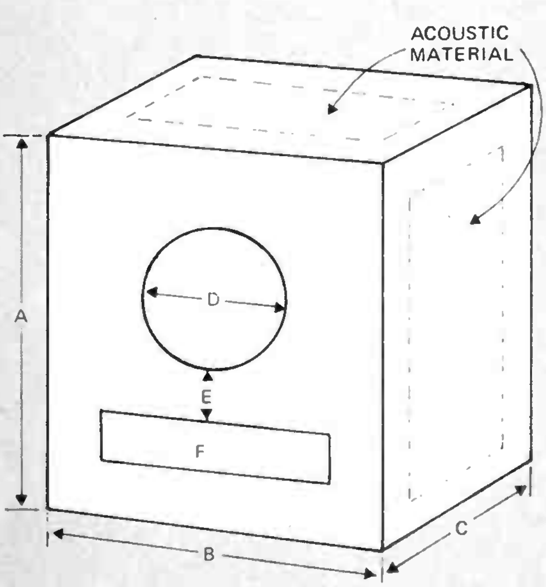
As explained above, a bass-reflex cabinet *can* be designed using the speaker diameter as a basis for the enclosure dimensions. The dimensions for a number of enclosures of this type are given in Table 1. These enclosures are based on the nominal speaker diameter — e.g., the diameter that is quoted by the manufacturer. The actual cone diameter will be less than this — probably by an inch or so. The area of the vent is shown in Table 1 and this is calculated from the *actual* measured speaker cone diameter.

The shape of the vent is not important; it may be circular, square or rectangular (as long as the ratio of length to height does not exceed five) or even divided into two or three separate vents whose total area equals that of the single correctly sized vent.

The position of the vent is also relatively unimportant, although it should not be located closer than 2" to the main speaker opening.

Whilst this design approach will result in a speaker enclosure with generally excellent performance — a more elegant approach is that based on the known (or calculated) free air resonance of the speaker.

This figure is generally quoted in the manufacturer's literature. But it is quite easy to determine — if one has (or can borrow) a suitable oscillator and ac voltmeter. All that is necessary is to connect the speaker as shown in Fig. 2 and with the speaker suspended from a cord (and well away from walls or other reflecting surfaces), to sweep the oscillator frequency *very slowly* from 10 Hz to about 150Hz. The



Nominal Speaker Diameter	Height 'A'	Width 'B'	Depth 'C'	'D'	'E'	Vent Area 'F'	Volume (cubic feet)
8"	24"	18"	11"	7"	3"	½ Area of 'D'	2.75
10"	28"	22"	12"	9.5"	3"	½ Area of 'D'	4.27
12"	31"	24"	13"	11"	3.5"	½ Area of 'D'	5.6
15"	34"	26"	14"	13.5"	4"	9/16 Area 'D'	7.17
18"	40"	27"	14"	16"	4.5"	5/8 Area of 'D'	8.25

Table I — this table and the associated drawing shows how to design a reflex cabinet if no data other than speaker diameter is known.

Nominal diameter of speaker	Frequency
8"	45 – 150 Hz
10"	40 – 100 Hz
12"	30 – 85 Hz
15"	25 – 55 Hz
18"	20 – 40 Hz

Table II — Typical free-air resonances of various size speakers.

voltage indicated by the meter will rise steeply at the free air resonance point. The frequency at which this occurs depends upon the design of the speaker — typical figures for various size speakers are shown in Table II.

(The free air resonance of a speaker changes slightly as the speaker ages — the greatest change takes place within the first few hundred hours — some constructors 'run-in' their speakers in sound proof enclosures before measuring the free air resonance.)

Once the free air resonance has been established, the enclosure dimensions can be determined from the data shown in Table III.

The important factors are the free air resonance, the internal volume of the enclosure, and the size of the vent. The shape should be vaguely rectangular, but providing the width and height are at least twice the diameter of the speaker, and the depth is at least one third the width, then the enclosure may be shaped to fit on a shelf, against a wall or as required.

The internal dimensions shown are fairly critical, and the necessary allowance must be made for panel thickness, stiffeners, crossover networks, and other internal enclosures. Do not make any dimensional allowance for the Fibreglass or Innerbond liner.

