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30 AUDIO PROJECTS

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25 Watt Amp

General Purpose Preamp

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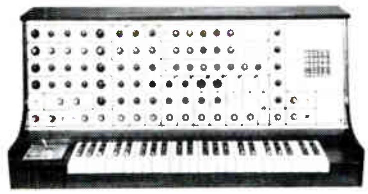

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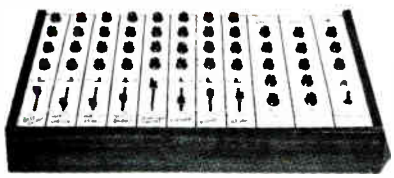

414 Master Mixer

- 8 input channels with volume, bass, treble, pan, echo send and sensitivity select.
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This compressor expander restores much of the dynamic range missing from records and tapes. Can be used as an effective noise reduction unit.

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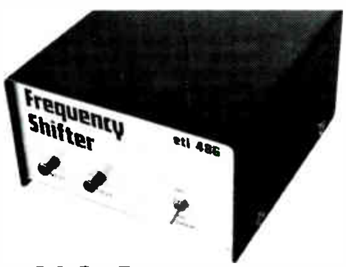
One of the best performing kits ever offered.

- 10 adjustable centre frequency controls per channel.
- Gain control for each channel.
- Tape monitor facility.



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30 AUDIO PROJECTS

From Electronics Today International

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Inside

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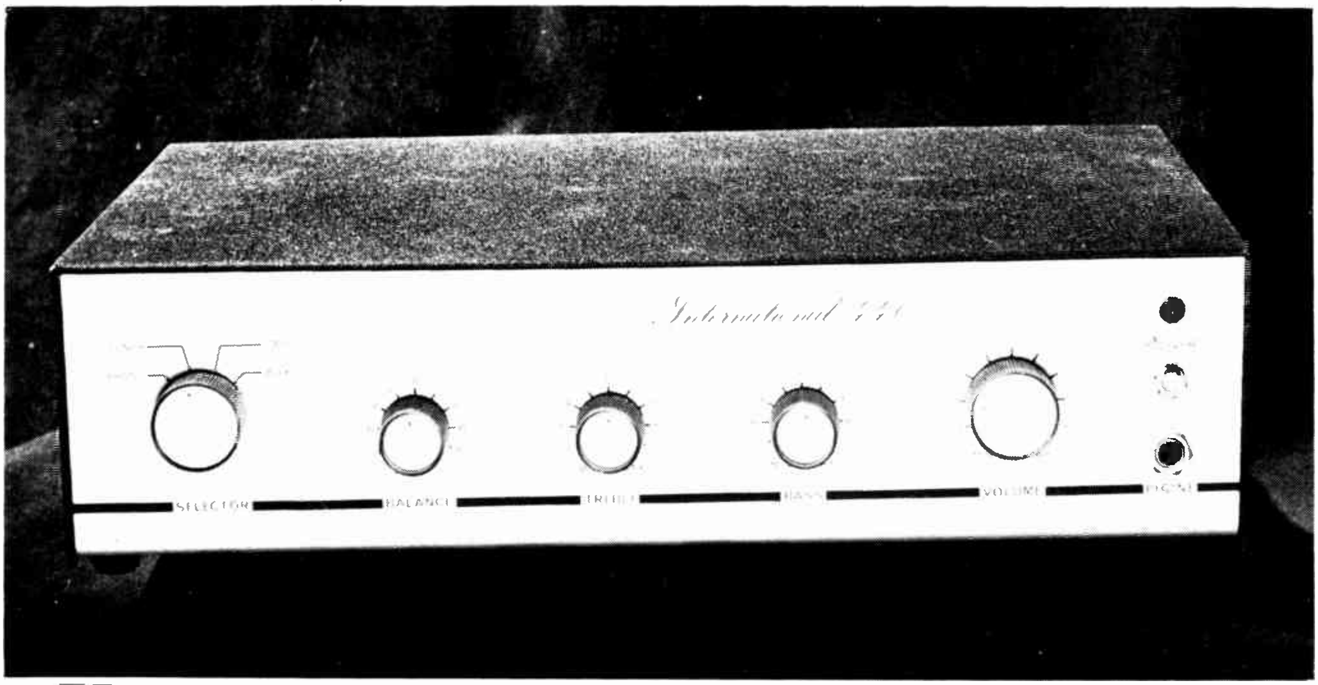


eti PROJECT 440

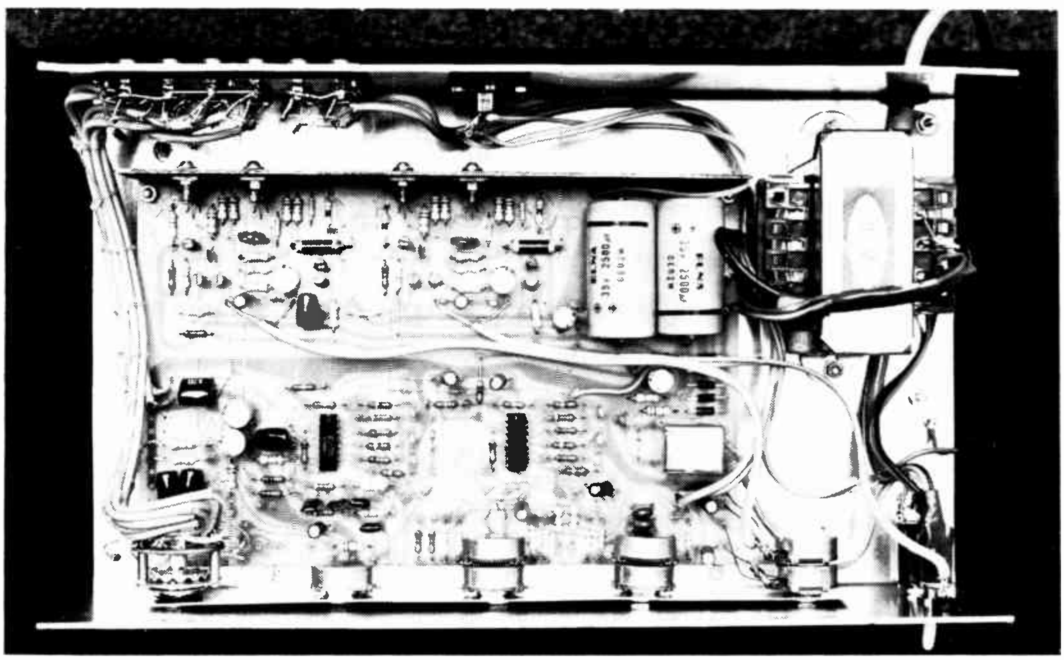
SIMPLE 25 WATT AMPLIFIER

Big performance at a low price.

SINCE A HIGH percentage of the cost of an amplifier is in the hardware, (eg chassis, potentiometers, switches etc) and this cost does not vary greatly relative to amplifier power output, we aimed at the highest possible power for reasonable cost. Thus the amplifier gives 25 watts RMS per channel which is about as much as can be obtained without component costs increasing dramatically.

We also wanted the amplifier to be simple enough for a beginner to build so we used a single printed circuit board, to hold as much as possible of the electronics, thus keeping external wiring down to a minimum.

The result is a 25 watt-per-channel amplifier which has a distortion of



Internal view of the amplifier showing location of the major components.

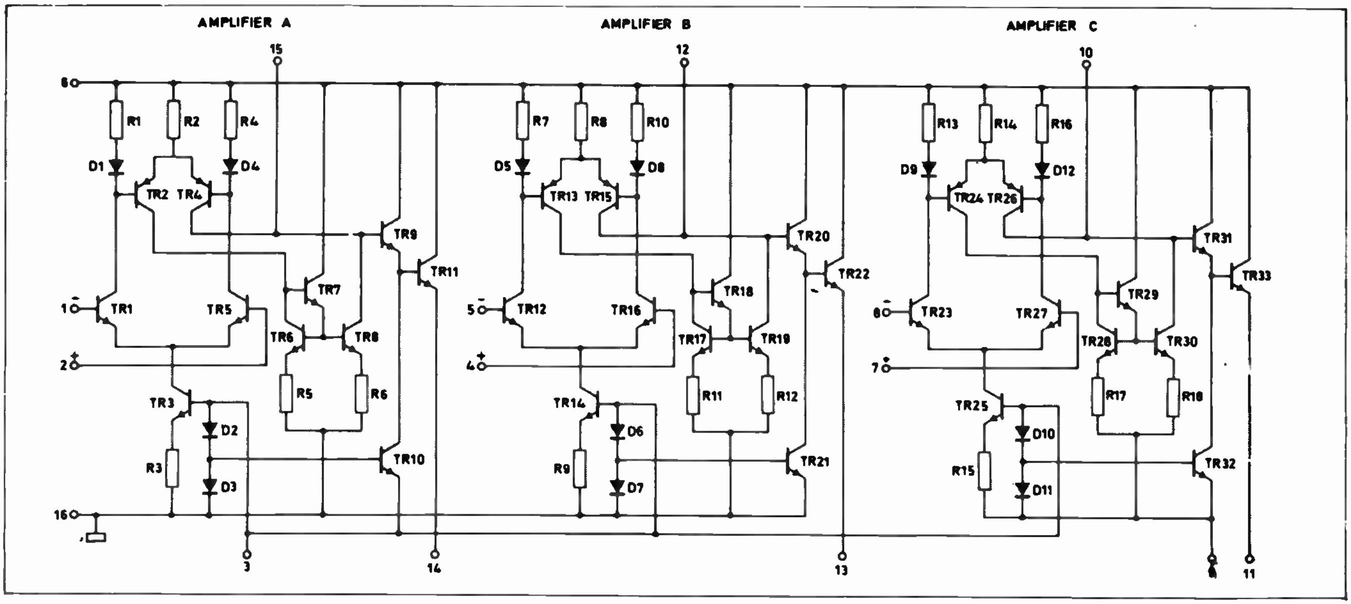


Fig. 1. Circuit diagram of the TCA220 triple-operational amplifier.

around 0.1%, costs about the same as some 12 watt-per-channel kitsets at present on the market and is extremely easy to build and set-up.

CONSTRUCTION

The single printed-circuit board construction greatly simplifies things for the beginner. (See page 12 for circuit board pattern). The heatsink is attached to the rear of the board to hold the power transistors, and a bracket at the front holds the potentiometers. Before attaching these brackets assemble the components to the printed circuit board, as shown in the component overlay diagram Fig. 3, doing all the linking required first. There are two links which go under resistors R49 and R50 in the power amplifier and these should be insulated with 1 mm spaghetti. All other links may be tinned copper provided that they are kept straight and flat on the board.

Although the components can be mounted in any order it is usually easier to mount the smallest (lowest height) components first, ie, resistors and diodes. These should be mounted flush on the surface of the board. The capacitors may now be mounted taking care not to damage the small ceramic capacitors by bending the leads too close to the body of the device. Make sure that electrolytic capacitors are orientated correctly, i.e., the polarity is correct.

The transistors, apart from Q7,8,9,10,15,16,17 and 18 (which are on the heatsink) may now be fitted to the board. With the BC548 there are two different lead connections. The Philips type has a bent centre leg (the base) and these are the types shown on the overlay. If a different brand is used, ie one with the pins all in line, they must be inserted 180° around from that orientation shown. Transistors Q7,8,9 and 10 MUST be the Philips type. Hence, if you have a mixture, keep the Philips types for

(Text continued on page 8)

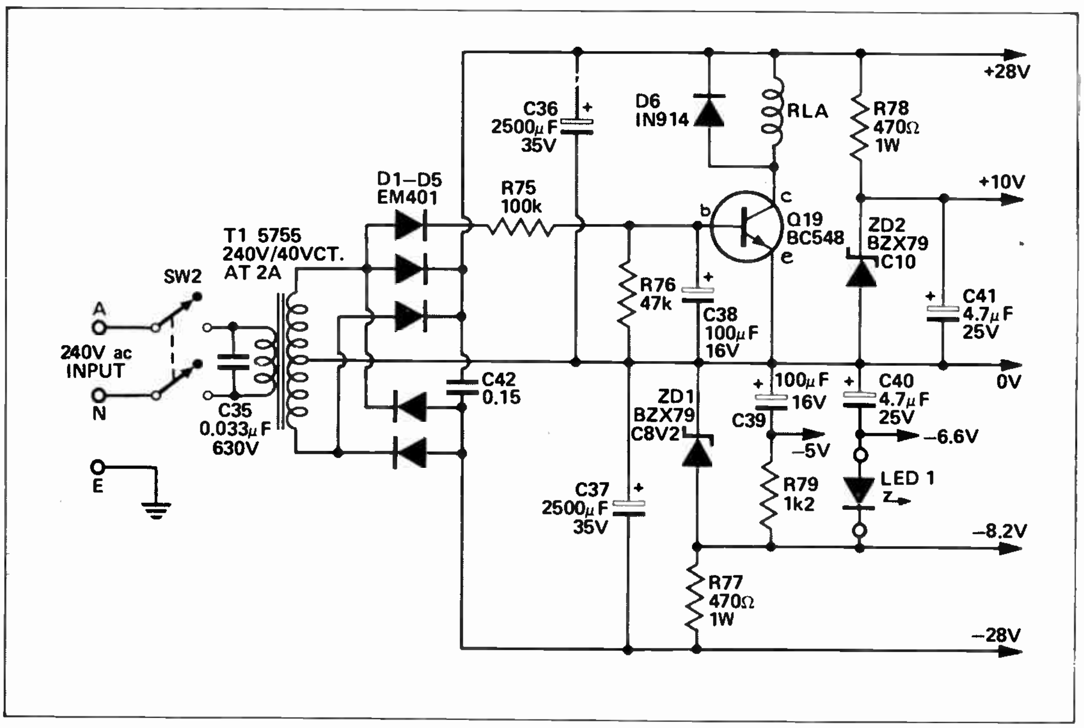
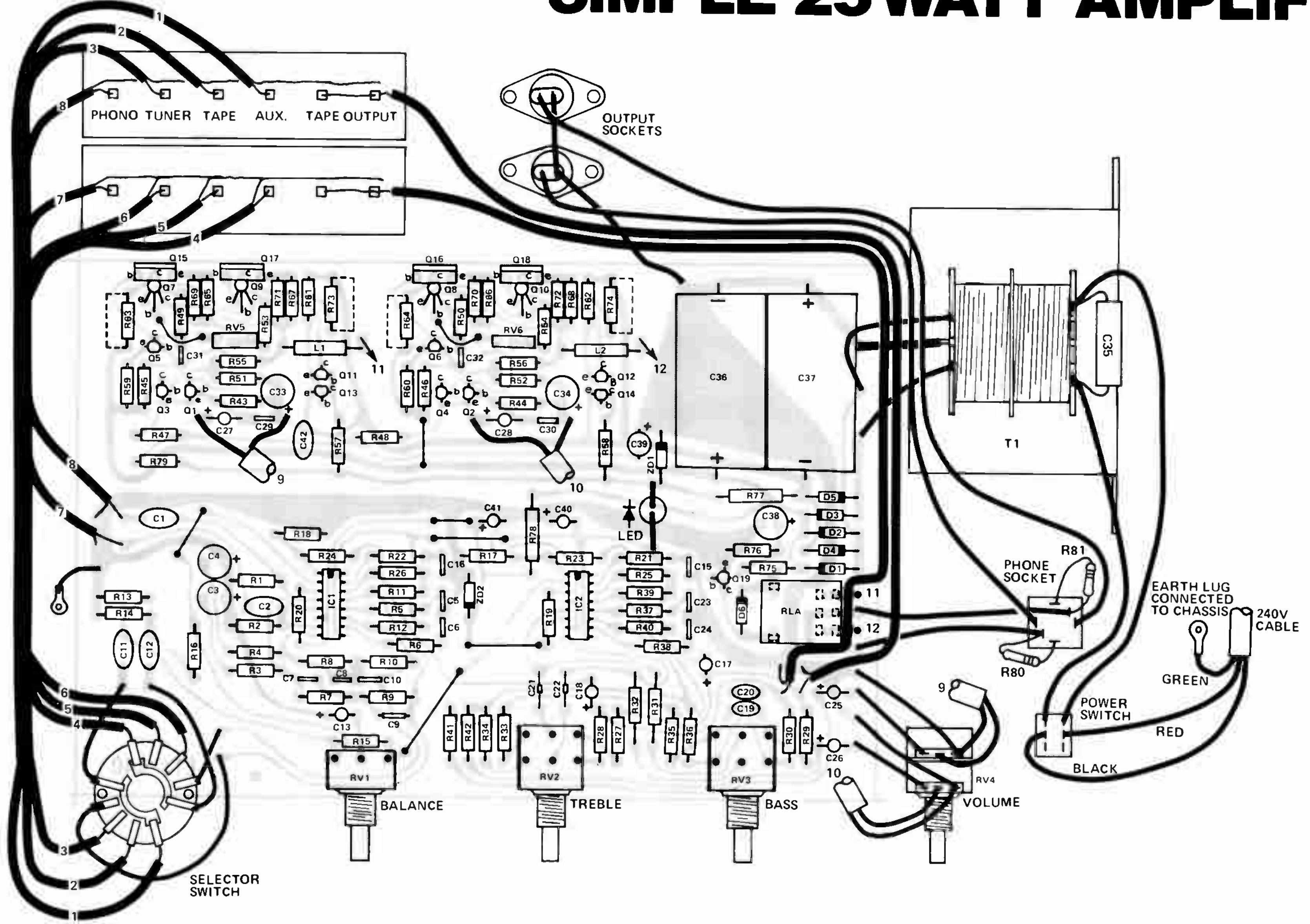
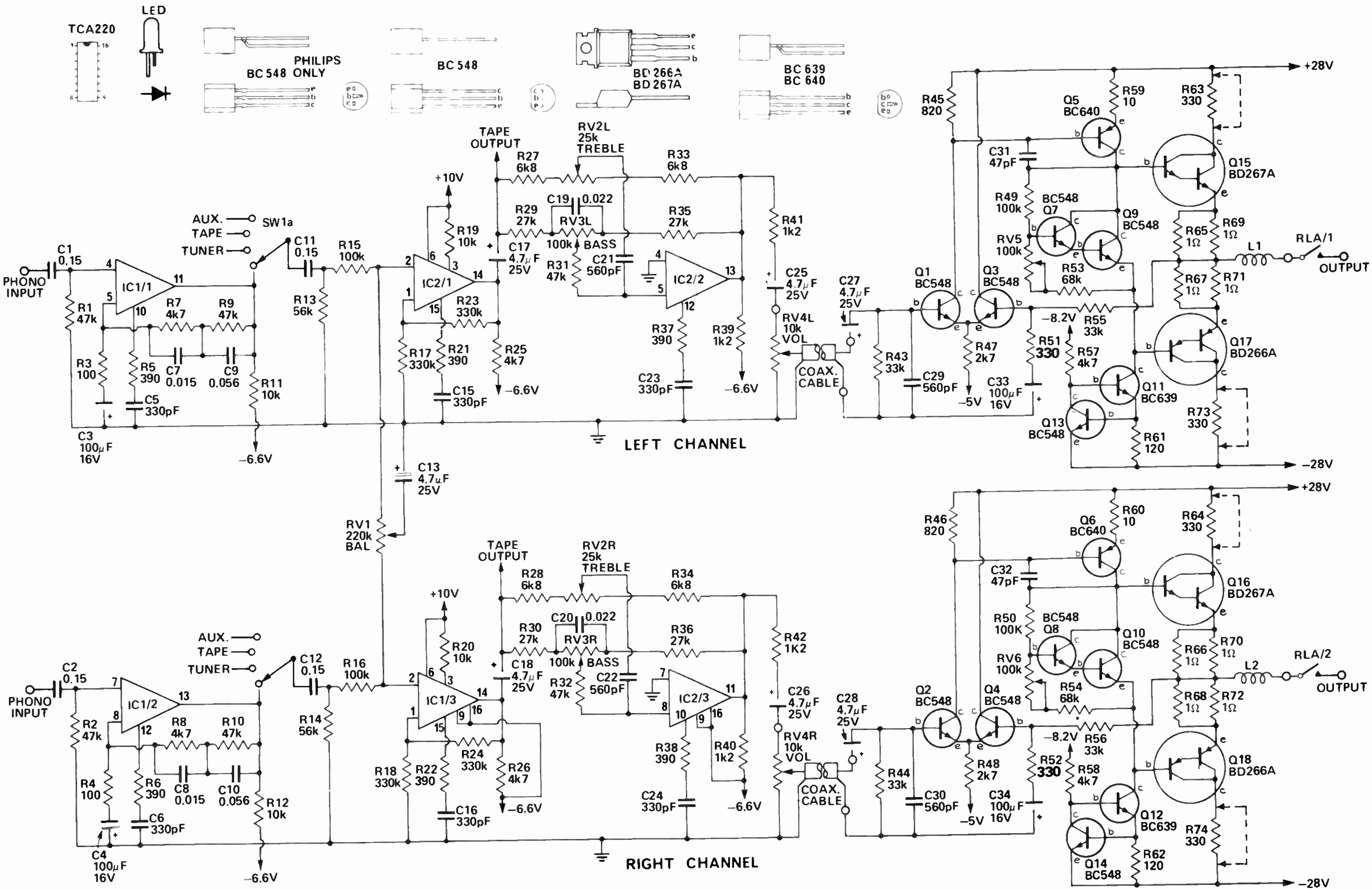


Fig. 2. Power supply for the 25 watt amplifier.

SIMPLE 25 WATT AMPLIFIER





TCA220
LED
PHILIPS
BC 548 ONLY
BC 548
BD 266A
BD 267A
BC 639
BC 640
PHONO INPUT
AUX.
TAPE
TUNER
SW1a
TAPE OUTPUT
RV2L 25k TREBLE
RV3L 100k BASS
RV4L 10k VOL
RV1 220k BAL
COAX. CABLE
RLA/1
OUTPUT
+28V
−28V
+10V
−6.6V
−8.2V
−5V
LEFT CHANNEL
RV2R 25k TREBLE
RV3R 100k BASS
RV4R 10k VOL
RLA/2
RIGHT CHANNEL
IC1/1
IC2/1
IC2/2
IC1/2
IC1/3
IC2/3
Q1 BC548
Q3 BC548
Q5 BC640
Q7 BC548
Q9 BC548
Q11 BC639
Q13 BC548
Q15 BD267A
Q17 BD266A
Q2 BC548
Q4 BC548
Q6 BC640
Q8 BC548
Q10 BC548
Q12 BC639
Q14 BC548
Q16 BD267A
Q18 BD266A
L1
L2

SIMPLE 25 WATT AMPLIFIER

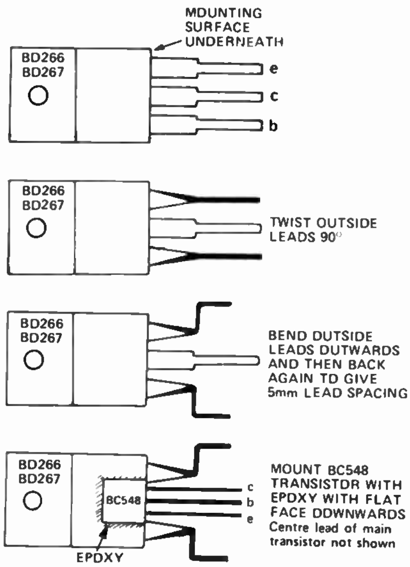


Fig.5. How to prepare the power transistor leads for installation.

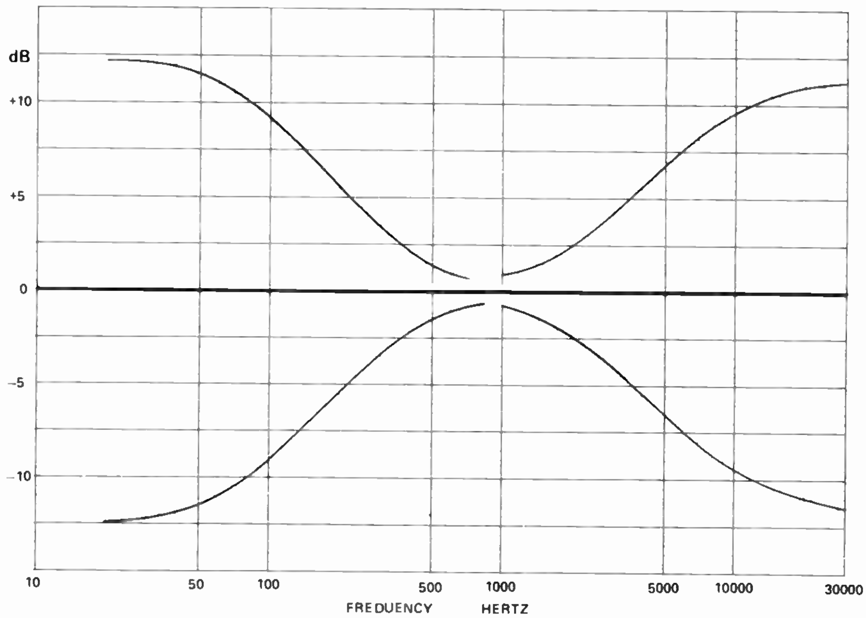


Fig.6. Tone control characteristics of the amplifier.

MEASURED PERFORMANCE OF ETI 440 AMPLIFIER

POWER OUTPUT	25 + 25 watts into 8 ohms		
FREQUENCY RESPONSE	+0 / −0.5 dB	15 Hz – 30 kHz	
	+0 / −3 dB	6 Hz – 80 kHz	
CHANNEL SEPARATION	1 kHz – 46 dB		
HUM AND NOISE (with respect to 25W)			
Phono (10 mV)	67 dB (unweighted)		
Other inputs	68 dB (unweighted)		
INPUT SENSITIVITY			
Phono	2.5 mV	47k	
Other inputs	200 mV	47k	

TOTAL HARMONIC DISTORTION

Power	Frequency	One channel only	Both channels
12.5W	100 Hz	0.1%	0.13%
	1 kHz	0.08%	0.16%
	10 kHz	0.12%	0.17%
20W	100 Hz	0.14%	0.5%
	1 kHz	0.12%	0.6%
	10 kHz	0.17%	0.8%
25W	100 Hz	0.5%	5.2%
	1 kHz	0.6%	4.8%
	10 kHz	0.7%	4.3%

TONE CONTROLS	
Bass	12 dB boost at 50 Hz
	12 dB cut at 50 Hz
Treble	9 dB boost at 10 kHz
	9 dB cut at 10 kHz
DIMENSIONS	340 x 88 x 210 mm

these positions. If a substitute is used for the BC639 and BC640 carefully check the pin connections as these types are unusual.

The integrated circuits may now be installed making sure that orientation is correct as indicated by the mark on the IC which is at the pin 1 end of the IC. Then mount the relay by passing the pins through the holes provided in the board and then bend the leads flush with the copper and solder them to the tracks.

The chokes L1 and L2 are made by winding about 25 turns of 0.4 mm copper wire (insulated) onto the body of a 10 ohm 1 watt resistor terminating the ends of the wire on the resistor leads. These may now be mounted on the board.

The balance, treble and bass controls should now have lengths of copper wire soldered to each of the terminals. They are then mounted, by passing the leads through the holes in the board, but are not soldered in position as yet. The front bracket should now be attached to the *component* side of the printed-circuit board and the potentiometers mounted to the panel. The leads from the potentiometers should then be drawn through the board as far as possible and then soldered in position. Then mount the heatsink bracket to the rear of the board using 9.6 mm spacers and countersunk screws.

The output transistors have to be prepared in a couple of ways before installation. The leads are too close

HOW IT WORKS – ETI 440

PREAMPLIFIER

In the preamplifier we have used two TCA220 integrated circuits each of which contain three identical operational amplifiers. These work similarly to the conventional op amp like the 709, 741 or 301 except the output is an emitter follower and needs a pull down resistor. An internal schematic diagram is given in Fig. 1. for those interested. Frequency compensation is accomplished by a 390 ohm resistor in series with a 330 pF capacitor connected to the appropriate terminal. The maximum voltage allowed on this IC is 18V. Since the output swing in the positive direction is less than that in the negative direction we have used +10V and −6.6V supplies to ensure equal clipping.

The magnetic pickup used on most good turntables has a low output and also needs equalization to perform correctly. We used part of the TCA 220 (IC1-1 and IC1-2) to amplify this signal (about 60 times or 35 dB at 1 kHz) and to provide the equalization required (+13 dB at 100 Hz and −14 dB at 10 kHz referred to the gain at 1 kHz). The output of this amplifying stage connects to the switch SW1 which selects the desired input. The signal from the cartridge is amplified before the selector switch to improve the signal-to-noise ratio.

After the selector switch we have the balance control (RV1) which attenuates either left or right channel as desired. The signal is then amplified, by a factor of two, to recover what is lost in the balance-control network and also to buffer the signal to give a low impedance output. The output drives the tone-control network and also the tape-output sockets.

The tone-control section uses the last sections of the TCA220 (IC2/2, IC2/3) with the bass and treble controls in the feedback network. These controls provide about 10 dB of boost and cut of both bass and treble. Resistors R27 and R33 set the limit of the treble boost and cut, while C21 controls the actual frequency where the treble control starts. Resistors R29 and R35 control the bass limits while C19 sets the frequency. The output of the stage is connected to the volume-control potentiometer RV4.

POWER AMPLIFIER

The power amplifier is of conventional design using a differential pair Q1 and Q3 followed by a common-emitter amplifier stage, Q5, working at a constant current (5 mA) supplied by Q11 and Q13. The output of Q5 is buffered by the output transistors Q15 and Q17. These are darlington transistors and have a current gain (Hfe) of over 750 at 3A. These transistors are biased on slightly (10 mA) to remove cross-over distortion and the bias is set by measuring the voltage across R63 or 73 (3V) while adjusting RV5. After bias adjustment is completed these resistors are shorted out to allow full power capability. Transistors Q7 and Q9 are physically joined onto Q15 and Q17 to provide accurate temperature indication and to ensure thermal stability.

The gain of the power amplifier stage is 100 and is set by the ratio of R55/R51. The earth reference for the power-amplifier input stage is supplied via the coax cables connecting to the preamplifier.

POWER SUPPLY

The power supply is a full wave rectifier with a centre-tapped transformer supplying ± 28V to the main amplifiers. The supplies for the preamplifier are obtained from a 10 V zener ZD2 and a 8.2V zener ZD1. The actual negative supply to the preamplifier comes via the LED on the front panel and is about −6.6 volts (1.6V across LED). A smooth −5V is also derived from the −8.2V and is used for the differential pair in the main amplifier.

The relay RLA is used to prevent the switch on transient reaching the speakers. After switch on there is a delay due to C38 of about 4 seconds before the speakers are connected. On switch-off the delay is only about 1 second.

together, and since they are mounted close to the board the transistors may be damaged if the leads are just pulled apart. Figure 5 shows the lead bending process which should be done carefully with a pair of long nose pliers. After bending, a BC548 (Philips only) should be epoxyed with flat side onto the face of these transistors.

It is preferable to use one of the slow drying epoxies as they appear to withstand the elevated temperature better. If such epoxy is dried in the 100-130°C range it will normally dry in about 30 minutes. Before glueing, however, it is best to scratch the type number on to the side of the output transistor to aid later identification.

When dry, the transistors can be mounted using insulation washers and a smear of silicon grease if available. The leads of the BC548 have to be bent out a long way but they should be long enough. If a small soldering iron is used these transistors can now be soldered in without removing the heatsink.

The rotary switch and volume control can now be mounted on to the front bracket. There are four links from the board to the rotary switch as shown in Figure 4, the rest of the connections going to the rear panel. There are also four links to the volume control and two coax cables which go

PARTS LIST

R65, 66	Resistor	1Ω	½W	5%
R67,68	"	1Ω	"	"
R69,70	"	1Ω	"	"
R71,72	"	1Ω	"	"
R59,60	"	10Ω	"	"
R3, 4	"	100Ω	"	"
R51,52	"	330Ω	"	"
R61,62	"	120Ω	"	"
R80,81	"	220Ω	1W	"
R63,64	"	330Ω	½W	"
R73,74	"	330Ω	"	"
R5,6,21	"	390Ω	"	"
R22,37,38	"	390Ω	"	"
R77,78	"	470Ω	1W	"
R45,46	"	820Ω	½W	"
R39,40,41	"	1k2	"	"
R42,79	"	1k2	"	"
R47,48	"	2k7	"	"
R7,8,25	"	4k7	"	"
R26,57,58	"	4k7	"	"
R27,28	"	6k8	"	"
R33,34	"	6k8	"	"
R11,12	"	10k	"	"
R19,20	"	10k	"	"
R29,30	"	27k	"	"
R35,36	"	27k	"	"
R43,44	"	33k	"	"
R55,56	"	33k	"	"
R1,2,9,10	"	47k	"	"
R31,32,76	"	47k	"	"
R13,14	"	56k	"	"
R53,54	"	68k	"	"
R15,16,49	"	100k	"	"
R50,75	"	100k	"	"
R17,18	"	330k	"	"
R23,24	"	330k	"	"
RV1	Potentiometer	220k lin single gang rotary		
RV2	"	25k lin dual gang rotary		
RV3	"	100k lin dual gang rotary		
RV4	"	10k log dual gang rotary		
RV5,6	"	100k trim pot		
C31,32	Capacitor	47pF ceramic		
C5,6,15	"	330pF "		
C16,23,24	"	330pF "		
C21,22	"	560pF "		
C29,30	"	560pF "		
C7,8	"	0.015μF polyester		
C19,20	"	0.022μF "		
C35	"	0.033μF 630 V		
C9,10	"	0.056μF polyester		
C1,2	"	0.15μF "		
C11,12,42	"	0.15μF "		
C13,17,18	"	4.7μF 25V electro		
C25,26,27	"	4.7μF 25V "		
C28,40,41	"	4.7μF 25V "		
C3,4,33	"	100μF 16V "		
C34,38,39	"	100μF 16V "		
C36,37	"	2500μF 35V "		
L1,2	Choke	25 Turns 0.4mm Cu Wire on a 10Ω 1W Resistor		
D1 — D5	Diode	EM401 or similar		
D6	"	IN914 " "		
LED1				
ZD1		BZX79 8V2		
ZD2		BZX79 C10		
Q1,2,3	Transistor	BC548 " "		
Q4,13,14,19	"	BC548 " "		
Q7,8,9,10	"	BC548 Philips only		
Q5,6	"	BC640		
Q11,12	"	BC639		
*Q15,16	"	BD267A or B		
*Q17,18	"	BD266A or B		

*insulation washers needed

IC1,2 Integrated Circuit TCA220

RLA Relay 2c/o contacts 1250Ω coil
T1 Transformer 40V cT @ 2A A&R 5755
SW1 Switch Rotary 2 pole 4 position
SW2 Switch miniature toggle 240V
Stereo Phone Socket
Two 6way RCA sockets
Two 2pin DIN sockets
CHASSIS
HEAT SINK
POT: SUPPORT BRACKET
COVER
ESCUTCHEON - rear panel escutcheon — 3 small knobs — 2 large knobs — 4 rubber feet — 2 9.6mm spacers — 3 core flex & plug rubber grommets.

SIMPLE 25 WATT AMPLIFIER

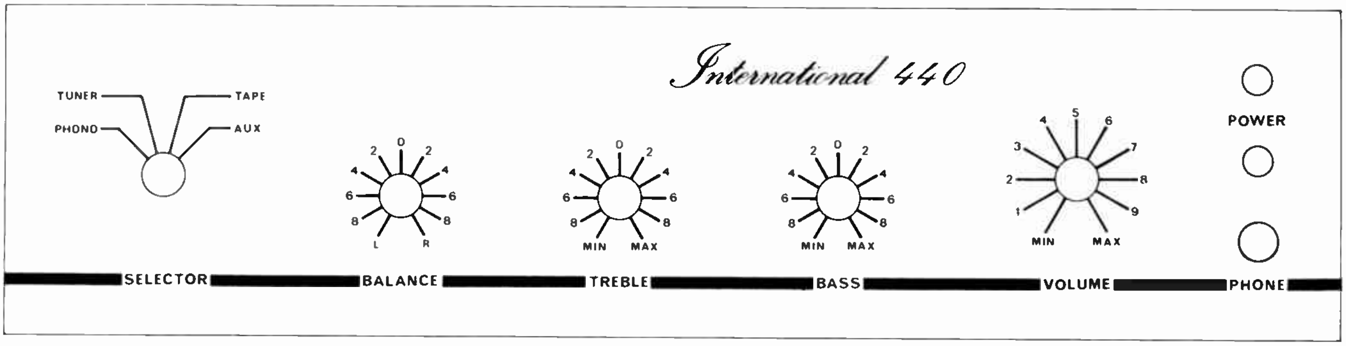


Fig. 7. Front panel artwork. Full size 335 x 83 mm.

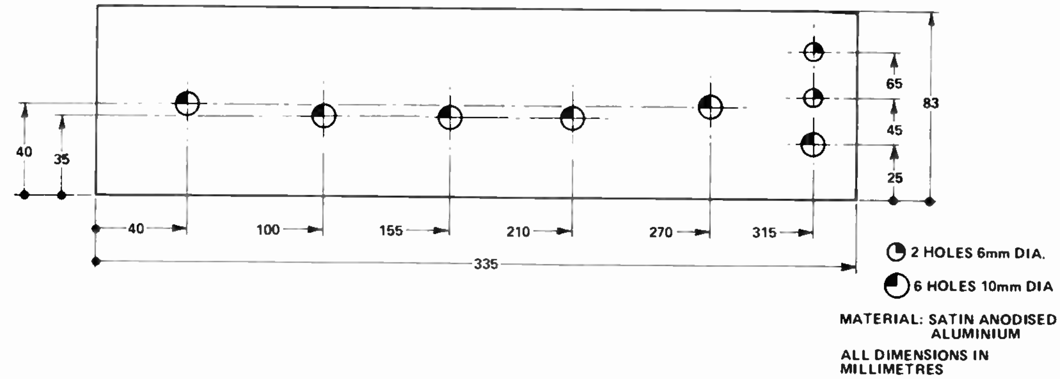


Fig. 8. Front panel details.

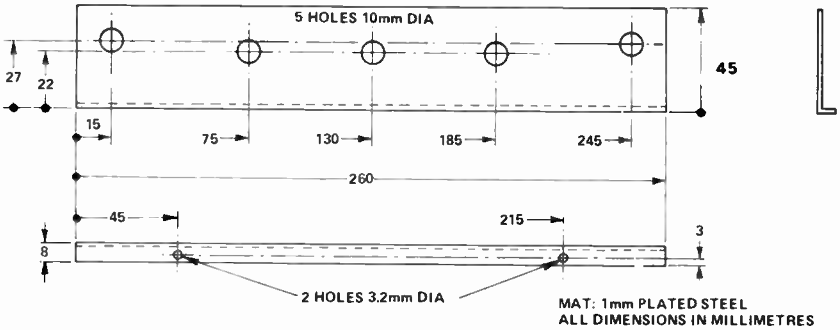


Fig. 9. Potentiometer support bracket.

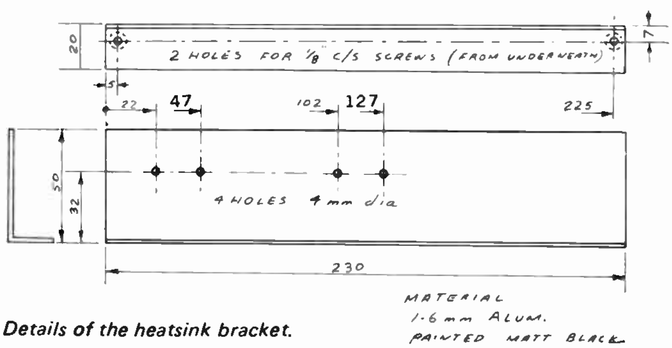
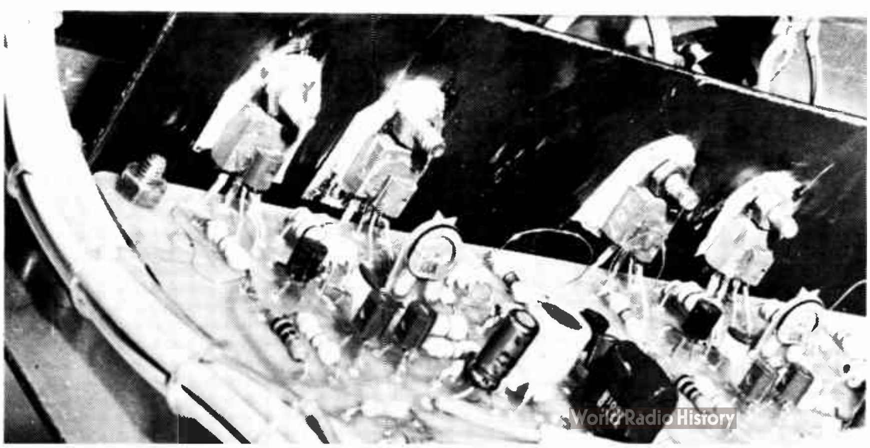


Fig. 10. Details of the heatsink bracket.



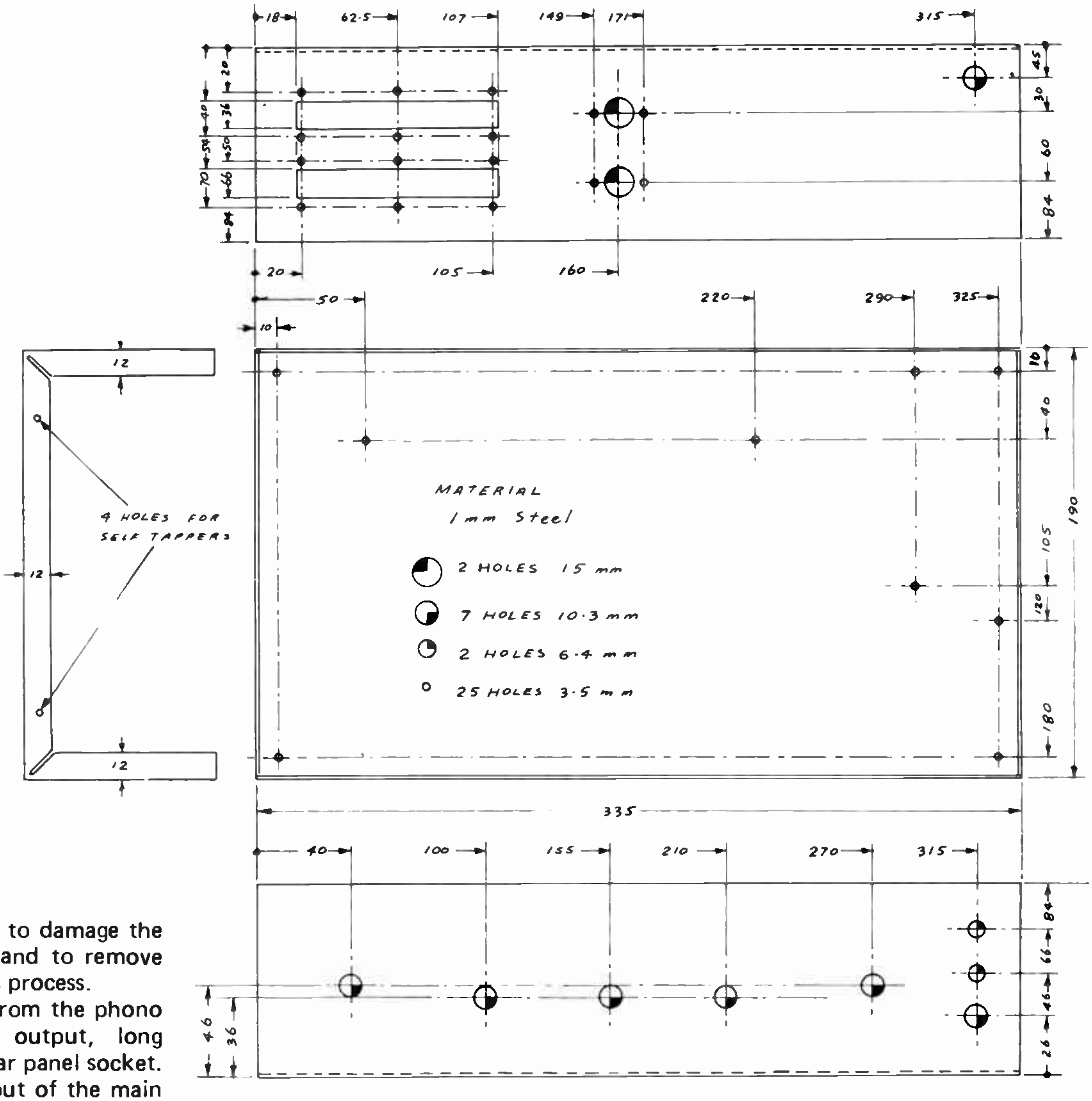
from the volume control to the main-amplifier inputs.

The chassis can now be assembled by mounting the transformer (terminals on the outside), the front panel, the phono socket, LED, speaker sockets, the 6-way RCA sockets, the rubber feet, the grommet for the power cord and the power cord itself. The screw for the cable clamp also mounts one of the rubber feet.

The printed-circuit board module can now be temporarily installed. If the potentiometers used have a long threaded portion (this depends on the brand) there may be room for extra nuts to hold the module and front panel on. If not, the nuts will have to be removed and refitted on the outside of the front panel. The module is held in by the potentiometer and by two self-tapping screws into the heatsink from the underside. Due to the variations in alignment of the mechanical parts, the location of the holes in the heatsink cannot be accurately determined. Therefore these holes have been left undrilled and can now be marked through the holes in the chassis. The unit can now be removed to facilitate drilling these holes to a size suitable for the self

Detail of power transistor assembly and installation. Note compensation transistors glued to output transistors (see text) and mica insulators between power transistors and chassis.

Fig. 11. Chassis details



tappers. Be careful not to damage the printed circuit board, and to remove any shavings during this process.

Connect coax cable from the phono input and the tape output, long enough to reach the rear panel socket. Leads to join the output of the main amplifier to the relay, and leads from the relay long enough to reach the phono socket can be installed along with the lead from the speaker common and the LED leads. To facilitate the assembly pins should be installed to the board where the transformer is connected.

The 240 V input cable can now be joined to the switch and then to the transformer primary along with the capacitor C35. The earth wire shall be bolted directly onto the chassis as shown. To prevent possible personal injury the switch and the transformer primary terminals should be taped up with insulation tape.

The printed-circuit board module can now be permanently reinstalled. The transformer secondary can now be connected and the rest of the wiring installed. The phono socket along with R80 and R81 can be wired according to Fig. 3.

This completes the assembly of the unit which is now ready for testing.

TESTING

Providing all components are in the correct place and all interconnections

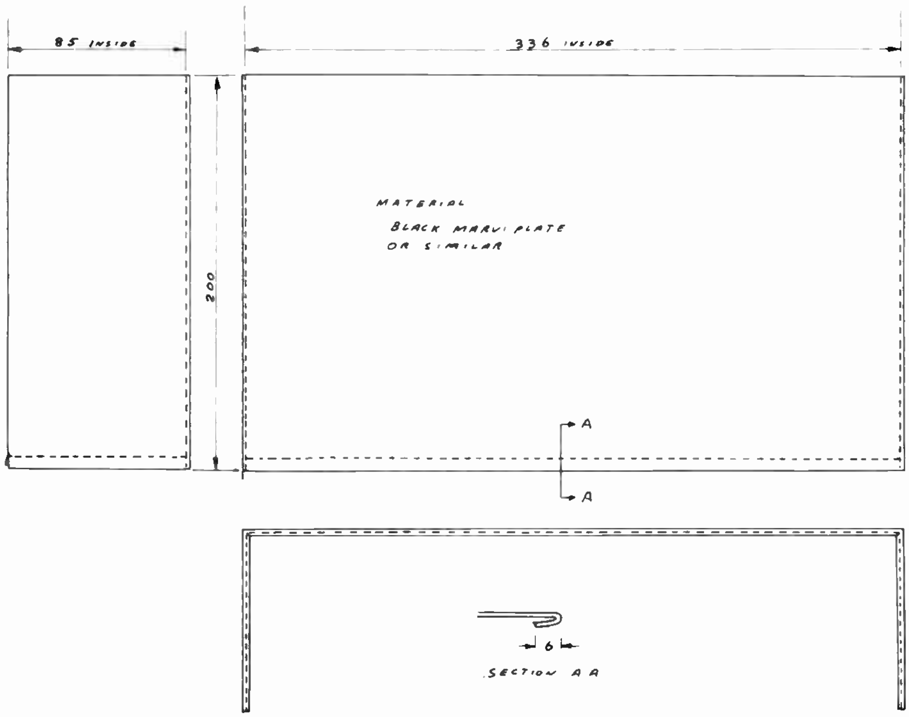
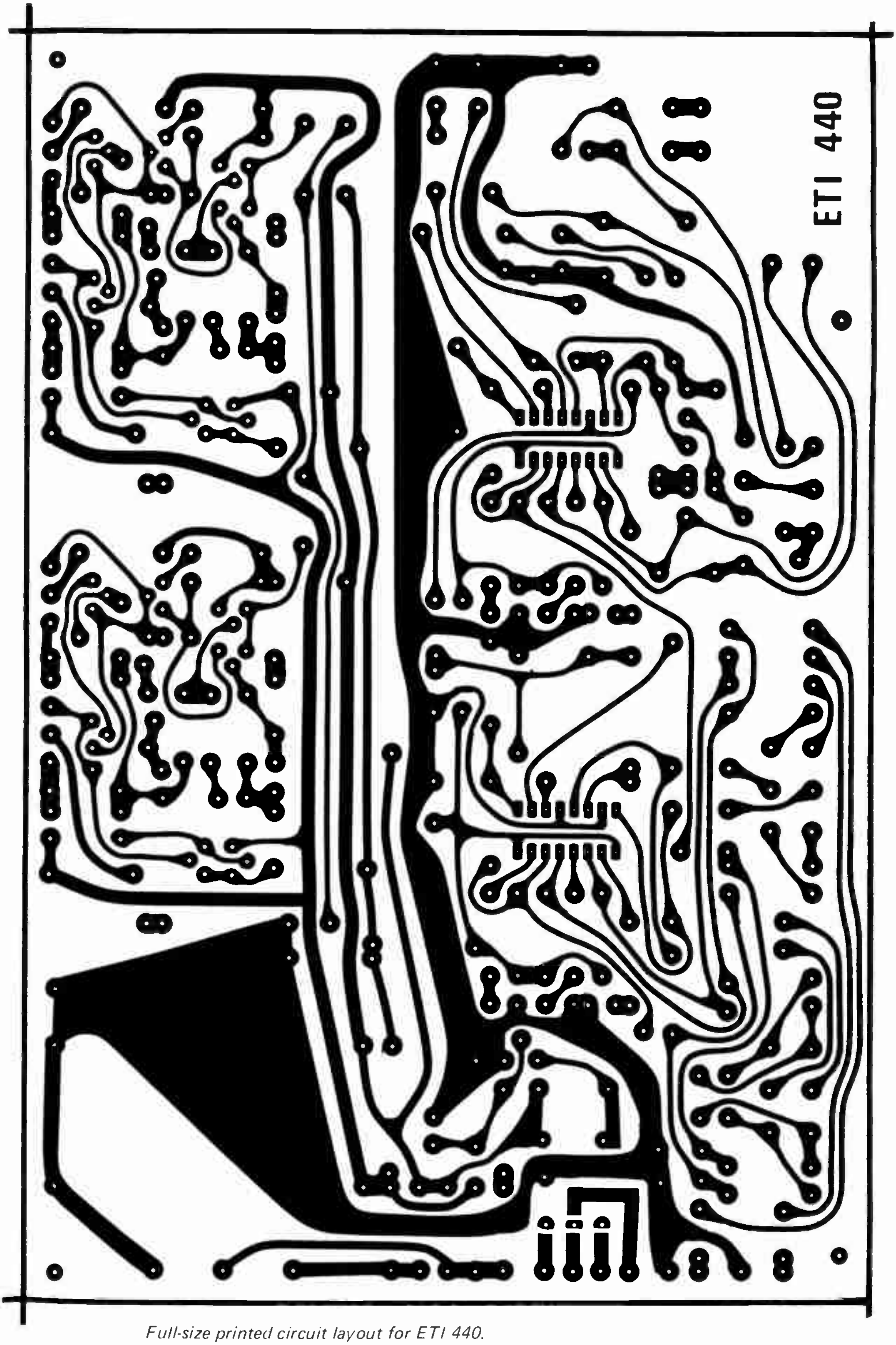


Fig. 12. Cover for the amplifier.



Full-size printed circuit layout for ETI 440.

SIMPLE 25 WATT AMPLIFIER

are correct the only adjustment is that to set the bias current in the output transistors.

Before switching on rotate the trim potentiometers, RV5 and 6, fully clockwise i.e. toward the transformer. Switch on without speakers connected and measure the voltage across R63 and adjust RV5 to give about 3 volts. Repeat the process with the other channel and R64 and RV6. The resistors R63, 64, 73 and 74 can now be shorted out (after switching off) by short links of wire soldered onto the leads of the resistor.

If a fault exists in the output stage, either a transistor is shorted to the heatsink or the bias setting is faulty etc. In such a case the resistors R63, 64, 73 and 74 will overheat and may burn out. This effectively protects the output transistors. ●

FAULT FINDING

PROBLEM	POSSIBLE FAULT AND CHECKS
R63 or R73 gets hot (only one)	shorted insulation on Q15 or Q17
R63 and R73 gets hot (both)	bias current too high
Bias current not adjustable down to within limits	Q7 and/or Q8 faulty or wrong polarity. Voltage between base of Q15 and base of Q17 should be about 2.3 Volts
Bias current too low or zero	check output voltage, if about 0V then possible shorted Q7 or Q8
Output voltage high (near supply rail)	check current source Q11 is working Voltage across R61 should be about 0.65V. Check voltage across R45 it should be almost 0V (output high) if it is suspect Q5. If not check voltage at base of Q1 and Q3. Q3 should be higher than Q1 if so suspect Q1 or Q3
Output voltage low	check voltage across R45 should be about 0.7V if >0.7V suspect Q5. If less than 0.5V measure voltages at base of Q1 and Q3. Q3 should be lower than Q1 if so suspect Q1 or Q3
Main amplifier has no gain	faulty or disconnected C33, R51 or R53 wrong value
Main amp appears OK but pre amp does not work	check supply voltages or pin 6 (+10 V) and pins 9 and 16 (–6.6 V) Check output voltage of each individual amplifier. They should all be about 0V if not check components in local area.

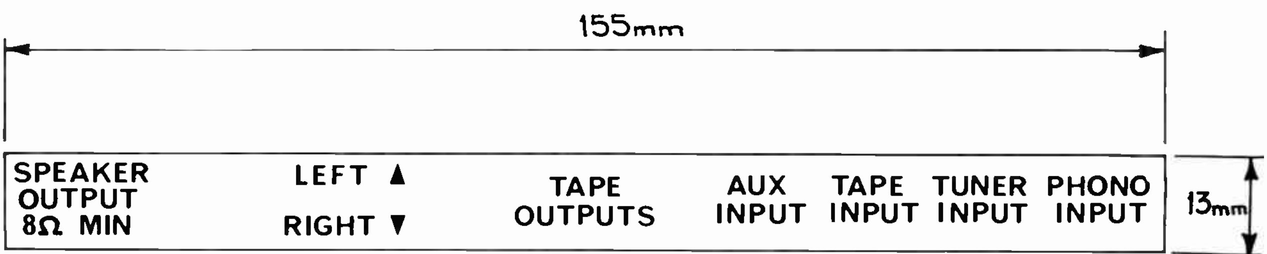


Fig. 13. Artwork for rear panel.

Loudspeaker Protection

IT IS generally safe to drive a loudspeaker from an amplifier of several times the speaker's power rating — as long as the volume control is not wound fully up — or much treble boost applied, there is little danger of burning out drive units. Nevertheless it is advisable to insert fuses in the speaker leads.

This chart, prepared by Altec Lansing indicates the fuse values that should be used.

Ideally, start with the lowest rated fuse in each category — increase to the 'good' value if the 'safest' one blows out too often. Never exceed the 'maximum' though.

FUSE VALUE SELECTION

3 Ag fuse ratings in amperes. (Do not use slow-blow-fuses).

Power rating of speaker		4-ohm speaker	8-ohm speaker	16-ohm speaker
15-25 watts	Safest	1	½	¼
	Good	2	1	½
	Maximum	4	2	1
25-35 watts	Safest	1½	¾	3/8
	Good	3	1½	¾
	Maximum	6	3	1½
35-50 watts	Safest	2	1	½
	Good	4	2	1
	Maximum	8	4	2
50-75 watts	Safest	2½	1½	¾
	Good	5	3	1½
	Maximum	10	6	3

GENERAL PURPOSE PREAMPLIFIER

A general purpose stereo preamplifier using a single LM382 IC which can be tailored for use with magnetic pickups, tape recorders or microphones by changing a few components.

WE HAVE HAD MANY REQUESTS for the circuit of a simple preamplifier module suitable for fitting into an existing system. The requirements differed — many people required a module to amplify a magnetic pickup, whilst others wanted a unit that could be used for a tape recorder or microphone.

Whilst these requirements usually require different circuitry, a preamplifier based on the LM382 IC can be made to do any one of these jobs simply by changing a few components around the basic amplifier circuit. As a straight preamplifier the frequency response extends to well beyond 20 kHz and gains of 40, 55 and 80 dB can be selected by means of simple component changes.

To use the preamplifier for your application select the appropriate component values as detailed in Table 1.

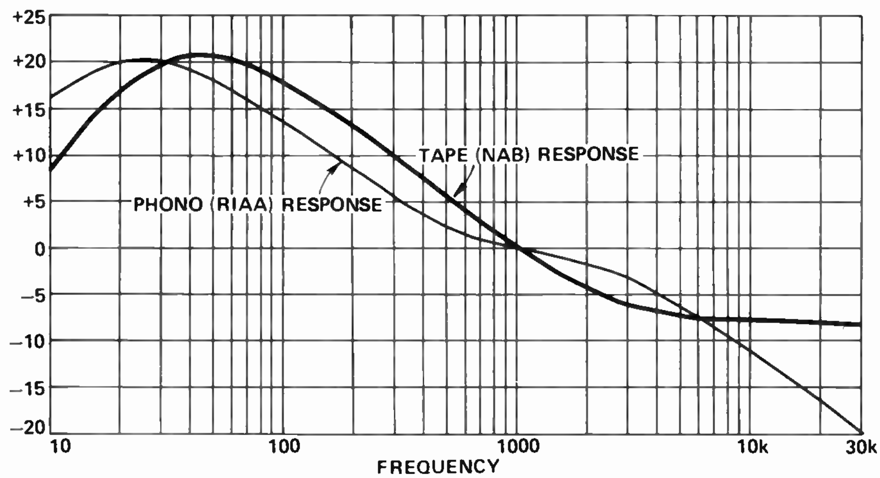


Fig. 1. Frequency response of the NAB and RIAA versions of the preamplifier.

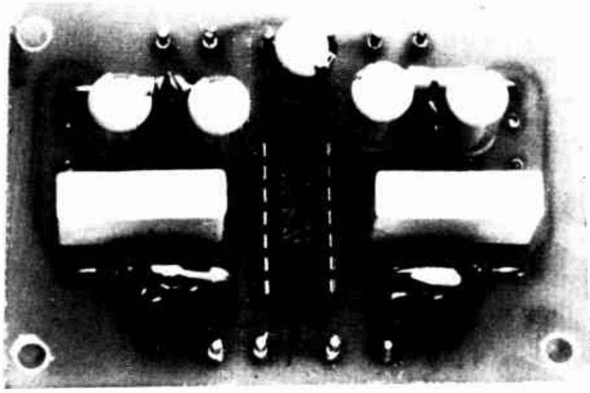
TABLE 1.

FUNCTION	C3, 4	C5, 6	C7, 8	C9, 10	R1, 2
Phono preamp (RIAA)	330n	10µ	10µ	1n5	1k
Tape preamp (NAB)	68n	10µ	10µ	—	—
Flat 40dB gain	—	—	10µ	—	—
Flat 55dB gain	—	10µ	—	—	—
Flat 80dB gain	—	10µ	10µ	—	—

Construction

Strictly speaking a printed circuit board is not necessary and any method, such as Veroboard or Matrixboard, may be used if desired. However, the neatness and ease of construction offered by the use of a proper printed-circuit board cannot be matched.

After determining what components are required from Table 1, assemble the board as shown in the component overlay diagram. The input cables must be shielded as the signals at the input are at very low levels. If trouble with hum pickup is encountered it may be necessary to mount the whole preamplifier in a metal box to shield it.



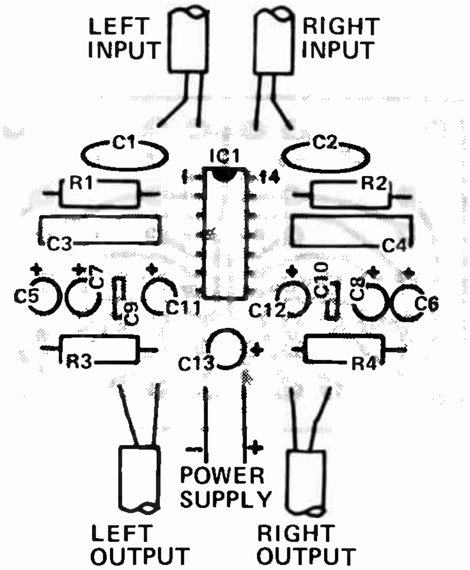


Fig. 4. Component overlay.

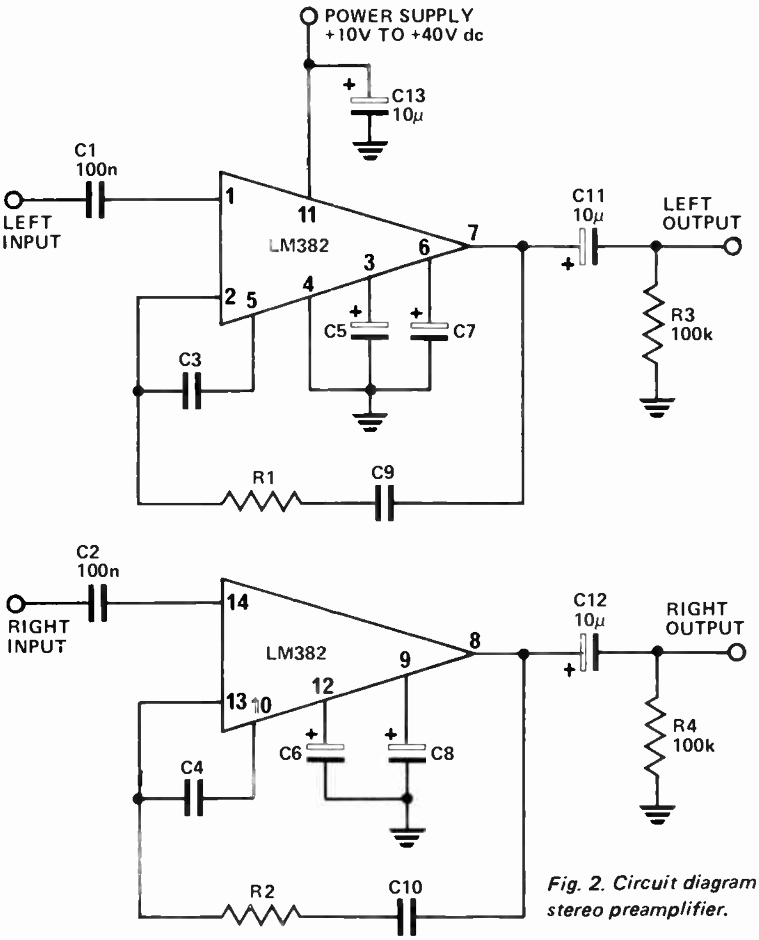


Fig. 2. Circuit diagram of the stereo preamplifier.

PARTS LIST – ETI 445

Resistors

R1, 2	see table 1		
R3, 4	100k	½watt	5%

Capacitors

C1, 2	100n polyester
C3 – C10	see table 1
C11-C13	10μ 25V electro

Note that C13 should be rated at 50 V for supply voltage above 24V

IC1	integrated circuit LM382
PC board	ETI 445

10 PC board pins.

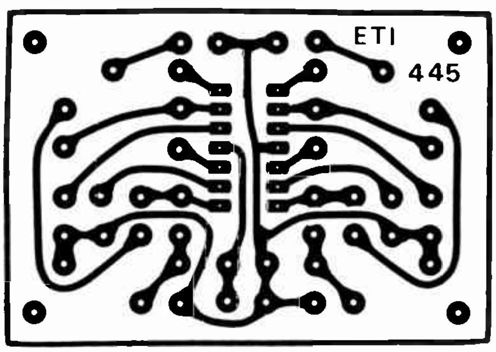


Fig. 3. Printed circuit layout. Full size.

How it works

Not much can be said about how the LM382 works as most of the circuitry is contained within the IC. Most of the frequency-determining components are on the chip – only the capacitors are mounted externally.

The preamplifier may be powered by any dc voltage between 10 and 40 volts, the output being automatically biased to about +6 volts. Due to this bias the output must be decoupled from the following stages and this is done by C11, 12 and R3, 4.

The LM382 has the convenient characteristic of rejecting ripple on the supply line by about 100 dB, thus greatly reducing the quality requirement for the power supply. Thus the power rails of the main amplifier may be used if accessible.

Bucket Brigade Audio Delay Line

This audio delay line uses the latest in IC technology, the 'Bucket Brigade' to give a simple unit suitable for various effects. However this is a project for the experimenter as full details of how to use it for any particular use are not given.

ANYONE WHO has been in an anechoic chamber will appreciate the need for some reverberation. In music the use of artifical reverberation or echo can compensate for a 'dead' room or create a new effect. Up until recently reverberation was normally obtained by mechanical means such as a spring or plate which is vibrated or excited by an electrical signal; a pickup elsewhere on the plate or spring receives the delayed signal. Due to the nature of resonances in springs, multiple echos occur giving the effect of reverberation.

A single echo is obtainable by using a tape loop, recording the signal on one head and playing back through a second. The distance between the heads and the tape speed determines the delay. Echo can also be obtained acoustically by a long tunnel with a microphone and speaker.

When the price of digital ICs started to come down a number of digital delay lines were developed. These used an A–D (analogue to digital) converter, a long shift register and finally a D–A converter. To accomodate the wide dynamic range required very good, fast, A–D, D–A converters along with a large shift register. Even with the low price of ICs these units still cost around $500.00 or so (this is the main reason we have not published one as a project).

A number of years ago several IC manufacturers started playing with a 'digital' delay line which works by storing an analogue voltage on a capacitor and then transferring this voltage to another and then successive capacitor. This is accomplished by switching FETs on and off under digital control. The circuit became known as a bucket brigade and this name has stuck.

The IC we have chosen is the MN3001 which is a dual 512 step device. This was chosen mainly for its availability through Elcoma. Brief specifications of other devices we know about are given below. All the devices except the SAD 1024 (Reticon) are handled by Elcoma.

Uses of BBD

Variable or fixed delay of analog signals
Reverberation
Echo
Tremolo, vibrato, flanging or chorus effects
Voice control of tape recorders
Time compression of telephone conversations
Voice scrambling

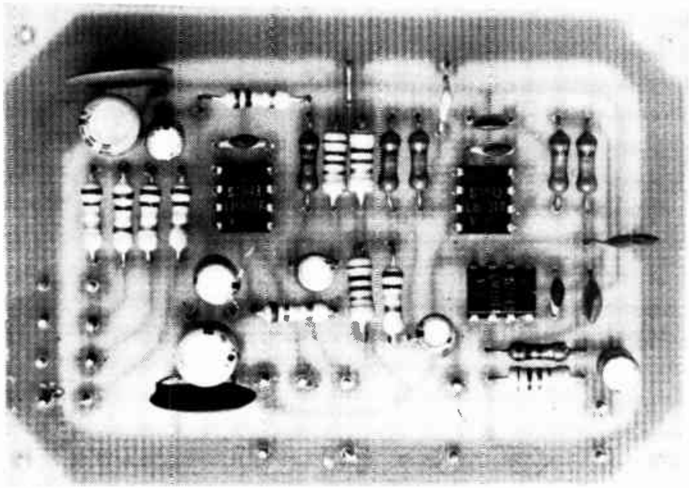
Construction

As we are describing no mechanical arrangement our description of construction is limited to the assembly of the PC board. It is recommended that a socket be used for the BBD IC as it is an expensive MOS device. The inputs are protected but it should be handled with care. The same care should be taken with the CMOS IC but as a socket costs more than the IC it cannot be recommended!

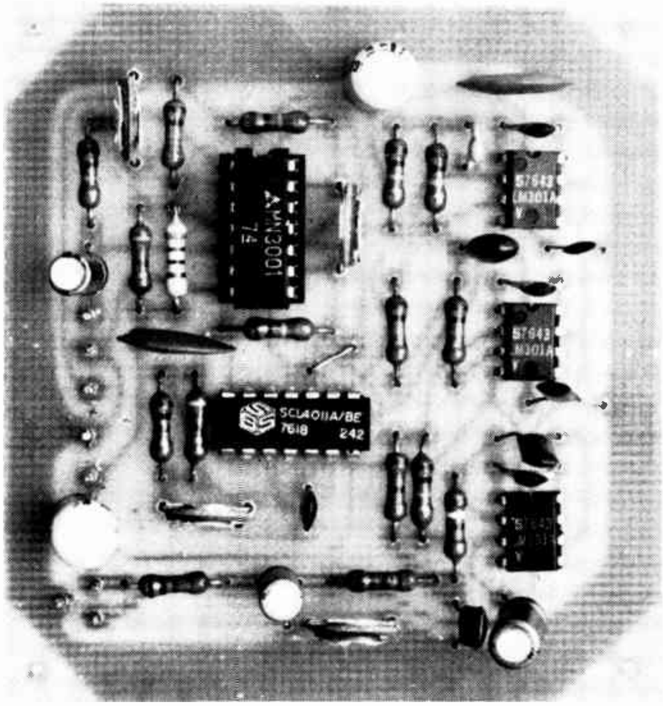
The interconnection between the pc boards depends on the effect needed.

SPECIFICATION – ETI 450

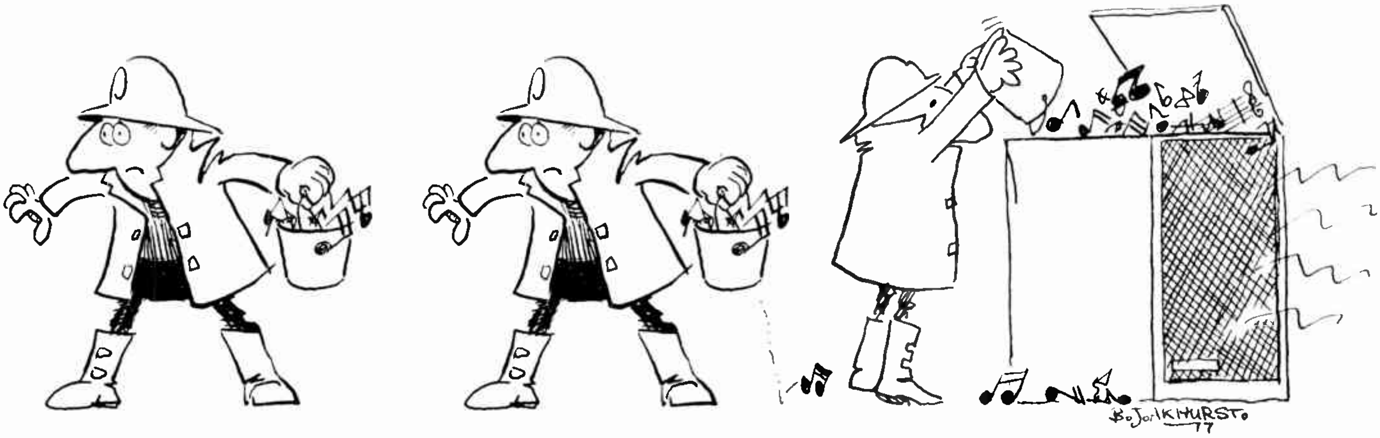
Maximum input < 3% distortion		2.0V RMS
Delay time internal oscillator		6 – 30ms
Frequency response		see graph
Distortion 1V in 1kHz		0.3%
Signal to noise re 2V input		67dB
Supply current (A)	+ 5V	6mA
	– 15V	9mA
(B)	+ 5V	6mA
	–15V	6mA



The mixer, filter board ETI 450B.



The bucket brigade board ETI 450A.



Bucket Brigade Audio Delay Line

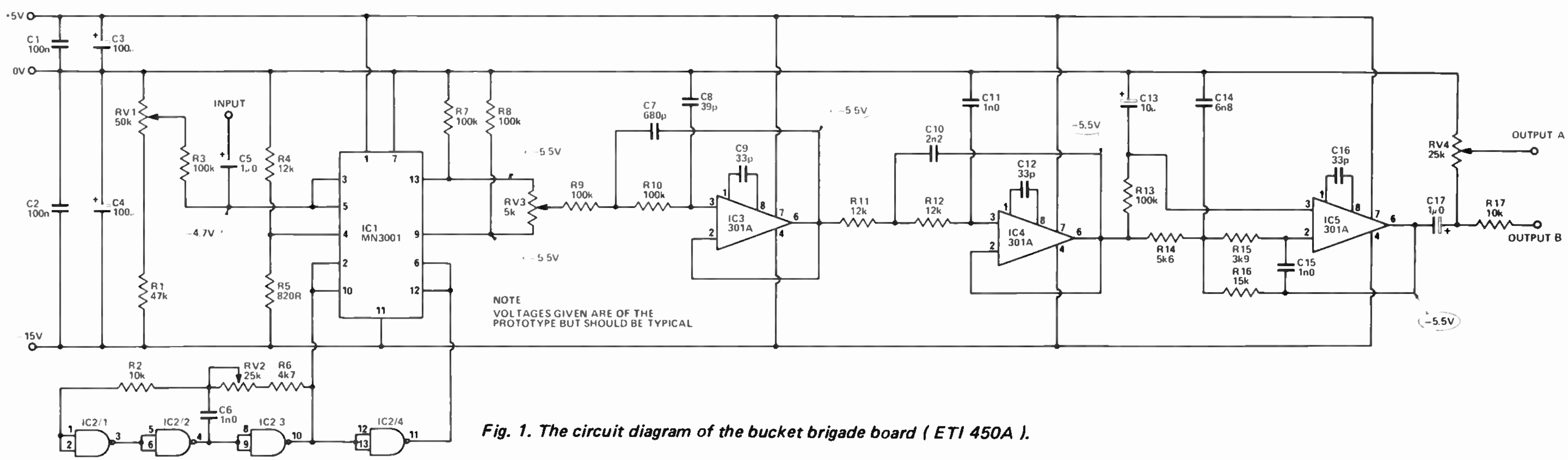


Fig. 1. The circuit diagram of the bucket brigade board (ETI 450A).

HOW IT WORKS – ETI 450

The bucket brigade device is an analogue delay line which samples the input waveform at an instant in time and stores this voltage on a capacitor. As we need more than just one point on the waveform we sample the input at least 3 times faster than the highest frequency required. A single capacitor cannot store more than one voltage at one time and so a series of capacitors is used. Before the second sample is taken the energy in the first capacitor is transferred to the second capacitor thus freeing the first to sample the input again. Then before the third sample the energy in the second capacitor is transferred to the third. The first into the second and the first again samples the input. This process continues on each sample with the energy in each capacitor being transferred to the next. Eventually we run out of capacitors and this then becomes the output. The number of capacitors, or stages, and the sample (clock) frequency determine the time it takes an input sample to appear at the output.

In the device we have used there are 512 stages in each of two identical and independent sections. The internal circuit diagram of the initial part and of the output stage is shown below (there are over 1000 capacitors and 2000 FETs in the IC!)

The transfer of energy is done using FETs which are controlled by the two clock lines CP1 and CP2. These are complementary square wave signals. Using a 40 kHz signal the input is sampled every 25μs then 'remembered' and transfered every 25μs. On the output, from stage 509 on, the signal is divided into two paths, one having an extra stage. This is needed as the signal on the output is only there for half the 25μs period. By adding these two out-of phase outputs a continuous output results.

All of this transferring of energy does however waste energy and the output is of a lower amplitude than the input. In the MN3001 it is about 8.5dB lower. To increase the delay it is normal to connect two sections (or more if needed) in series. However the output has then twice the loss and even with an intermediate amplifier this results in a lower signal to noise ratio.

A second method of obtaining a large delay is to run the two sections in parallel with each sampling on alternate half cycles of the clock wave-form giving effectively two sampling periods per clock pulse. This allows the clock frequency to be halved for the same frequency response giving twice the delay with only one attenuation loss. However as you never get anything for nothing the lowering of the clock frequency increases the low frequency energy content of the noise, making the filter do more work.

Getting back to the circuit diagram we see that the input signal is coupled to the input of both halves of the BBD with dc biasing being provided by RV1. IC2 is used as an oscillator with frequency adjustable from about 20kHz to 90kHz giving delays of 6-30 ms. The output of IC2/3 is inverted by IC2/4 giving the two complementary clocks required by the BBD. The outputs of the BBD are mixed with RV3 being used to remove the clock frequency before the 6 pole filter IC2 – IC4 removes all the other hash generated by the clocking. The first two sections of this filter have unity gain while the third stage has a gain of 8.5dB to compensate for the loss in the BBD. These gains are of course below the cut off point!

The second board used is simply a mixer and 4 pole filter which can be used together or in separate parts of the unit. Due to the sampling done by the BBD, the frequency of an input signal must not exceed the clock frequency otherwise it will appear at the output at some other frequency lower than the clock frequency. This is due to the BBD input circuit sampling almost corresponding points on successive cycles of the input waveform. For this reason the 4 pole filter is used before the BBD.

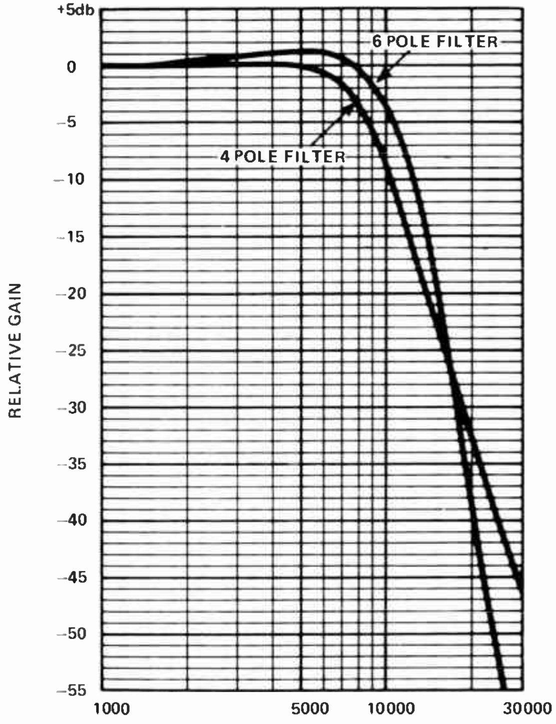
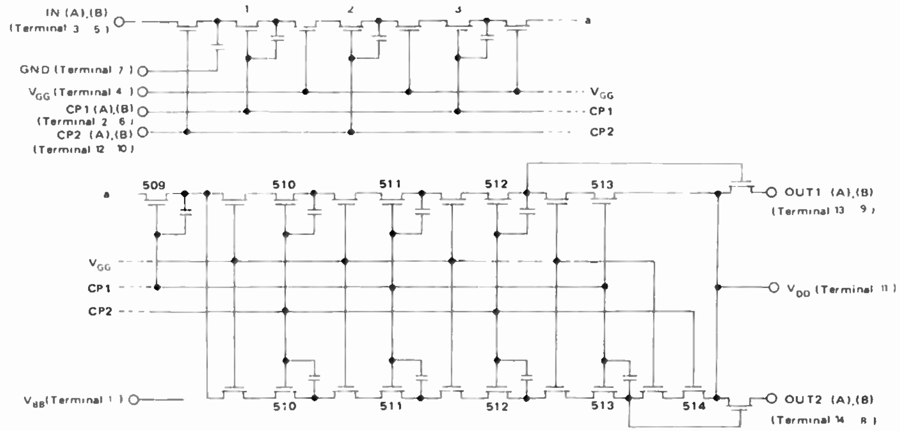


Fig. 3. The frequency response of the two filters. The overall response is approximately the sum of these two filters provided the clock frequency is at least 20 kHz.



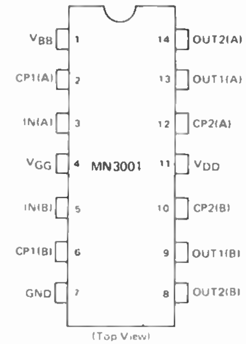
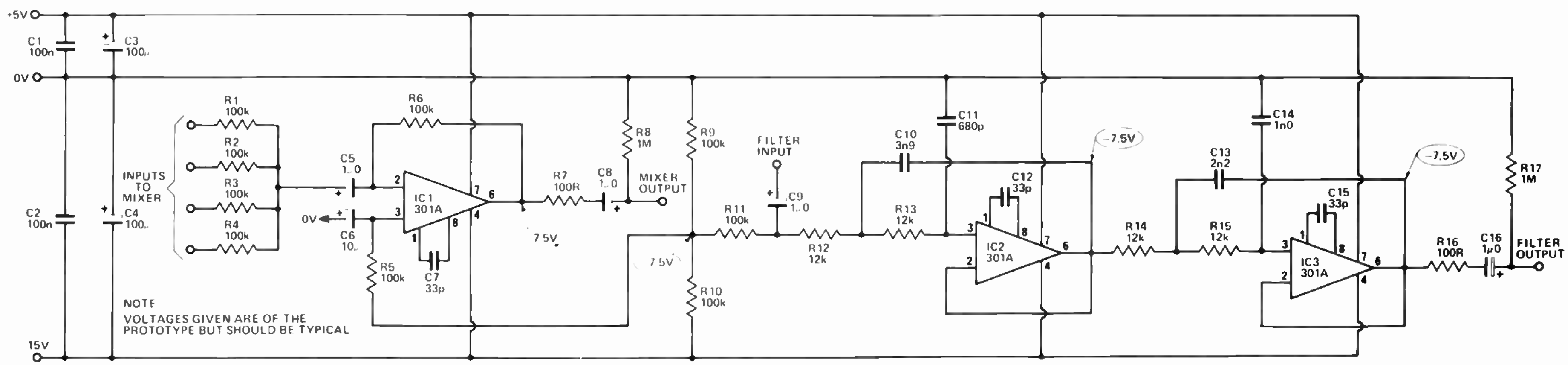


Fig. 2. The internal circuit of the MN3001 showing the first three and last four stages.

TYPE	MN 3001	MN 3002	MN 3003	MN 3004	TDA 1022	SAD 1024
NO OF STAGES	2 X 512	512	2 X 64	512	512	2 X 512
DELAY (ms)	1–25	1–25	0.16–3.2	2.5–3.2	0.5–50	0.2–170
INSERTION LOSS (dB)	8.5	8.5	3.5	1.5	4.0	
DISTORTION (%)	0.4	0.4	0.5	0.4	0.4	1.0
SIGNAL TO NOISE (dB)	70	70	> 68	85	74	> 70
SUPPLY VOLTAGE (V)	+5, −14, −15	+5, −14, −15	−8, −9	−15	−15	−15

Fig. 4. Summary of the bucket brigade ICs which we know exist.

Fig. 5. The circuit diagram of the mixer, 4 pole filter board (ETI 450B).



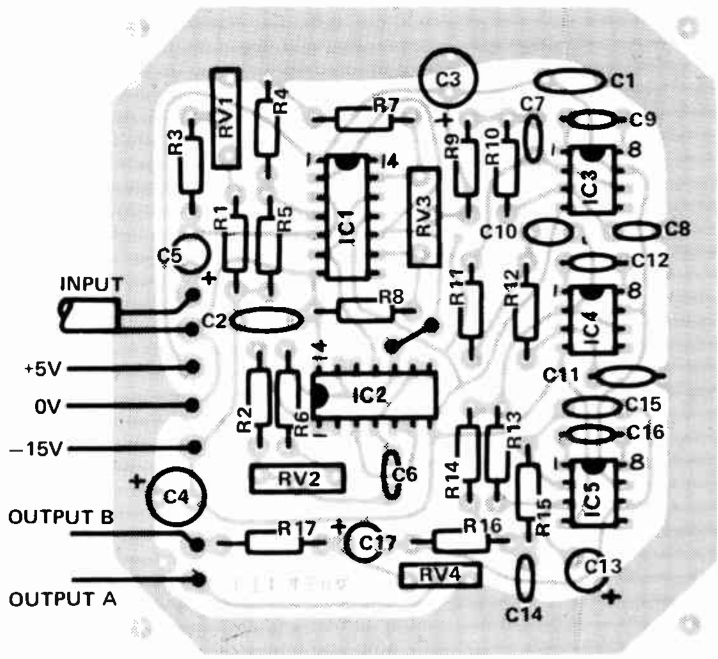


Fig. 6. The component overlay of the bucket brigade board.

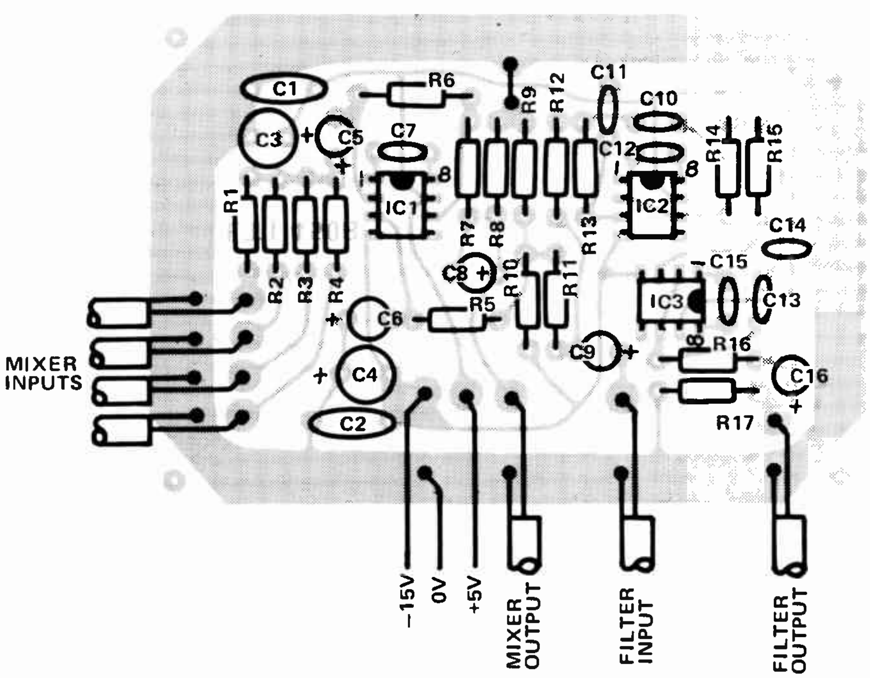


Fig. 7. The component overlay of the mixer — filter board.

Adjustment

RV1 is used to set the bias voltage. If an oscilloscope is available look at the output of the board while feeding in a sine wave signal. Adjust RV1 to allow the maximum input signal without clipping. RV2 adjusts the delay while RV4 sets the output level to compensate for differences in the loss of the BBD sections. RV3 is used to remove the clock frequency from the output. If an oscilloscope is available look at the wiper of RV3 and adjust to give the smoothest output. The switching transients at this point are very high but these are removed by the filter.

PARTS LIST – ETI 450A

Resistors all ½W 5%

R1	47k
R2	10k
R3	100k
R4	12k
R5	820R
R6	4k7
R7–R10	100k
R11,12	12k
R13	100k
R14	5k6
R15	3k9
R16	15k
R17	10k

Potentiometers

RV1	50k trim
RV2	25k trim
RV3	5k trim
RV4	25k trim

Capacitors

C1,2	100n polyester
C3,4	100μ 25V electro
C5	1μ0 25V electro
C6	1n0 polyester
C7	680p ceramic
C8	39p ceramic
C9	33p ceramic
C10	2n2 polyester
C11	1n0 polyester
C12	33p ceramic
C13	10μ 25V electro
C14	6n8 polyester
C15	1n0 polyester
C16	33p ceramic
C17	1μ0 25V electro

Semiconductors

IC1	MN3001
IC2	4011 (CMOS)
IC3–IC5	301A

Miscellaneous
PC board ETI 450A

PARTS LIST – ETI 450B

Resistors all ½W 5%

R1–R6	100k
R7	100R
R8	1M
R9–R11	100k
R12–R15	12k
R16	100R
R17	1M

Capacitors

C1,2	100n polyester
C3,4	100μ 25V electro
C5	1μ0 25V electro
C6	10μ 25V electro
C7	33p ceramic
C8,9	1μ0 25V electro
C10	3n9 polyester
C11	680p ceramic
C12	33p ceramic
C13	2n2 polyester
C14	1n0 polyester
C15	33p ceramic
C16	1μ0 25V electro

Semiconductors
IC1–IC3 . . . 301A

Miscellaneous
PC board ETI 450B

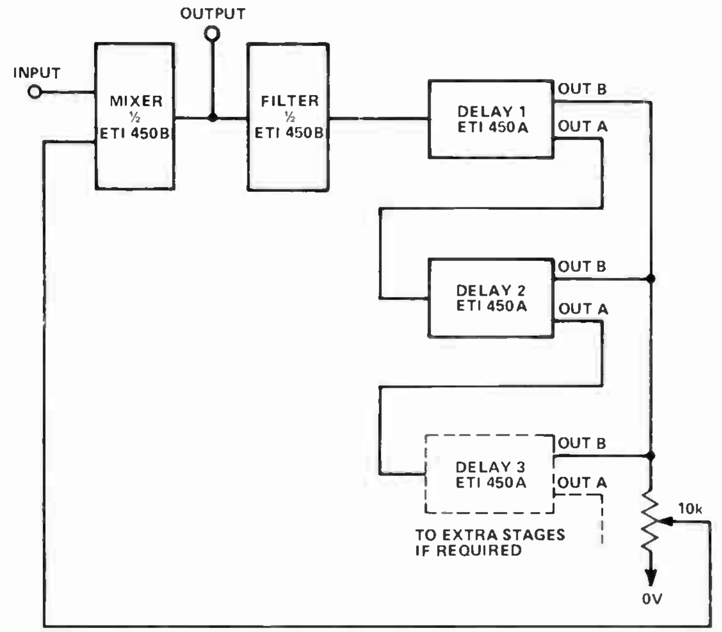


Fig. 8. The interconnection for reverberation.

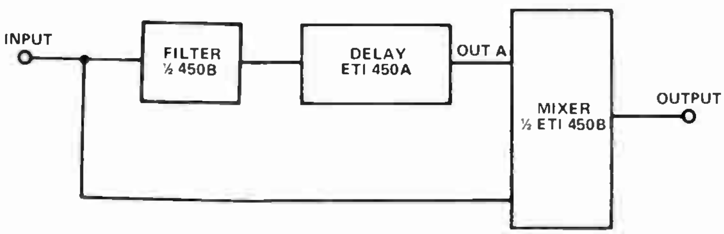
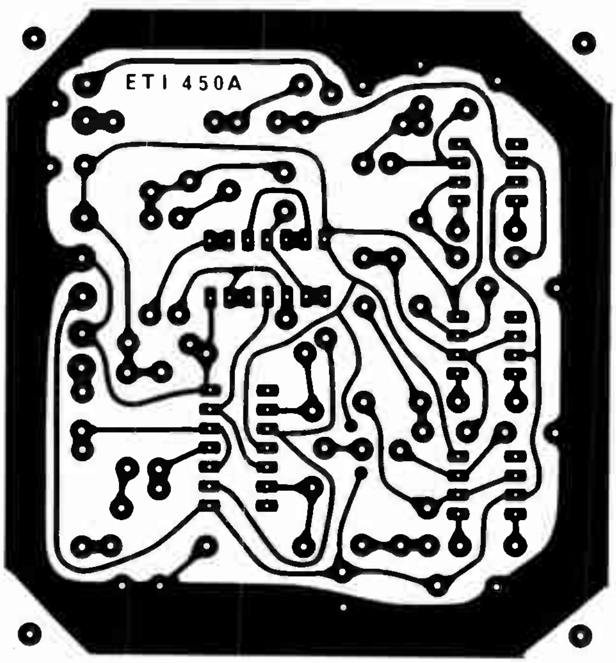
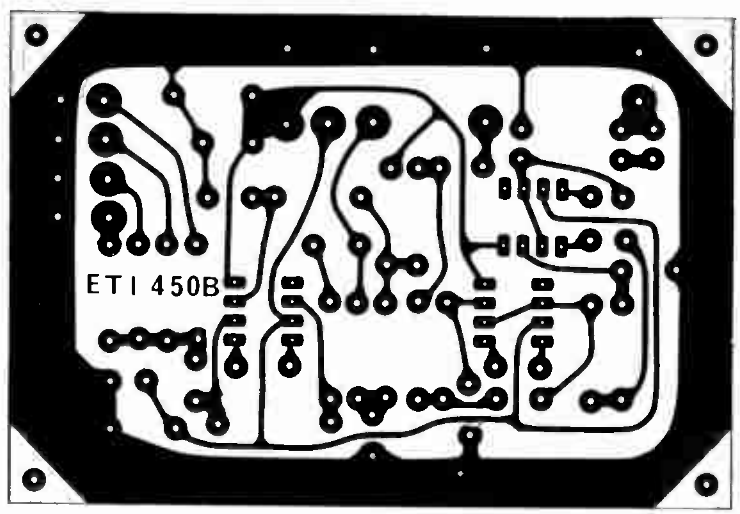


Fig. 9. Connections for a single echo. With a short delay this becomes a phaser.