As Table III indicates, many of the enclosures are fitted with tuned ducts, rather than just plain vents. These ducts can be made from standard cardboard mailing tubes — obtainable from many office supply companies — or may readily be made by winding glue-coated brown paper tightly around a pre-waxed former of the correct diameter. The wall thickness of the duct should be between 1/8" and 1/4". (Note that at the extremes of frequency and volume shown in Table III — no duct or vent is used — the enclosure is, in effect, an infinite baffle).

As with the first design approach described in this article the position of the duct, or vent, is not critical.

That's basically it. There are other, far more complex, ways to design bass reflex enclosures and many of these methods may well result in marginally improved performance — especially if the duct is subsequently tuned to obtain the flattest possible bass response. But the method outlined in this article will provide a basis for producing enclosures with at least the performance of most professional designs. ●

Free-air resonance	Volume in Cubic Feet 2.0	2.5	3.0	3.5	4.0	5.0	6.0	8.0
25Hz	(A) 5"	(A) 3.75"	(A) 2.75"	(B) 6"	(B) 5"	(B) 3.25"	(C) 8.75"	(C) 5.5"
30Hz	(A) 3"	(B) 5.75"	(B) 4.5"	(B) 3.5"	(C) 9.25"	(C) 6.5"	(C) 4.75"	11 sq. ins.
40Hz	(B) 3.5"	(C) 7.75"	(C) 5.75"	(C) 4.5"	(C) 3.25"	13 sq. ins.	18 sq. ins.	28 sq. ins.
50Hz	(C) 5.5"	(C) 3.5"	13 sq. ins.	16 sq. ins.	18 sq. ins.	29 sq. ins.	39 sq. ins.	62 sq. ins.
60Hz	11 sq. ins.	16 sq. ins.	20 sq. ins.	29 sq. ins.	35 sq. ins.	50 sq. ins.	75 sq. ins.	Closed
70Hz	18 sq. ins.	26 sq. ins.	35 sq. ins.	46 sq. ins.	58 sq. ins.	90 sq. ins.	Closed	Closed
80Hz	28 sq. ins.	41 sq. ins.	60 sq. ins.	80 sq. ins.	96 sq. ins.	Closed	Closed	Closed
90Hz	42 sq. ins.	64 sq. ins.	89 sq. ins.	117 sq. ins.	Closed	Closed	Closed	Closed

Duct Tubes (A) = 2" inside diameter
(B) = 3" inside diameter
(C) = 4¾" inside diameter

Thus (A) –2.5" is a duct 2" inside diameter by 2.5" long.

Where a measurement is given in square inches — this implies that a vent is required — not a tuned duct.

Table III — This table provides the design data for a given speaker free-air resonance and various enclosure volumes (in cubic feet)

DIN CONNECTORS

Many amplifiers and tape recorders of European or Japanese origin are equipped with DIN connectors. The pin connections for these connectors are standardized in accordance with IEC* recommendations which are given here.

Most equipment will be wired to this convention. The type numbers given are those designated by the IEC and may be different to those assigned by individual manufacturers.

*International Electrotechnical Commission.

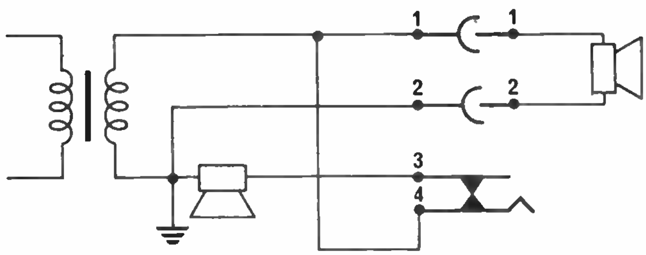


Fig. 1a. Example of using the type 05 connector inserted in position A. Both speakers are operational.

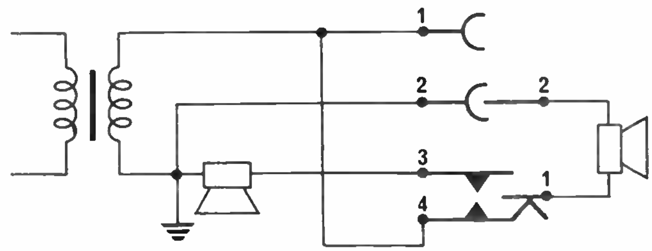


Fig. 1b. With the connector reversed, in position B, the internal speaker is disconnected.