Reverberation

If the audio signal is fed via a mixer into the delay line and its output fed back into the mixer we have a feedback system which will repeat a single sound many times. This is reverberation. If several different delays are used the effect will seem more natural. With all feedback systems if the sum of all the delayed outputs exceeds the original sound uncontrolled oscillations will result. This is similar to howl-round in PA work and careful adjustment is needed if long reverberation times are required.

Echo

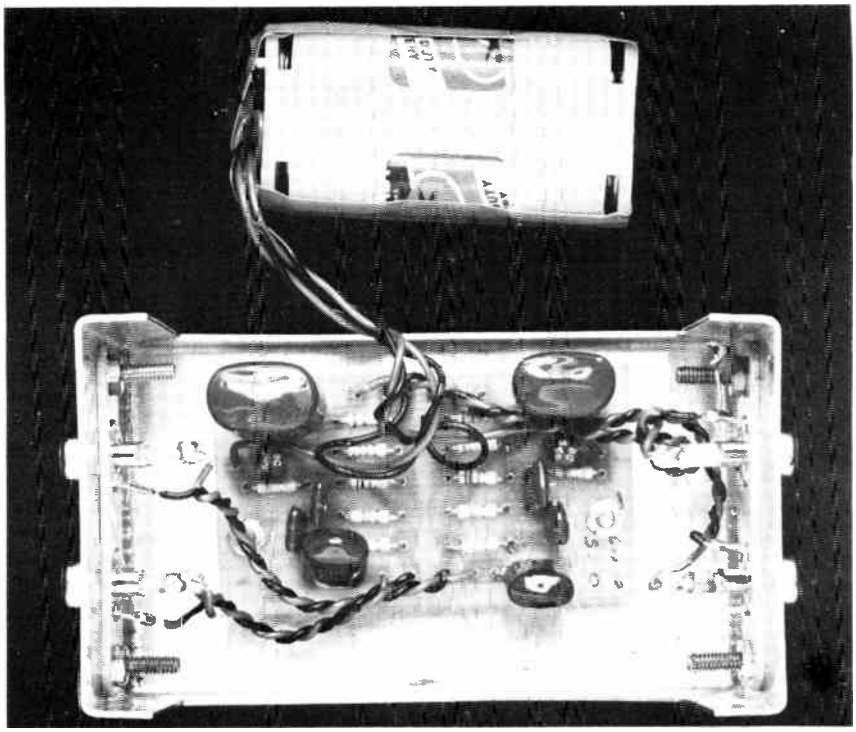
This is similar to reverberation except the delayed signal is not fed back to its own input. A single echo only results (from a single delay) and it can be of any amplitude in relation to the original signal.

Phasing (Flanging)

By varying the delay times and by mixing in the right proportions total cancellation of some frequencies can occur. Now if the clock frequency is made variable a phasing or flanging effect occurs. A variable clock can be made by replacing potentiometer RV2 by an LDR and illuminating it with a globe the brilliance of which is controlled (try a 555 timer). We must leave details of this to the individual constructor.

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

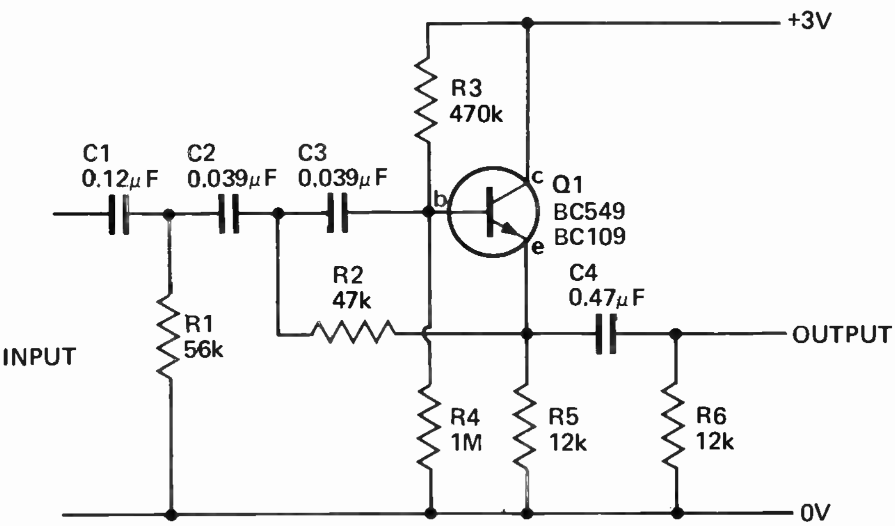


Fig. 1. Circuit diagram of the rumble filter. Two required for stereo.

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 át 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,

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).

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.

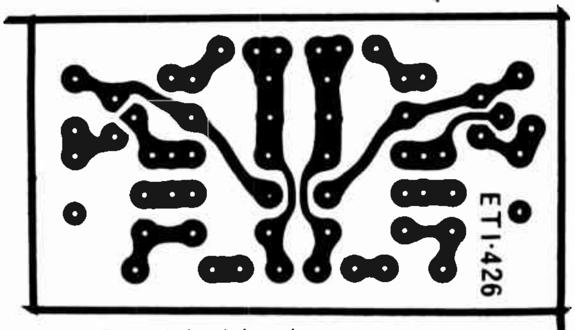
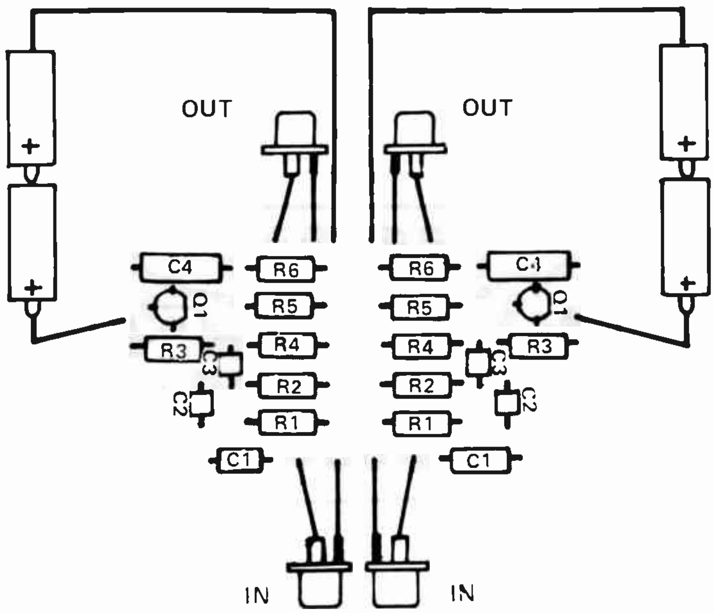
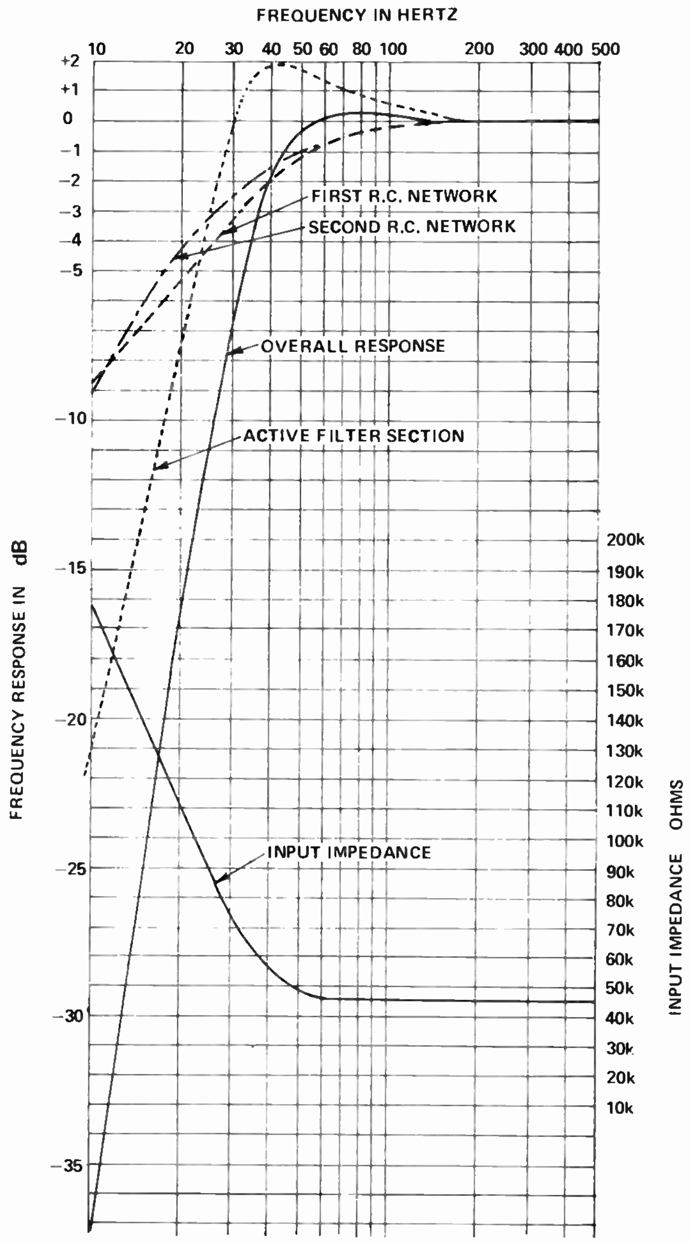


Fig. 2. Printed circuit board layout for the rumble filter 40mm x 70mm.



STEREO RUMBLE FILTER



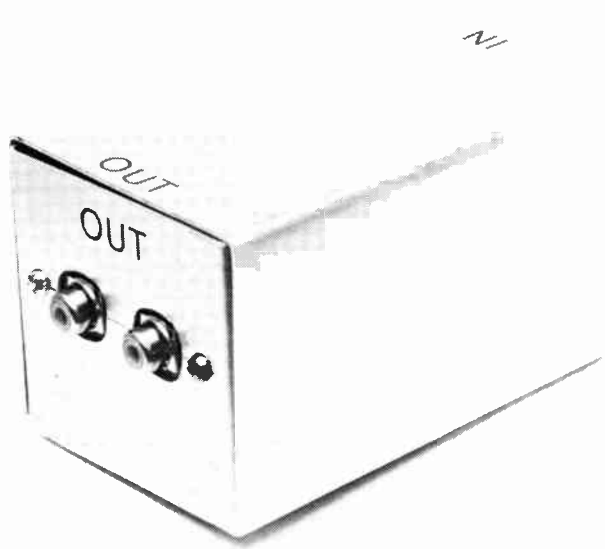


Fig. 4. Characteristics of the rumble filter.

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. ●

OVER LED

Unit flashes a light when your hi-fi system is overloading.

MOST PEOPLE ARE aware of distortion when they turn up the volume control on their hi-fi system too far – but rarely know the cause.

Nine times out of ten the distortion is caused by 'clipping'. That is, the amplifier has insufficient reserve power to handle the peak music 'transients' at the volume required. During such peaks – which may demand power outputs of 100 watts or more – the amplifier is driven into overload and as a result the music peaks are clipped.

A few modern amplifiers are equipped with peak limit indicators. These consists of a 'VU' meter or an arrangement of lights which flash when the peak limit is exceeded. Peak limit indicators are interesting devices because they show very effectively just how much power is required for realistic non-distorted reproduction (much less than you might think for most of the time and a very great deal more occasionally).

The 'Over-Led' indicator described here flashes a warning light if the power level at which clipping occurs is exceeded. It can be built into any existing hi-fi amplifier – or made as a separate unit.

Two completely independent channels are shown so that each channel of a stereo system may be monitored separately. Nevertheless we have designed the unit so that you can make it up if you want to use it tc monitor just a single channel (mono) system.

Figure 1 shows one complete single channel – there are leads coming from the speaker terminals of one channel, and there is one LED. If you look at

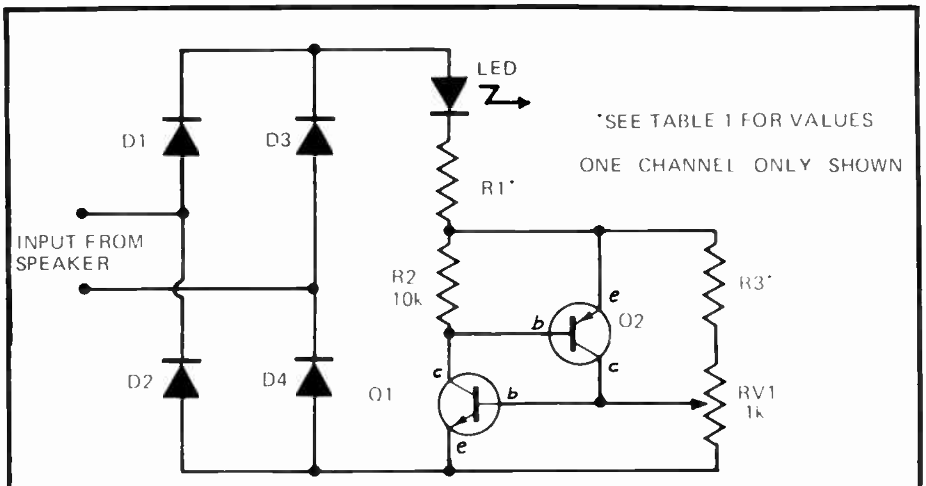


Fig. 1.

the component overlay board shown in Fig 2 you will see that there are two identical sets of components. Each set of components is for one channel. So if you want to build a single channel version just build up half of the board. If you know someone else who is building up a single channel unit you can cut the board in half and share the cost.

CONSTRUCTION

Usual precautions. Make sure all the diodes are the right way round, particularly the LEDs. The LEDs won't be damaged (in this circuit) by wrong polarity – but they won't work. Note that the values of R1 and R3 must be chosen specifically for each application from Table I. Most speakers used in hi-fi systems are 8 ohms but do check first to see. The finished board may be mounted as you wish. It's very small – our component overlay is shown here full-size and can be mounted inside most existing amplifiers. The LEDs can be located so that their tips just protrude through holes in the front panel. LEDs last for ever so you can secure them permanently in position – once you're sure the device is working!

The leads shown on the drawing as 'input from speaker' are connected across the amplifier's speaker terminals. Polarity is not important but make sure you don't mix up the leads between channels. It's best to twist each pair to make sure.

Another way of housing the 'Over-Led' is to build one into each speaker so that each has an LED visible near the top of the front panel — or mounted in a small separate enclosure sitting on the top of each enclosure. If you do this simply connect the input to the speaker leads where they enter the speaker enclosure.

CALIBRATION

The best way to calibrate this unit is to connect an audio oscillator across both input channels of your amplifier. Then, with the amplifier's volume control set low, adjust the oscillator to 1 kHz sine wave. Set both RV1 trimpots on each channel so that their wipers are nearest R3. Increase amplifier volume until you hear a sudden harshness of tone — i.e. the onset of clipping.

Don't leave the volume control at this setting for more than a second or two since few amplifiers or speakers can tolerate sine-wave inputs at this level for extended periods. Once the clipping point has been established, turn the volume down and then up again momentarily meanwhile adjusting both RV1s until the point is reached where each LED just comes on.

Repeat this procedure a few times finally arriving at settings where the LEDs come on just before clipping.

If you don't have access to an audio oscillator you can use a record which has a track of a solo instrument such as a flute. A recording of the human voice is also very effective. In such cases follow the procedure described above.

NOTE: This is an updated version of a project originally published as ETI 417 in Electronics Today International August 1973. The original version used transistors which are now less readily available than those shown here.

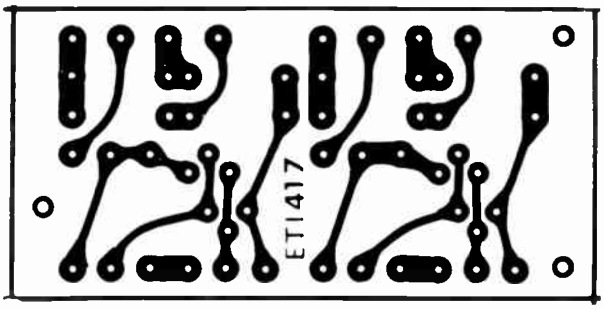
PARTS LIST — ETI 087

Note: one set required per channel.

R1 resistor see Table 1.
R2 resistor 10 k ½ watt 5%
R3 resistor see Table 1

RV1 trim potentiometer 1 k

D1-D4 diodes EM401 or similar
Q1 transistor BC548
Q2 transistor BC558
LED light emitting diode FLV110, MV5025, HP4403 or similar
Printed circuit board ETI 417
(1 only required — each board has provision for two channels).
Or Veroboard.



Printed circuit board layout — shown full size.

TABLE 1

SPEAKER IMPEDANCE						
RMS watts per channel	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

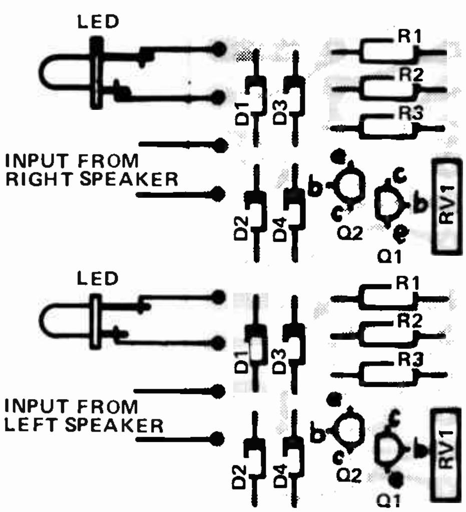
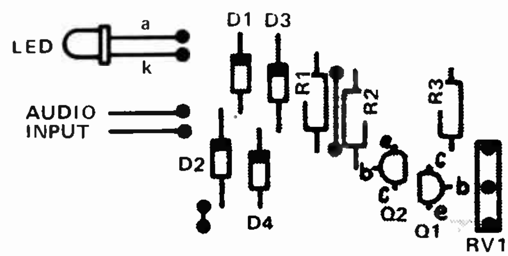
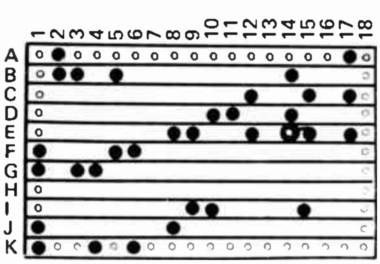


Fig. 2. This is the component overlay of the printed circuit version. Note that the board contains two separate but identical circuits.





This is the drilling details for the Veroboard version. Do make sure that the breaks required (shown as black dots) are clean, and that no 'whiskers' remain.

HOW IT WORKS

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

Transistors Q1 and Q2 are (together) equivalent to a sensitive gate SCR. If the voltage at Q2's base 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 falls to zero.

When transistors Q1 and Q2 are on, the current flowing through them also flows through the LED causing it to light up. Resistor R1 limits peak current through the LED to about 100 mA.

The range of calibration potentiometer RV1 is set by R3. The values of R1 and R3 should be chosen from Table 1. These values are not critical. If your amplifier's output is other than shown the nearest values will do.

ACTIVE CROSSOVER

Optimize your multi-way system with this electronic approach to crossover design.

NO SINGLE loudspeaker can adequately handle the whole range of audio frequencies in high-fidelity reproduction. Thus to obtain the best possible fidelity we must resort to multiple speaker systems where each driver is designed to cover one portion only of the audio spectrum.

This means that some method must be used to divide the audio spectrum, from the amplifier, so that an individual driver only receives the band of frequencies for which it was designed. This is especially important for midrange and tweeter drivers for they are seldom capable of handling frequencies lower than a specified limit without being damaged.

PASSIVE CROSSOVERS

In simple systems a single capacitor may be used to block low frequencies and pass only highs to a tweeter. But unfortunately such a capacitor only provides 6 dB per octave attenuation. With some tweeters this attenuation is not sufficient to suppress the resonant frequency of the tweeter. The driver could thus be damaged when operated at high power levels. Additionally, the presence of frequencies other than those in the desired pass – band leads to high levels of intermodulation distortion and a general 'muddiness' of reproduction.

Hence all good multi-way systems use networks which provide at least 12 dB per octave attenuation, in the stop band, to control the audio band presented to each drive unit. A typical network for a three-way system is given in Fig.1. To keep power losses down in such networks the coils must have dc resistances of less than one ohm. This means that heavy gauge wire must be used, making the coils large and expensive. Additionally the high value of capacitance required would normally call for the use of non-polarized electrolytics, however, there are several disadvantages with these, Firstly, the tolerance on non-polarized electros is plus or minus 50%! This means that a crossover using them could quite easily give a system which had peaks and/or deep holes in the response. Additionally such capacitors have disadvantages such as limited life, fairly low working voltages and problems due to leakage. Thus all good crossovers use polyester capacitors which, again, are rather expensive.

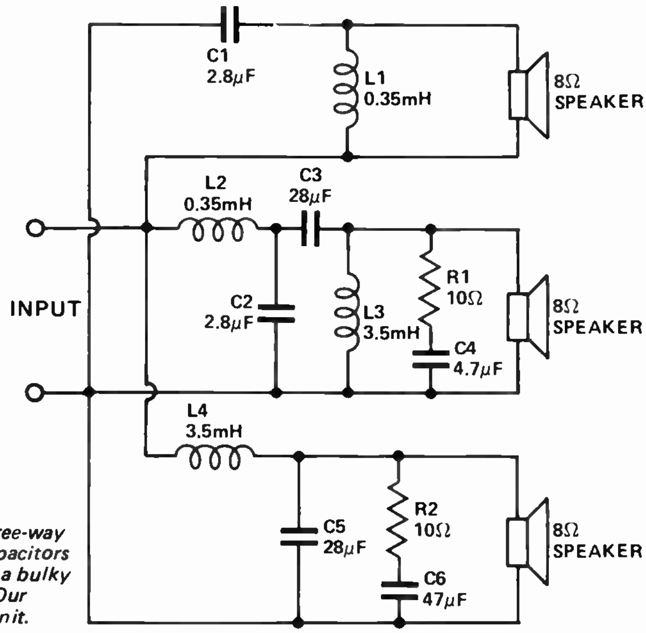


Fig. 1. A conventional three-way crossover system using capacitors and air-cored inductors is a bulky and very expensive unit. Our drawing shows a typical unit.

This all leads to the fact that, for a multi-way high-fidelity system, the crossover can and should be quite expensive. In fact it can cost almost as much as the bass driver!

Many people try to save money by trimming crossover cost – they use lighter wire and electros – and then wonder why an otherwise expensive system does not sound right. *The crossover design is one of the most important features of the whole system – it is better to compromise on a less expensive woofer than to compromise on the crossover.*

(Main text continued page 30)

SPECIFICATION

Cutoff Slope	(High pass)	12 dB / octave
	(Low pass)	6 dB / octave
Maximum Output		2 V rms.
Distortion (at 2 V out)		< 0.05%
Noise (Below 2 V)		86 dB
Cutoff Frequency		As required
Input Impedance		47 k
Output Impedance (Buffered)		< 10 ohm
Minimum Load (Buffered)		500 ohm
Frequency Response (Sum of all outputs) 20 Hz to 20 kHz		± 1 dB

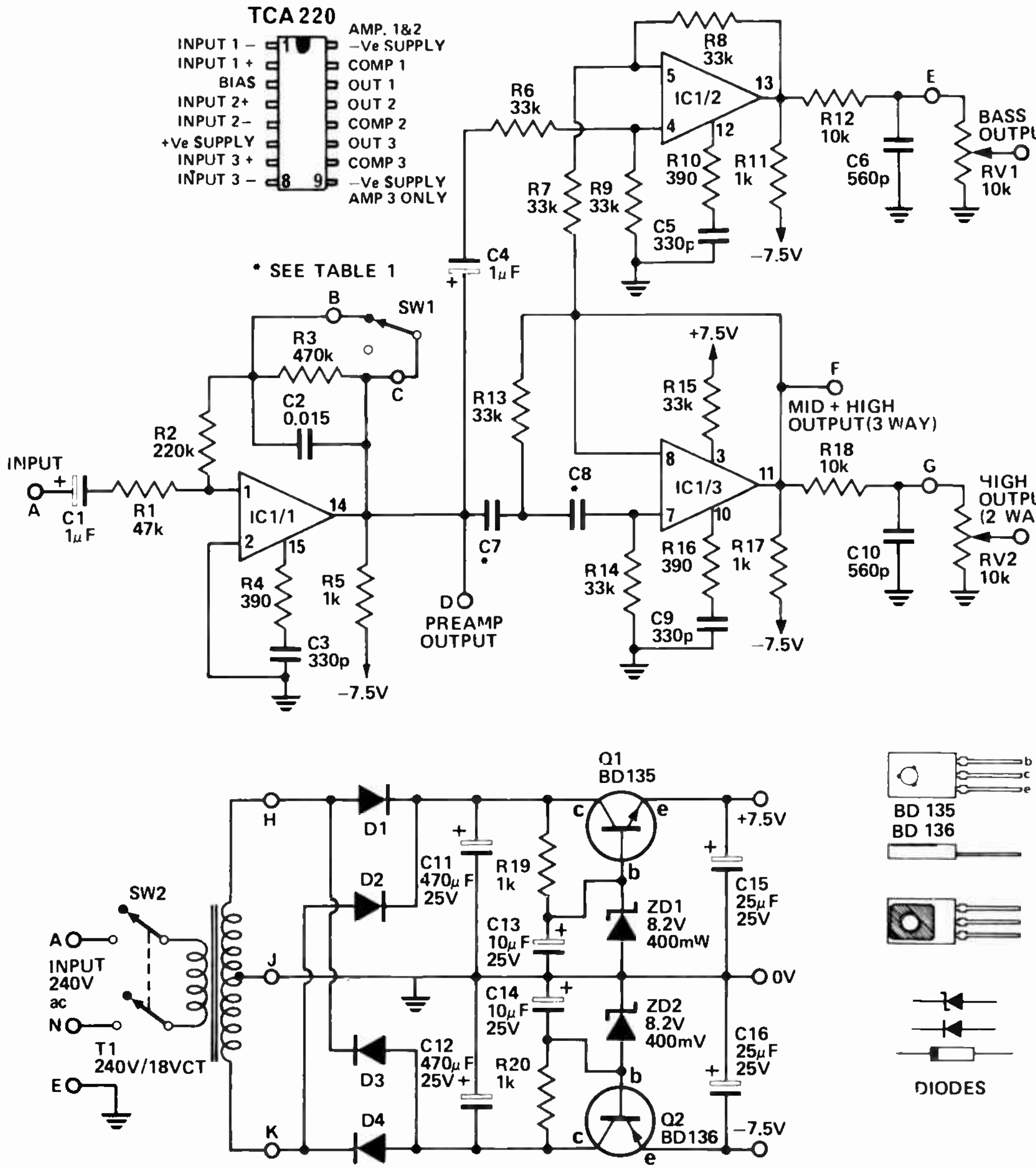


Fig. 2. Circuit diagram of the basic two-way electronic crossover and its power supply.

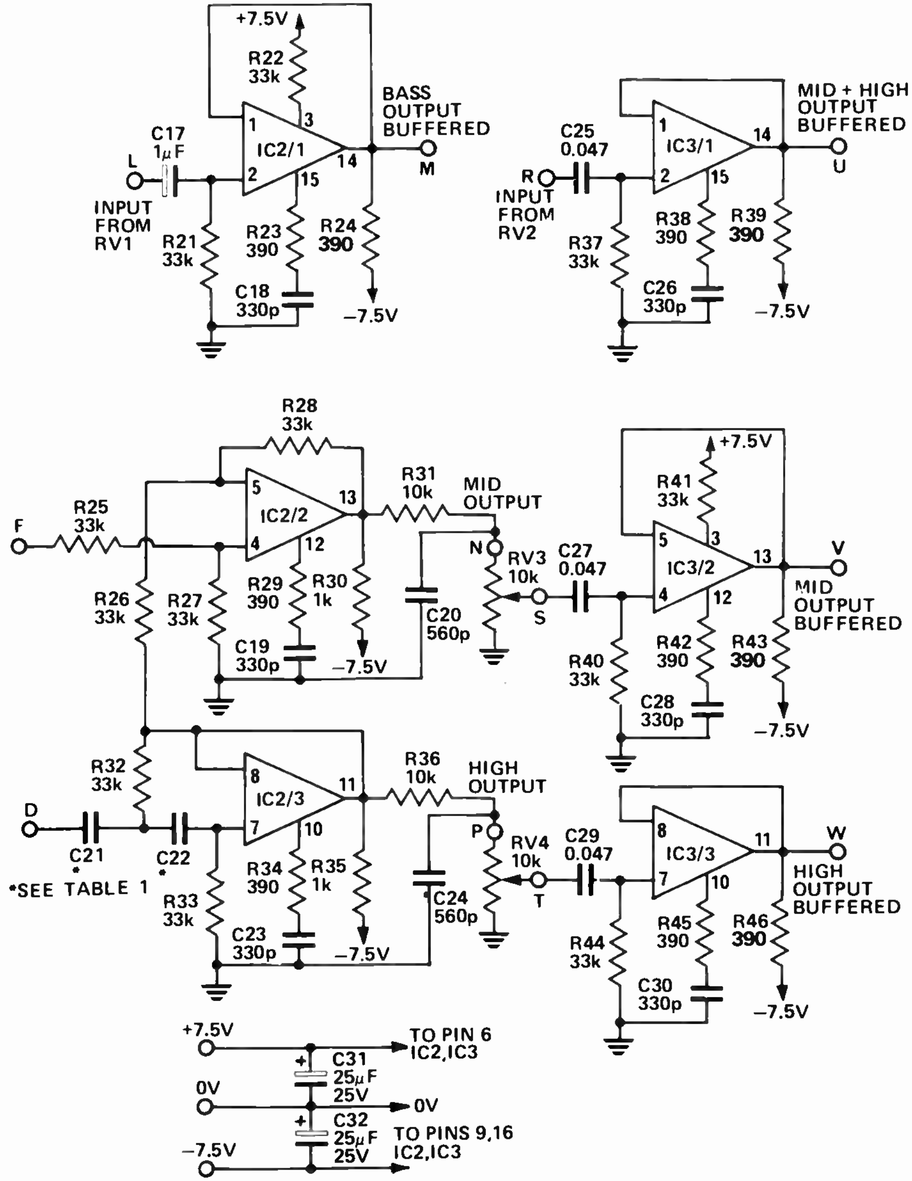


Fig. 3. Circuit diagram of the mid/high crossover board which provides four output buffer amplifiers.

HOW IT WORKS – ETI 433

The input signal is initially amplified by IC1/1. Switch SW1 together with R3 and C2 provide a maximum of 10 dB of boost below 50 Hz at a rate of 6 dB per octave. The frequency at which the boost comes in may be altered by selecting a value of C2 such that its reactance is 220k at the frequency where the woofer is normally 3 dB down. Thus if the turnover frequency is required to be 100 Hz the value of C2 should be halved.

If the boost facility is not required R3, C2 and SW1 should be deleted and a link installed between points B and C. The mid frequency gain is set by R2/R1 to about 13 dB and the input impedance is equal to the value of R1, that is, 47 k.

The first high-pass filter consists of IC1/3 where R13, R14, C7 and C8 set the cut-off frequency. The values of C7 and C8 required may be found from Table 1. This output is the high range in a two way system, or the mid plus high of a three-way system. This signal, when subtracted from the input signal by IC1/2 gives the bass range output. A second high-pass filter, where C21, C22, R32 and R33 form the frequency determining network, gives the output for the tweeter in a three-way system. This when subtracted from the mid-plus-high signal leaves the mid only as required.

Each of these outputs goes to a level set potentiometer and then is buffered by amplifiers IC2/1 and IC3/1,2,3. These outputs are now capable of driving loads in excess of 500 ohms. If the crossover is to be used to drive a constant and known load (that is, it is to be used on only one type of amplifier) the buffer amplifiers may be omitted and the outputs taken directly from the potentiometers.

The full-wave power supply provides plus or minus 13 volts which is regulated down to plus or minus 7.5 volts, by series regulators Q1 and Q2, where zeners ZD1 and ZD2 provide the necessary reference. If the unit is to be powered from the power amplifier C11, 12, and D1 to D4 should be deleted. Resistors R19 and R20 are altered to suit as shown in Table 2. The collector of Q1 now goes to the positive supply rail of the amplifier and the collector of Q2 to the negative supply rail. If the amplifier supply rail is above plus and minus 20 volts, or if both printed circuit boards are being used, (that is it is a buffered three way system) a heatsink must be added to Q1 and Q2.

PARTS LIST – ETI 433A

2-WAY SYSTEM

R4,10,16	Resistor	390	¼W	5%
R5,11,17	"	1k	¼W	5%
R19,20	"	1k	¼W	5%
R12,18	"	10k	¼W	5%
R6,7,8,9	"	33k*	¼W	2%
R13,14,15	"	33k	¼W	5%
R1	"	47k	¼W	5%
R2	"	220k	¼W	5%
R3	"	470k	¼W	5%

*These may be any value between 15k and 82k provided they are all the same value and preferably 2%.

RV1,2 Potentiometer 10k lin.

C3,5,9	Capacitor	330 pF ceramic
C6,10	"	560 pF ceramic
C2	"	0.015 μF polyester
C1,4	"	1 μF Tag tantalum
C13,14	"	10 μF 25V Electro
C15,16	"	25 μF 25V Electro
C11,12	Capacitor	470 μF 25V Electro
C7,8	"	See Table 1.

D1-D4 Diode EM401, IN4005 or similar
ZD1,2 Zener Diode 8.2 volt 400 mW

Q1 Transistor BD135 or similar
Q2 Transistor BD136 or similar

IC1 Integrated Circuit TCA220

T1 Transformer 240V/18V CT 150 mA
SW1 toggle or slide switch SPDT
SW2 Toggle switch DPDT 240V rated
PC Board ETI 433A

PARTS LIST – ETI 433B

3-WAY WITHOUT BUFFERS
All 2-way system PLUS

R29,34	Resistor	390	¼W	5%
R30,35	"	1k	¼W	5%
R31,36	"	10k	¼W	5%
R25,26,27,28	Resistor	33k*	¼W	2%
R22,32,33	"	33k	¼W	5%

* These may be any value between 15k and 82 k provided all are the same value and preferably of 2% tolerance

RV3,4 Potentiometer 10k Lin

C19,23	Capacitor	330 pF ceramic
C20,24	"	560 pF ceramic

C21,22 See Table 1.

IC2 Integrated Circuit TCA220

PC board ETI 433B

3-WAY SYSTEM WITH BUFFERS
ADD

R24,39,43,46	Resistor	390	¼W	5%
R23,38,42,45	"	390	¼W	5%
R21,37	"	33k	¼W	5%
R40,41,44	"	33k	¼W	5%

C18,26,28,30	Capacitor	330 pF ceramic
C25,27,29	"	0.047 μF polyester
C17	"	1 μF TAG Tantalum
C31,32	"	25 μF 25V electro

IC3 Integrated Circuit TCA220

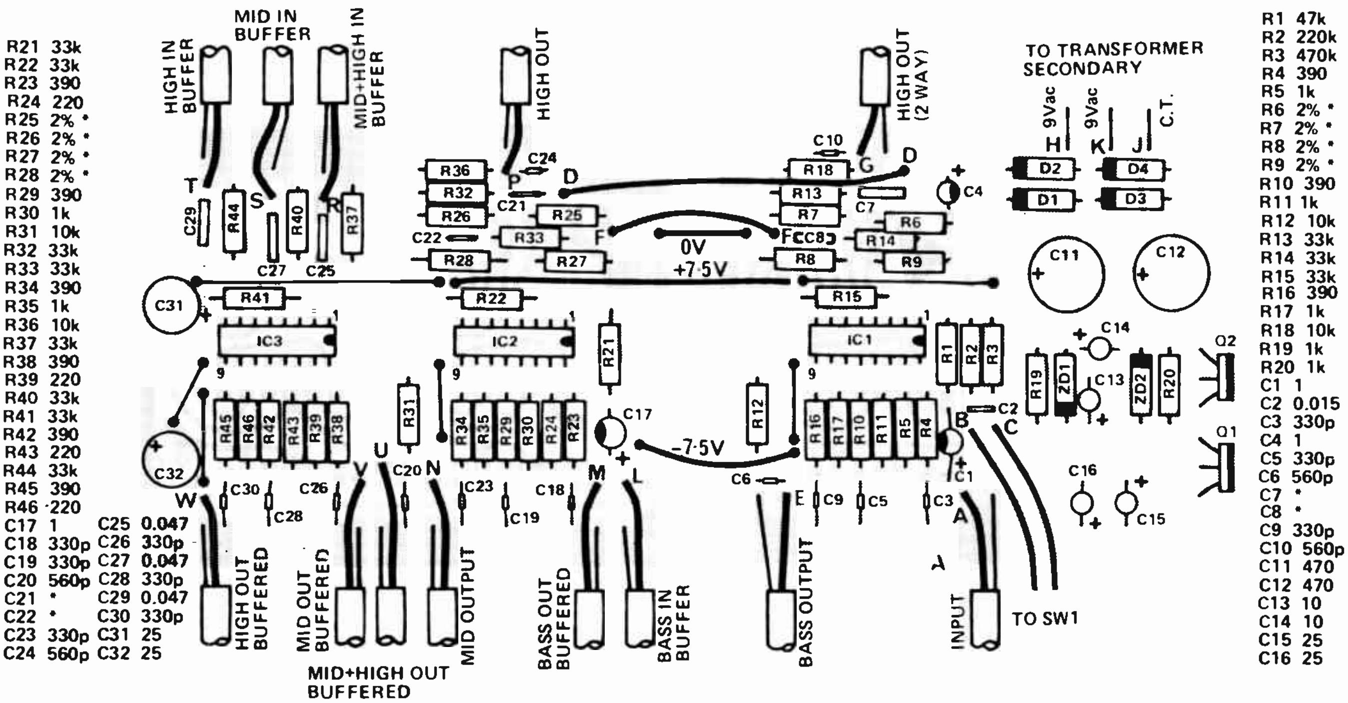
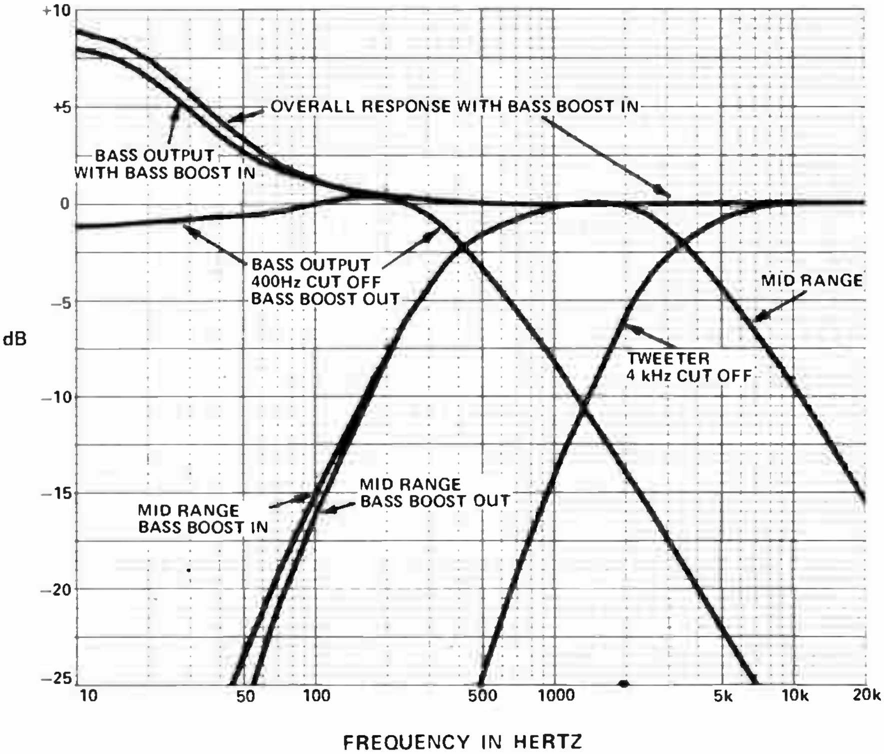


Fig.4. Component overlay for complete three-way system capacitance values are in microfarads except where otherwise noted.



Response curves of the active filters.

TABLE 1	
CROSS OVER FREQUENCY IN HERTZ	VALUE OF C7,8 or C21, 22 in μF
100	0.082
130	0.068
150	0.056
200	0.047
230	0.039
270	0.033
330	0.027
400	0.022
500	0.018
600	0.015
750	0.012
1000	0.0082
1300	0.0068
1500	0.0056
2000	0.0047
2300	0.0039
2700	0.0033
3300	0.0027
4000	0.0022
5000	0.0018
6000	0.0015
7500	0.0012
9000	0.001

ACTIVE CROSSOVER

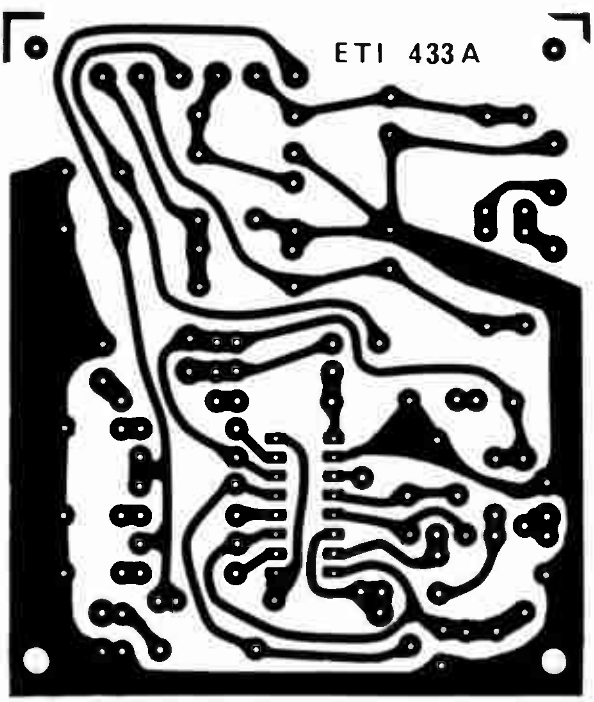


Fig. 5. Printed-circuit layout for the two-way board. Full size 77 x 90 mm.

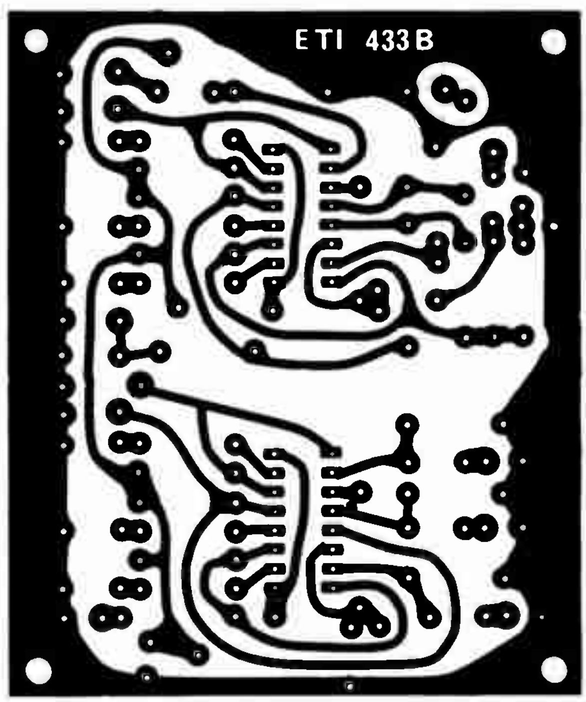
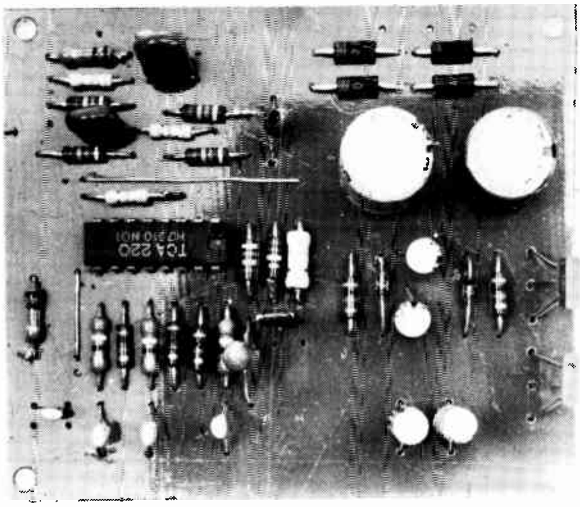


Fig. 6. Printed-circuit layout for the add-on three-way board. Full size 77 x 90 mm.

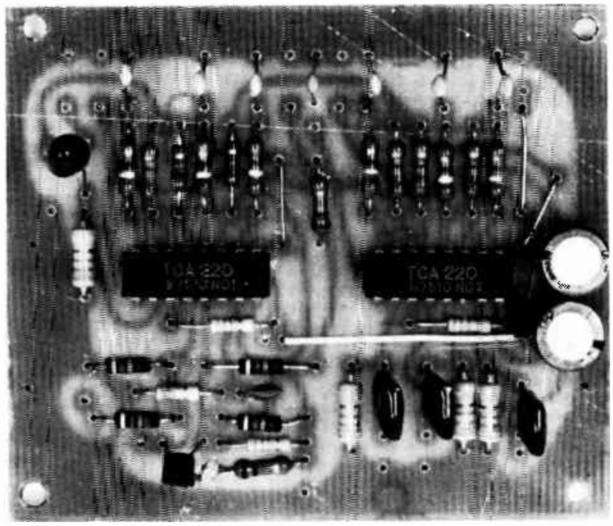
ACTIVE APPROACH

Having now established that effective conventional crossovers cost money, we may now wonder if that money could be spent in a better way by using a completely different approach. There is a better way, but until recently it has been much too expensive to be generally used. The method is to use an electronic crossover, after the preamplifier, followed by separate power amplifiers for each driver. This is feasible because a power amplifier can now be built at a cost which is about the same as that of the passive crossover. Indeed quite a few manufacturers are bringing out systems based on this principle.

Even well-designed crossovers have several serious disadvantages. As we have already said they are expensive, they waste power, they reduce damping factor (in the crossover region damping factor may drop to less than unity) and they only perform correctly into their designed load impedance. Practical drivers exhibit



The basic two-way electronic crossover.



This board provides three-way crossover plus output buffers if required.

their nominal impedance only over a very small portion of their passband, and impedance may well increase to several times the nominal value at the high end of the range. It is possible to compensate for this, to some extent, by using extra networks across the driver (the series RC networks in Fig. 1) — but this adds even more expense. Further, it is very difficult to alter the crossover frequency and also difficult to trim the crossover for best results.

However, if we were to use an electronic crossover incorporating active filters, we overcome most of the problems mentioned in a single stroke. The bulky and expensive inductors and the large and expensive capacitors are eliminated. Damping factor is restored (due to separate amplifiers being used to drive each speaker directly) and it is quite easy to change or trim the crossover frequency as desired.

Further, as electronic crossovers may have gain, it is quite a simple matter to match the various drivers of the system for sensitivity. This can be only achieved, in passive designs, by attenuating the more sensitive units down to the level of the least sensitive unit. A process which can be quite wasteful of amplifier power.

Of course with active crossovers, as with anything, there are disadvantages. In active filters we generally use operational amplifiers to implement the filters and therefore, bandwidth and noise become considerations. Further, as said before, a separate amplifier is required for each driver or group of drivers — and this can be expensive.

Nevertheless the technique is now quite feasible and is certainly worthwhile. Consequently we have developed a minimum-expense method of building a very fine system based on active filter techniques.

This article describes a basic two or three-way active filter system which may be incorporated into existing amplifiers such as the ETI type 480 50/100 watt modules, the constructional details of which are on pages 102 to 108 of this book.

A suitable three-way speaker system for use with this system is described on pages 49 to 53 of this book.

DESIGN FEATURES

There are several different approaches which may be used in the design of active filters. The first and most commonly used method, is to use separate filters for the bass, mid and high range speakers. This method is capable of compensating for amplitude, if the components are chosen correctly, but not for phase. In fact there has to be a phase change of 180° between filters to eliminate the hole that would otherwise occur at the crossover point. This is the reason for the tweeter being reversed in phase when a conventional crossover is used in a two-way system.

Another design approach, and the one that we have elected to use, is to use an active high-pass filter to generate the signal for the tweeter, and to subtract this signal from the input signal in a differential amplifier in order to generate the bass output. This substraction process generates the required crossover characteristic with both amplitude and phase taken in to account.

Initially we were worried because the bass output had a slight peak before the cutoff point but the peak is necessary to maintain that response when phase is taken into account. When the output of all channels are summed the combined response is within plus or minus one quarter of a dB of being flat over the whole range.

With this type of active filter the initial slope can be varied by adjusting the feedback resistor (R13, R32) to give a slow rolloff (Bessel filter) or to give a slight peak and fast cutoff (Chebishev). The sharper the initial cutoff the greater the apparent peak in the bass response.

As several operational amplifiers are required to implement this design we elected to use the TCA 220 triple operational amplifier. This IC, as well as containing three op-amps in the same package, is cheaper than using three separate op-amps of the 741 type or similar. Unlike the 741 type of op-amp, the TCA 220 requires a pull-down resistor on each output and a compensation network. An additional resistor is required to bias each complete IC. The use of the TCA 220 simplifies and cheapens the construction of the filter system considerably.

With active filter crossovers it is a relatively simple matter to alter the gain-versus-frequency characteristic of the filter, within its pass-band, in order to compensate for non-linearities in the associated driver. An example of this kind of compensation is our inclusion of low frequency equalisation for the woofer. Most woofers begin to drop off in the 50 to 100 hertz region. This may be corrected to some extent by adding boost below this turnover frequency. In our design we have provided 6 dB of boost which may be switched in when desired and which is limited to a maximum of 10 dB. The 10 dB limit is necessary to prevent the amplifier being over driven at low frequencies even at fairly low average listening levels.

The turnover frequency may be

(Continued on page 33)

SILENT A-B SWITCH

Speakers may be A-B tested using this simple modification to our tone-burst generator

WHEN evaluating speaker systems in A-B listening tests, the first few seconds of listening convey the truest impression of sound quality. Listening for longer than a few seconds not only fails to give further information, but may well give a false indication. For this reason it is usual to switch rapidly between the reference speaker and the speaker under test. This is generally done by using the amplifier's A/B speaker selector switch, or by wiring a change-over relay in the speaker wiring.

Whilst such switching methods are simple and reliable they have one major drawback. That is that switching may take place at any point in the waveform and as a consequence switching transients may be introduced which tend to mask the subtle differences for which one is listening. Hence a method of switching at zero-crossing points would be of great value.

When the ETI Tone-Burst Generator was constructed it was realised that it contained all the circuitry needed to performance this switching task and that it could be modified to do so very simply.

The switching must be done at low level and hence the unit is used at the input of a stereo power amplifier. The reference speaker and the speaker under test are each connected to one channel of the amplifier and the silent switch switches the input to the amplifiers as required. Thus the arrangement is mono only but this is all that is required to assess the transient response and performance of a speaker in comparison to a reference speaker.

HOW IT WORKS – ETI 124 AB

As this unit is based on the operation of the tone-burst generator ETI 124 described on page 149 that article should be thoroughly read first. Only the changes necessary to that unit are detailed in this article. A-B switch would be a little simpler if designed specifically for that purpose, the modifications required to the tone-burst generator are so simple that we thought it not worth while to design a special circuit.

To make the generator act as an A-B switch it is necessary to disable the existing mode switch. We do this by plugging in an external control switch, SW6, via a stereo phone socket. The phone socket has two change-over contacts fitted which are used to disconnect the plus and minus six volts supplies from SW3 when the jack is inserted. One of the phono contacts also disconnects the plus six volts from the common of the socket when the jack is removed. As the common of the socket is required to be at plus six volts the phono socket must be insulated from the front panel which is at 0 volts.

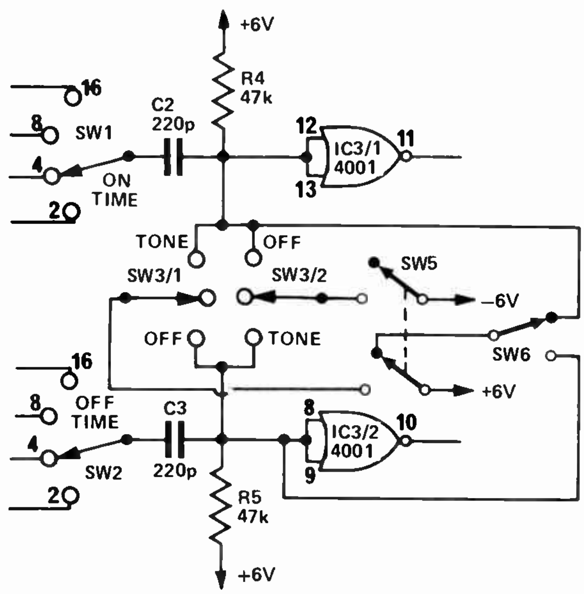
The control switch, SW6, effectively shorts either R4 or R5 thus stopping the pulses from C2 or C3 triggering the flip-flop. When the switch is actuated there is a delay until the number of cycles as set by the front panel switch have occurred and then, at the next zero crossing, the change-over occurs. The delay is necessary to ensure that any contact bounce of the SW6 contacts does not cause unwanted switching of the circuit.

CONSTRUCTION

The ETI 124 Tone-burst Generator should first be constructed as detailed on page 153 except that the wiring to SW3 is changed as detailed in Fig. 1 and 2 of this article. The dual-RCA socket and the phono socket are then mounted on one side of the box. If a metal box is used make sure that the phono socket is insulated from the case of the box as it is at a potential of six volts. The switch, SW6, should be mounted in a small pill container or similar housing and fitted with a three-core cable that is terminated at the other end by a stereo phone jack. Note that the common of the switch should be connected to the common of the jack but that the other wires may be wired to either of the remaining contacts.

USING THE SWITCH.

The audio switch requires a reasonably high level of signal to ensure correct zero-crossing switching. There are two suitable points in a conventional amplifier. The first position is between the tape-in and tape-out sockets but the second and preferable position is between the pre and main amplifiers provided that the main amplifier has a volume control that is independent of the preamplifier.



PROJECT 124 AB

Fig. 1. Partial circuit diagram of the tone-burst generator modified to perform AB switching.

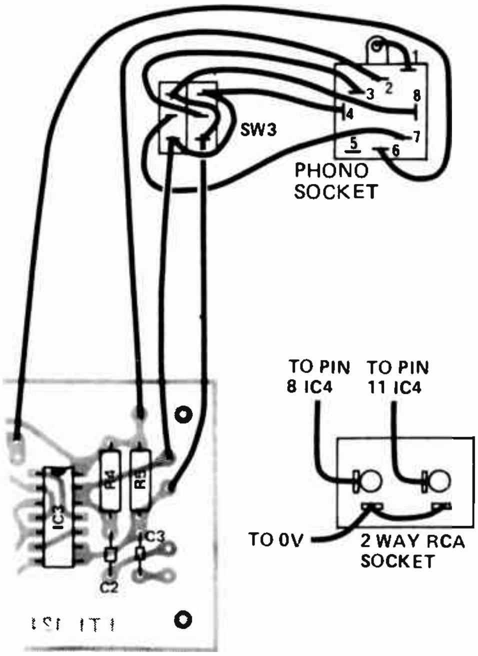
To connect the unit for AB testing apply a single input, from the preamplifier (switched to mono), to the normal input socket of the generator. The normal output socket of the generator is not used but the two RCA output sockets are connected back to the left and right channel inputs of the main amplifier. When SW6 is operated the mono input will be silently switched between right and left channel speakers.

If using the tape sockets the monitor switch should be in the 'monitor' position and the balance control should be adjusted so that the levels from the two speakers are apparently the same. Make sure that the tone controls are in the flat position, as they can cause phase shifts which prevent the switching occuring at the zero-crossing point.

If the pre and main amplifier terminals are used the preamplifier volume should be adjusted to about half way and separate volume controls used to balance for the difference in efficiencies of the two speakers. If the main amplifier does not have separate volume controls then external ones must be added if balance is to be achieved. In this case the tone controls may be used if required without upsetting the crossover point.

Change over may be effected by using either a toggle switch or a push button. The tone-burst generator controls should be set for eight cycles on and off as this position will effectively remove any contact bounce. ●

Fig. 2. Interconnection diagram to phono socket and RCA output sockets of AB switch.



ACTIVE CROSSOVER

(Continued from page 31)

selected by means of a simple component change to suit the driver in use. This equalisation technique can effectively extend the low frequency response by another octave, eg, from 50 hertz down to 25 hertz.

CONSTRUCTION

The configuration of the electronic crossover used will depend very much on the system into which it is to be built. The prospective builder should therefore carefully determine his individual requirements before commencing to build a system.

If a fixed load is to be driven (ie, numbers of amplifiers) as would be the normal case, the buffer amplifiers are not required, and the output may be taken directly from the potentiometers.

It must also be decided whether you want a two-way or a three-way system. Rather than use three separate amplifiers to drive the woofer, mid and tweeter drivers separately, it may be better to use a conventional crossover for the mid/high crossover and a two-way electronic crossover for the bass/mid.

Mono or stereo? If a stereo unit is to be built only one power supply is required and the bass-boost switch and the level potentiometers can all be dual units.

If the amplifier has a dual power supply with voltages exceeding ± 10 volts it may be used to power the crossover. This course of action will save one transformer, four power diodes and the filter capacitors.

Mechanical layout is not given as the unit will most probably best be mounted within the amplifier case. Keep it well clear of the power transformer and mount it using insulated spacers. This is necessary to avoid the possibility of earth loops which will cause a high hum level.

Full component overlays are given for all alternatives but only the circuitry required should be assembled. In a three-way system without buffers one section of IC2 is not used. In this case just leave out the components associated with the unused section in order to reduce power consumption.

If the unit is being powered from the main amplifier, or a three-way system with buffers is being used, a heatsink is required. The heatsink recommended is a piece of aluminium 60 x 85 mm bent into a 'U' shape and mounted vertically on the end of the board. The transistors should be insulated from the heatsink.

For a stereo system delete the power supply components on one of the boards (up to C15 and C16) and just link the two boards together. ●

TABLE 2

MAIN AMPLIFIER SUPPLY VOLTAGE	VALUE OF R19,R20
± 10-15 V	1 k
± 15-20 V	1.8 k
± 20-25 V	2.7 k
± 25-30 V	3.9 k
± 30-40 V	5.6 k
± 40-50 V	8.2 k

SIMPLE COMPRESSOR EXPANDER

Our new compressor expander uses a single IC to replace several components in a previous design, and features a 2:1 compression ratio.

CASSETTE RECORDERS are becoming more acceptable in the hi-fi situation as the use of narrow gap heads and special tapes improves frequency response. In this respect the modern deck rivals the reel-to-reel machine. However, the reel machine and disc recording still offer a better dynamic range, a result of the signal to noise ratio of the cassette equipment not being high enough to blank out background noise in quiet passages.

When recording tapes there has to be a compromise met between signal to noise ratio and clipping the peaks of the music due to tape saturation. Many systems have been devised to help alleviate this problem with the most commonly known one being the Dolby system. This effectively gives an additional 10 dB or so of dynamic range. Limiters are used on a lot of recorders to prevent tape saturation but these alter the dynamic range which is not normally acceptable to the hi-fi listener.

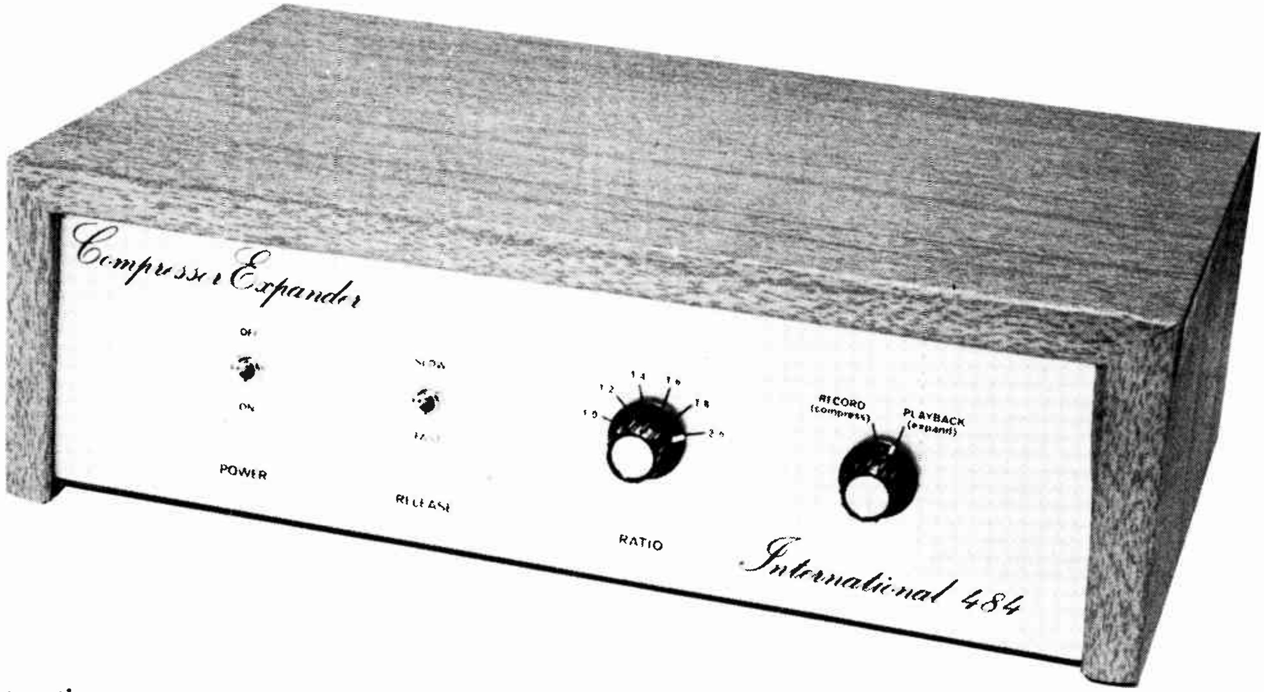
One other system used professionally but not a great deal in the domestic situation is the compressor expander. The best known system here must be the dbx unit. With this type of system the full dynamic range, say 80 dB, is compressed to perhaps 40 dB (compression ratio of 2), then it is recorded. If the signal to noise ratio of the recorder is 50 dB and our peak recording level is 5 dB below maximum our minimum level is still 5 dB above the noise. On replay we now expand by the same factor giving us our full 80 dB dynamic range with the noise 10 dB lower.

We have already published the design of a compressor expander (in ETI, April 1976) which worked well but was complex and used a double sided printed circuit board with eight ICs and four dual transistors. This new design is simplified by the use of a special IC which takes the place of all these separate components reducing the cost and complexity.

SPECIFICATION – ETI 484

Compression ratio	1.0, 1.2, 1.4, 1.6, 1.8, 2.0
Expansion ratio	1.0, 1.2, 1.4, 1.6, 1.8, 2.0
Attack time	
fast	10ms
slow	40ms
Maximum input voltage *	
R25–R28 = 0Ω	1 volt
Distortion 1 volt out	
untrimmed max.	2%
untrimmed prototype	0.25%
trimmed max.	0.2%
trimmed prototype	0.09%
Signal to noise ratio re 1V	
2.0 compression	45dB
2.0 expansion	90dB

* The max. input voltage can be increased to 3 volts using R25,26 = 22k and R27,28 = 10k

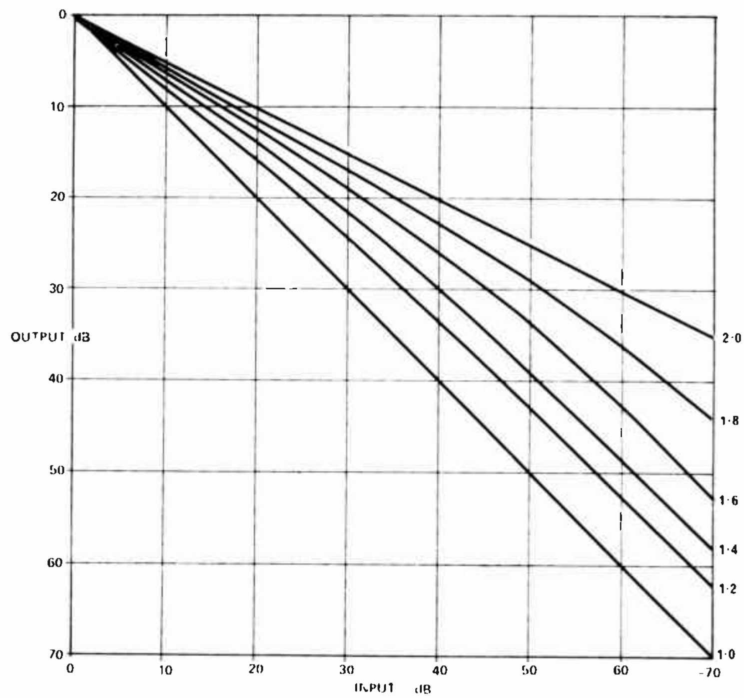


Construction

Commence assembly with all the components which are mounted flat on the printed circuit board. If, and only if, you have distortion measuring equipment add RV1, 2 and R29–R32. If these are not adjusted correctly the distortion may well be higher than without them (it should be less than 2%). Now add to each rotary switch ½ inch long 6BA spacers on the bolts holding the switch together. It may be necessary to remove the rear nuts to give enough thread to hold these spacers. Now bolt the switches onto the printed circuit board (the 6 pos. one is the nearest the IC). Take note of which contact is the wiper on each of the switches. On the 6 pos. one there is a normal contact as well as the wiper in the same position except on the opposite side of the wafer and this normal contact is not used.

There is a series of holes in the printed circuit board around the switches in two rows, one slightly outside the other. The inner row connects to the wafer closest to the printed circuit board. Start connections by the wiper contact (marked W on the printed circuit board) using tinned copper wire and then the other contacts by the appropriate resistor or link. For the links to the top wafer it is recommended that insulation be used over the wires.

The release time switch can now be wired and the printed circuit board mounted into the chasis. The transformer input sockets etc. can now be mounted and wired.



Graph showing relationship between input and output for the various compression ratios.

Distortion Adjustment

Distortion can only be adjusted with a meter. Set the ratio switch to 2 and feed about 1 to 1.5 V at about 1 kHz into the socket marked 'to tape output on amplifier' and measure the distortion at the socket marked 'to tape recorder input'. By adjusting RV1 and RV2 depending on which channel you are measuring it should be possible to adjust the distortion to under 0.2%. This can be repeated with the second channel.

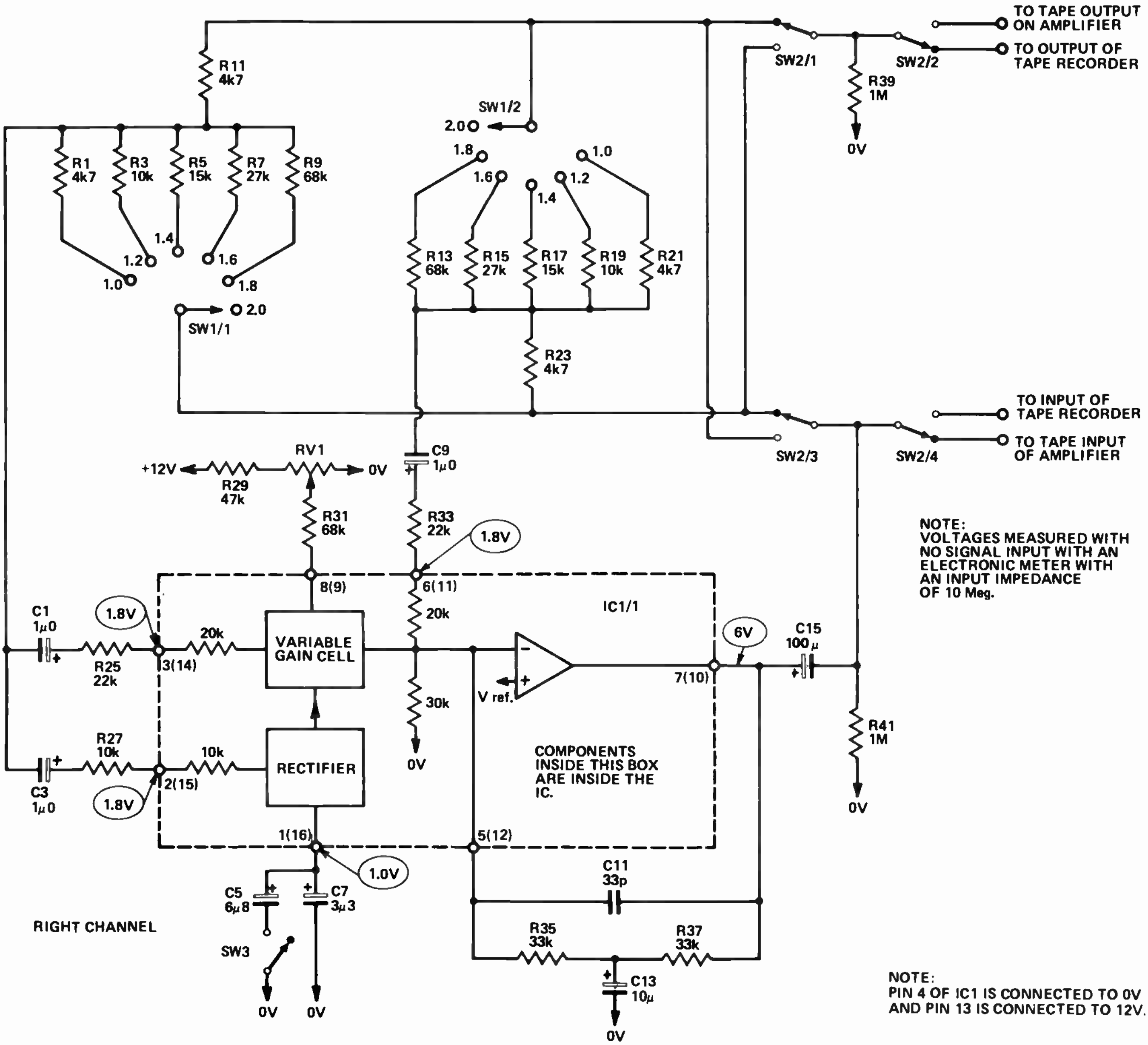


Fig. 1. Circuit diagram of the right channel.

Input Levels

The maximum input level the IC can handle is one volt. However by including resistors R25-R28 the maximum input level is raised to three volts. These resistors also affect unity gain voltage. If input levels higher than one volt will not be used these resistors should be replaced by links. Resistors R33 and R34 should also be replaced by links if R25-R28 are removed.

How It Works – ETI 484

As most of the work is done inside the IC we must look inside the IC to explain the operation. The IC contains a rectifier circuit which is used to measure the actual signal level, a variable gain block which is controlled by the output of the rectifier so that the gain is proportional to the input signal, and an amplifier. By connecting the IC in various ways either a compressor or expander can be formed. We can do either by switching and also by mixing the two by a series of resistors we obtain ratios other than the preset 2. However due to the mixing being done before the logarithmic control of the variable gain cell the ratio is only true in the top 30-40 dB range reverting to a ratio of 1 below this level. Both compressor and expander however follow the same curve and compensate for each other.

We have provided two release times in the unit. With a fast release time there is distortion created at low frequency while if it is too slow the unit appears to 'breathe'. The slow time is slow enough to give reasonable low distortion while minimising breathing. However the distortion created by a fast release time is compensated in the expansion mode provided it is recorded and played back at the same settings.

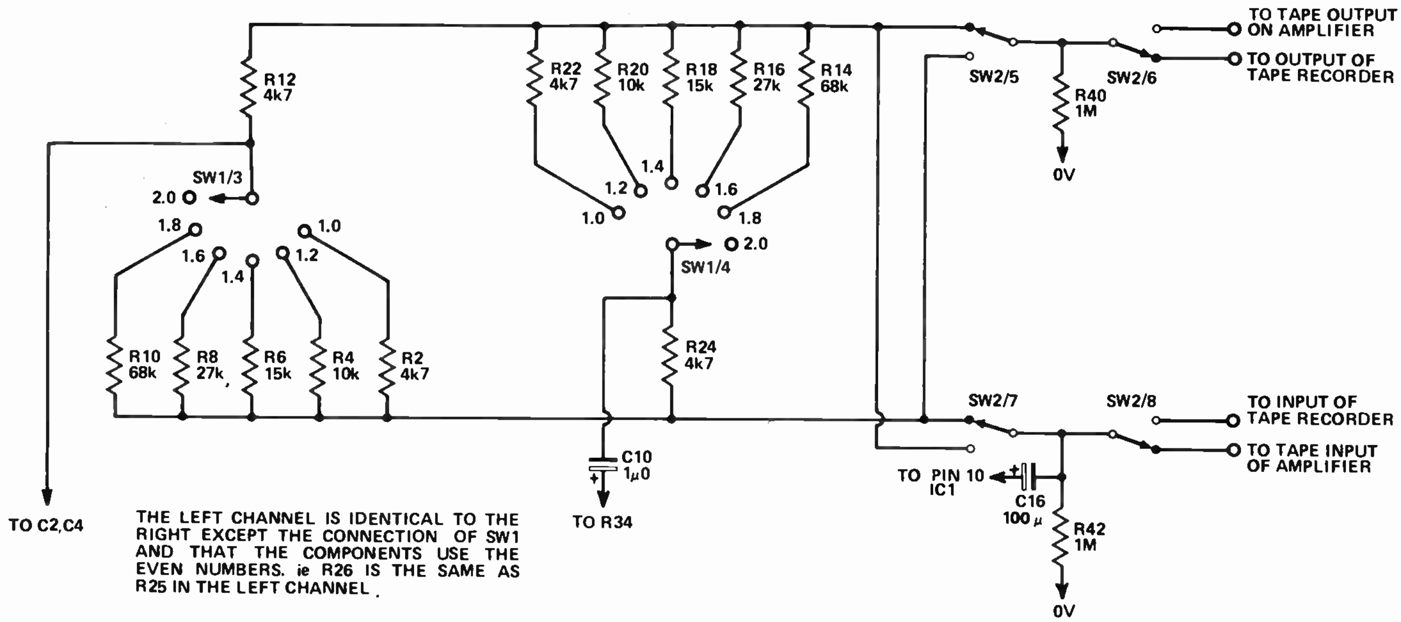


Fig. 2. Changes in the circuit for the left channel. The changes are only to simplify the PCB layout.

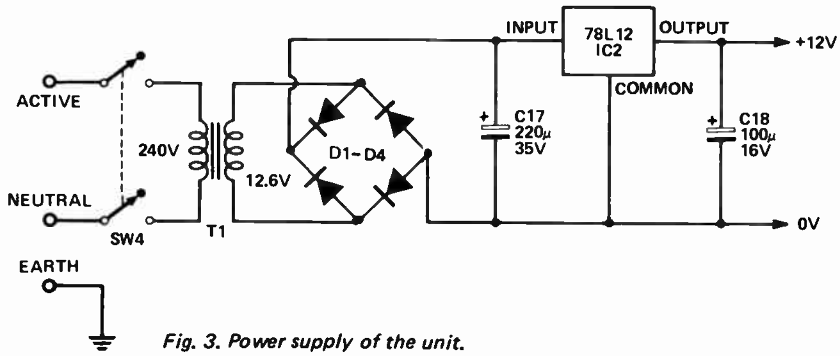


Fig. 3. Power supply of the unit.

Note: A number of people building this unit have been puzzled by a phenomenon inherent in its operation.

In the expansion mode the output voltage can never exceed the input voltage — it will just equal it when the input is 1.0 volts. Thus if the unit is bypassed whilst in the expansion mode there will be a false impression of expansion simply because the output level will now be higher.

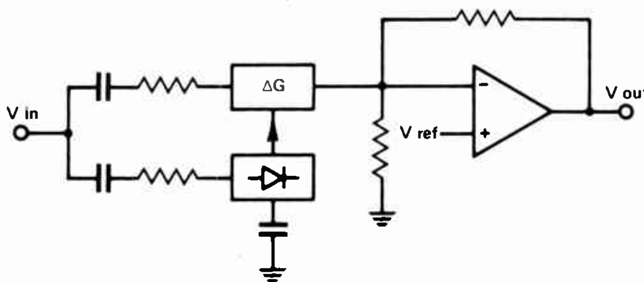
It should be understood that the unit's function is to expand dynamic range and because of this it is necessary to increase amplifier volume when using expansion.

This effect may be minimised by maintaining maximum input level as close as possible to one volt (or three volts —see main text).

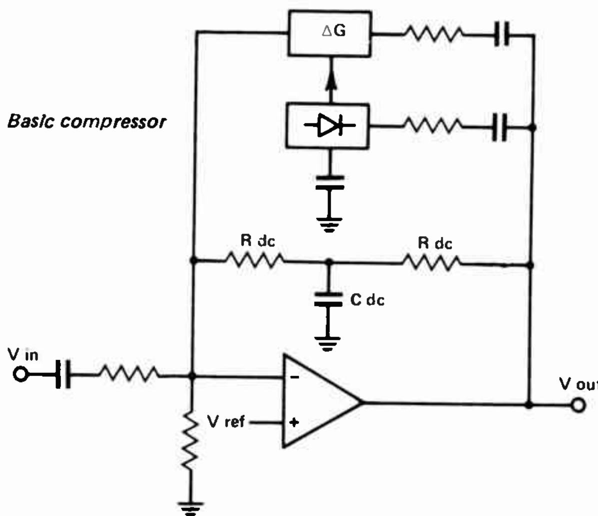
Expander or Compressor

These diagrams show how the IC is connected to operate as either a compressor or expander with a fixed ratio of 2.0.

Basic expander.



Basic compressor



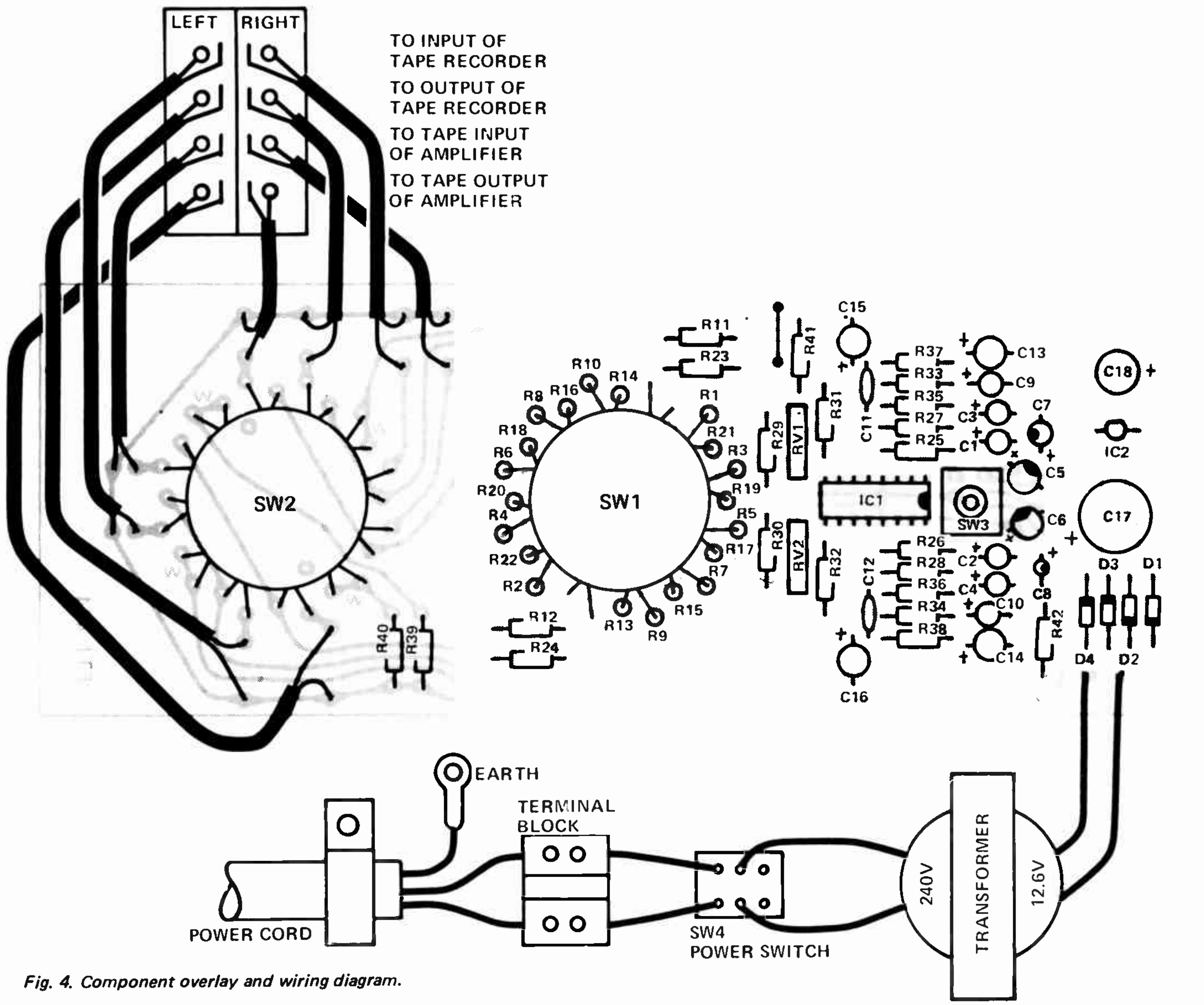


Fig. 4. Component overlay and wiring diagram.

PARTS LIST – ETI 484

Resistors all ½W 5%
R1,2 4k7
R3,4 10k
R5,6 15k
R7,8 27k
R9,10 68k

R11,12 4k7
R13,14 68k
R15,16 27k
R17,18 15k
R19,20 10k

R21,22 4k7
R23,24 4k7
R25,26 22k
R27,28 10k
R29,30 47k
R31,32 68k

R33,34 22k
R35-R38 33k
R39-R42 1M

Potentiometers
RV1,2 25 k trim

Capacitors
C1-C4 1µ0 50V electro
C5,6 6µ8 10V tantalum
C7,8 3µ3 10V tantalum
C9,10 1µ0 50V electro
C11,12 33p ceramic

C13.14 10µ 16V electro
C15,16 100µ 16V electro
C17 220µ 35V electro
C18 100µ 16V electro

Semiconductors
IC1 NE571
IC2 78 L 12
D1-D4 1N4001

Miscellaneous
PC board ETI 484
Transformer 240-12.6V 100mA
SW1 4 pole 6 position OAK switch (2 sec. 2 poles 6 pos.)
SW2 8 pole 2 position OAK switch (2 sec. 4 poles 2 pos.)
SW3,4 DPDT toggle
Two, four-way RCA sockets
Chassis, cover and front panel
3 core flex, plug and clamp

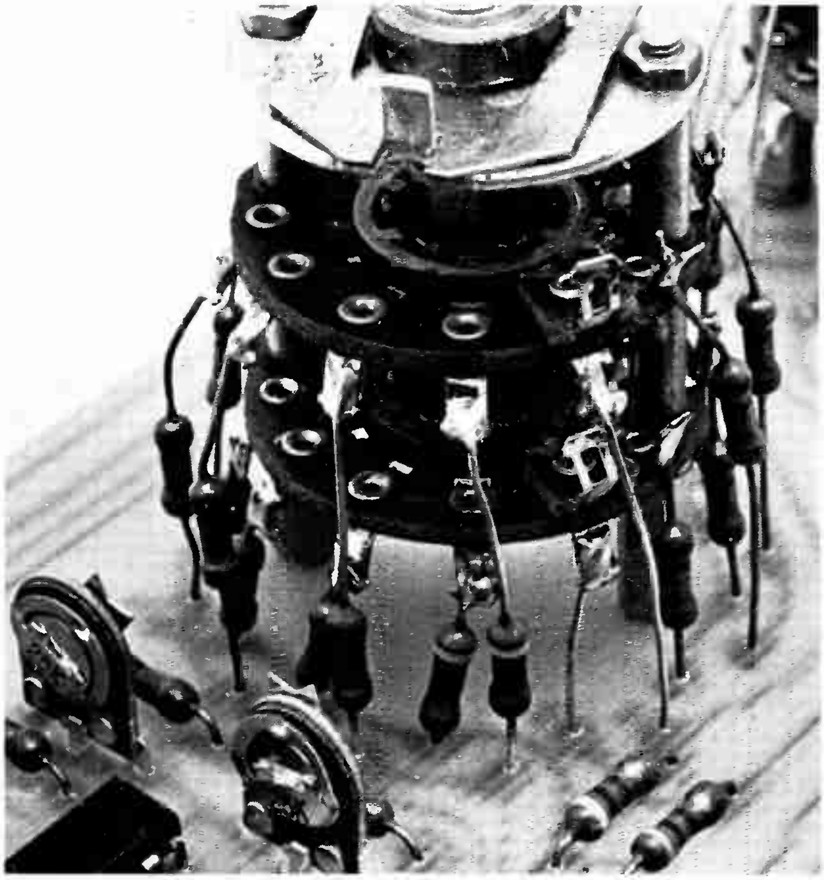
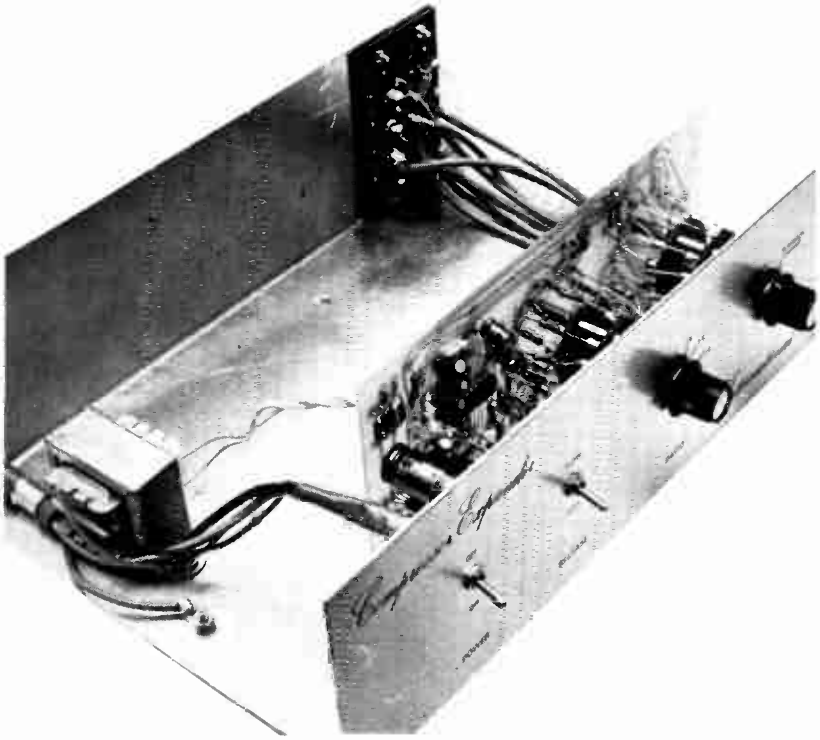


Photo showing how the resistors are connected to the rotary switch.



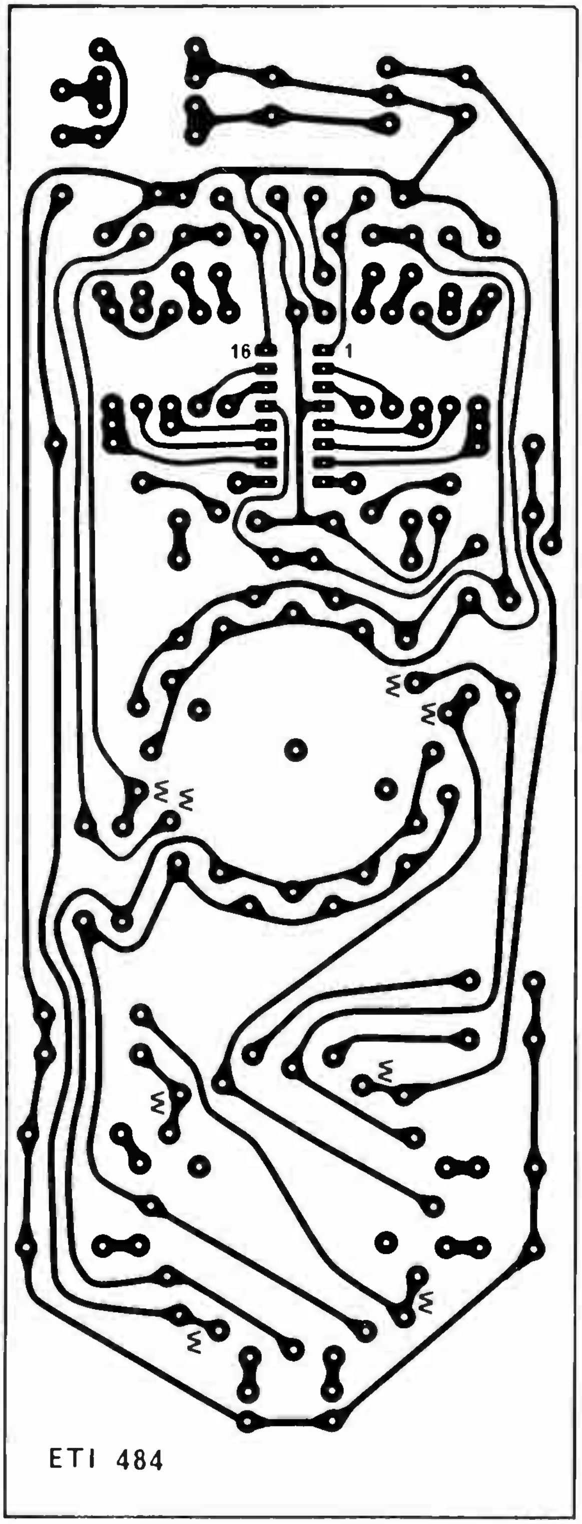


Fig. 5. Printed circuit layout.
Full size 200 x 75 mm.

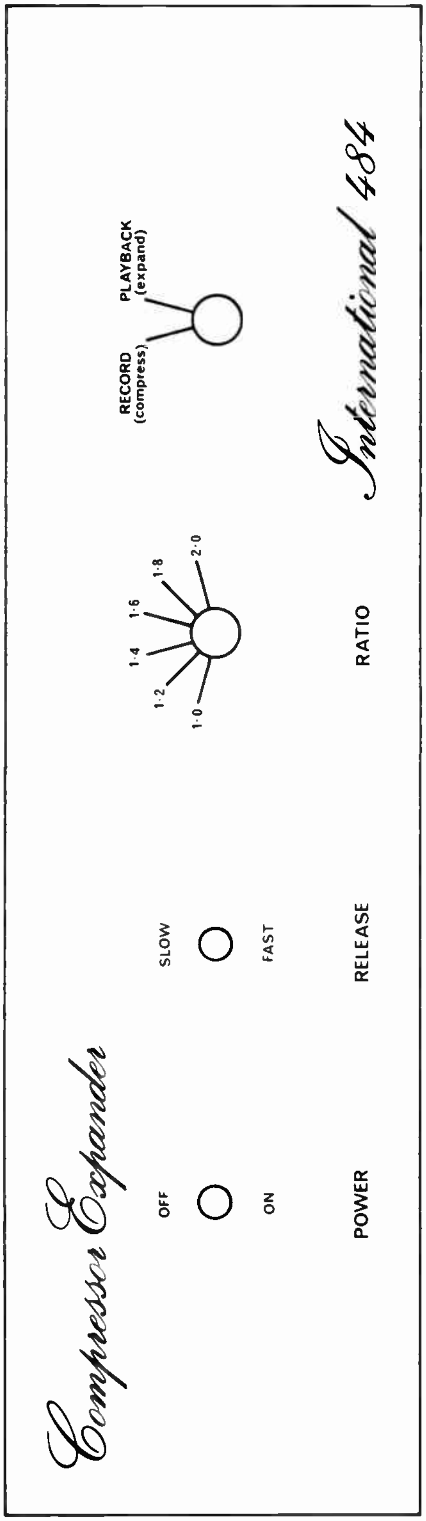


Fig. 6. Front panel artwork. For dimensions see Fig. 9.