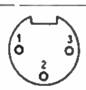
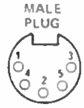
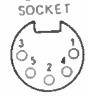
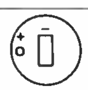
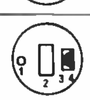
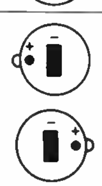
Notes 1. — The numbering of the contacts is shown as seen on the mating face of the connector.

2. — The same connectors are used for monaural and stereophonic systems.

3. — Normally, it is recommended to connect the shell of the plug to terminal 2 of the connector to ensure that the screen is earthed.

4. — The pin connector Type 05 can be inserted in a socket connector type 08 in either of the two positions A or B. The switch is actuated by the short round pin 1, when the pin connector is inserted in position B.

TABLE 1

Contact arrangement See Note 1	IEC Type designation* Pin connector	Socket connector	Application		Connections 1	2	3	4	5
	01	02	Microphone	Monaural system (balanced)	Hot lead	Screening; earth	Return lead		
				Monaural system (unbalanced)	Hot lead				
MALE PLUG  FEMALE SOCKET 	03	04	Microphone	Stereophonic system (balanced)	Hot lead of left-hand channel		Return lead of left-hand channel	Hot lead of right-hand channel	Return lead of right-hand channel
				Stereophonic system (unbalanced)	Hot lead of left-hand channel			Hot lead of right-hand channel	
			Record player	Monaural system			Hot lead		Connected to 3
			See Note 2	Stereophonic system			Hot lead of left-hand channel		Hot lead of right-hand channel
			Tape recorder	Monaural system	Input signal		Output signal	Connected to 1	Connected to 3
			See Note 2	Stereophonic system	Input signal of left-hand channel	See Note 3	Output signal of left-hand channel	Input signal of right-hand channel	Output signal of right-hand channel
	06	07 09	Low impedance loudspeaker	Loudspeakers	Hot lead	Return lead			
		08 See Note 4		Loudspeakers					
	05 See Note 4			Loudspeaker with or without switch					

* all connectors have the prefix 130-9 IEC

How often would you have liked to add extension speakers and not been quite sure what to do? This article tells you how.

extension

Some amplifiers have provision for connecting and switching extension speakers in and out of circuit. Many do not.

Yet this facility is most useful — for music is often required in more than one room, but without the expense of duplicating equipment, or lugging bulky speakers about the house.

At first sight, wiring extension speakers is a simple enough task — but there is more to it than at first appears. Here then — in answer to requests from many readers — is how to do it.

Modern transistor amplifiers operate best with load impedances between four and eight ohms, and as with these amplifiers the power developed in the load is proportional to load impedance, one cannot necessarily just wire extra speakers in parallel with the existing ones without possibly overloading the amplifier.

Generally, load impedances of less than four ohms may damage the amplifier, whilst impedances greater than 16 ohms will prevent the amplifier developing full power — it is for this reason that 16 ohm speakers are rarely made nowadays — and will not be discussed in this article.

The circuits shown here assume that the amplifier has an output impedance of between four and eight ohms. Most amplifiers have nowadays — if in doubt, the output impedance is almost invariably quoted in the manufacturer's handbook. Amplifiers with switched (or tap selected) output impedances — such as transformer coupled units — should be set to four ohms when these extension speaker circuits are used.

BASIC CONNECTIONS

The basic rule for connecting extension speakers is that — providing the amplifier is not overloaded — any combination of speakers of similar or different impedances may be connected in parallel, but speakers connected in series must have similar impedances. If speakers of differing impedances are connected in series, each will affect the distribution of power/frequency connected to he remainder, for example one speaker may end up with predominantly high frequency response and another may have little high frequency response.

Fig. A. Symbol for a resistor.

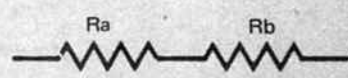


Fig. B. Two resistors in series.

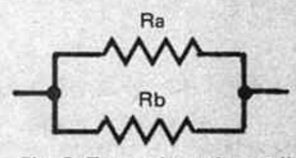


Fig. C. Two resistors in parallel.

RULES OF COMBINATION

Some readers who wish to add extension speakers may not be familiar with electronics. For their benefit here are the rules for calculating the combined impedance of series and parallel speaker combinations.

For this purpose, each speaker may be considered as a resistance, the symbol for which is shown in Fig. A. Series connected resistors are shown in Fig. B and parallel connected resistors in Fig. C.

The combined value of two resistors connected in series is given by simply adding the individual resistance values. Thus:

The combined resistance $R_s = R_a + R_b$.