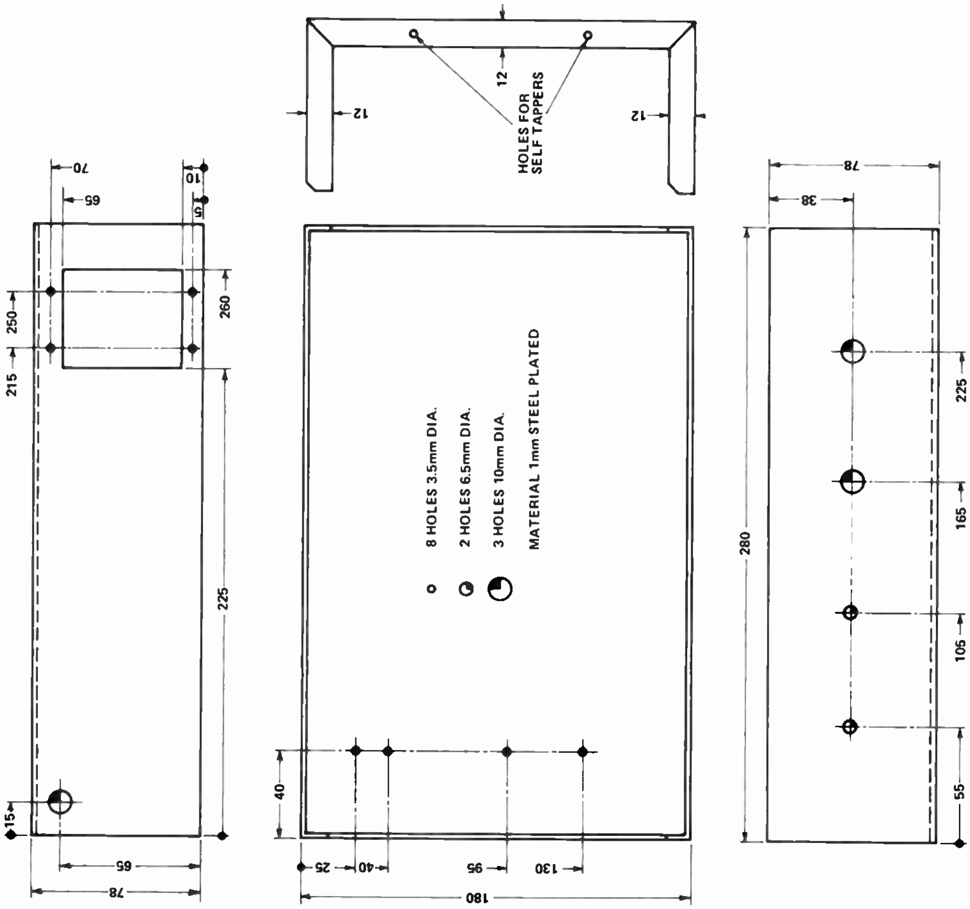


Fig. 7. Chassis details.

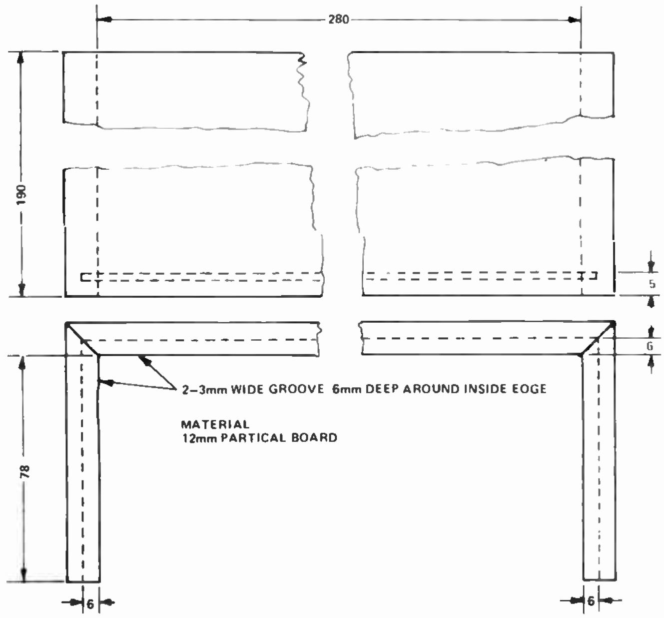


Fig. 8. Cover used on the unit.

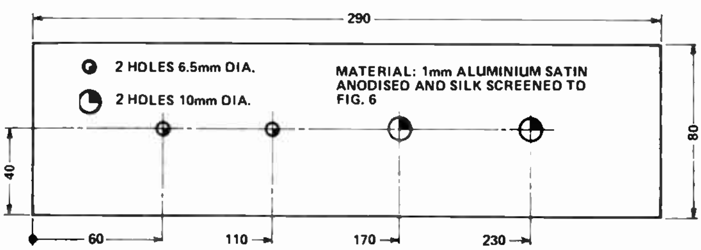


Fig. 9. Dimensions of the front panel.

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.

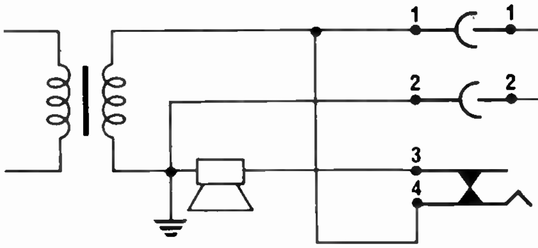


Fig.1a. Example of using the type 05 connector inserted in position A. Both speakers are operational.

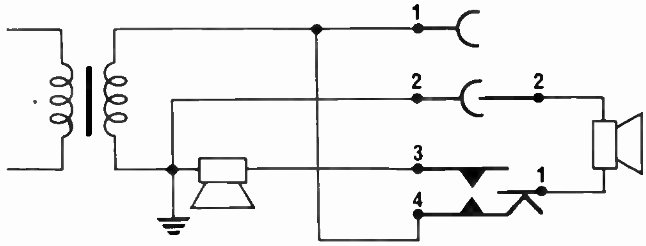


Fig.1b. With the connector reversed, in position B, the internal speaker is disconnected.

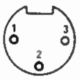
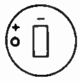
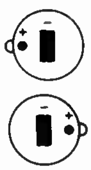
Notes 1. — The numbering of the contacts is shown as seen on the mating face of the connector.

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.

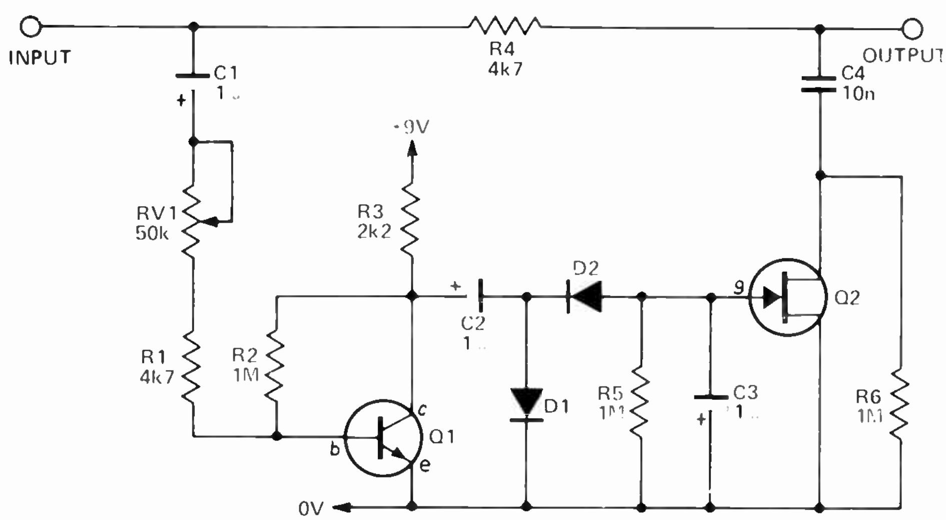
TABLE 1

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 	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
FEMALE SOCKET 			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 130-91EC

TAPE NOISE LIMITER

Cut down tape hiss by adding this unit to your cassette recorder.



DESPITE the small size, the performance obtainable from a cassette tape in a good recording deck is quite remarkable. In fact the latest top quality decks are so good that it is difficult to tell the difference between the recording and the original sound.

Unfortunately this is not true of the cheaper units – in which 'tape hiss' can be very prominent. Tape hiss is caused by random irregularities in a tape's surface coating. The effect is common to all tapes but some are marginally worse than others.

The annoying characteristic of tape hiss delayed the acceptance of cassette tape recorders in hi-fi systems for some years – until the advent of the Dolby system which was primarily developed as a cure for the phenomenon.

The Dolby system is often misunderstood – *it only works if the cassette tape itself has been recorded using the Dolby process* – and few commercially produced tapes are. Unless the tape cassette says specifically that it is Dolby processed then it's not! You can of course record your own tapes using Dolby if you own a Dolby machine.

To overcome this limitation a number of cassette recorders are fitted with noise reduction circuitry which reduces the level of hiss on non-Dolby recordings. Most of these noise reducing circuits work by progressively reducing all high frequency signals when the output level falls below a preset minimum. Above that minimum level all sounds are allowed through because tape hiss cannot be heard once the sound level is substantially louder than the hiss. This effect is called 'acoustic masking'.

The circuit described in this project is a simple but very effective unit which may be used with any cassette recorder which is connected to a hi-fi system.

The unit should preferably be connected between the cassette recorder and the amplifier input – using short lengths of screened cable and suitable connecting plugs. If you really know what you're doing it may be actually built into the tape recorder or amplifier. Alternatively it may be connected between the pre-amplifier and power amplifier on those units which are so separated (note that many apparently integral amplifiers still have 'pre-amp out' and 'power-amp in' connectors on the rear panel. These connectors are normally bridged by 'U' shaped links– which should be removed to enable this unit to be plugged in).

CONSTRUCTION

As with most projects in this series you can use either Veroboard or the special printed circuit board shown here.

Take the usual precautions about inserting components the right way round – taking particular care with the field effect transistor Q2. Note that the cathode lead of the diodes (shown as a horizontal bar on the circuit diagram) will be identified on the component by a black band or similar marking.

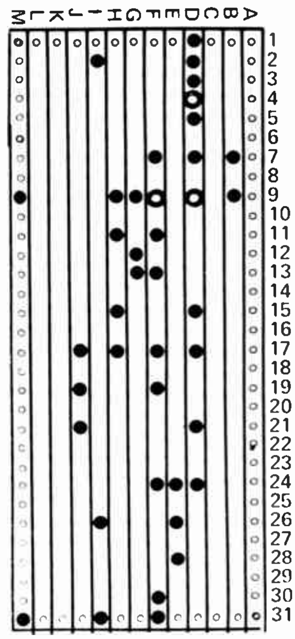
Unless the leads between this unit and the tape deck and amplifier are very short it is advisable to connect it via screened cable. Note that the 0V line shown on the circuit is also the 'earthy' side of the input/output connections.

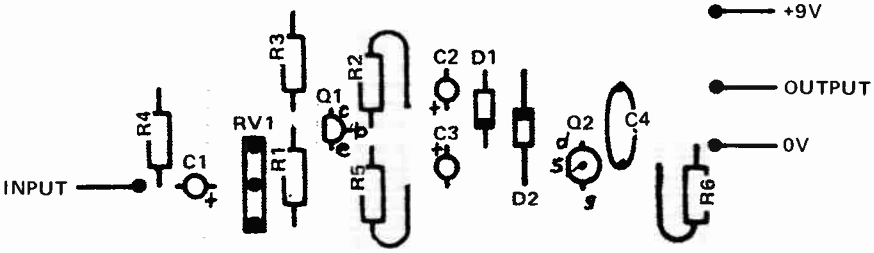
To set up the unit simply choose a recording with a longish quiet passage and then adjust RV1 for the best compromise between tape hiss reduction and minimum loss of high frequency programme content.

NOTE: If you listen only to hard rock – where there aren't any quiet passages – then this unit will be of little value to you. Its main effect is to reduce annoying tape hiss during otherwise quiet programme material.

SPECIFICATION

Input level –	up to 2 Vrms
Min level for flat response –	about 10 mV
Input impedance	depends on Q1 gain but > 4.7 k
Output impedance	impedance driving the input + 4.7 k
Output impedance of drive device	– preferably 600 ohms.



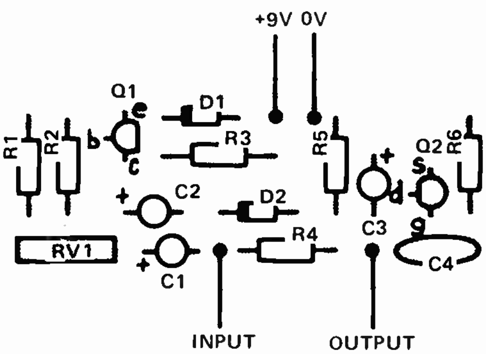


Component layout of Veroboard version.

PARTS LIST
ETI 071

R1	Resistor	4k7 0.5 W	5%
R2	"	1M "	"
R3	"	2k2 "	"
R4	"	4k7 "	"
R5,R6	"	1M "	"
RV1	Potentiometer 50 k trimpot		
C1-C3	Capacitor	1 uF 25 V	
C4	"	10 n polyester	
Q1	Transistor	BC548	
Q2	"	2N5459	
D1-D2	Diode	1N914	

Nine volt battery and clip
Veroboard or pc board ETI 071.



Component layout of printed circuit board version.

Note difference in order of source(s) and drain (d) of Q2 in the Veroboard version and pc board version of this project. This is in fact correct as the source and drain of this transistor are interchangeable in this circuit.

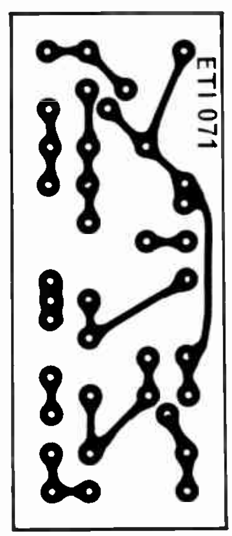
HOW IT WORKS

The circuit passes all frequencies (without attenuation) if the incoming signal is above a set minimum level. Signals below the preset minimum are progressively attenuated from 1 kHz upwards. The maximum attenuation of about 10 dB is applied at approx 10 kHz.

Resistor R4 and capacitor C4 form a filter in which Q2 is used as a variable resistor with the degree of resistance dependant on gate voltage. Thus, if the input voltage is at or near 0V then Q2 appears as a low resistance and C4 is in circuit. If on the other hand the input signal is higher than (say) four volts negative, Q2 has a very high resistance and C4 is effectively out of circuit.

The voltage applied to the gate of Q2 is that derived from Q1 – after rectification by D1 and D2. Transistor Q1 amplifies the input signal and with RV1 in minimum position, input signals above 10 mV or so will cause Q2 to be off.

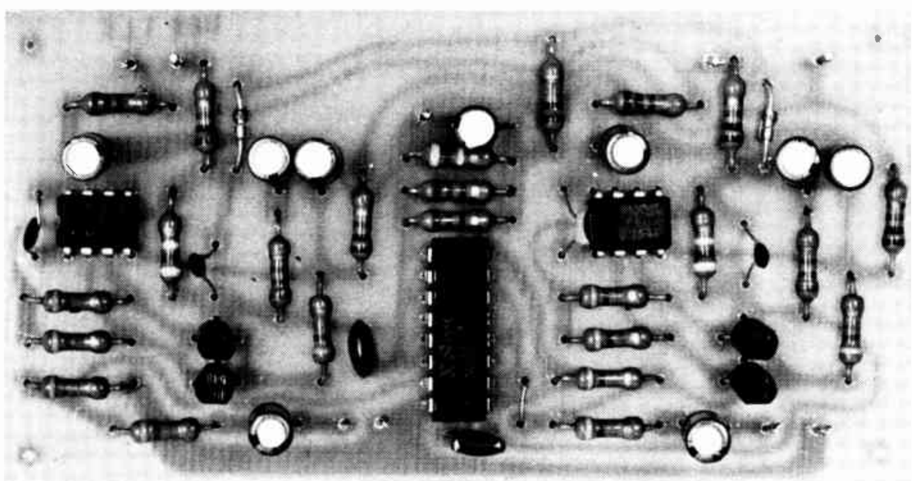
Increasing RV1 raises the level below which high cut will occur. The change from full to zero cut occurs over a range of approx 5 dB input level change.



Foil pattern for pc board – shown full size.

AUDIO LIMITER

This simple but effective unit can be used as a limiter, automatic volume control or voltage controlled amplifier.



A LIMITER IS A FORM of compressor which operates only when the signal exceeds a certain predetermined level. For example signals which do not exceed say 80% of the predetermined maximum are not compressed at all and are amplified with their full dynamic range. For signals above the 80% level the limiter begins to operate and very large input signals are required to obtain the extra 20% of output.

Another use of a limiter is in the continuous-limit mode such that it acts as an automatic volume control (AVC). In this mode a 60 dB change in input level can be limited to say, a 6 dB change in output level.

Finally the limiter may also be used as a voltage controlled amplifier having a range of about 55 dB. A typical application of such a device would be a remote volume control. It should be noted, however, that although the transfer function of such a voltage-controlled amplifier is fairly sharp, two of them may not necessarily track perfectly due to differences in the FETs in the ICs. Thus on our prototype the difference between channels when used as a stereo volume control was up to 5 dB at some points with any given input.

Design Features

When FETs are used in voltage controlled amplifiers it is essential that the voltage across them is kept as low as possible if the distortion is also to be kept low. This means that

Specification ETI 446

Input voltage range	1 mV – 10 V
Frequency response	± 3 dB 10 Hz – 20 kHz
Limiting point set by R2/16	3mV
Equivalent signal-to-noise ratio	70 dB re 1 V out
Distortion	see graph
Input impedance	47 k
Maximum gain	
R2/16 = 4k7	26 dB
R2/16 = 47k	40 dB
Maximum attenuation as voltage controlled amplifier	55 dB
Supply voltage	± 8 V to ± 16 V dc at 5 mA

the FET must be used as an attenuator where the voltage across the FET can be kept low irrespective of input voltage. The most suitable type of FET for this purpose is the enhancement-mode device but these are not readily available. The commonly available types are junction FETs which unfortunately require a negative voltage to turn them off. However, there is a suitable alternative, the 4049 CMOS IC which contains six inverting buffers. By suitable interconnection the IC may be made to provide six enhancement-mode FETs and this is the approach we decided to use.

To restore the signal level an amplifier is required and originally we intended to use the LM382 but, because of cost and availability considerations, we finally decided to use an LM301 or 741 operational amplifier together with a transistor pair at the front end. The noise performance of this arrangement was found to be as good as the LM382's and supply voltage to be less critical (although a dual supply is required). If only a single-ended supply is available then a 382 may be used, although a different board layout would be required.

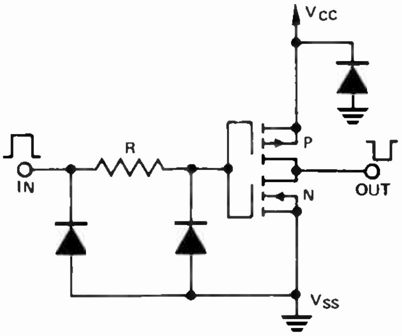


Fig. 6 Internal circuit diagram of one of the six inverter stages in the CMOS 4049 IC

USES OF A LIMITER

Peak Limiting. In this mode only signals above 85% of maximum level are attenuated. This is useful for preventing amplifier clipping (for pop groups or other live shows) which gives rise to objectionable distortion. It may also be used when tape recording the same type of programme material as above, to prevent the tape being saturated, which again would give rise to distortion.

AVC. In this mode, the limiter is used typically to drastically reduce the dynamic range of a programme being recorded. For example, when recording a lecture the 60dB dynamic range of lecture room speech may be compressed to 6dB.

Voltage Controlled Amplifier. As a voltage-controlled amplifier the unit lends itself to a variety of remote or automatic control applications. For example, it may be used as a remote control for stereo amplifier volume. Alternatively, it may be adjusted to increase car radio volume as ambient noise level rises.

Special Effects. The limiter may also be used to modify the sound of musical instruments. For example, such a limiter is often used to eliminate the attack transient on a bass guitar to give a smoother mellower sound.

The uses of such a circuit are wide indeed, and we are sure our readers will think of many more applications for this interesting circuit.

Construction

Although a printed-circuit board is not essential it certainly makes construction very much easier. Before assembly decide whether a limiter or an AVC is required as the values of R2 and R16 will vary accordingly. Use 47k for R2 and R16 in the AVC mode and in limit mode, depending on limit point, between 470 and 4k7. The transistor type specified is available from a number of different manufacturers but pin connections are different – the overlay shows connections for the Philips type. If a different brand is used the transistor should be reversed (emitter and collector interchanged). The overlay also shows the arrangement for using the LM301 ICs – these may be directly replaced by 741s simply by omitting the 33 pF capacitors.

Although the CMOS ICs 4449 and 4009 are electrically similar to the 4049 and are interchangeable with it when the devices are used as hex-inverters, they cannot be used as replacements in this circuit. The 4049 must be used. The 4449 and 4009 have different circuitry and will not work in this mode.

As this unit will normally be used in association with another piece of equipment, and most likely built in to it, a case has not been described. When installing the unit make sure that the input cables are coaxial or shielded cable – outputs are not important and can be normal hookup wire.

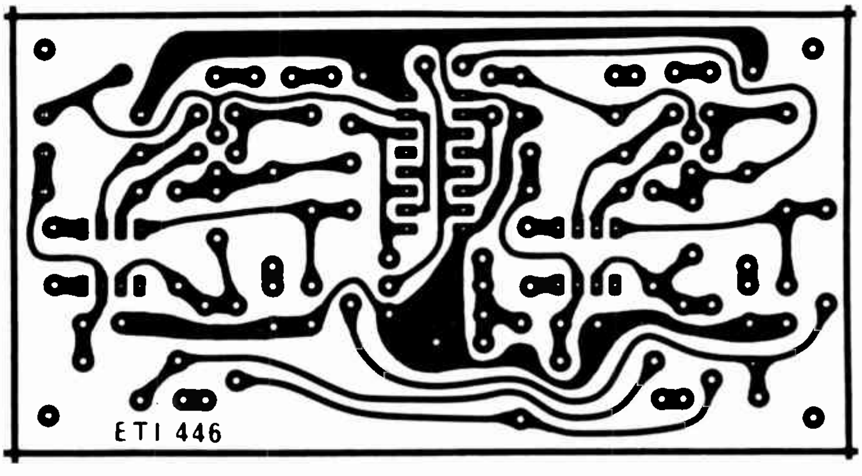
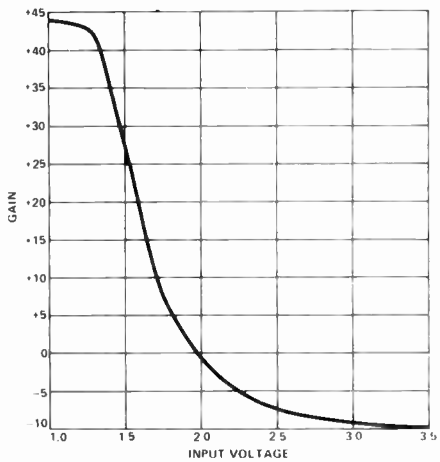


Fig. 5. Printed-Circuit layout for the limiter. Full size 58 mm x 110 mm.

Fig. 4 Gain versus control voltage with R2 = 47k



AUDIO LIMITER

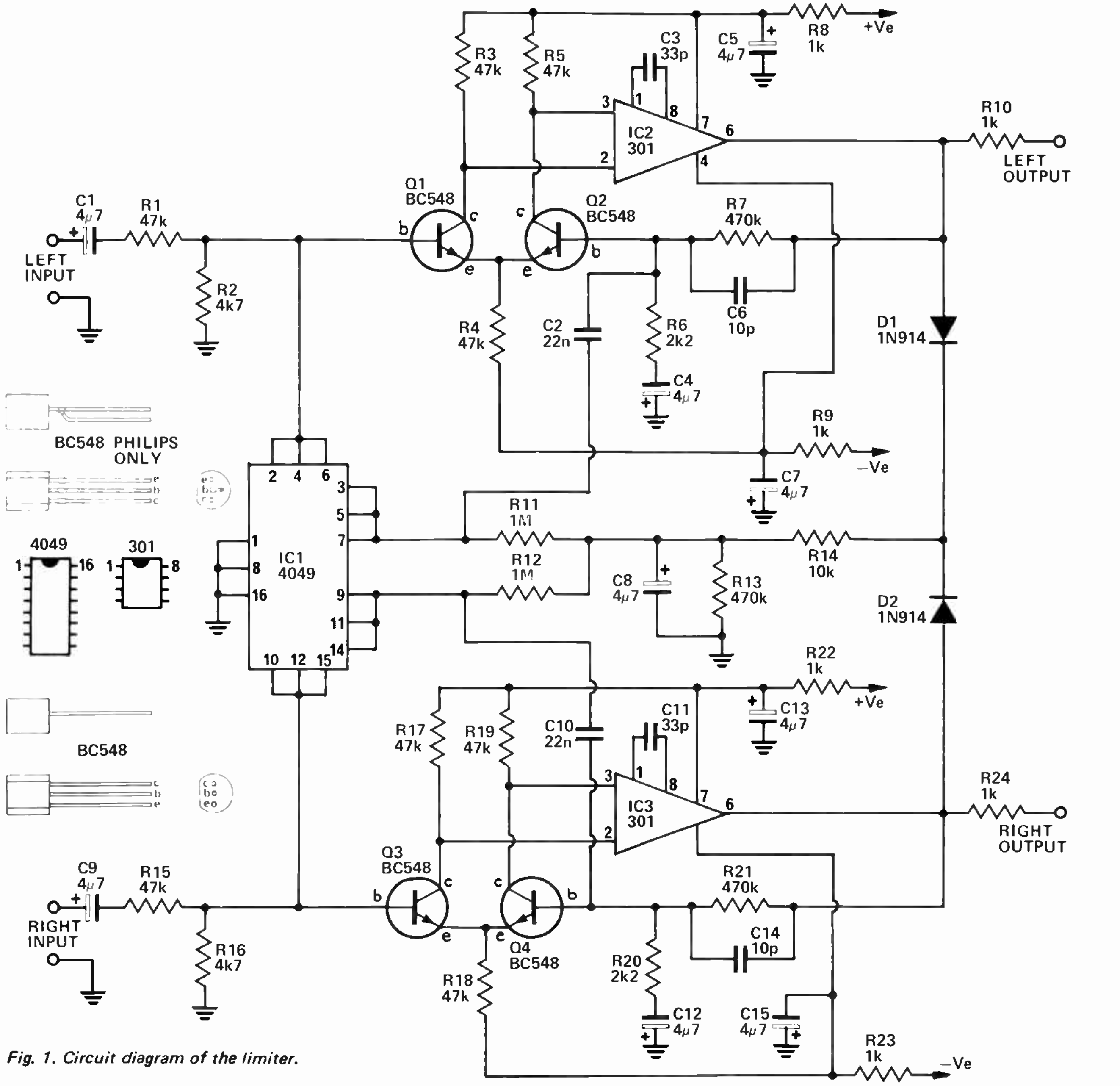


Fig. 1. Circuit diagram of the limiter.

How it works ETI-446

The circuit basically consists of a voltage-controlled attenuator followed by a low-noise amplifier with a gain of 46 dB. The output of this amplifier is rectified to generate a dc voltage which is used to control the attenuator.

The variable element in the attenuator is an enhancement mode FET. This is made from a CMOS hex-inverter IC, the 4049, by special interconnection. The difference between enhancement mode FETs and the normally available depletion-mode junction FETs is as follows: The enhancement mode FET has a high resistance between source and drain when the gate is at zero volts, but this decreases as the gate is taken more positive. A JFET (N type) is hard-on with the gate at zero volts and turns off as the voltage is taken negative.

The amplifier is required to have high open-loop gain and have fairly low noise. The gain requirement is provided by an LM301 operational amplifier and the low-noise requirement by a pair of transistors (connected as a differential pair) placed before the operational amplifier. The gain is set, by the combination of resistors R6 and R7, to 215 (or 46 dB). The lower 3 dB point is set at 15 Hz by C4 and R6 whilst the upper 3 dB point is set at 33 kHz by C6 and R7.

The outputs of both channels are summed and rectified by diodes D1 and D2 to charge C8 via R14. The voltage on C8 is coupled to the gate of the FETs (three in parallel on each channel) via R11 and R12.

As the input voltage increases the output also tends to increase and voltage on capacitor C8 also increases and this increase is applied back to the gates of the FETs. This reduces

PARTS LIST ETI 446

Resistors

R1	47k	½ W	5%
R2	4k7	"	"
R3-R5	47k	"	"
R6	2k2	"	"
R7	470k	"	"
R8-R10	1k	"	"
R11,12	1M	"	"
R13	470k	"	"
R14	10k	"	"
R15	47k	"	"
R16	4k7	"	"
R17-R19	47k	"	"
R20	2k2	"	"
R21	470k	"	"
R22-R24	1k	"	"

Capacitors

C1	4μ7 25 V electro
C2	22n polyester
C3	33p ceramic
C4,5	4μ7 25 V electro
C6	10p ceramic
C7-C9	4μ7 25 V electro
C10	22n polyester
C11	33p ceramic
C12,13	4μ7 25 V electro
C14	10p ceramic
C15	4μ7 25 V electro

Semiconductors

Q1-Q4 Transistors BC548
D1,2 Diode 1N914
IC1 Integrated circuit 4049 *
IC2,3 " " LM301

Miscellaneous

PC board ETI 446
9 PC board pins

**Do NOT substitute a 4009 or 4449 as the input protection is different. Nor should a Philips or Signetics type be used as these have a buffet output and cannot therefore be connected for use in the linear mode required for this project.*

the resistance of the FETs and thus increases the attenuation, tending to prevent the output from changing as much as the input does.

With all FETs the resistance changes with applied voltage and this gives rise to distortion. However by modulating the gate voltage with a signal equivalent to the voltage across the FETs the distortion is greatly reduced (3.5% down to 0.8%).

The attack and release times can be adjusted by varying R14 for attack and R13 for release.

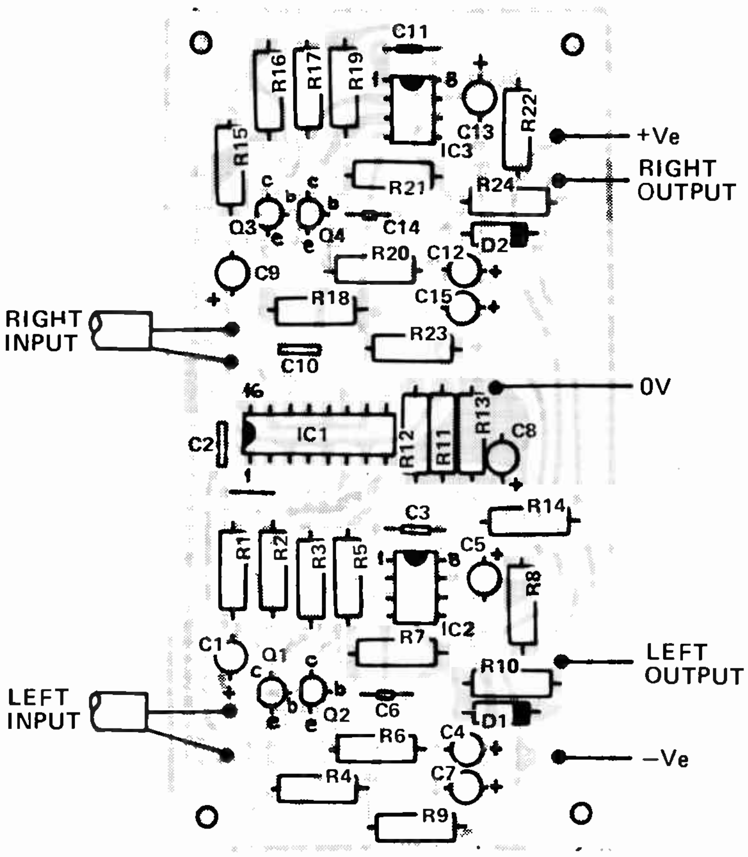
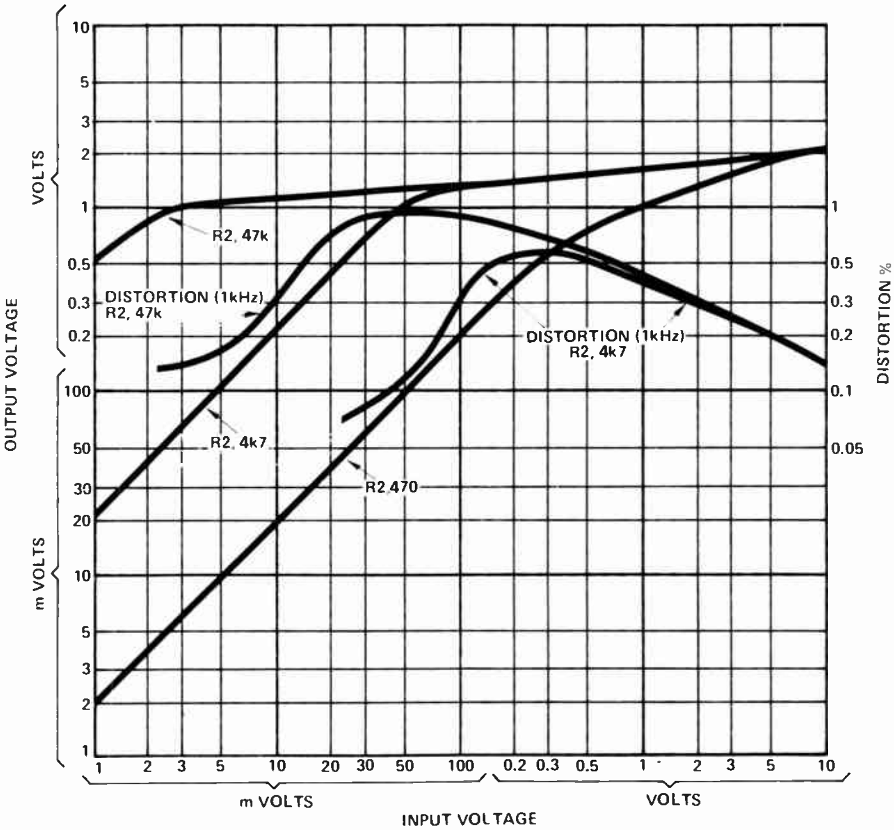


Fig. 2. Component overlay.

Fig. 3 Input versus output voltage for various values of R2 (and R16) Distortion at 1kHz for R2=4K7 and R2=47K are also shown.



ETI PROJECTS

Reprints of many of our most popular projects are available in book form. Top Projects Vols 4 and 5 and our Test Gear book are available from most newsagents or directly from us. Our address is:— Electronics Today International, 15 Boundary Street, Rushcutters Bay, NSW. 2011. The Synthesizer book is available only from us and a limited number of specialist suppliers — it is not sold by newsagents.

AUDIO EXPANDER COMPRESSOR. 50-100 WATT AMPLIFIER MODULES. AUDIO LIMITER. SELECTA-GAME. AUDIO PHASER.

POOL ALARM. TRAIN CONTROLLER. ACTIVE ANTENNA. GSR MONITOR. DYNAMIC NOISE FILTER. SELECTA-GAME. 'SCOPE TEST YOUR CAR. TEMPERATURE METER. UNIVERSAL TIMER. KITS FOR ETI PROJECTS. 50-100 WATT AMPLIFIER MODULES. GENERAL PURPOSE POWER SUPPLY. AUDIO LIMITER. TEMPERATURE ALARM.

TOP PROJECTS VOL 4
Published in June 1977. Projects include Audio Expander/Compressor, 50-100 Watt Amp Modules, Stereo Amplifier, Dynamic Noise Filter, Audio Phaser, Audio Limiter, TV Game, Swimming Pool Alarm, Train Controller, Car 'Scope Testing. Temperature Alarm, Active Antenna, GSR Monitor.
$3.00 plus 45 cents post and packing.

TOP PROJECTS VOL 5
Published in 1978. Projects include Shutter Speed Timer, Ultrasonic Switch, Accentuated Beat Metronome, Marine Gas Alarm, House Alarm, White Line Follower, Induction Balance Metal Detector, Photographic Strobe, Simple Compressor Expander, CB Power Supply.
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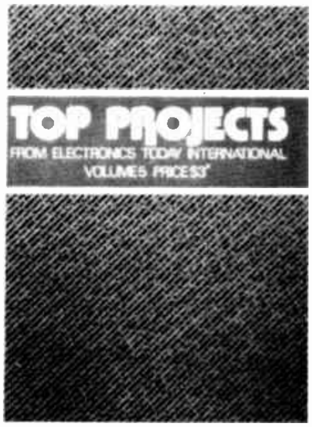

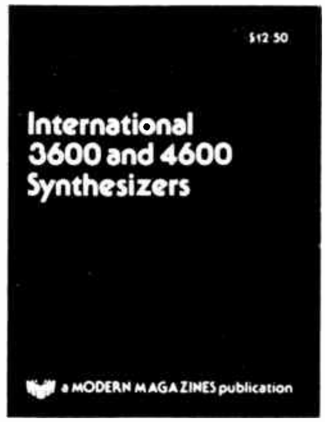

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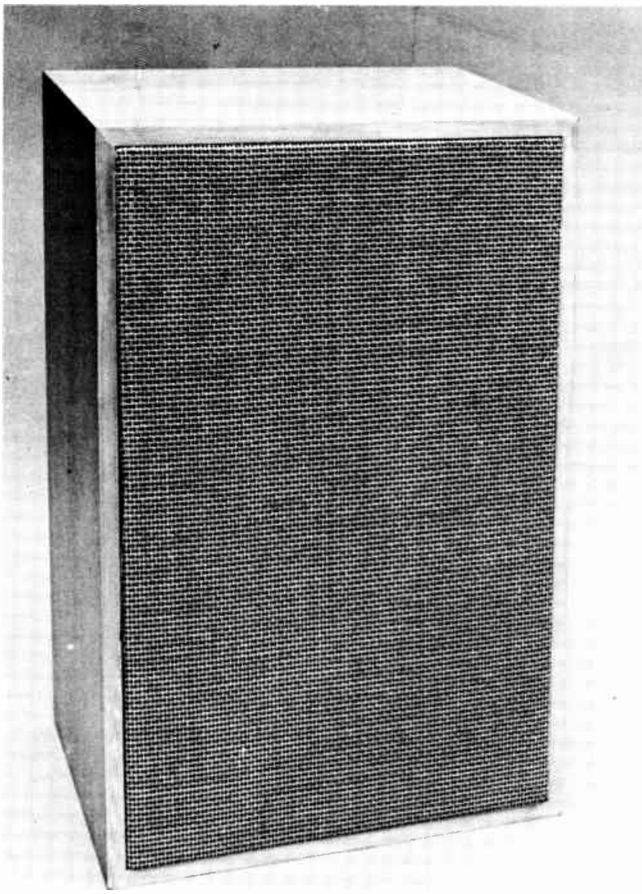
THREE-WAY SPEAKER SYSTEM



A high quality system suitable for use with conventional hi-fi amplifiers to 100 watts — or with ETI's active crossover and separate amps.

PROJECT 439

MANY of our readers have asked us to publish a design for a medium to large sized three-way speaker system.

As we were in any case planning to design an enclosure suitable for our previously described active crossover we decided to kill two birds with one stone and to design an enclosure suitable for either purpose.

We originally started our investigations by using a design supplied to home constructors by one of the major speaker manufacturers. Surprisingly we found that the original design had a number of failings — to the extent that little has been retained except for the original cabinet dimensions.

Several problems had to be overcome. Firstly, the original enclosure was found to cause bad colouration due to flexing of cabinet walls. Consequently the first modification was to brace the box walls. It was then determined that the sensitivities of the three drivers were different, the main problem being that the mid-range was about 4 to 5 dB down on the tweeter and the woofer. This problem was overcome by adding an attenuator network to the tweeter and by winding coil L1 with two to three ohms resistance for the woofer. The increased dc resistance of the coil attenuates the woofer slightly as well as allowing a much cheaper coil to be used. Both coils in the bass mid crossover (L1 and L2) were wound this way as a higher resistance in the coil L2 does not have any serious effects and avoids the necessity of using two different coils.

Tests then showed that the crossover frequencies were higher than they should be and there were deep holes in the response. This was caused by the rising impedance of the drivers in the crossover region which caused

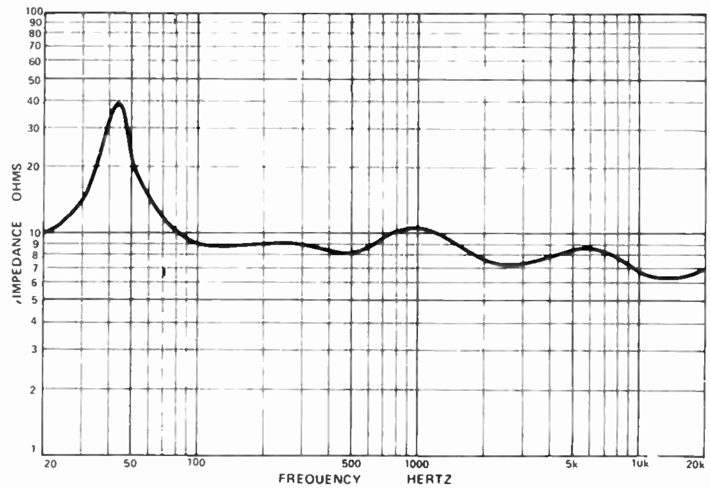


Fig. 1. Impedance versus frequency of the ETI 439 system (conventional cross-over).

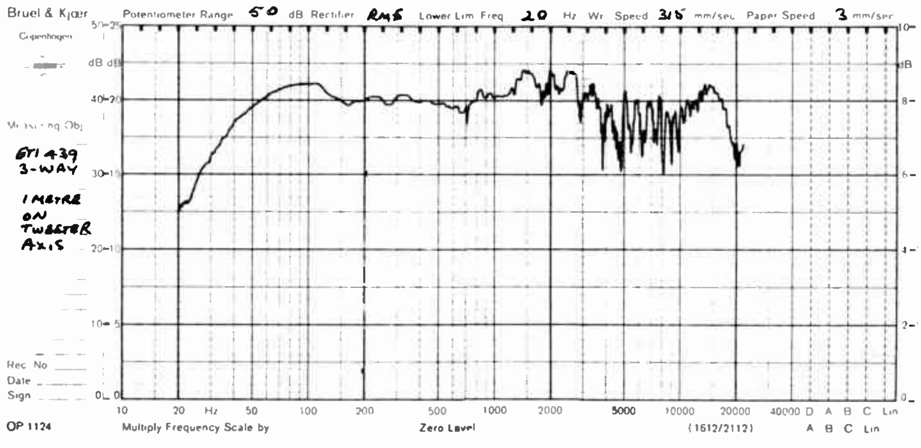


Fig. 2. Frequency response of the conventional system. Large dips in the response between 3 kHz and 10 kHz are due to measuring microphone positioning not to speaker deficiency.

THREE-WAY SPEAKER SYSTEM

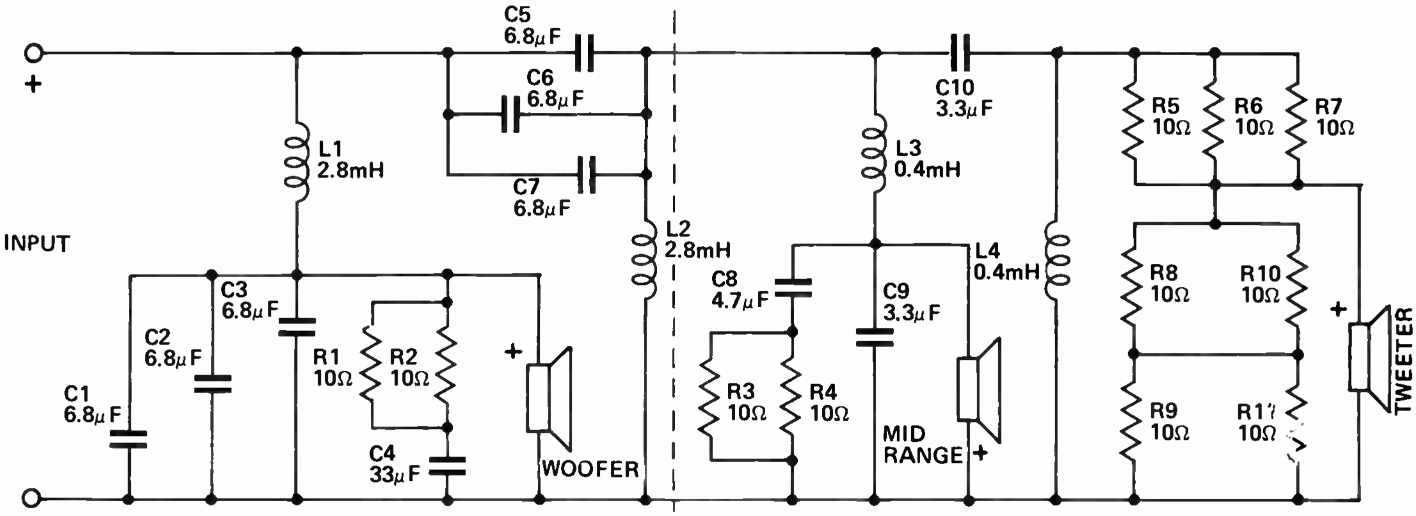


Fig. 3. Circuit diagram of the three way crossover. If crossover amplifier is used only the components to the right of the dotted line are required.

impedance mismatch and phase errors. This was cured by adding compensating networks, R1, R2 and C4 for the woofer and R3, R4 and C8 for the midrange, to control the impedance presented to the crossover network. This restored the crossover frequencies to their proper positions and smoothed out the response considerably. Further tests showed that the positioning of the drivers in the original system was causing cancellation between woofer and mid, and mid and high off-axis and the drivers were therefore positioned in a vertical line. This improved the smoothness of the off-axis response considerably.

Finally a new 50 mm dome midrange (AD 0210/SQ8) was substituted for the 125 mm cone type (AD 5060/SQ8) specified for the original system. Although the 50 mm unit is much more expensive it has similar sensitivity but wider dispersion and considerably lower distortion. If expense is a limiting factor however the cone type will give quite acceptable results. Note however that the mounting hole for the 50 mm dome is 112 mm and is *larger* than that required for the 125 mm cone type (108 mm).

CONSTRUCTION

The Crossover.