Parallel combinations present slightly more difficulty.

For the same two resistors in parallel:

The combined resistance

$$R_p = \frac{R_a \times R_b}{R_a + R_b}$$

Hence if we parallel an 8 ohm resistor with a 15 ohm resistor

$$R_p = \frac{8 \times 15}{8 + 15} = \frac{120}{23} = 5.2 \text{ ohms}$$

approx.

The term resistance, strictly speaking, may only be applied when speaking in terms of direct current (dc), whereas speakers are driven by alternating currents (ac). The term impedance is used to denote the effective resistance of a speaker (or any other device) to alternating current.

When putting speakers in parallel or series the same rules as above apply but we talk in terms of impedances rather than resistances.

EXAMPLE 1

Four ohm main speakers — one pair of four ohm extension speakers.

Switching arrangements:

(1) Main speakers only
(2) Auxiliary speakers only
(3) Main and auxiliary speakers

The extension speakers cannot be simply paralleled across the main speakers because four ohms paralleled with four ohms is two ohms — and a load such as this will probably damage an amplifier.

Here the only way to connect the extra speakers is by wiring them in such a way that, when switched into circuit, they are in series with the main speakers.

As their combined impedance is now eight ohms the subsequent mismatch will cause some loss of power to each speaker. Fortunately the response of the human ear is logarithmic and so the subjective drop in sound level is somewhat less than one might at first expect.

Figure 1 shows how the speakers should be connected.

EXAMPLE 2

Eight ohm main speakers and one pair of eight ohm extension speakers.

Switching arrangements:

(1) Main speakers only
(2) Auxiliary speakers only
(3) Main and auxiliary speakers

In this example the main and extension speakers are wired in parallel. If they were wired in series the resultant impedance would be 16 ohms and hence amplifier power would be reduced considerably. If the speakers are paralleled (as shown in Fig. 2), the resultant impedance of four ohms will provide adequate output. The overall sound level will not alter appreciably when the extra speakers are switched into circuit, and so this arrangement is preferable to that described in Example 1.

EXAMPLE 3

Four ohm main speakers and one pair of eight ohm extension speakers.

Switching arrangements:

(1) Main speakers
(2) Auxiliary speakers

In this example the only practicable

speakers

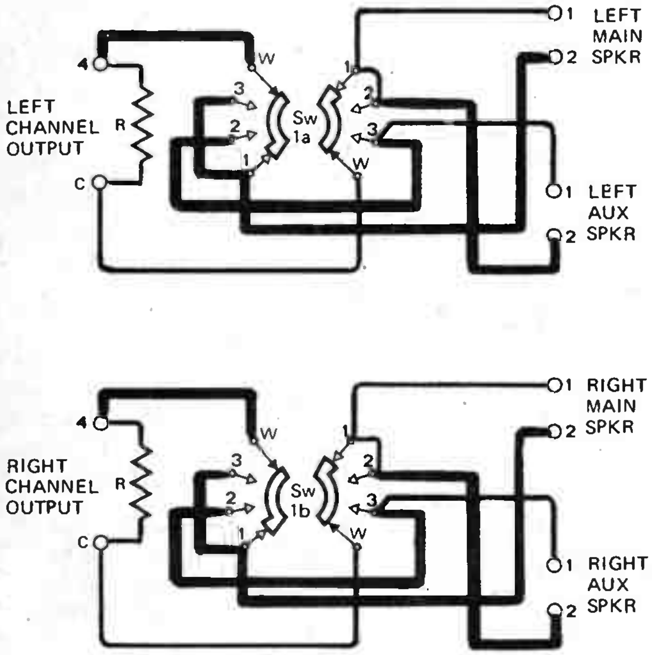


Fig. 1. Four ohm main speakers with one pair of four ohm extension speakers.

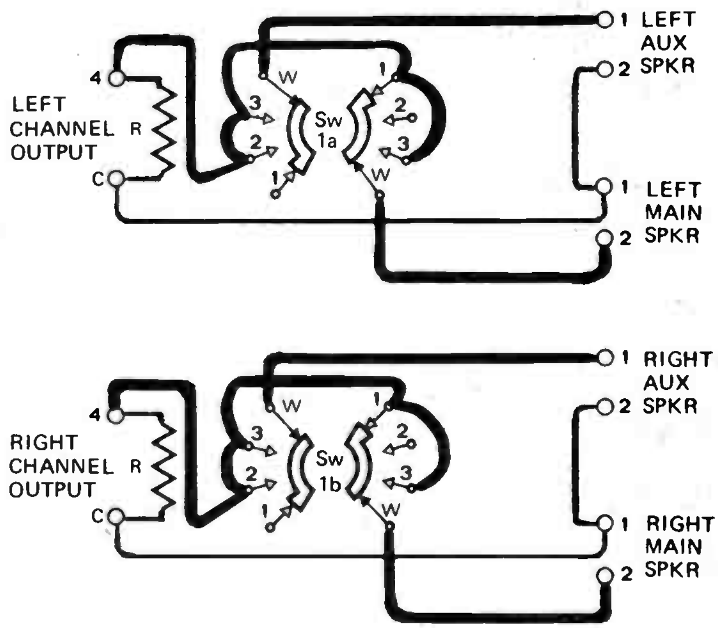


Fig. 2. Eight ohm main speakers and one pair of eight ohm extension speakers.

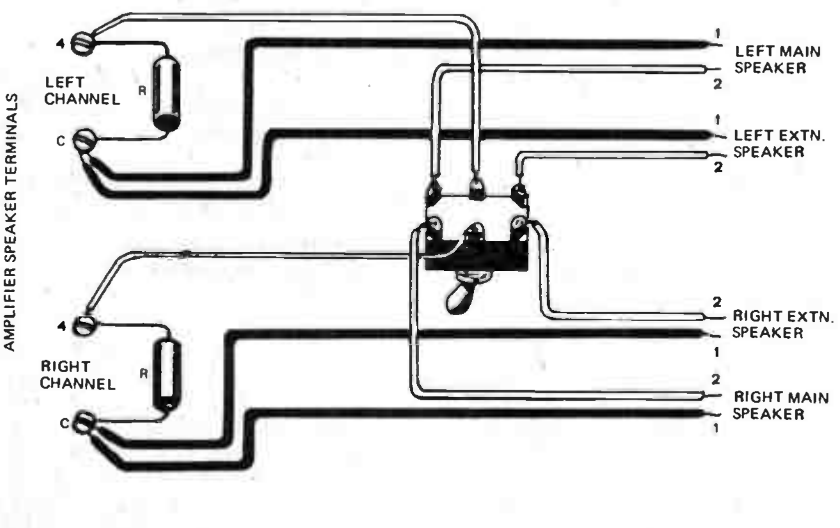


Fig. 3. Four ohm main speakers and one pair of eight ohm extension speakers.

way of interconnecting main and auxiliary speakers is by wiring them in series, as a parallel connection will reduce the combined impedance to a possibly unsafe level. In the arrangement shown in Fig. 3 we can use the main speakers alone, and the auxiliary speakers alone — but not both sets simultaneously — remember that any attempt to series connect speakers of different impedances will cause distortion.

EXAMPLE 4

Four ohm main speakers and two sets of eight ohm extension speakers.
Switching arrangements:
(1) Any pair of speakers may be used alone
(2) Both pairs of extension speakers may be used together
(3) Neither pairs of extension speakers can be used at the same time as the main speakers
Figure 4 shows how the interconnections are made.

EXAMPLE 5

Eight ohm main speakers and two pairs of eight ohm extension speakers.
Switching arrangements:
(1) Any pair of speakers may be used alone
(2) Any two pairs of speakers may be used together
(3) All speakers must not be used simultaneously
Wiring arrangements are shown in Fig. 5.

INSTALLATION

In all cases 50 ohm 5 watt resistors are wired across the speakers outputs. These reduce the annoying click that is otherwise experienced when switching from one speaker combination to another. Apart from this function, many amplifiers will oscillate at very high frequency (in the Megahertz range) if the output is open-circuited. The resistors will prevent this occurring whilst switching.

Speaker wiring should be run in twin-core flex (23/0076 is ideal). If possible try to obtain flex that has each conductor colour coded — or identified in some way — as it is most important to maintain the connection sequence shown in the diagrams. Crossed-over wiring will result in incorrect speaker phasing and consequent cancellation of some frequencies.