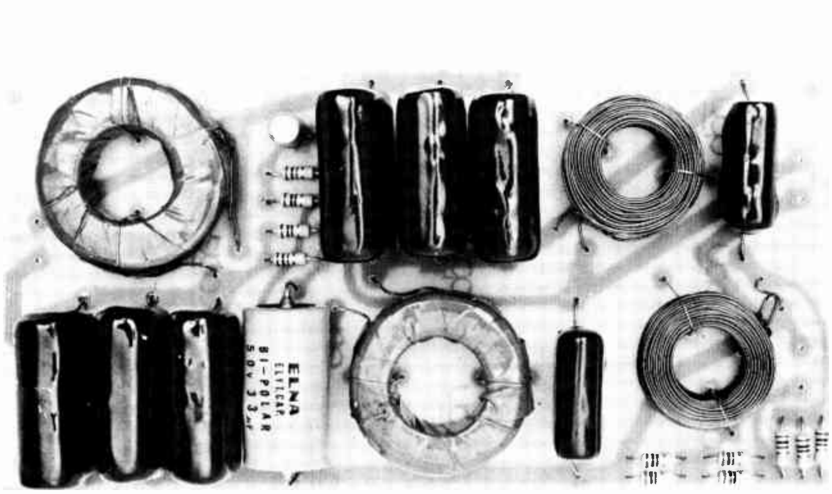
All the crossover coils are wound with 0.063 mm (22 B & S) wire. Coils L1 and L2 are wound on a 13 mm former and coils L3 and L4 are wound on a 10 mm former. These coils do not have many turns and are quite easily wound by hand. Try to keep the turns in layers as far as possible but some jumbling of turns will not materially affect results.

Polyester capacitors should be used in all positions except for C4 and C8 which may be non-polarised electrolytics. Note that the resistors are made up by series – parallel combinations of 10 ohm half-watt resistors. This is cheaper than using individual high-power resistors for these positions.

The polyester capacitors are available from many sources including Philips and Soanar. Space has been allowed for the physically largest type.

After mounting all components on to the boards, attach leads to the

The completed crossover network contains all the components for the conventional three-way system. The same board may be used if crossover amplifier is used simply by leaving off the unnecessary components.



PARTS LIST — ETI 439

R1-R11	Resistor	10 ohm ½ W 5%
C1-C3	Capacitor	6.8 μF polyester
C-4	"	33 μF non polarised electro
C5-C7	"	6.8 μF polyester
C8	"	4.7 μF non polarised electro
C9,10	"	3.3 μF polyester
L1,2	Inductor	2.8 mH see table 1
L3,4	"	0.4 mH " " "

PC Board ETI 439

Philips AD12100/W8 woofer
Philips AD0210/SQ8 or AD5060/SQ8 mid range
Philips AD0160/T8 tweeter

Wood Box to Fig. 5.

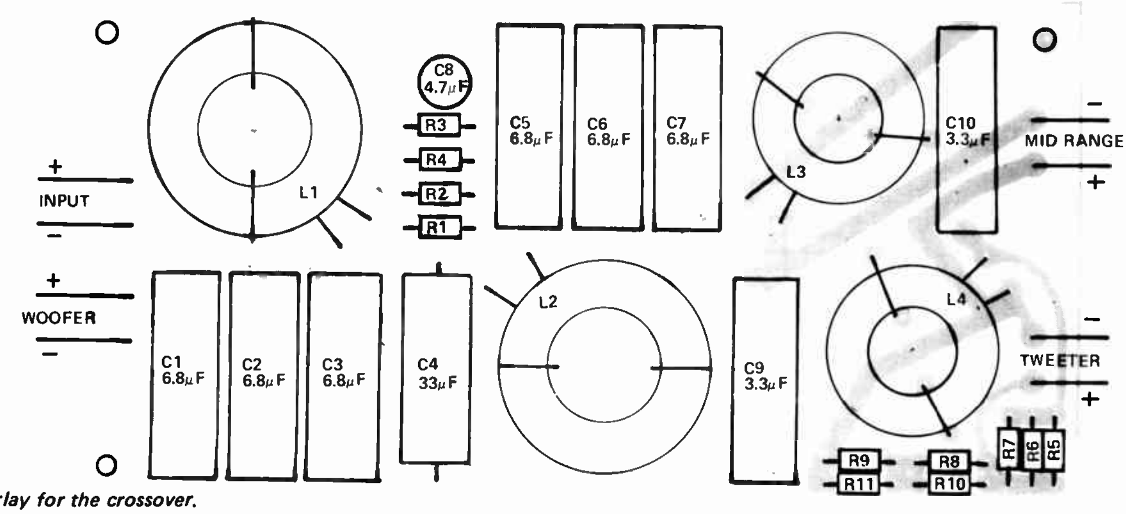


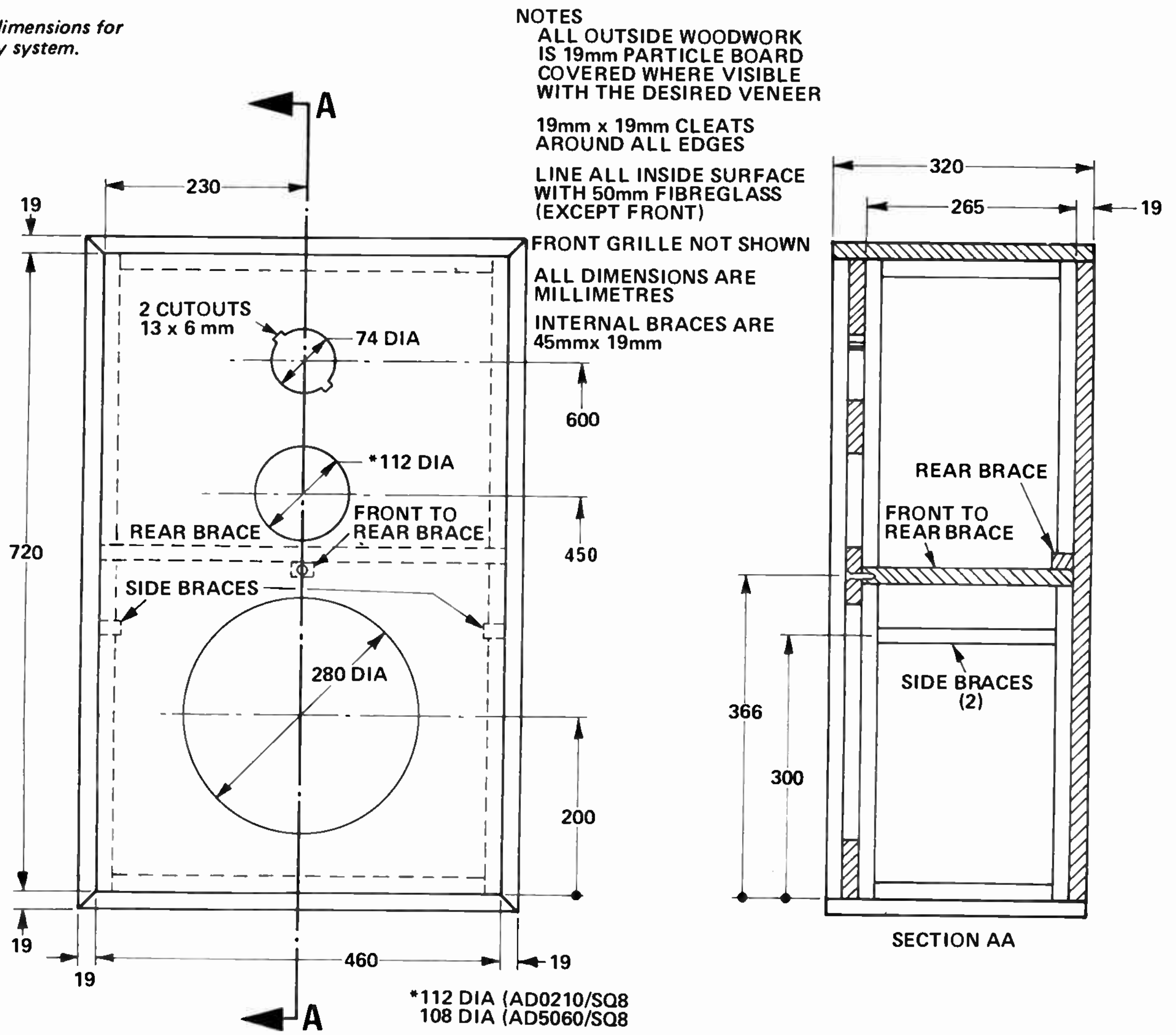
Fig. 4. Component overlay for the crossover.

crossover long enough for later connection to the drivers, and to the rear panel terminals.

The Box.

The enclosure volume is 82 litres (2.9 cubic feet), the outside dimensions of the box are 756 x 496 x 320 mm. Two boxes may both be cut from a single 1800 by 1200 mm sheet of 18 mm veneered pine board except for the front and rear panels which are cut from plain 19 mm pineboard. The internal bracing is 45 x 19 mm hardwood glued on edge in the positions shown in the drawings. A further brace between centres of front and rear panels may also be added. Cleats of 19 mm square timber should be glued into all internal corners of the box and care should be taken to ensure that the box is absolutely airtight.

Fig. 5. Box dimensions for the three-way system.



THREE-WAY SPEAKER SYSTEM

TABLE 1

INDUCTOR WINDING DETAILS

L1, 2	2.8 mH 300 turns 0.063 mm (22 B&S) wire 25 mm inside diameter 13 mm long
L3-4	0.4 mH 130 turns 0.063 mm (22 B&S) wire 20 mm inside diameter 10 mm long

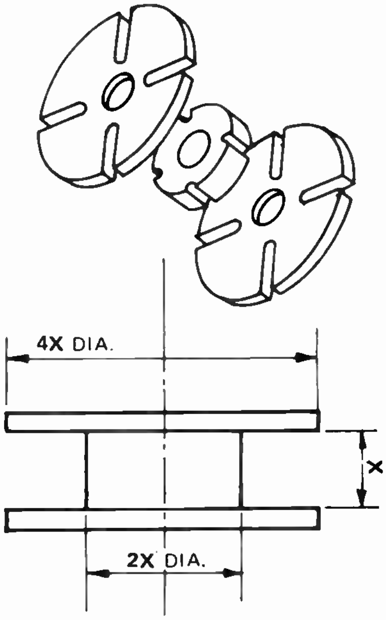
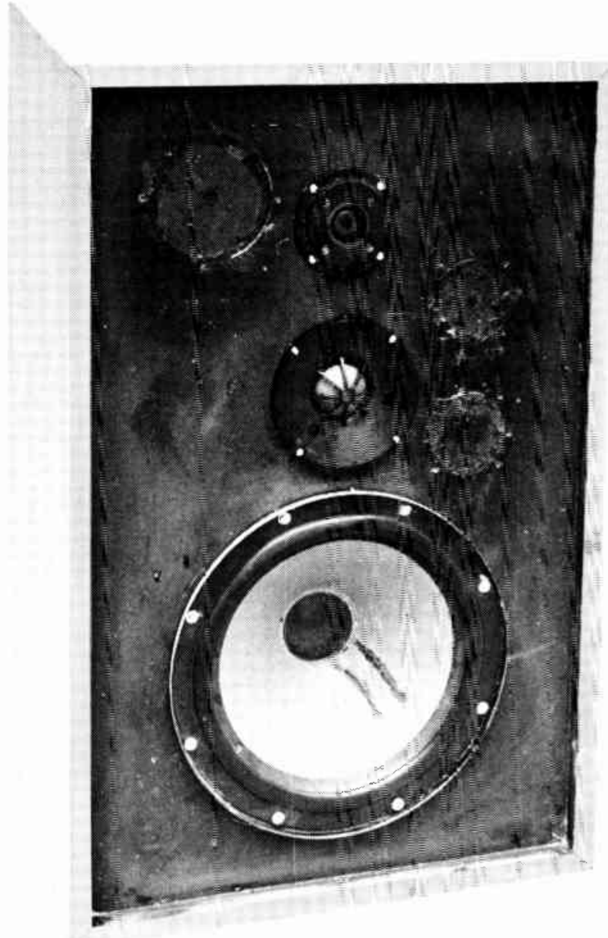


Fig. 6. Dimensions of bobbins used to wind coils L1 to L4. For L1 and L2 'x' = 13 mm, for L3 and L4 'x' = 10 mm.



The prototype speaker with grille removed. The plugged holes were for trial positioning of midrange and tweeter.

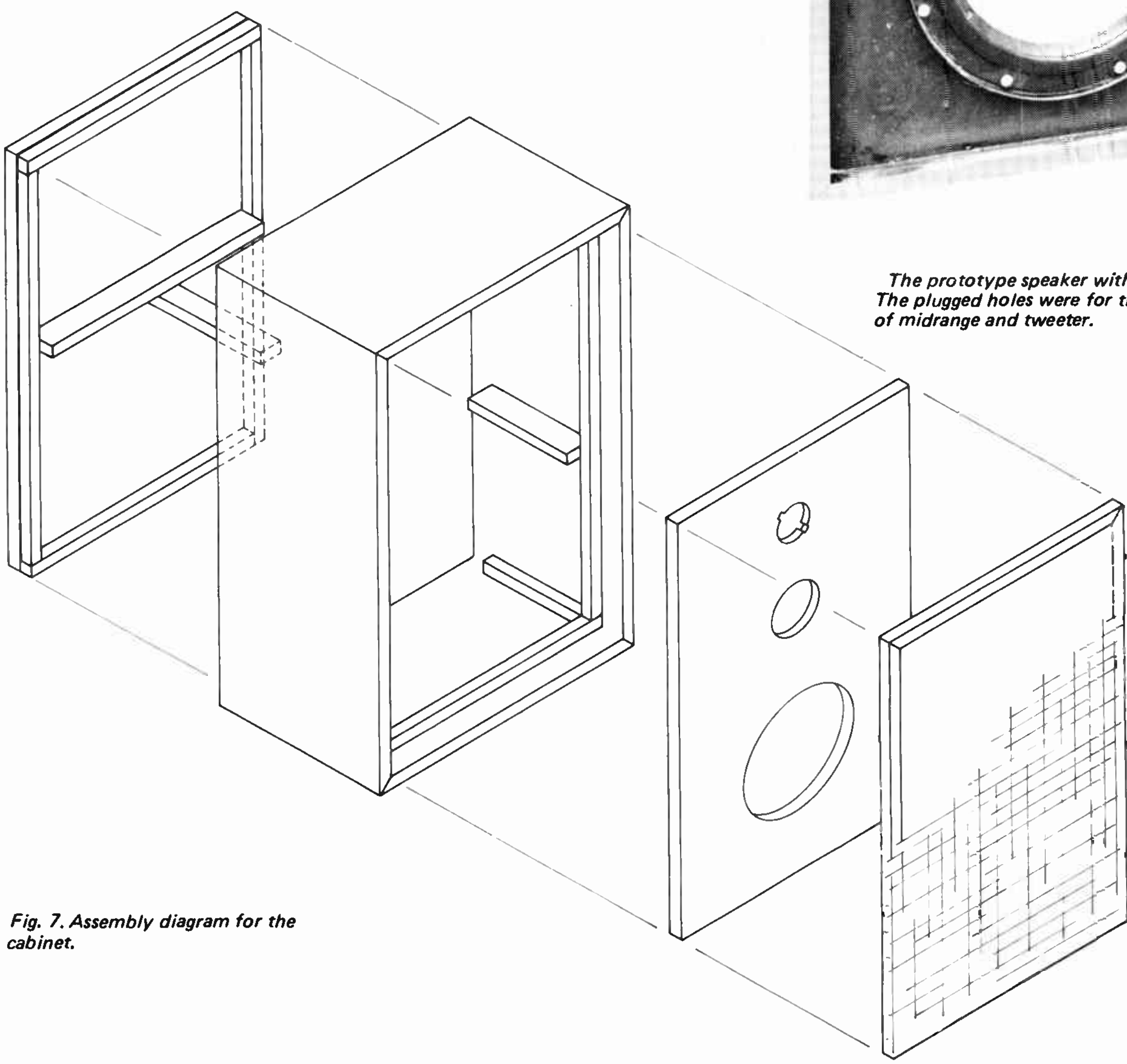


Fig. 7. Assembly diagram for the cabinet.

The box should be lined on all sides and the back with 50 mm – thick fibreglass and the cross-over network mounted in the vicinity of the woofer. Mount the terminals to the rear panel, connect them to the crossover network and drape the leads for the drivers out of their respective holes. Roll out some plasticine such that its diameter is about 2 or 3 mm and apply it around the circumference such that a good seal will be obtained when the driver is mounted in position. Finally, carefully fit the drivers into their holes and secure them with wood screws.

If an oscillator is available drive the speaker with a low frequency, about 20 to 30 Hz, and listen carefully to determine if there are any air leaks. These will be evident as whistling or hissing sounds. If any leaks are found they must be cured as the performance of the box will otherwise be adversely affected.

Make the grille frame from 18 mm square timber and staple or glue the grille cloth to it. Finally polish the boxes by any suitable method – we found that a Scandinavian-oil finish is easy to apply and looks very professional.

Electronic Crossover.

This speaker system may be used in conjunction with ETI's active cross-over network (pages 27 to 33) and separate bass and mid/high range amplifiers. The ETI 480 amps described on pages 102 to 108 of this book are particularly suitable for this purpose.

The woofer should be driven from one channel and the mid/high range drivers from the other. Thus the mid/high drivers still require a passive cross-over (three separate amps may of course be used but we have found that the increase in performance by so doing is absolutely marginal). Thus three terminals will be required on the input panel: one for woofer input, a common, and one for mid/high.

The crossover required for the high is that to the right of the dotted line in Fig. 3. Note however that with this system the woofer must be connected in the same phase as the mid-range rather than in anti-phase as shown in Fig. 3.

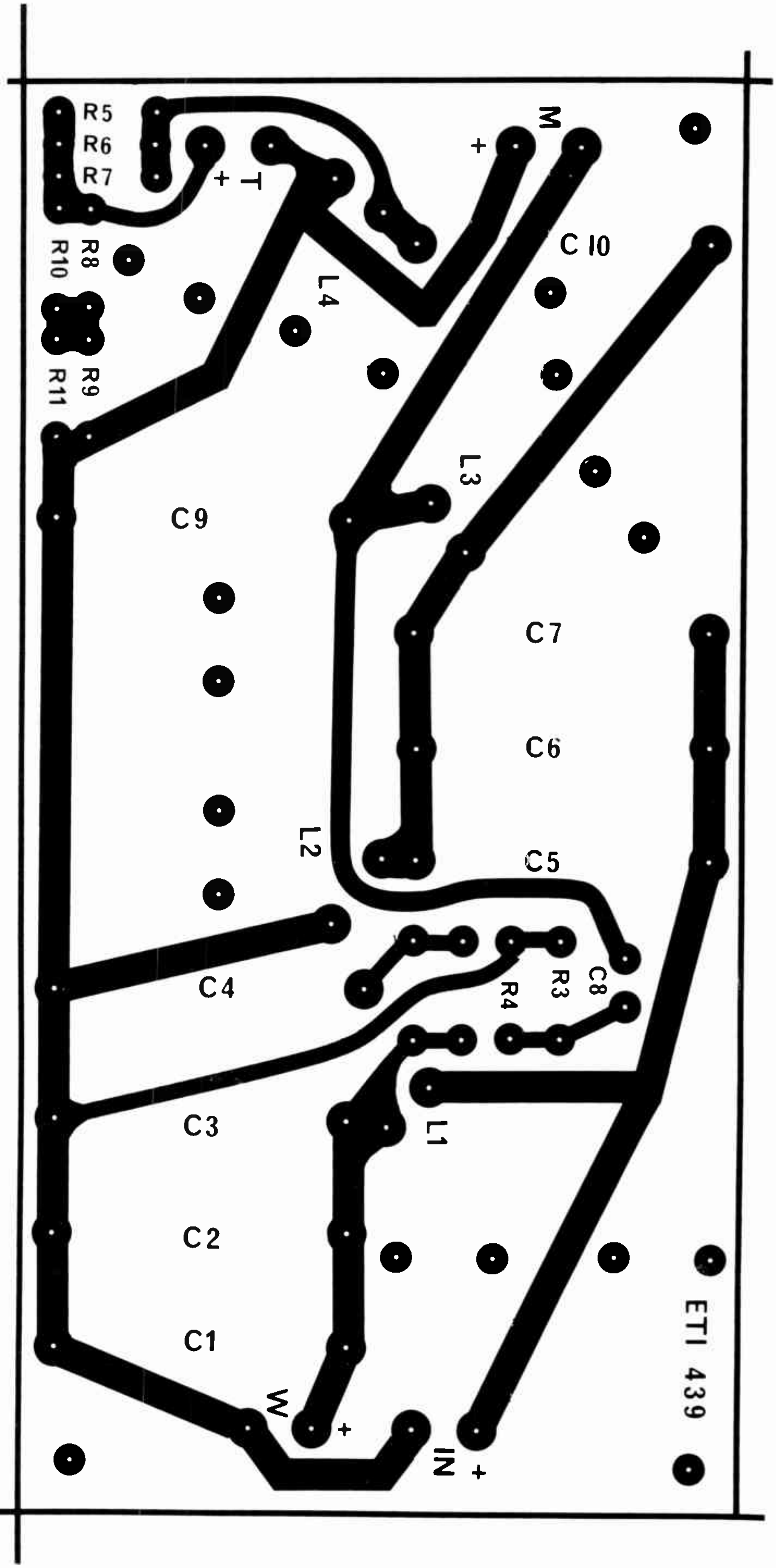
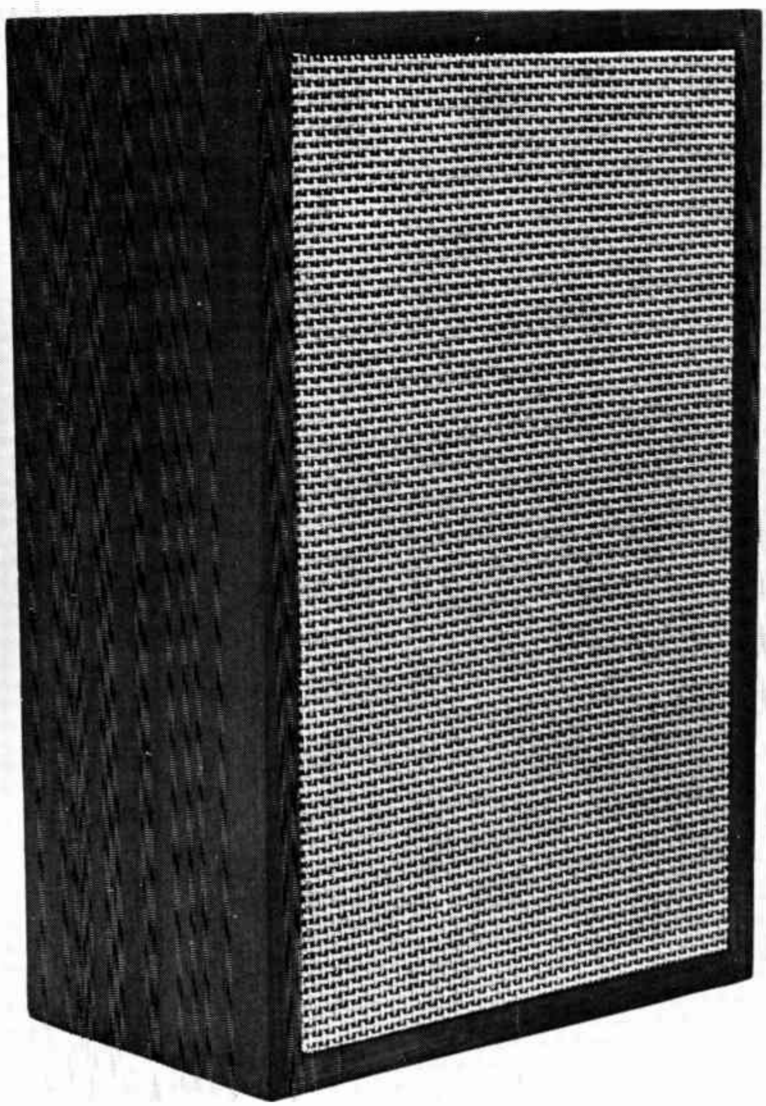


Fig. 8. Printed circuit layout for the crossover board. Full size 224 x 113 mm.

ETI 400 SPEAKER SYSTEM

Acoustic suspension design has big sound.



The cross-over network described in this article must be used exactly as specified if the intended performance is to be obtained.

A simpler version of this network – or a simple series capacitor – is not 'almost as good'. Suppliers are warned that to market the ETI 400 design in any other than the form specified here is a breach of the Trades Practices Act.

EVER since we published our extraordinarily successful Magnavox 8-30 speaker design some five years ago many readers have asked us to design a loudspeaker system that had at least equivalent performance but of smaller overall dimensions.

The design published here will we are sure satisfy the needs of these readers. It is an 'infinite baffle' design based on Philips drive units.

When we first started to investigate this project, we based our prototype on a system described in the Philips Elcoma publication 'High Fidelity Loudspeakers and Enclosure Designs'.

The Elcoma design is simple and effective but the basic cross-over network does not really do justice to the truly excellent design of the drive units specified. It just is not possible to obtain really top-class performance from a multi-speaker enclosure unless a good cross-over network is used. And a good cross-over network cannot be built cheaply.

Initial experiments showed that truly excellent performance was obtainable using a better cross-over. This being so we concentrated on designing the enclosure before finalising the cross-over network described later in this article.

THE ENCLOSURE

The design of an enclosure for an acoustic suspension speaker system is determined by making compromises on three basic quantities:
(1) The volume of the enclosure.
(2) The efficiency of the system.
(3) The low frequency cutoff.

There is no mandatory volume for an acoustic suspension system but tests over hundreds of different systems have shown that the optimum volume for a 200mm (8 inch) driver lies between 14 litres (0.5 cu ft) and 42 litres (1.5 cu ft). The 14 litre enclosure will sacrifice bass response and efficiency but will handle more power whilst the 42 litre enclosure has extended bass response is more efficient but will handle much less power. We opted for a 20 litre (0.7 cu ft) enclosure as the one which offered reasonable bass response and good power handling with the particular driver being used. (This is in contrast to the recently released kit from Philips using the same drivers that has a volume of 15.6 litres – which in our opinion sacrifices bass response for a smaller albeit cheaper enclosure).

The ETI 400 speaker uses the Philips 203 mm (8") bass driver and the 25 mm (1") dome tweeter.

The dome tweeter is known to be more efficient than the bass driver – in fact our measurements showed that this was of the order of 4 dB. We have therefore included a 4 dB resistive attenuator pad before the tweeter to match it to the woofer. (Philips have specified an 8 dB attenuator for their recently released kit but both measuring and listening tests confirm that 4 dB attenuation is better). The resistor pad has a fortuitous advantage in that it provides extra tweeter damping, considerably improving its sound — especially at the top end where the undamped tweeter (due to crossover impedance) tends to be a little harsh.

The resistive pad and tweeter is fed from C3 and L2 (see Fig. 3) which form a 12 dB per octave high pass filter allowing only frequencies above 2.2 kHz to pass — a 12 dB per octave crossover *must* be used if damage to the tweeter is to be avoided. Some people who built up the Magnavox 8-30 system, previously described in ETI, complained of tweeters burning out. We investigated many of these complaints and found that the problem was caused by using a single capacitor feed to the tweeter rather than the specified network. The Philips tweeter has a pronounced resonance around 900 Hz and if this is not adequately suppressed the tweeter will be damaged by excessive cone excursion at this frequency.

A 12 dB per octave network has also been provided for the woofer and again this should be used if proper mid-range response is to be obtained. Network R1 and C1 provides compensation for the rising impedance of the woofer (with frequency) and effectively keeps the response reasonably level up to the crossover frequency.

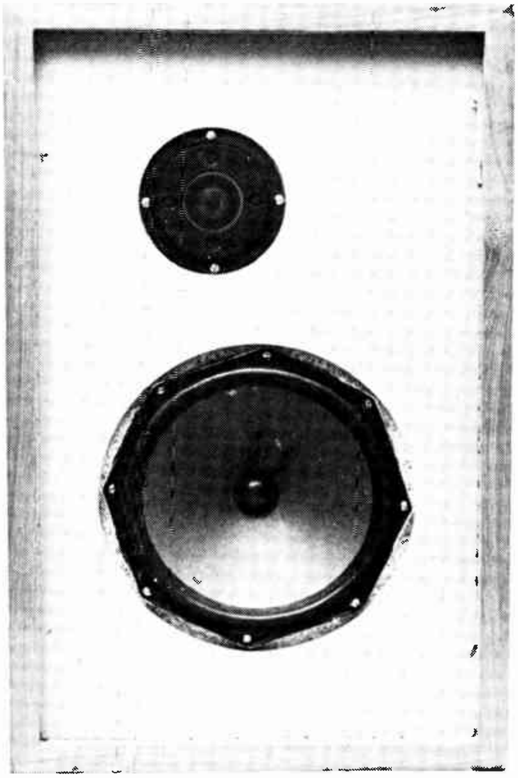
Capacitors C2 and C3 should be polyester types — not non-polarized electrolytics! However C1 may be a non-polarized electrolytic if desired.

The coils should be of air wound construction (see winding details) and *not* of the iron cored variety. Iron cored coils tend to saturate at high levels, producing a very nasty kind of distortion similar to amplifier clipping.

Resistors R2 and R3 may be constructed from jug element as follows. Measure out a length of jug element having the required resistance and wind it around the body of a 1 watt resistor (any value above 100 ohms) soldering one end to each of the resistor leads. The wire may be fixed in position on the resistor by a little 5 minute epoxy.

Note that the tweeter is connected out-of-phase. This is necessary due to phase shifts in the crossover. Conventional connection results in a deep hole in the response at about 3 to 4 kHz in addition to a 10 dB peak at around 2 kHz!

The coils may be hand wound, in accordance with Table I, on the former shown in Fig. 4.



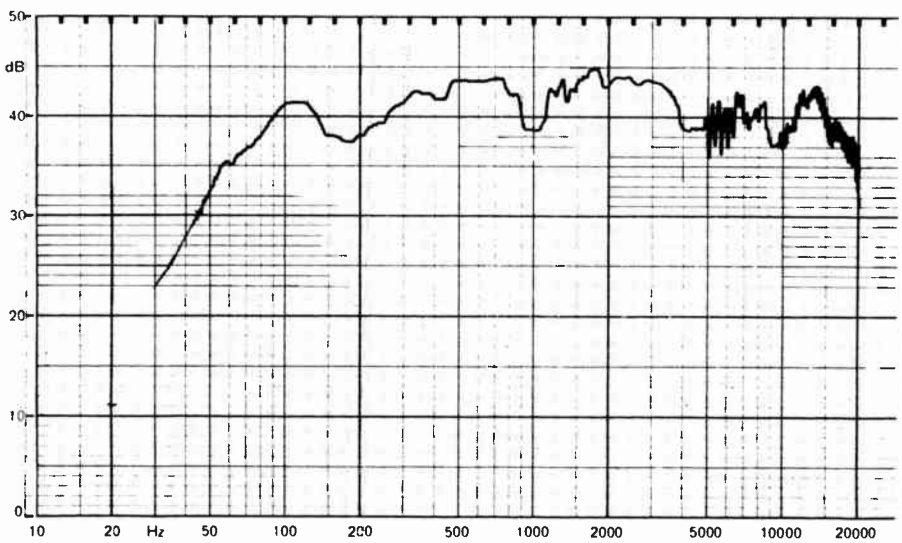
The completed speaker before front panel is painted or grille cloth fitted.

We must emphasize again that the crossover is the heart of any good speaker design. The circuit as described for the ETI 400 *must* be used if good results are to be obtained.

CONSTRUCTION

Dimensions of the enclosure and its assembly are illustrated in Figs. 1 and 2. Note that 19 mm square cleats should be glued into all corners. It is absolutely essential that all joints be airtight, for, if the enclosure leaks at all, the air rushing in and out will produce hissing sounds and the bass response will be seriously degraded.

Note that we used veneered pine-board for our prototypes and hence the drawings show mitred joints at the corners. If such joints are beyond your woodworking capability it may be well to use plain, unveneered pineboard and butt joints. The whole box may then be covered with iron-on veneer or with self-adhesive vinyl. Self-adhesive veneer or vinyl does not adhere too well to plain pine-board and tends to lift or bubble after some time. We found that adhesion could be improved by applying one coat of clear lacquer to the pine-board before veneering. This has the effect of



Frequency response of the ETI400 speaker system.

TABLE 1 CONSTRUCTIONAL OATA

CHOKES 0.8 mH, L1 & L2
170 turns of 0.8 mm (20 B&S) on former shown in Fig.4.

RESISTORS
Resistors can be made from jug element as follows. Standard jug element coils are 300 mm long and have a total resistance of 38 ohms. Therefore a 25 mm length (unstretched) will have a resistance of 3 ohms, and a 120 mm length (unstretched) will have a resistance of 15 ohms. That is use 8 mm of coil per ohm of resistance required.

CUTTING
Two speakers can be cut from one 1800 x 900 mm sheet of veneered pine board (including front and back panels if required).
Alternatively 4 speakers can be cut from one 1800 x 900 mm sheet of veneered pine board if the fronts and backs are cut from a separate sheet of 1800 x 900 mm plain pine board.

CAPACITORS
Any value of polyester capacitor between 5.6 and 6.8μF can be used for C2 and C3 if difficulty is experienced in obtaining the correct value. Both must be the same value. Such change will shift the crossover frequency slightly but will not have other serious effects.

sealing and strengthening the pine-board surface thus preventing veneer lift due to break-up of the pine-board surface.

When the box is completed paint the front panel matt black and finish the veneer surface as desired — a Scandinavian oil finish is easy to apply and very attractive.

Mount the input terminals to the back panel, the crossover network to the inside of the back panel and connect the terminals to the crossover. Attach leads to the crossover for the woofer and tweeter and drape these leads out through the respective driver holes.

Now line the box on all sides (except the front) with 50mm glass wool. Connect the leads to the drivers (watch polarities) and mount the drivers in position using plasticine or sealing putty around the rim of each driver to ensure an airtight fit.

Make the grille-cloth frame from 12mm square timber and attach the grille cloth by stretching it around the frame and then stapling it at the rear. If a stapler is not available the cloth may be glued on.

The speakers will safely handle 40 watts RMS. Although decidedly not recommended, our lab staff fed them with music peaking to 200 watts. Apart from our production department (located next door to the lab) stopping work no difficulty was encountered in handling this level. We therefore consider that the system is entirely safe with amplifiers which deliver 40 watts RMS per channel. ●

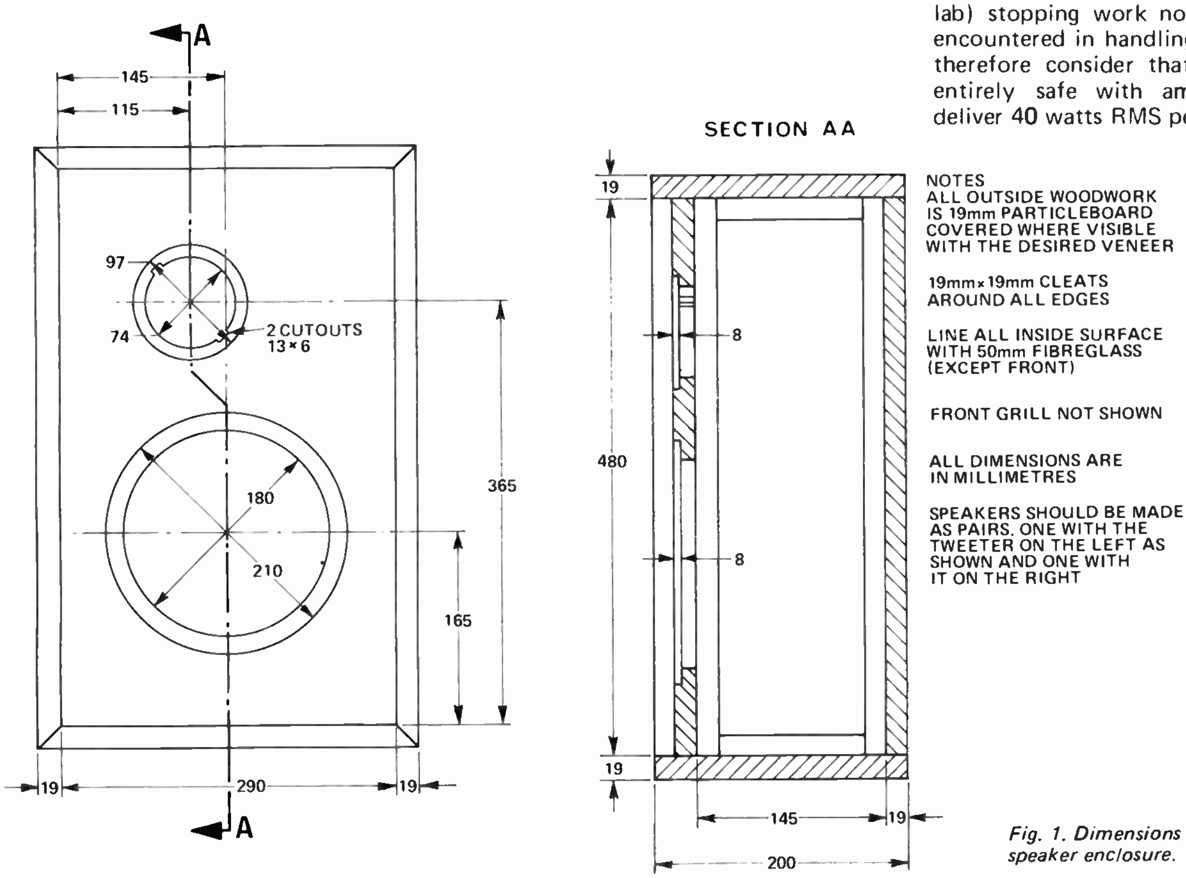


Fig. 1. Dimensions of the ETI 400 speaker enclosure.

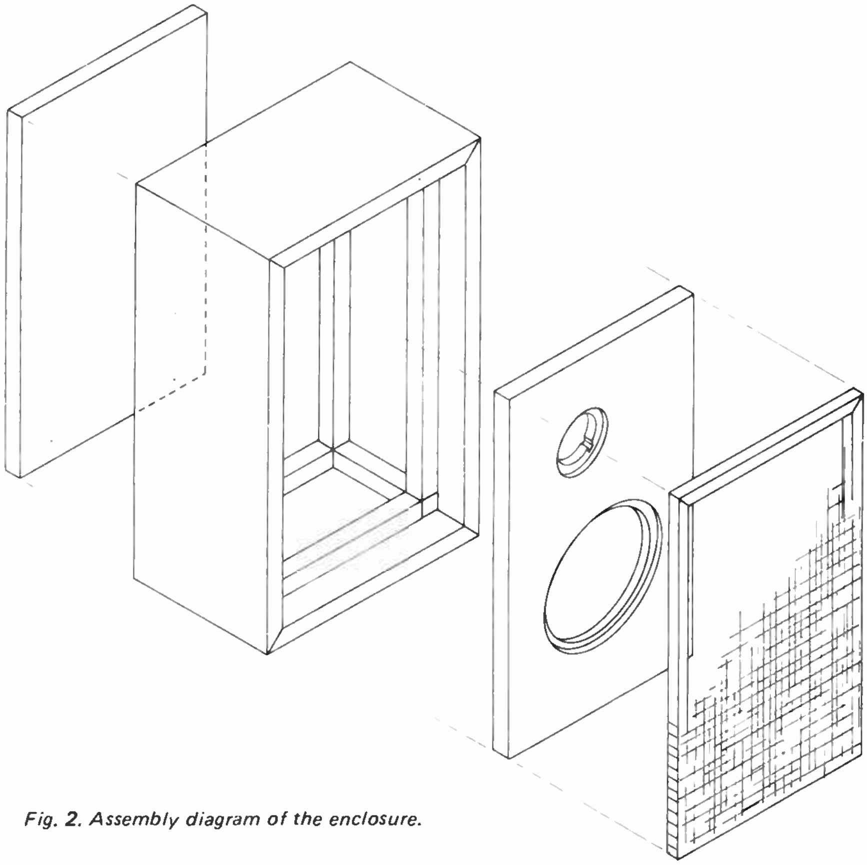


Fig. 2. Assembly diagram of the enclosure.

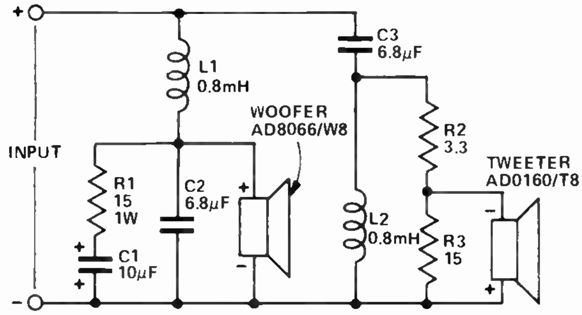


Fig. 3. Circuit diagram of the crossover network. This is the heart of the system and must not be changed if best results are to be obtained.

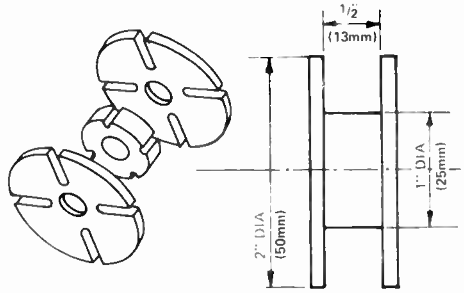


Fig. 4. Former for winding chokes L1 and L2. Chokes may readily be wound by hand, try and keep wire in uniform layers but a little jumbling will not appreciably affect the final value of inductance.

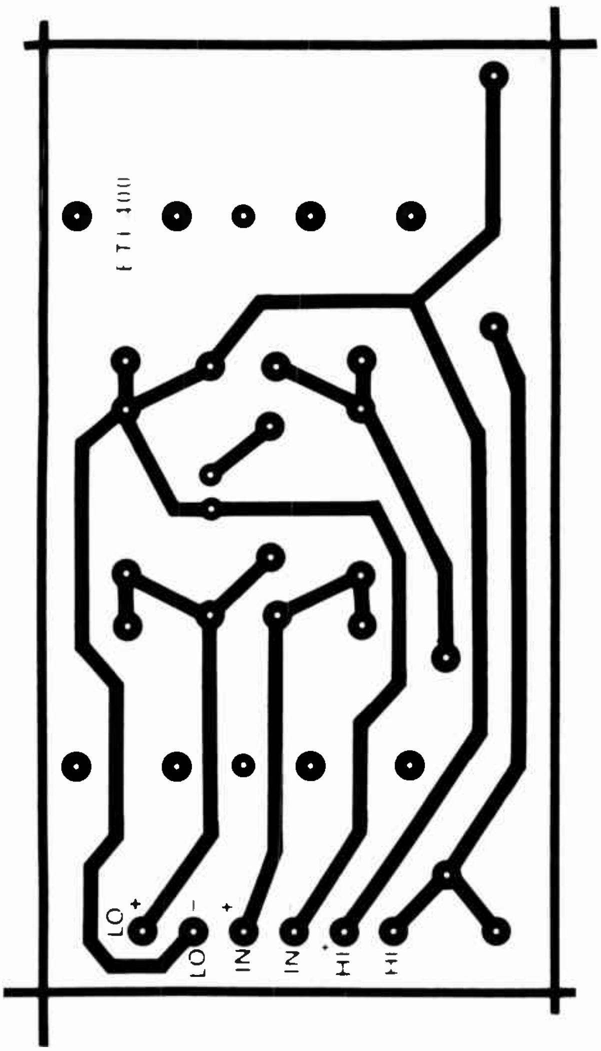


Fig. 5. Printed circuit layout for the cross-over network. Full size 145 x 78 mm.

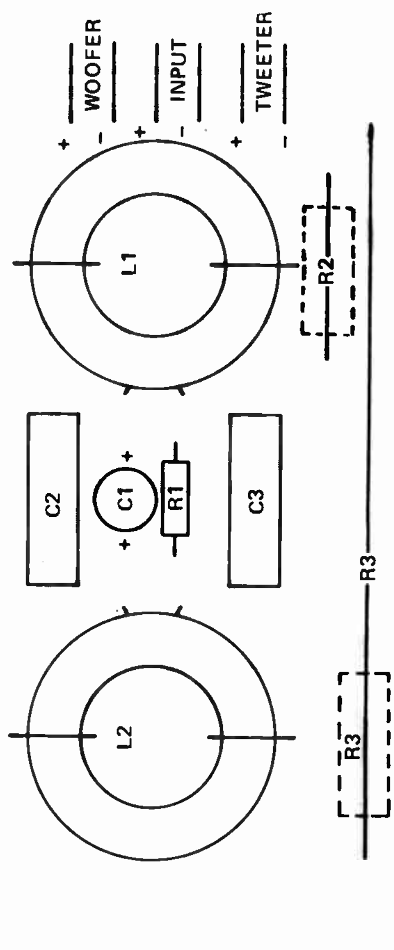
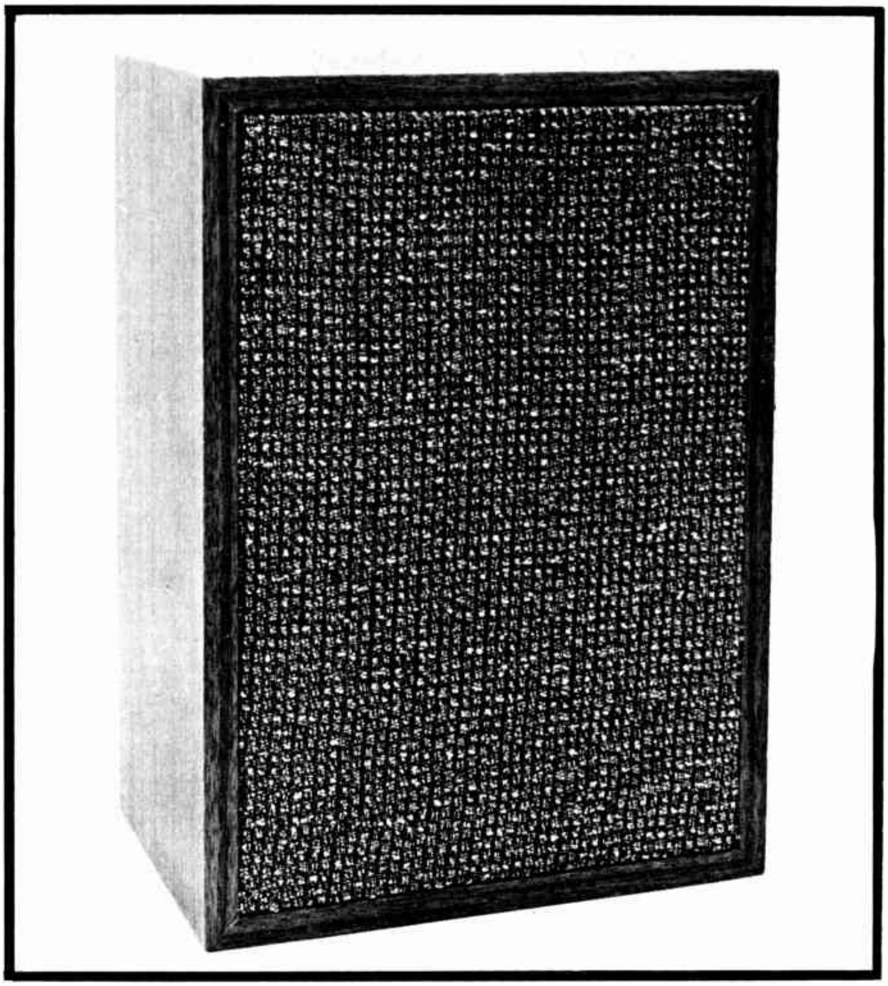


Fig. 6. Component overlay shows how to assemble the crossover network.