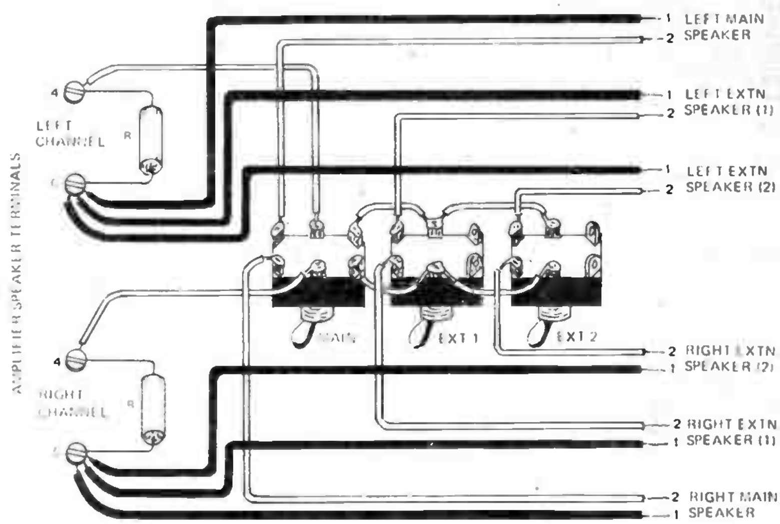


Fig. 4. Four ohm main speakers and two sets of eight ohm extension speakers.

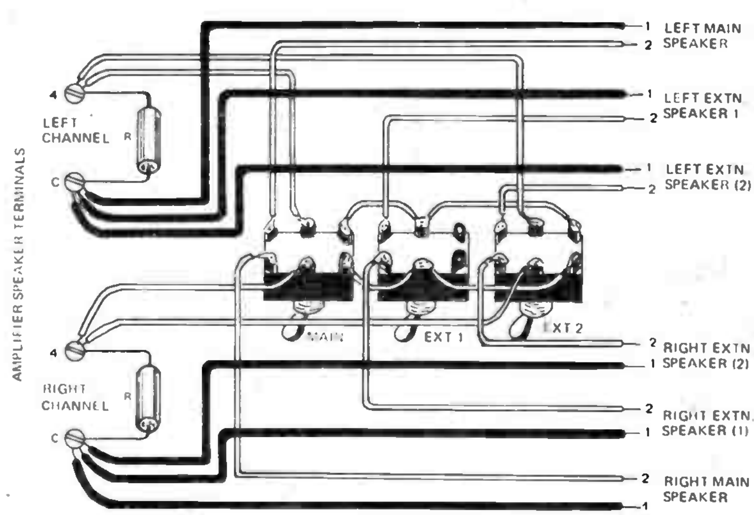


Fig. 5. Eight ohm main speakers and two pairs of eight ohm extension speakers.

A lighter gauge wire (14/0076) may be used for runs of less than 50 feet. However this lighter wire is not as readily obtainable as the 23/0076.

Ordinary hook-up wire may be used to interconnect the switch units — but again be quite sure to connect all leads exactly as shown. ●

No matter what you want in a speaker, hear it is.

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WOOFER & WIDE RANGE

LOUDSPEAKER	POWER HANDLING	FREQUENCY RESPONSE
KC6M	8W RMS	50Hz-11kHz
KC6MX	8W RMS	50Hz-20kHz
C60	10W RMS	35Hz-7500Hz
C60X	10W RMS	30Hz-17kHz
C80	20W RMS	35Hz-8kHz
C80X	20W RMS	35Hz-20kHz
C100	20W RMS	40Hz-11kHz
C100X	20W RMS	40Hz-20kHz
C12P guitar	30W RMS	55Hz-10kHz
C12P woofer	30W RMS	35Hz-10kHz
C12PX wide range	30W RMS	35Hz-13kHz
C12PX guitar	30W RMS	55Hz-13kHz
12U50	50W RMS	25Hz-11kHz
12UX50	50W RMS	40Hz-13.5kHz
MIDRANGE		
C6MR	20W RMS	450Hz-6600Hz
KC5MR	15W RMS	700Hz-14kHz
TWEETERS		
X20 horn	—	3kHz-30kHz
X30 dome	—	3kHz-30kHz
KC 3G X cone	—	1.5kHz-19kHz
5FX cone	—	4kHz-20kHz

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ERRATA

Due to a printing error, Figure 12 on page 34 has accidentally been over-printed. It is reproduced here in the correct size to match Figure 11 which is correctly reproduced on the same page.

We regret any inconvenience this may have caused the purchasers of this book.

Fig. 12. Artwork for rear-panel escutcheon half-size.