HI-FI SPEAKER

This hi-fi speaker can be built for only a few dollars yet has true hi-fi performance.

THE WAY a loudspeaker sounds is very largely determined by the inter-action between the actual speaker drive unit/s and the enclosure in which it is housed Unless the design is right the resultant sound will be unsatisfactory — regardless of the quality or cost of the drive unit used.

This point can be taken yet further. Some of the very best speakers made, costing hundreds of dollars each, and some of the very worst, use absolutely identical drive units.

It's not so much the drive units that matter — it's knowing what to do with them!

Providing the basic design is correct it is perfectly feasible to build good sounding speakers at quite low cost — as this project proves. The enclosures described are suitable for 150 mm or 175 mm diameter speakers of the 'wide-range' type. We chose the Philips AD 7063/M8 which is quite inexpensive, but almost any other wide-range speaker of similar characteristics and dimensions will do.

Most small speakers have a response which falls away at both the bass and treble ends of the audio range. It is for this reason that the more costly units have additional bass and treble drive units. Nevertheless the response of any simple unit can be greatly improved by adding a simple circuit which attenuates (cuts down) the mid-range. The effect is that bass and treble will now be reproduced at the correct level relative to the mid-range. Maximum sound output will be reduced slightly but the quality is enormously improved.

Construction

The enclosures should preferably be made from particle board — this is an easily worked 'dead' material, ideal for our purpose because of its inherent ability to dampen resonances. Use 13 mm board or thicker. If thicker, retain the *inside* dimensions shown on the drawings.

The box should be both glued and screwed together with the front panel temporarily in position so that the box is held square. When the glue has thoroughly set, sand the corners to obtain a smooth finish and then cover the box with wood-grained contact paper. Allow sufficient width of paper so that it may be folded around the inner edges and back a couple of centimetres or so onto the inner faces. The edges will be held firmly in place when the front and rear panels are fitted.

Screw the drive unit onto the rear of the front panel and then cover the panel with speaker cloth or any other material that is transparent to sound. Fit the front panel into position and secure by driving a few thin tacks through the front panel and into the cleats. The heads of the tacks may be hidden by driving them below the surface using a nail or centre punch.

Compensating network

The compensating network should be mounted directly into a hole cut in the rear panel and the choke glued onto the panel alongside the potentiometer. The bipolar electrolytic capacitor is mounted across the terminals of the potentiometer. The choke is connected to the same terminals. The whole network is then wired in series with one of the speaker leads – it doesn't matter which one.

Adjust the potentiometer for the most level response. Once adjusted you'll be surprised just how good this simple speaker sounds.

Note: The attenuating network described in this project will greatly improve the performance of *any* low priced speaker in which a single drive unit is used.

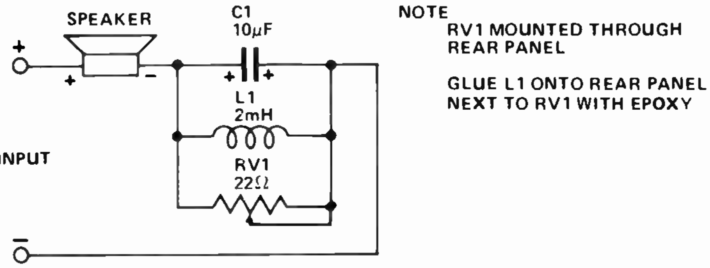
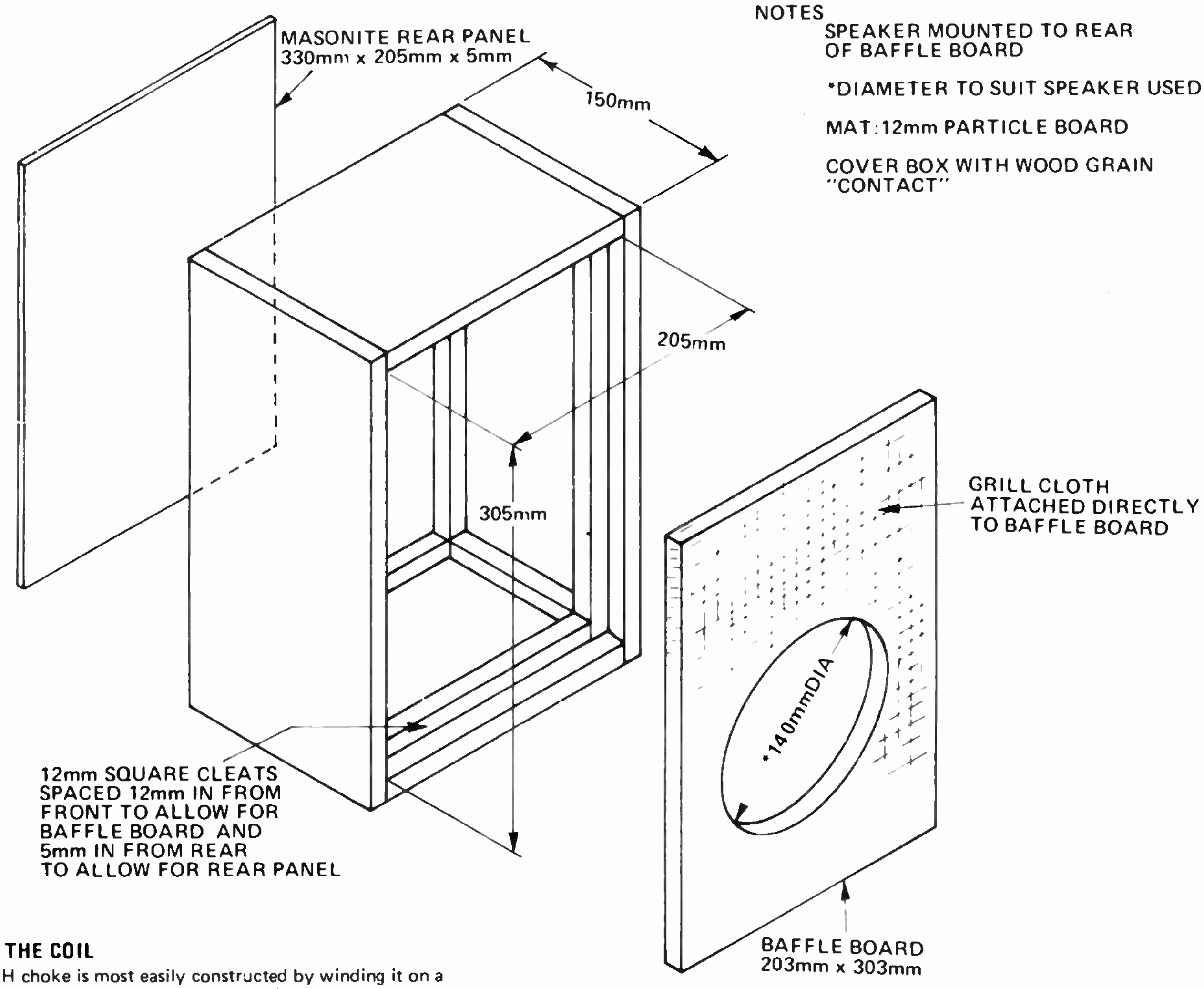


Fig. 1. Dimensions of the speaker box.



MAKING THE COIL

The 2 mH choke is most easily constructed by winding it on a Philips Elcoma P26 ferrite pot core. These P26 cores have different permeabilities which are marked on the core. Any of the types listed below may be used with the appropriate number of turns. Wire gauge is not critical. Anything over 0.4 mm, up to the maximum shown in the table may be used for any type core.

Do not use a bolt to hold the two halves of the cores together; use the Elcoma clip and then glue the completed coil into position.

TABLE 1

CORE TYPE	NO OF TURNS	MAX WIRE GAUGE
A_L 1600	20	1.0 mm
μ_e 330 or AL1000	55	0.8 mm
μ_e 220 or AL630	70	0.63 mm
A_L 400	70	0.63 mm
μ_e 150	65	0.5 mm
μ_e 100	80	0.5 mm
A_L 250	90	0.5 mm
μ_e 68	95	0.5 mm
A_L 160	110	0.4 mm

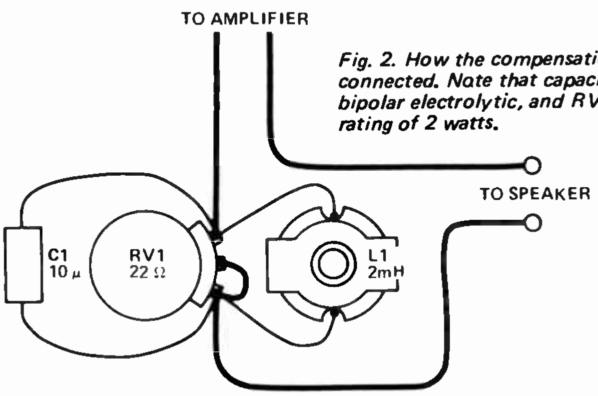
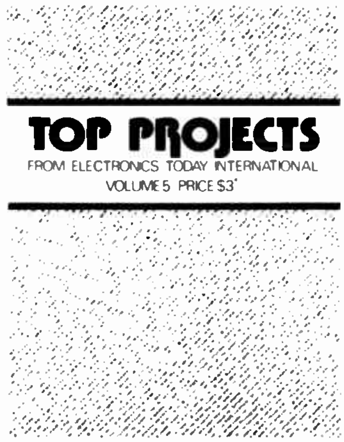


Fig. 2. How the compensation network is connected. Note that capacitor C1 is a bipolar electrolytic, and RV1 has a minimum rating of 2 watts.

ETI SPECIALS



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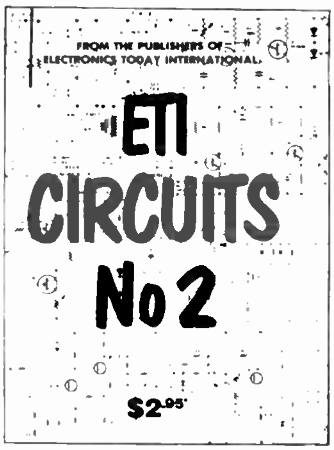

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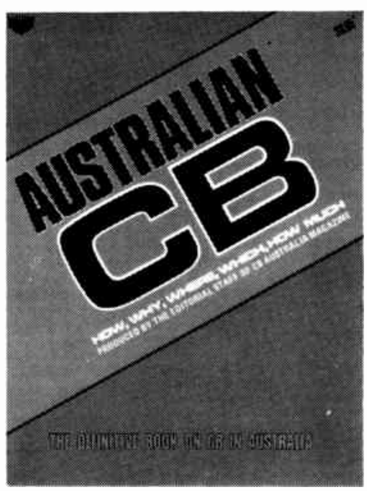

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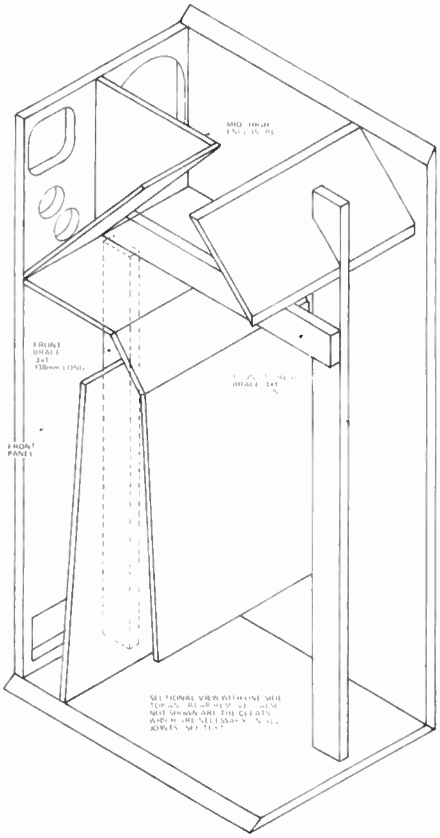
TRANSMISSION LINE SPEAKERS

These transmission line speakers have been designed and progressively developed by audio consultant Richard Timmins. In their final form they have been used as reference speakers by our sister publication Hi-Fi Review.



IN MANY respects transmission line speakers are an attempt to utilise the benefits of infinite baffle speaker enclosures but without the latter's inherent drawbacks – particularly that of restricted bass response.

Theoretically, transmission line speakers are essentially non-resonant over the entire low frequency register. In practice the need to fold the 'line' can introduce resonances and therefore colouration in the upper-bass and lower mid-range though these may be designed out by suitable techniques which are described later in this article.

Other advantages of the design include effective isolation of front and rear diaphragm output, effective control of diaphragm behaviour over the audible frequency range, bass response extended smoothly to the bass-driver's fundamental resonance (typically 25 Hz), and effective damping at that frequency.

As far as we can gather the first transmission line speaker was developed in 1936 by Benjamin Olney and demonstrated at the Acoustical Society of America's meeting in Chicago that year. Olney's enclosure was produced by Stromburg-Carlson for some years but was eventually eclipsed by less costly designs.

The transmission line principle appears then to have been largely neglected – particularly in the USA.

Arthur Radford worked on the principle from 1950 onwards – finally marketing his Radford Studio loudspeaker in 1964.

A.R. Bailey of Britain's Bradford Institute of Technology drew world-wide attention to the transmission line speaker in an article published in a 1965 issue of Wireless World. Bailey packed his labyrinth with long-fibre wool and this damped tube-resonance more effectively than Olney's lined walls of thirty years before.

Bailey compared his stuffed labyrinth to the ideal electrical transmission which is free of signal reflections – and test results indicated smooth, extended low frequency response and excellent transient performance.

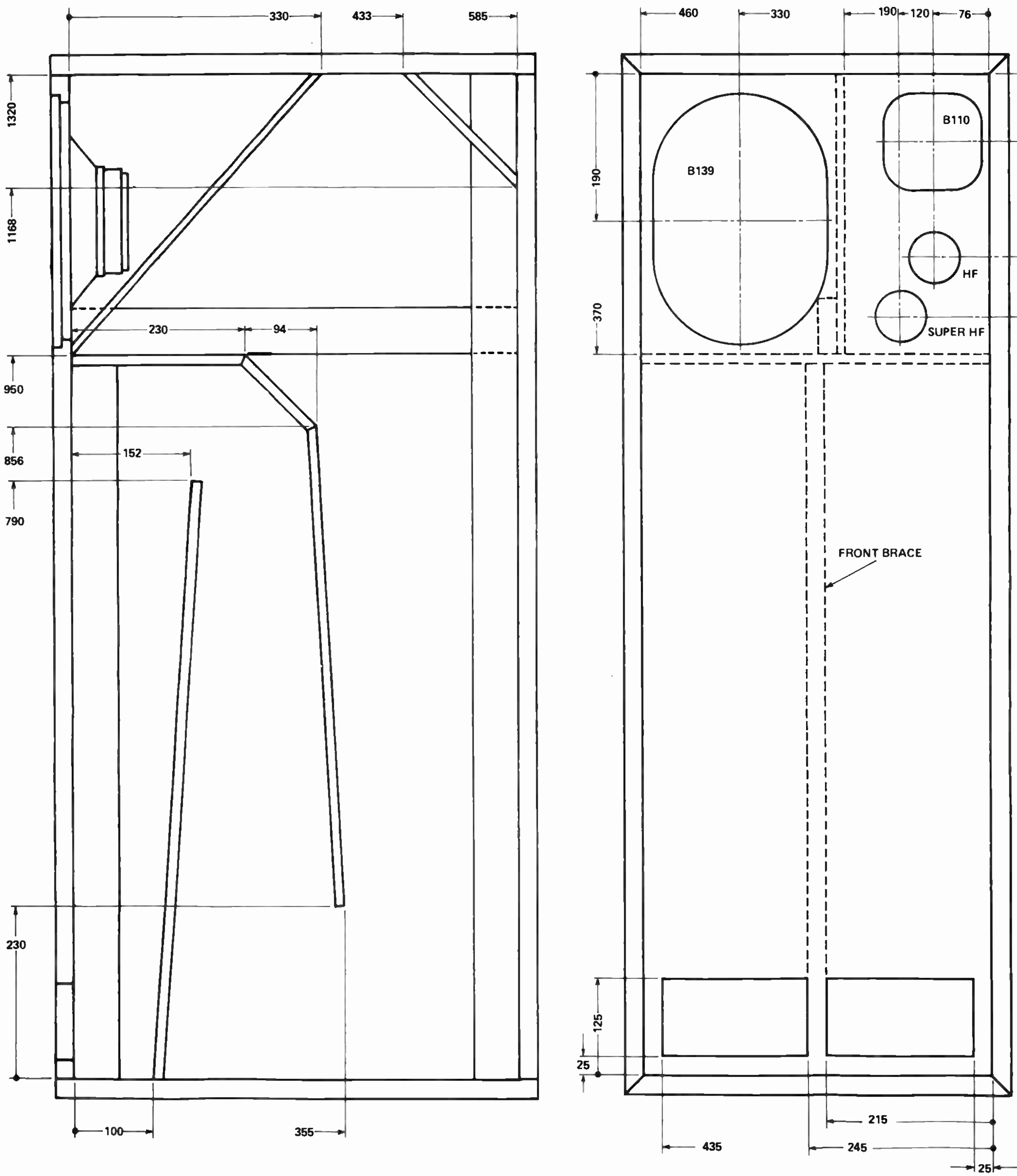
BEFORE BUILDING

Do read this . . .

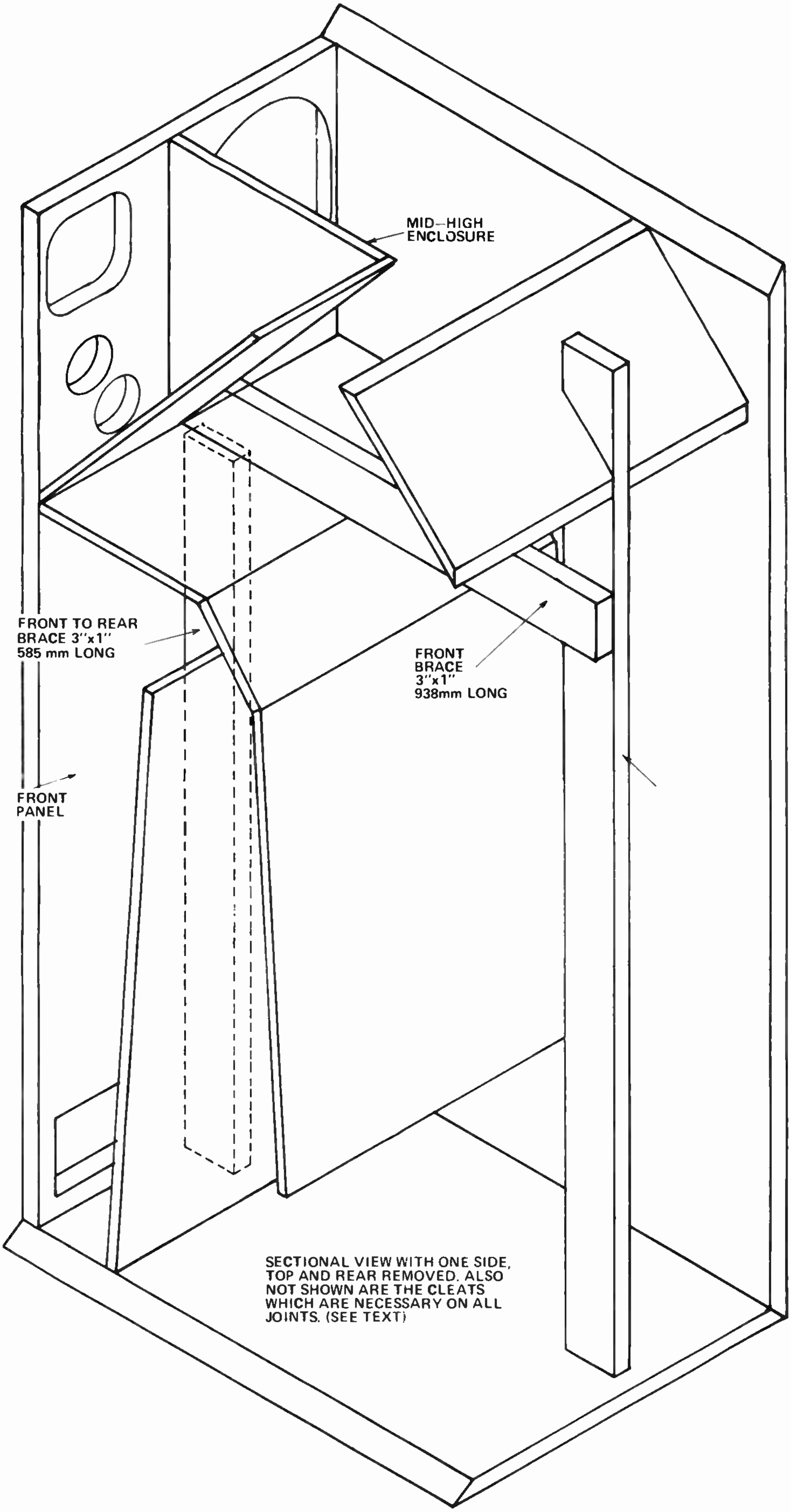
These speakers are costly to build and unless you have a medium to high power amplifier – preferably 75 watts or more – and a turntable and cartridge to match, the benefits of these speakers will not be obtained. They are larger than most and heavy to move around.

But if you accept all this you'll end up with a pair of speakers massively superior to most commercial designs that you could buy for their cost.

A final note: Transmission line loading can't necessarily be called the 'best'. Results from a good transmission line speaker can be almost unbelievably good – but so can the results from the very best reflex units, infinite baffles and horns. The point to bear in mind is not the loading principle itself – rather how it is applied.



Study these drawings in conjunction with the one on the following page. Note that ideally the speakers should be built as a 'mirror-image' pair — that is, so that when placed in the listening room the tweeters should be innermost.



Since those early days the transmission line speaker has to some extent become the only enclosure design seriously considered by many hi-fi people seeking the 'ultimate sound'.

The Operating Principle

The basic principle is simple. It is to load the rear of a bass driver by a tube of 'infinite' length.

For all realistic audio frequencies an adequate compromise is a tube which is one quarter wave-length long at the bass driver's fundamental resonant frequency.

For the drive unit recommended (KEF B139) this tube will be a little over 2.5 metres long and folding it over enables us to produce an enclosure of acceptable size without serious performance compromises.

It's not *quite* that simple for the pipe will produce resonances at its folds and will also have a high-Q resonance at a frequency associated with its length – for the quarter-wave example discussed this will be at about 100 Hz. Some way must be found to 'lose' both the energy causing the resonance and the resonance itself.

There are several ways by which such resonances may be minimized. One is to use many drive units, each having a different fundamental resonance. By careful design it is then possible to cancel out the worst effects of the 'staggered resonances' to give a remarkably smooth response. A form of this principle is used in IMF's ALS-40 which certainly isn't the simple infinite baffle device it appears to be at first sight!

Another method of reducing resonances is to fill the tube with a damping material – and this also increases the effective length of the tube by slowing down the sound travelling within it.

Various materials may be used for this damping. One of the best is long-haired sheep wool; glass-fibre may also be used, it is less effective than wool but tends to be more constant in its physical characteristics.

Resonances caused by the folds in the tube can be minimised by increasing the density of packing material at these points but a far more effective cure is to use a suitable mid-range driver which takes over well below the point at which the lowest resonance frequency occurs.

Sub-audible Noise

If the tube is correctly packed almost all of the sound radiated from the rear of the bass driver's diaphragm will be absorbed. Only those frequencies below the driver's bass resonance will reach the far open end of the tube. But those frequencies which are not absorbed cause problems, because at frequencies below resonance the diaphragm 'sees' a very much smaller load and even low level signals at such very low frequencies will produce large diaphragm excursions.

This sub-audible problem is the major drawback with transmission line speakers: even the quiestest turntables produce some sub-audible noise, and modern amplifiers of the quality and power output required to do justice to the speakers will provide a goodly amount of amplification of that noise. It's also most disconcerting to watch the bass diaphragms of transmission line speakers emulating the swoop of the pick-up arm as it traces a warped record. You may argue that your turntable is quiet, that you have optimized your pick-up arm and cartridge to reduce resonant effects — yet every record carries some sub-audible noise introduced during manufacture of the master by the cutting lathe itself and the cutter head mechanism.

In itself, reproduction of sub-audible noise isn't disastrous — it's too low to be heard. But it does affect reproduction indirectly by effectively restricting diaphragm movement and by creating intermodulation components and attendant harmonic distortion. The first problem is the greater — and you can visualise how the bass unit would 'bottom' if the diaphragm were close to its limit of movement due to a sub-audible noise whilst a high level musical note was simultaneously superimposed.

The cheapest, simplest and most effective cure is to ensure that the sub-audible noise doesn't reach the speaker in the first place. A high-pass or rumble filter, operative below 30 Hz and having a slope of at least 18 dB/octave is very effective.

Many high-power amplifiers are already fitted with the necessary filter network but for those who own units which aren't, a simple and very effective filter design was published in Electronics Today International in October 1974. (Photostats of this design are obtainable from ETI for $1.00.)

Selecting the Drivers

The first step is to select a suitable bass driver. A long-throw device is essential since a properly designed enclosure will maintain constant output down to the lowest audible frequencies — thus even the diaphragms of large drive units will be called upon to make long excursions.

The cross-sectional area of the tube must be equal to or greater than the radiating surface of the drive unit's diaphragm and therefore the size of the bass driver will largely determine the final size of the enclosure.

A suitable bass driver, combining all the required properties including an ultra-rigid low-mass diaphragm, is KEF's B139. This driver has a low fundamental resonance and its radiating surface and throw is sufficient to enable bass fundamentals to be reproduced at adequate listening levels in the home — but it is not so large that the enclosure becomes of unwieldy size. In addition matching mid-range (B110) and treble units (T27) are available from the same manufacturer and these require minima compensation for use with the loaded B139.

Our own reference units use the B139 and B110 but we use Celestion tweeters (HF1300) and super-tweeters (HF2000). These latter require slightly more attention to matching but provide marginally better performance in our own enclosures.

Construction

At this stage then we need a bass driver mounted in an enclosure which is in reality a folded tube stuffed with absorbent material. The effective length of this tube is related to the fundamental resonant frequency of the bass driver and is open at the far end. The tube has a cross-sectional area no smaller than the diaphragm surface of the bass.

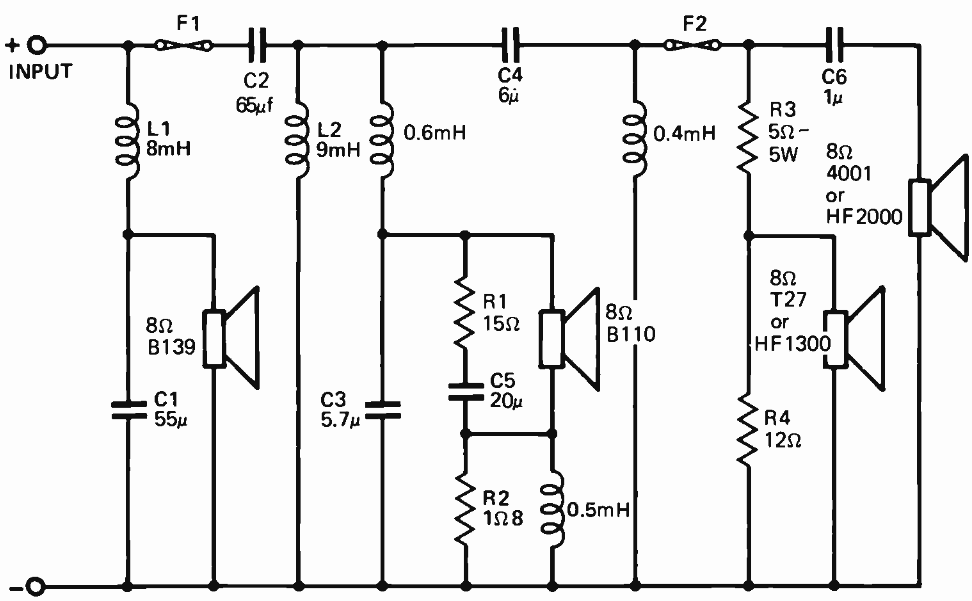
Making the tube isn't that difficult — anyone who's tried to produce a folded norn wouldn't think twice about making a transmission line. Even the legendary Jim Kelly who was once observed repairing a gas chromatograph with a 4lb coal hammer successfully built a pair — and they were magnificent!

Dimensions are not overly critical — except for length which should be within a couple of centimetres of the specified length. The tube should preferably be tapered — so as to reduce or preferably eliminate parallel surfaces and hence standing waves.

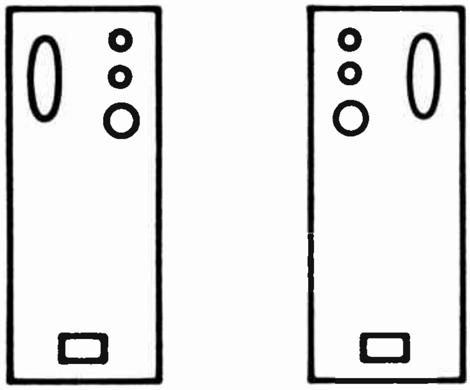
All panels should be cut as accurately as possible, particularly the internal partitions. Check each panel against the job before securing it; an error during the cutting stage could prevent the enclosure from fitting together correctly.

The most suitable material for the enclosure itself is 19 mm particle board, wood veneered preferably, otherwise with plain finish. Pre-veneered board is easier to finish, although ideally it should be mitred at the corners where top and bottom panels join the sides. All joints should be glued using a PBA woodworking adhesive such as Aquadhere, and should be pinned and clamped whilst the glue is setting.

The most secure way to fix the internal partitions is to groove the side panels and cut the partitions slightly oversize to rebate into the grooves. But



Components for the crossover network are both large and expensive. Don't try to economise though, as the design shown is vital for optimum performance. Note that R3 and R4 may be left out if the Celestion HF drive units are used.



Finished enclosures should be located in the listening room such that tweeters are innermost.

this will be beyond the means of most constructors (unless you have access to an understanding cabinet-maker). Cleats, made of offcuts of particle board, or suitable timber, should be provided to give good anchorage. Extra bracing is also an advantage; longitudinal bracing on the 13 mm internal partitions is worthwhile.

Our own units were built by first attaching top and bottom panels to one of the sides. The partitions and midrange sub-enclosure were then added, followed by the rear panel to which connecting terminals and a fuse, mounted on a laminated plastic panel, had already been glued. Wiring was also added at this stage. Next came the front panel — of plywood since this offers greater strength when apertures for drive units have been cut.

All drive units should fit flush; if a Celestion HF 1300 is used, this is designed to be fitted from inside and not from the front. The front panel apertures should therefore be rebated out to accept the drive unit fixing flanges. This involves some rather fiddly chisel work unless you have access to a router.

Once the five sides, partitions and midrange enclosure are in position, the drive units can be mounted and wired to the crossover, which can be placed either on the inside face of the rear panel or on the platform behind the bass unit. The latter position is probably best since it gives access to the network via the bass unit aperture — far easier to remove than the remaining side panel, which should, ultimately, be glued in position once the enclosure is complete.

Drive units originally chosen for our own units were the B139 for bass, KEF B110 for midrange, KEF T27 tweeter and STC 4001K (8 ohm). Later, the T27 and STC were swopped for Celestion HF1300 and HF2000 respectively, (available from the Australian distributor, M&G Hoskins). However, the latest version of the KEF T27 is a vast improvement over the earlier model, and for economy this driver could be used without use of a super-tweeter. The HF1300 exhibits roll-off above 15 kHz and should therefore always be allied to a super-tweeter. Eight ohm versions of both HF1300 and HF2000 should be used, and the drive unit positions indicated on the plan should be adopted, since correct phase relationships are preserved using the crossover network shown.

Our units were filled with fibreglass material — the slab type, not the rolls. This can be secured using suitable pins, or alternatively on small dowels inserted through holes in the partitions and subsequently glued and sealed. It is *essential that all joints* are fully airtight otherwise the enclosures will fail to work correctly. The fibreglass should fill all the available space in the 'line' yet should not be compressed. Density may be increased slightly at bends in the tube. Final adjustment is best done by careful listening and experimenting with packing density. That's why the remaining side panel should be secured by screws. Gaskets should be used to ensure the enclosure is sealed.

Long-fibre sheeps' wool (Dr. Bailey's long hair!) can be used, although this is more difficult to work with and may settle after a period of time, with a consequent change in performance. Bonded acetate fibre such as Innerbond may be used, tightly packed — although not overtightly — in the midrange enclosure. It should not be used in the bass section.

When the enclosures are correctly packed with fibreglass, bass performance should be smooth and extended, with no obvious constriction or colouration. However, *there may be an apparent lack of bass energy by comparison with many speakers*, although fundamentals will be clearly defined and 'tight' sounding.

Transmission line speakers accurately reproduce the bass that is in the original programme material. No more — and very little less. They don't manufacture bass in the form of resonances.

Our crossover network is based on air-cored coils supplied by Transcap (Orchard Road, Brookvale, NSW). All capacitors are paper or polyester, the 55 and 65 microfarad values being made up of oil-filled paper fluorescent lighting ballasts from Plessey Ducon.

Values for R3 and R4 can be altered to achieve correct balance between midrange and treble, and these values actually depend on the drive units chosen. These resistors might best be left out completely if Celestion HF drive units are used.

Fuse protection may be considered necessary if high levels are envisaged — 3 amp fusing should be adequate. The tweeters can be protected separately by a 1 amp fuse. Fuseholders should be fitted in some accessible position such as adjacent to the input terminals.

Our prototypes were used for a long period of time as a high-quality reference speaker system for evaluating the subjective performance of hi-fi equipment and assessment by comparison or other loudspeakers. They were used in mirror-image form, with the enclosures positioned so that each array of tweeters was innermost. This provides optimum stereo performance, since the main axis of each speaker projects into the room, away from boundary walls. Reduction of reflected sound by this means was found to provide a less anomolous stereo image.

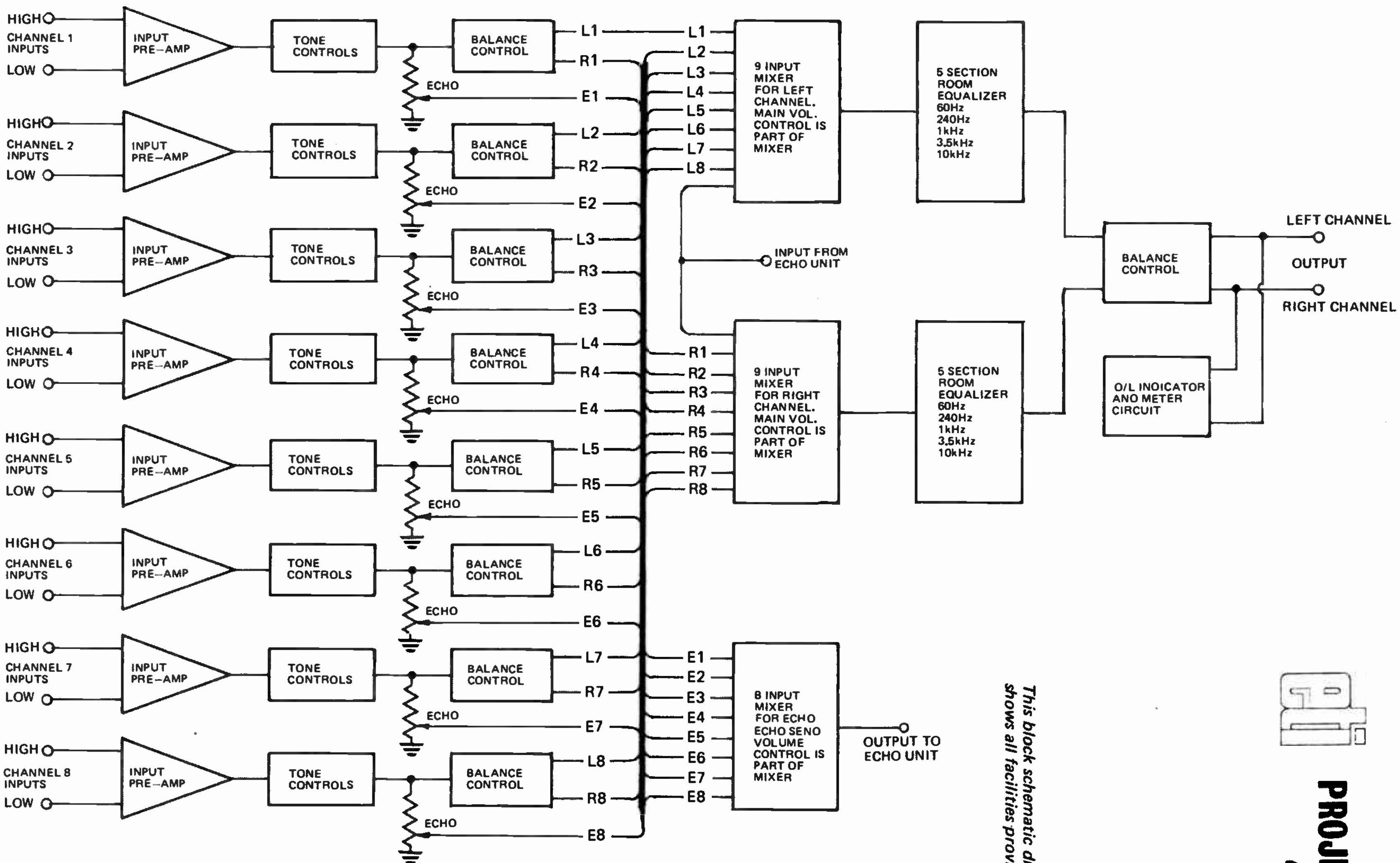
Reticulated foam is recommended for the grilles since this causes less colouration than frame/fabric grilles. The prototypes used open grilles constructed of aluminium channel, and these proved aesthetically pleasing and sonically satisfactory.

Once all internal adjustments have been made, the detachable side panels may be secured and sealed in position. It would be advisable, however, to leave these panels removeable in case access to the interiors is necessary in the future.

Overall, the systems as described performed admirably and despite their size, were found to take up little effective space due to their tall, tower-like format. *(Continued on page 83).*

ETI MASTER-I

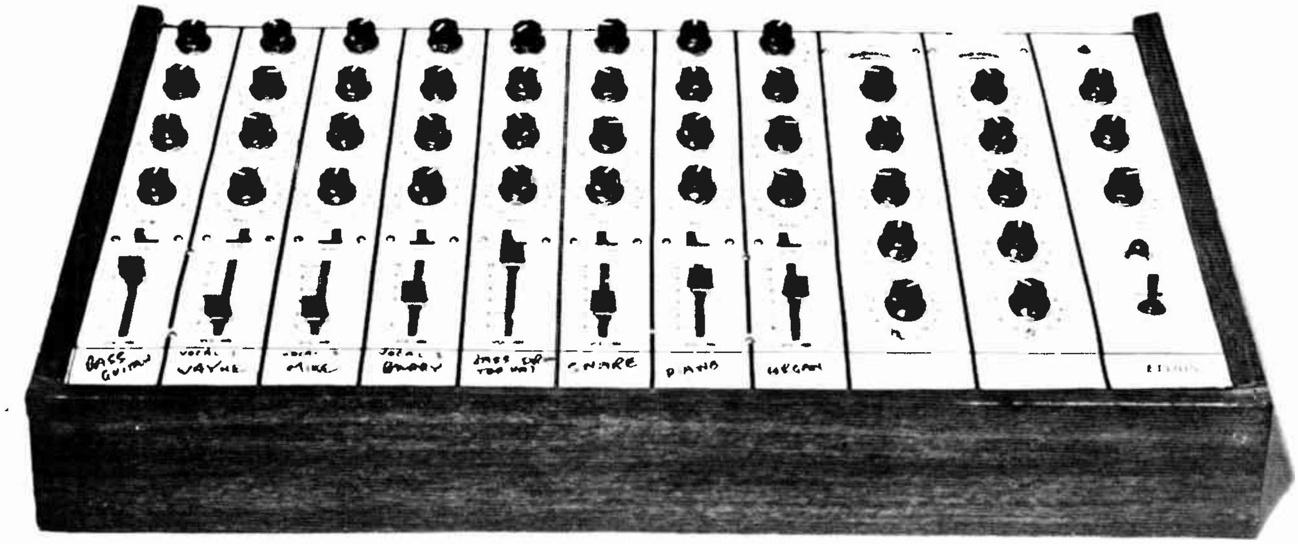
PROJECT 414



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.

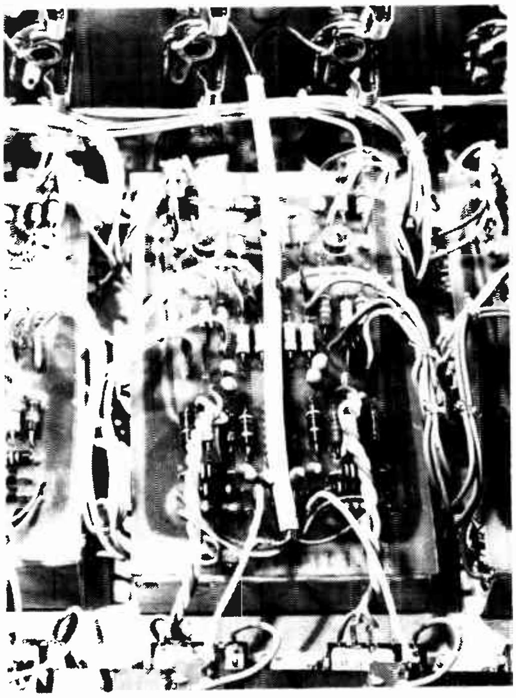
OUTPUTS

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.

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.

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.

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

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

Equalizer

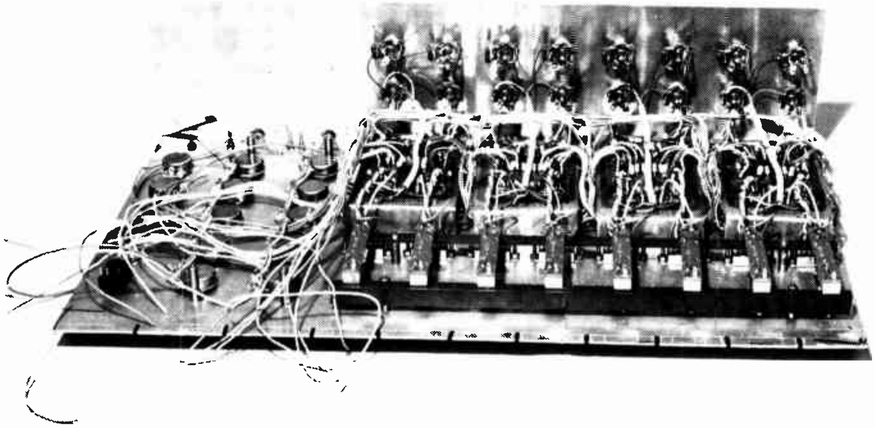
After the preamplifier boards are assembled, we can assemble the main mixer/equalizer boards of which there are two. The winding data for the inductors associated with this section is given in Table 1.

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.

ETI MASTER-MIXER



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

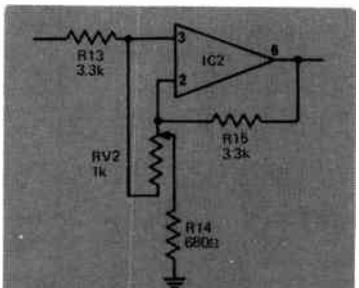


Fig. 2a – Equivalent circuit of the equalizer with potentiometer set for maximum boost at the resonant frequency of the network.

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

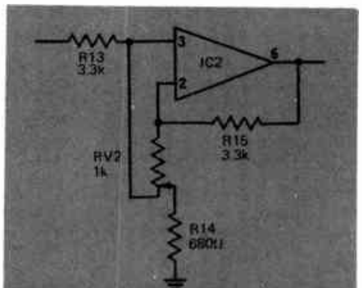


Fig. 2b – Equivalent circuit of the equalizer with the potentiometer set for maximum cut at the resonant frequency of the network.

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

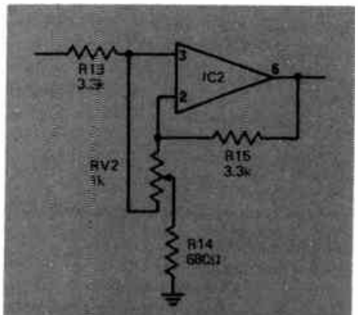
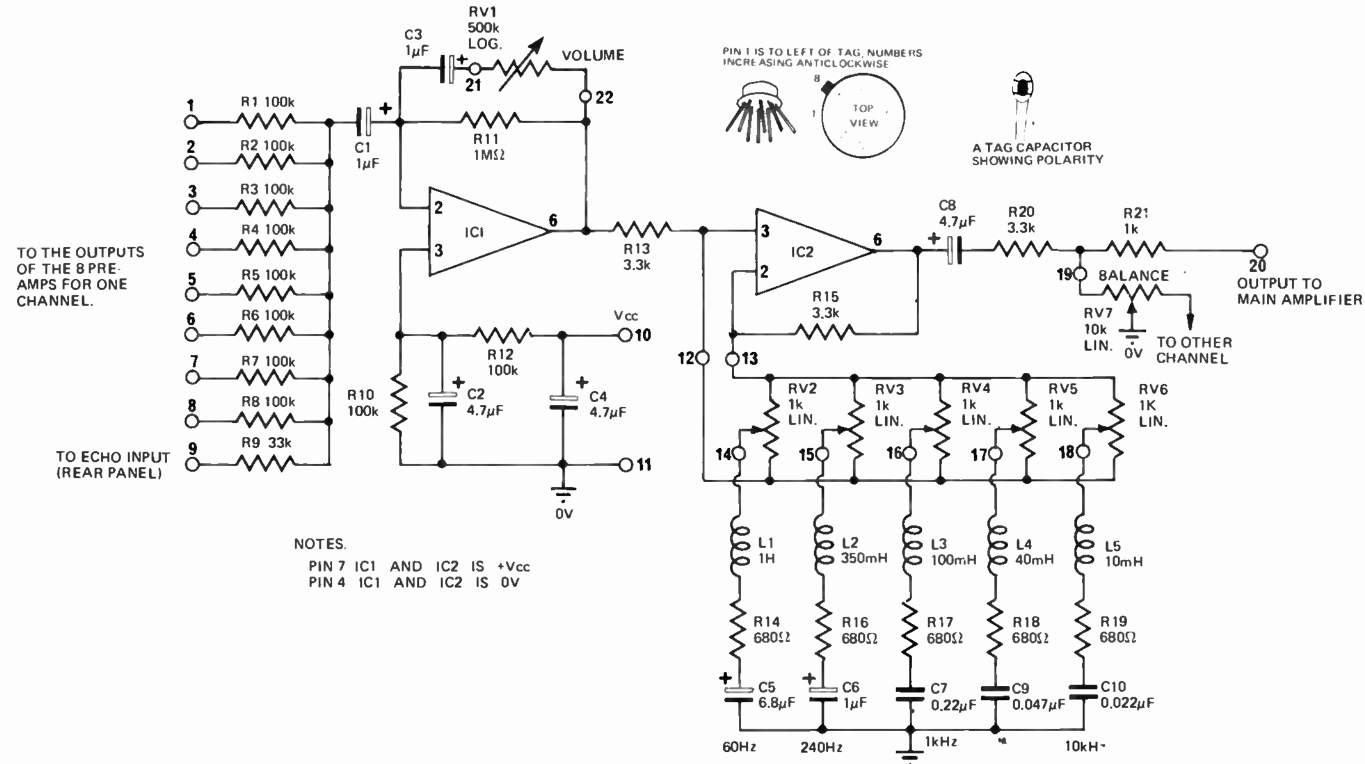


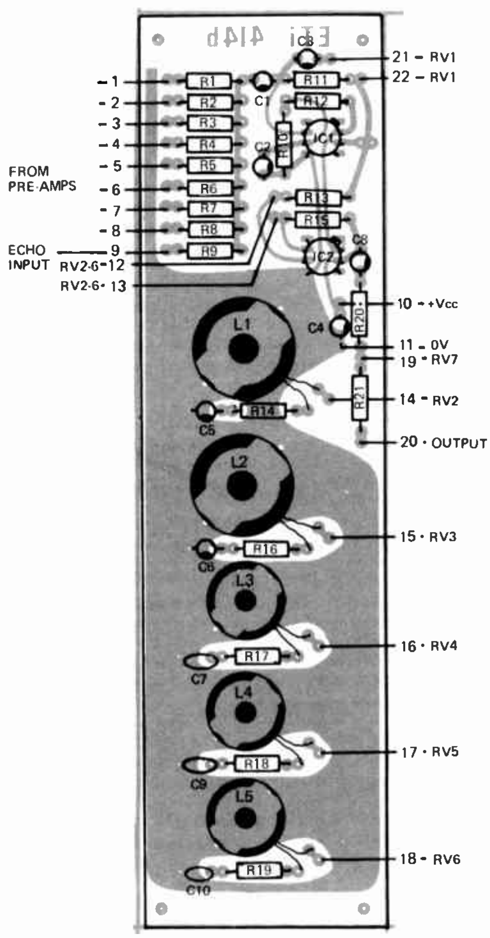
Fig 2c – Equivalent circuit of the equalizer with the potentiometer set for unity gain regardless of frequency.

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.



0V

The circuit diagram of the main mixer and equalizer boards.

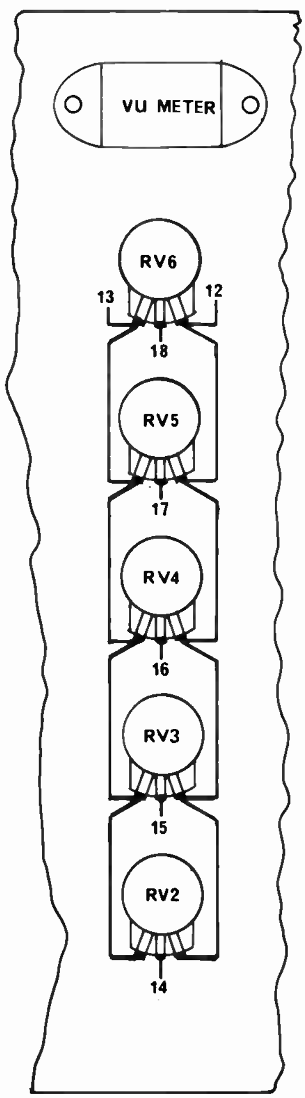


Component overlay for main mixer and equalizer.

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

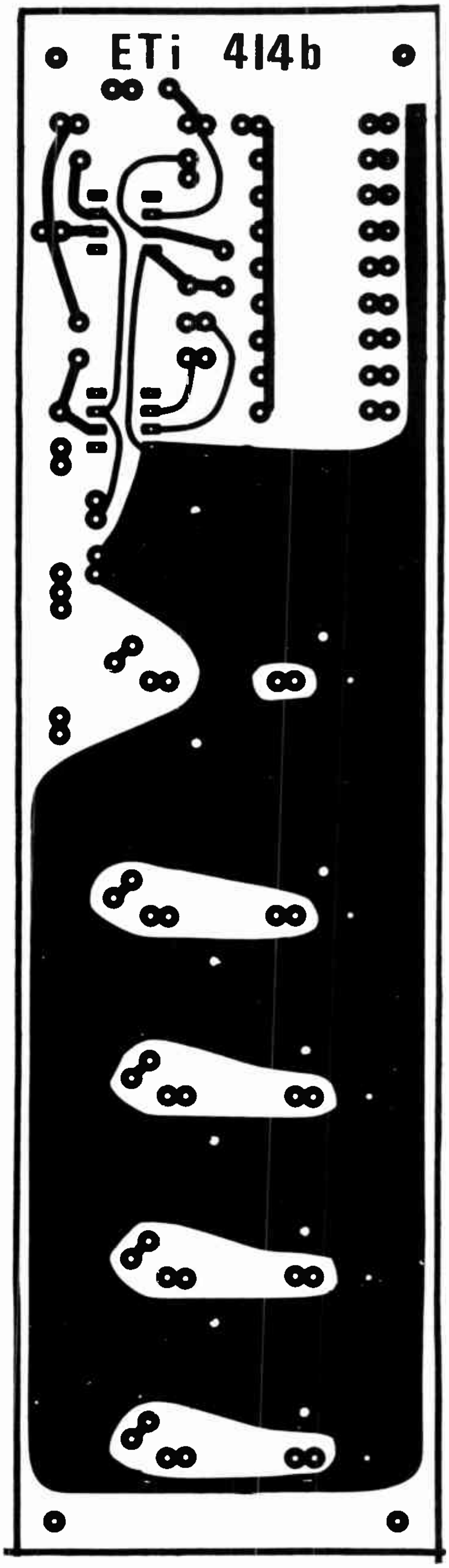
R1	resistor	100 k	5%	½ watt
R2	"	100 k	"	"
R3	"	100 k	"	"
R4	"	100 k	"	"
R5	"	100 k	"	"
R6	"	100 k	"	"
R7	"	100 k	"	"
R8	"	100 k	"	"
R9	"	33 k	"	"
R10	"	100 k	"	"
R11	"	1 M	"	"
R12		100 k	"	"
R13	"	3.3 k	"	"
R14	"	680 ohm	"	"
R15	"	3.3 k	"	"
R16	"	680 ohm	"	"
R17	"	680 ohm	"	"
R18	"	680 ohm	"	"
R19	"	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.

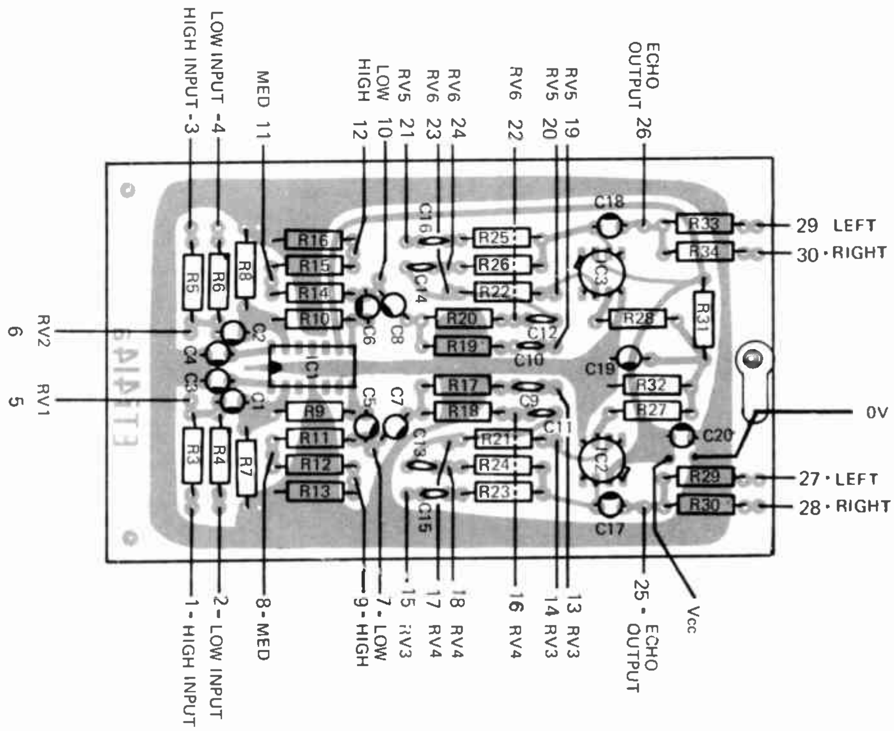
ETI MASTER MIXER



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, IC1 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

$R = (4700 \times Zin)/(4700 - Zin)$

where Zin is the desired input impedance.

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.

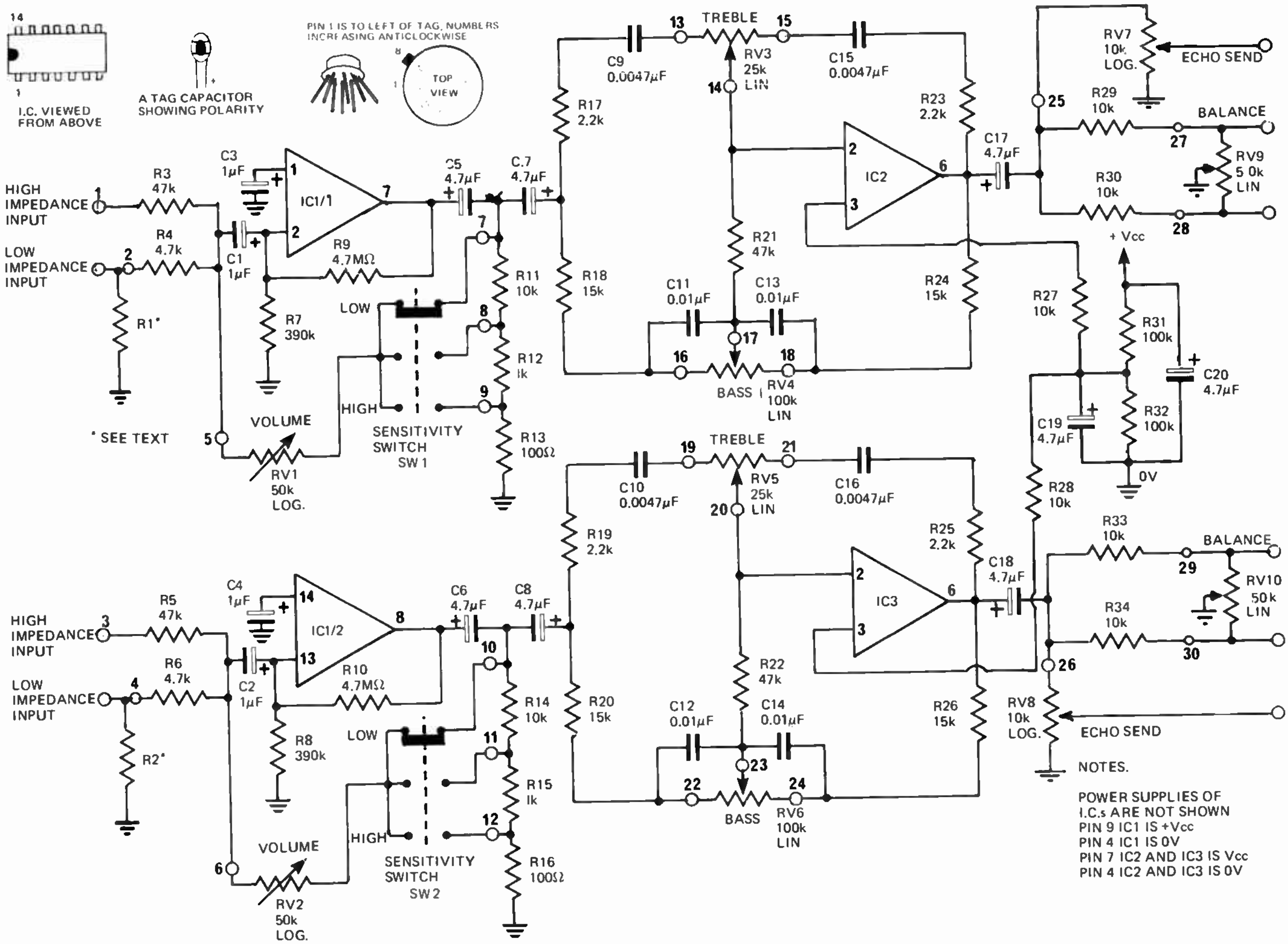
R1*	Resistor	200ohm	5%	½W
R2*	"	200ohm	"	"
R3	"	47k	"	"
R4	"	4.7k	"	"
R5	"	47k	"	"
R6	"	4.7k	"	"
R7	"	390k	"	"
R8	"	390k	"	"
R9	"	4.7M	"	"
R10	"	4.7M	"	"
R11	"	10k	"	"
R12	"	1k	"	"
R13	"	100ohm	"	"
R14	"	10k	"	"
R15	"	1k	"	"
R16	"	100ohm	"	"
R17	"	2.2k	"	"
R18	"	15k	"	"
R19	"	2.2k	"	"
R20	"	15k	"	"
R21	"	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

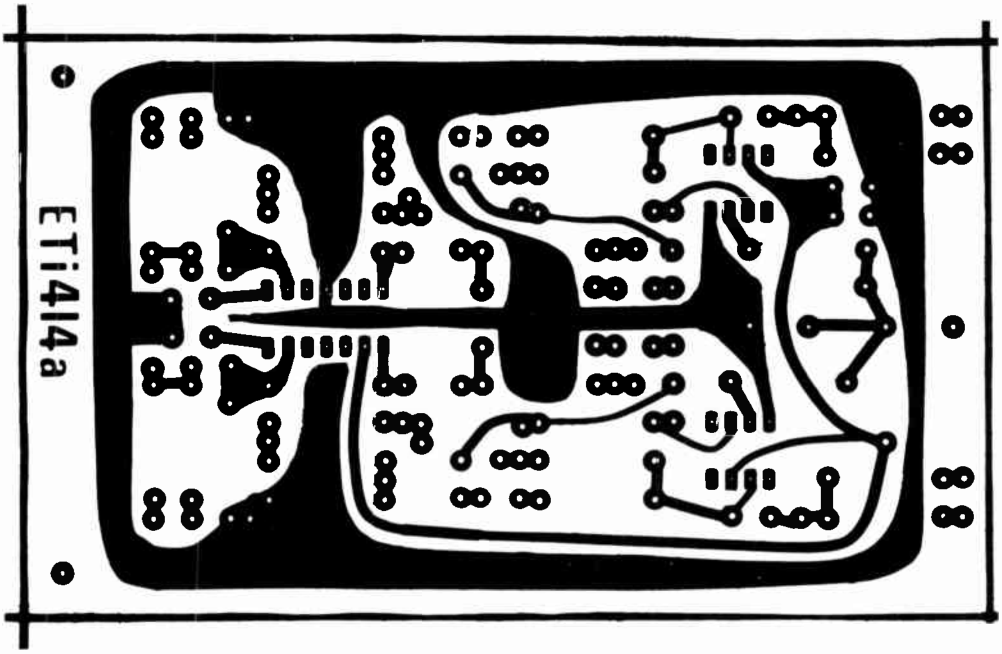
IC1	Integrated circuit	LM381
IC2	" "	uA741, LM307 (metal can or mini dip only)
IC3	" "	uA741, LM307 (metal can or mini dip only)

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.

ETI MASTER MIXER



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

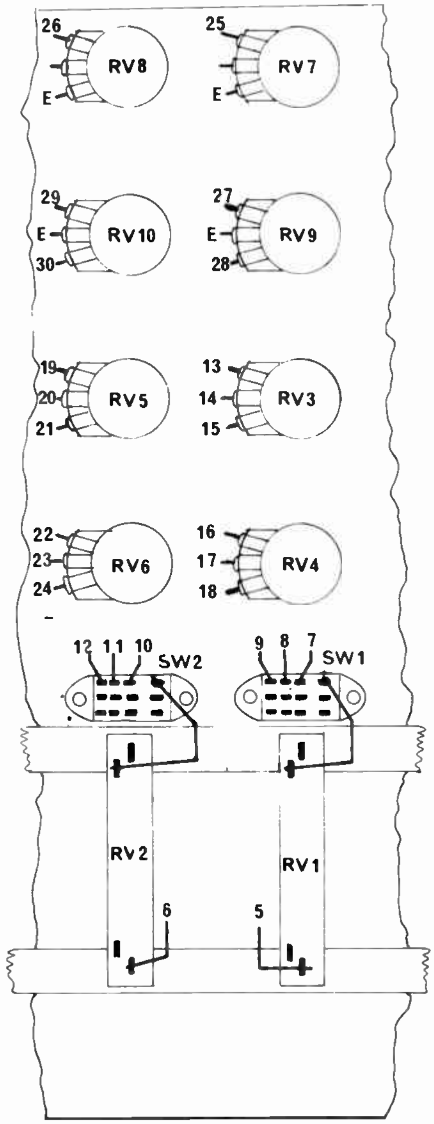


Fig. 1. This diagram shows connections to the potentiometers and sensitivity switches associated with the preamplifier boards. The numbers corresponds to those on the pre-amplifier circuit and overlay diagrams.

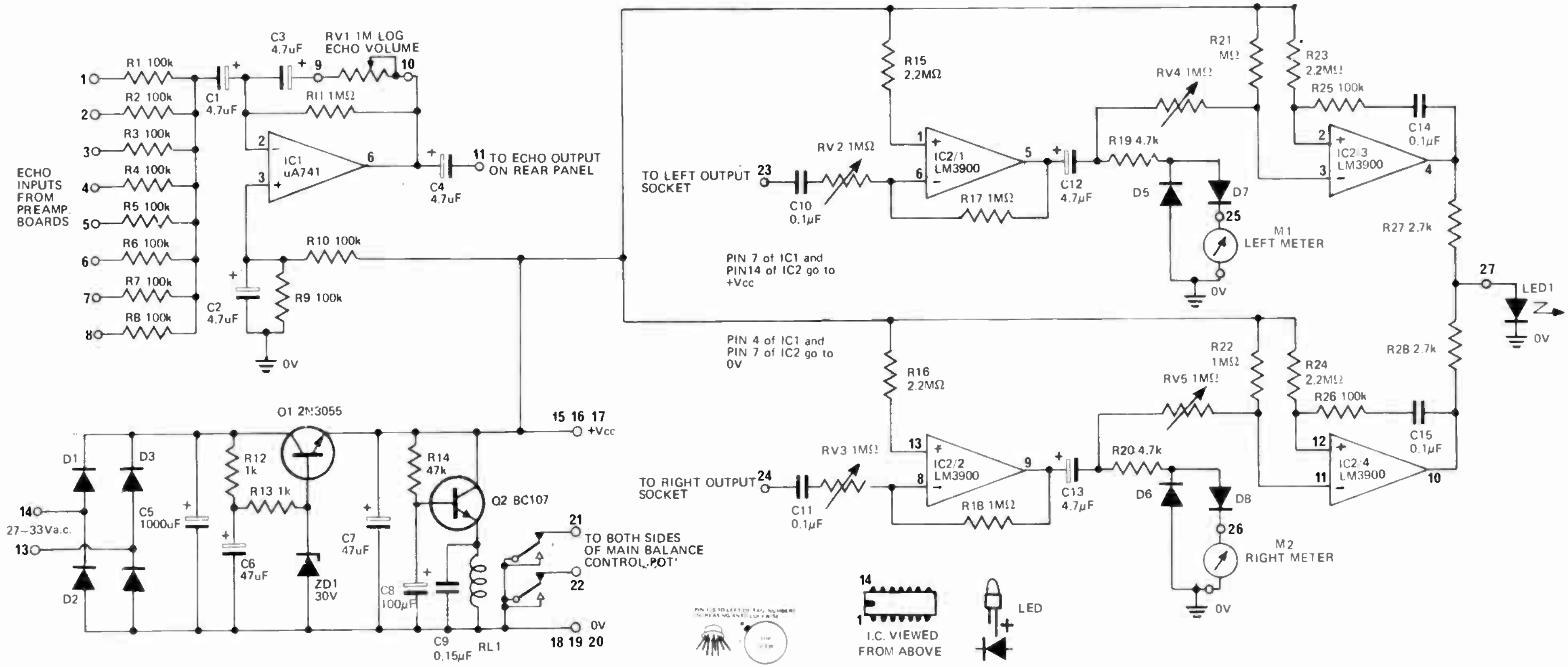


Fig. 1. Circuit diagram of power supply and metering board

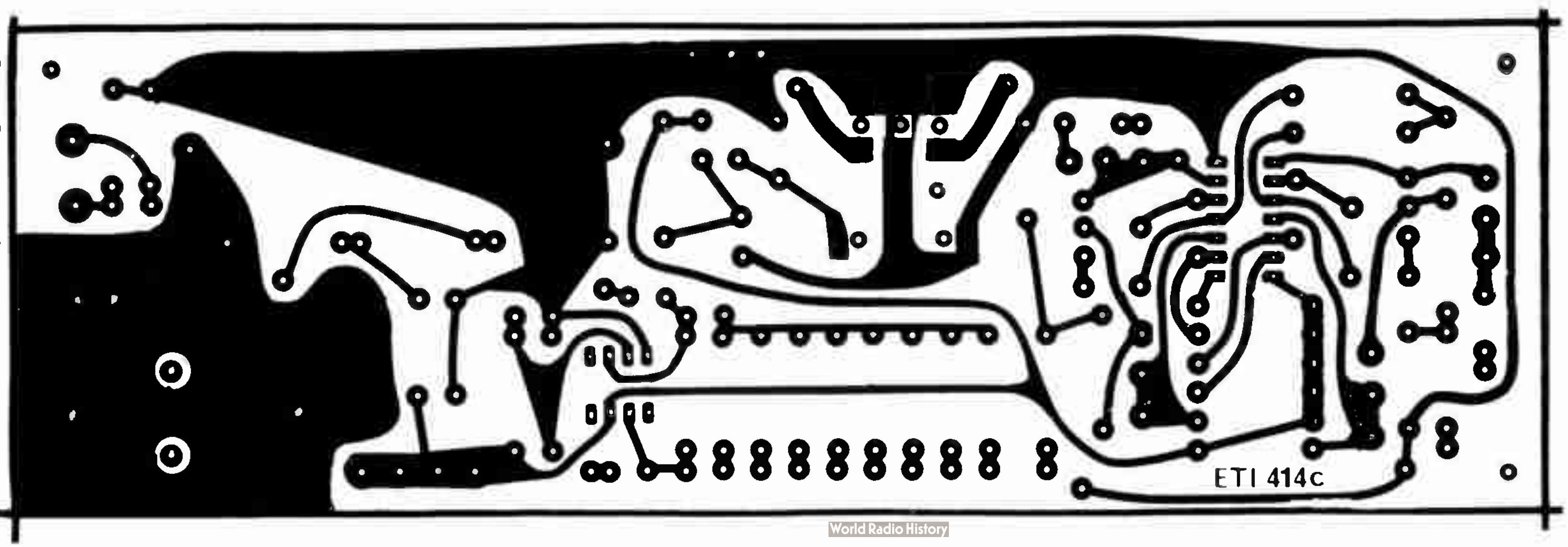


Fig. 2. Foil pattern for power supply board

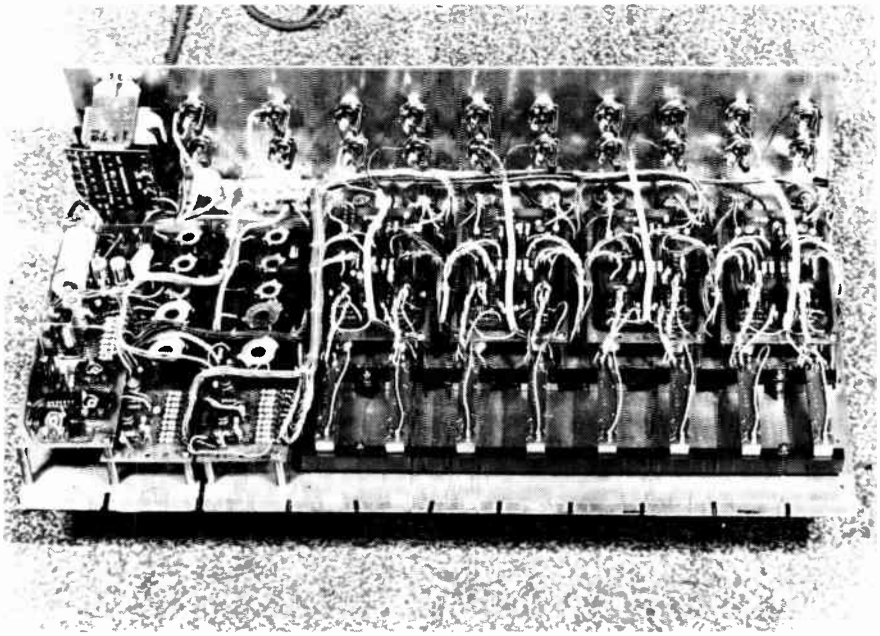
Assembly and operation of the power supply – overload indicator and metering circuits, and final assembly details are provided in this third article in the series.

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 74).

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.



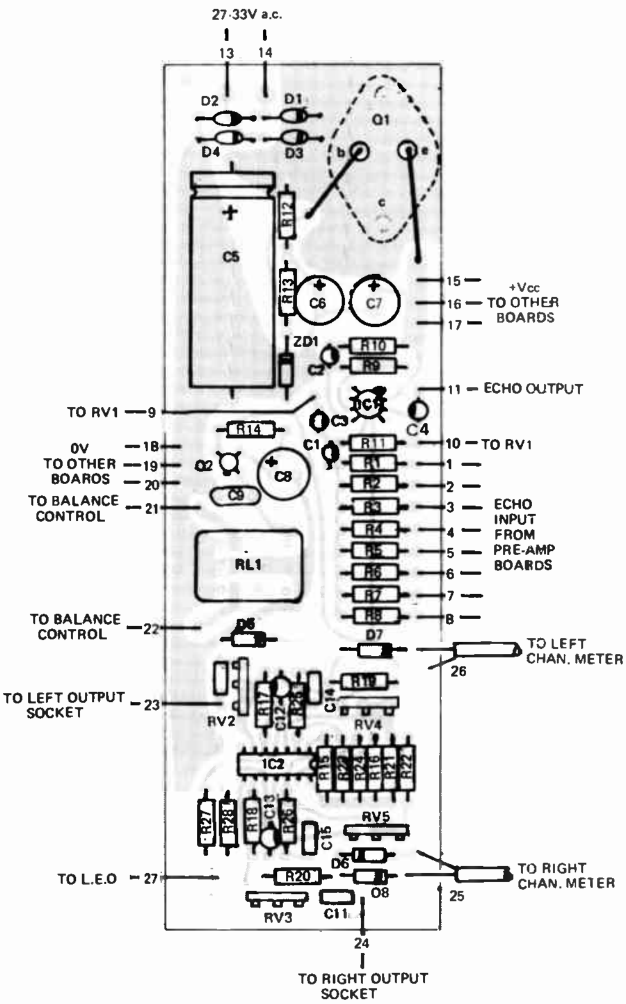


Fig. 3. Component overlay of power supply board.

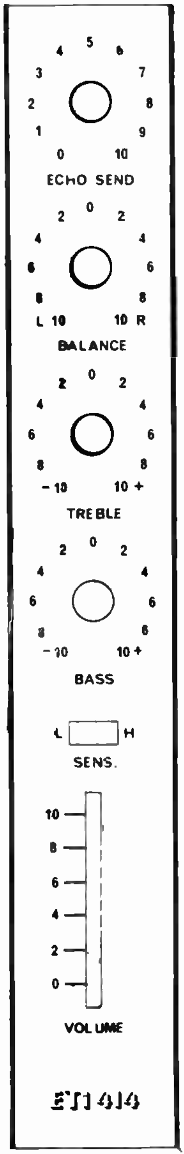


Fig. 4. Escutcheon for preamplifier (actual size 12" x 1¾").

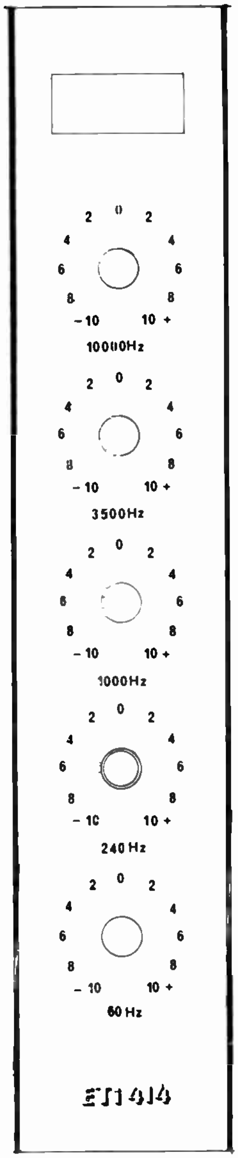


Fig. 5. Equalizer panel escutcheon (actual size 12" x 2¼")

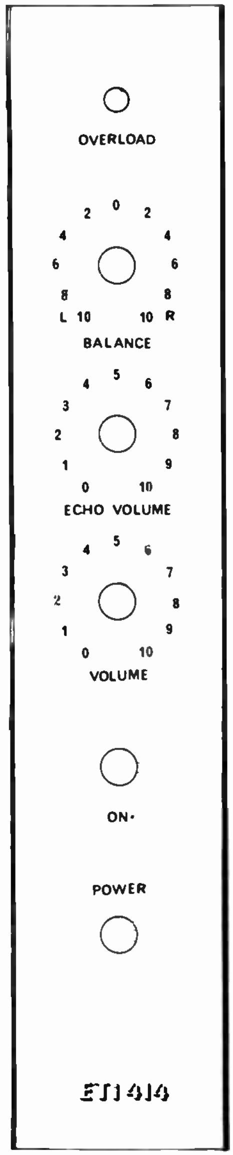


Fig. 6. Main control panel escutcheon. (actual size to be 12" x 2¼").

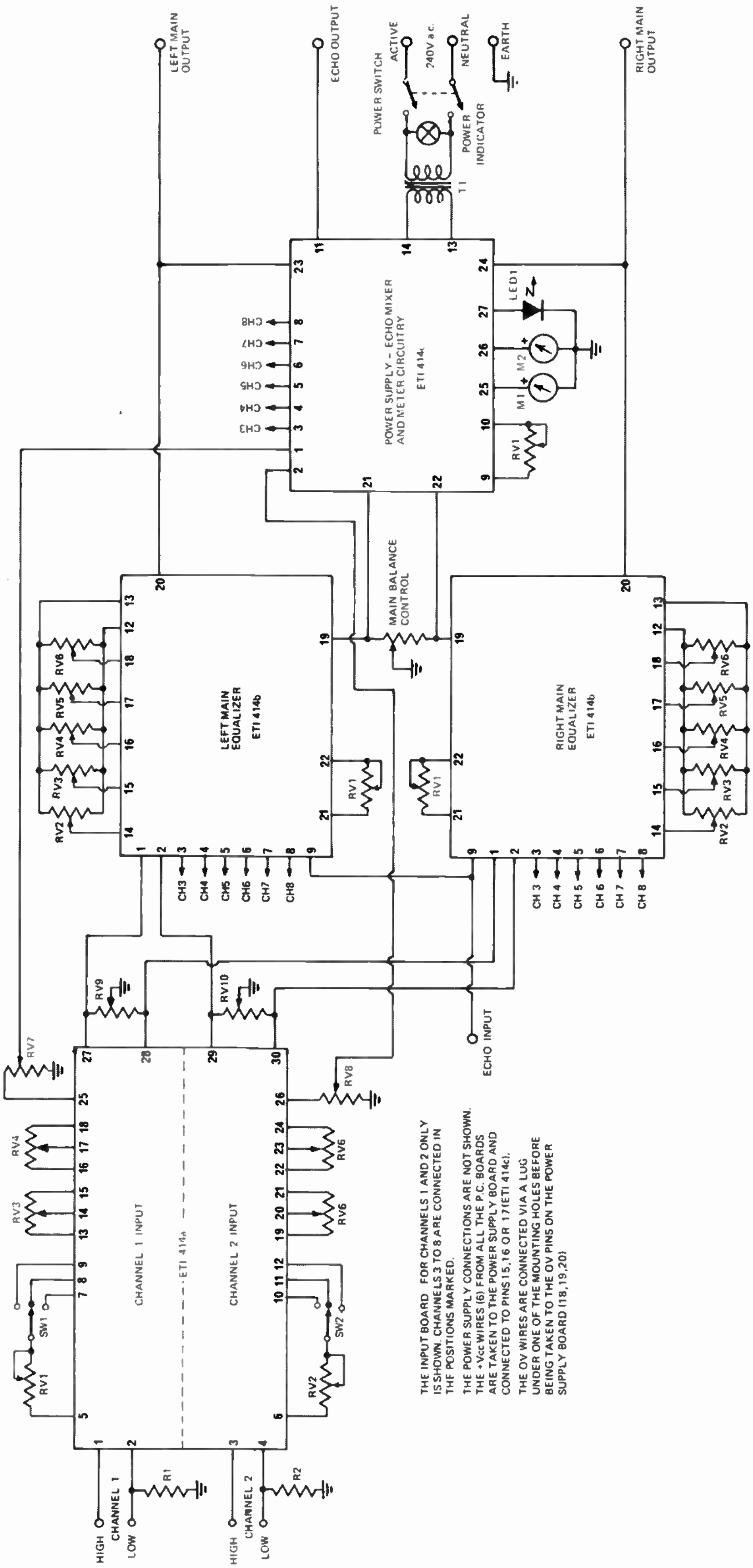


Fig. 8. Main interconnection diagram.

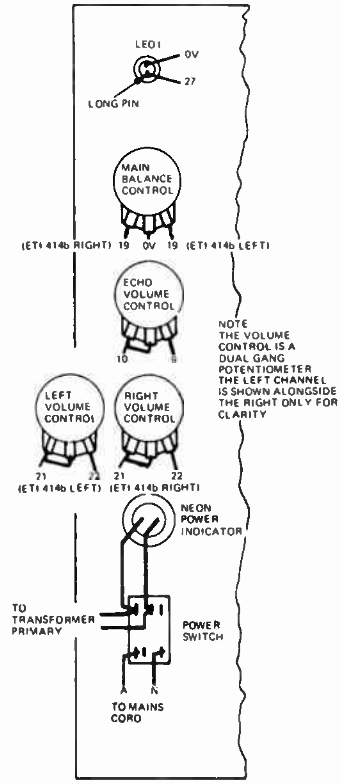


Fig. 7. Wiring to rear of main control panel.

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

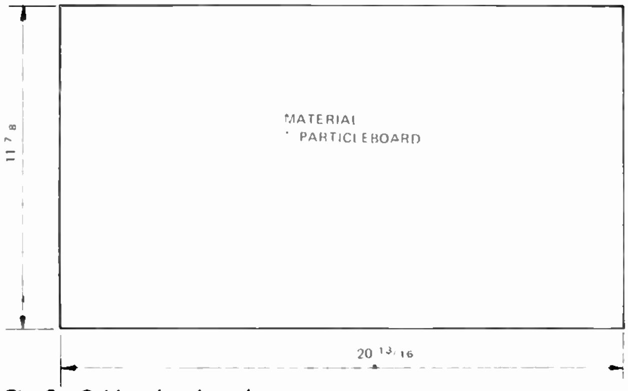


Fig. 9a. Cabinet baseboard.

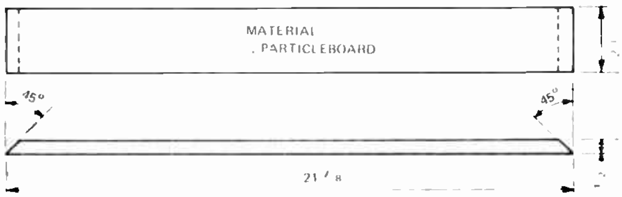


Fig. 9b. Cabinet front

Fig. 9c. Front panel support

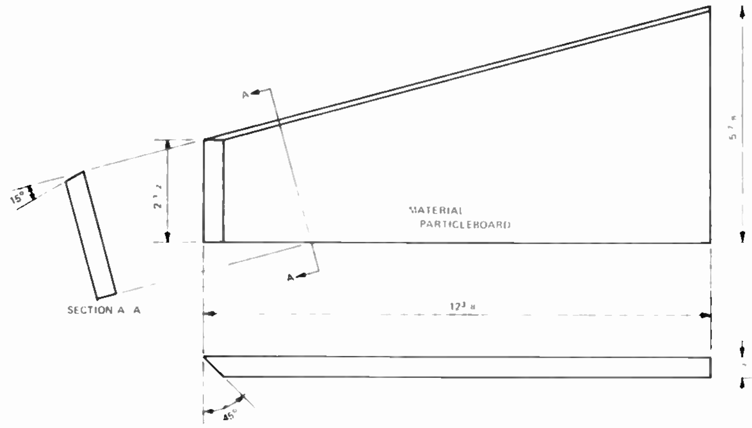


Fig. 9d. Cabinet sides — two required. Note that boards should be mirror image of each other, that is, chamfers should have opposite slope.

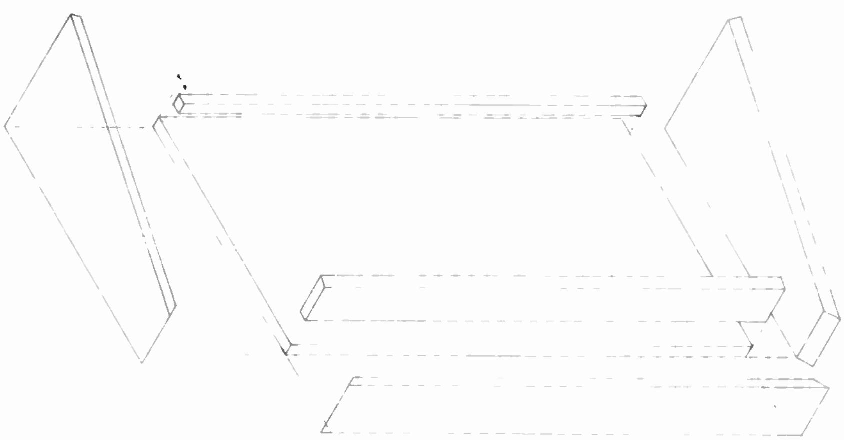


Fig. 10. Cabinet assembly details.

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.

WOOOWORK

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).

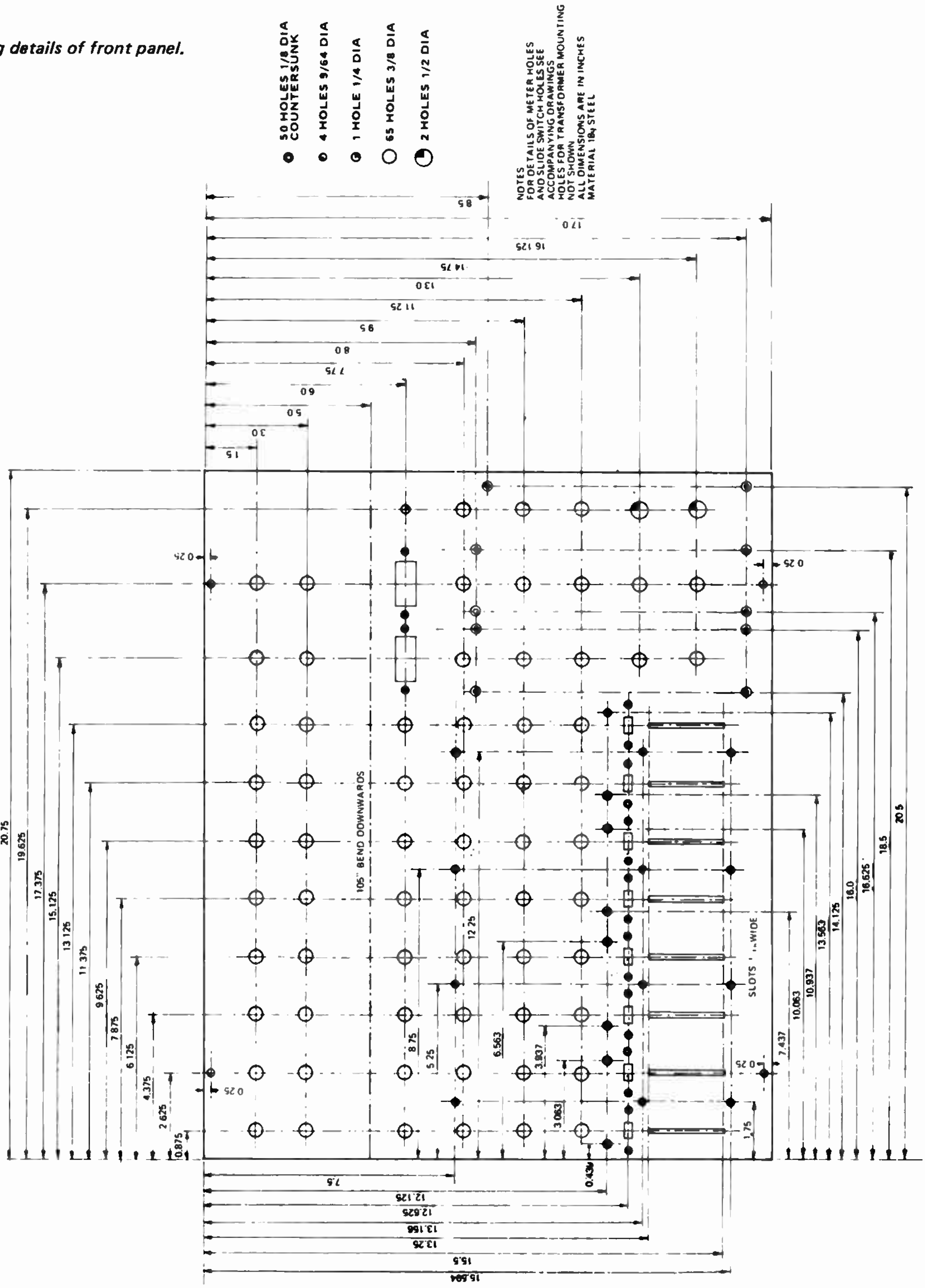
Assemble the box as per Fig. 10. Screws or nails should be used to hold the panels together while the glue sets. Take care to ensure that the sides are square to the base, otherwise the metal panel may not fit in place. In fact it is a good idea to use the panel as an assembly guide. The support piece (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).

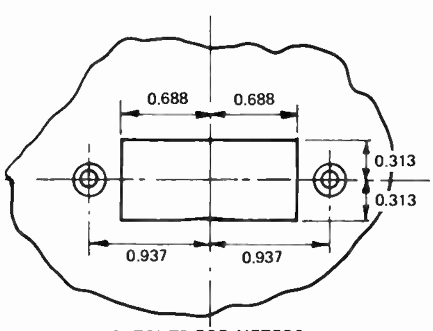
When the glue is set, the box can be sanded and all visible outside surfaces veneered, before final sanding and finishing operations are carried out. The inside of the box should be lined with "Alfoil", and this earthed to the metal chassis. If the Alfoil goes over the rear panel support, the metal panel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

TESTS AND AOJUSTMENTS

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

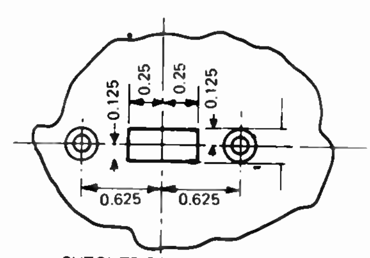
Fig. 11a. Drilling details of front panel.





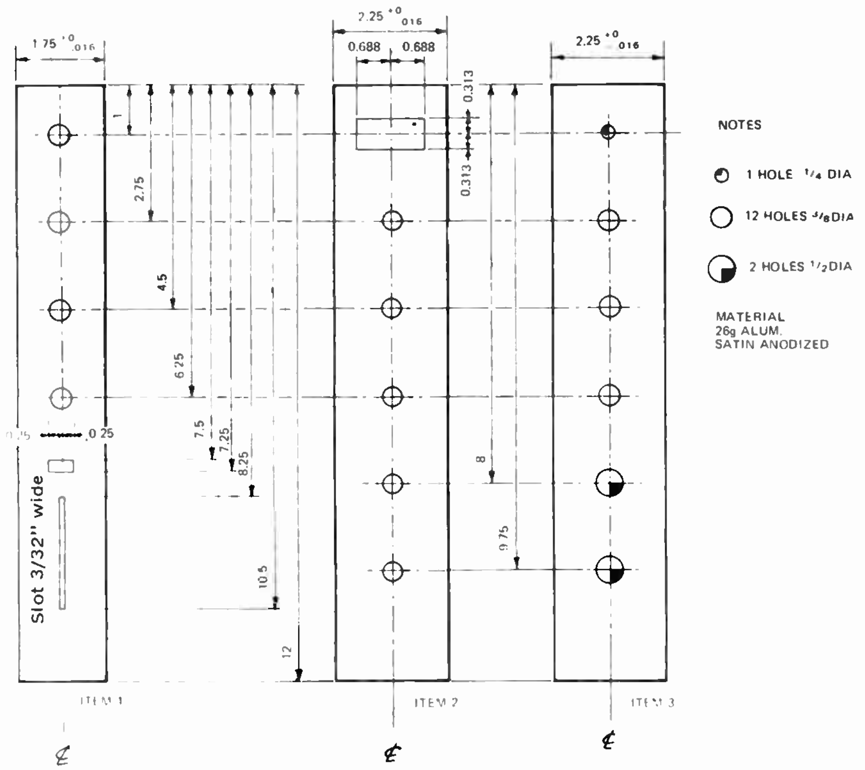
CUTOUTS FOR METERS

Fig. 11. Cutouts for meters and sensitivity switches.



CUTOUTS FOR SENS. 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.

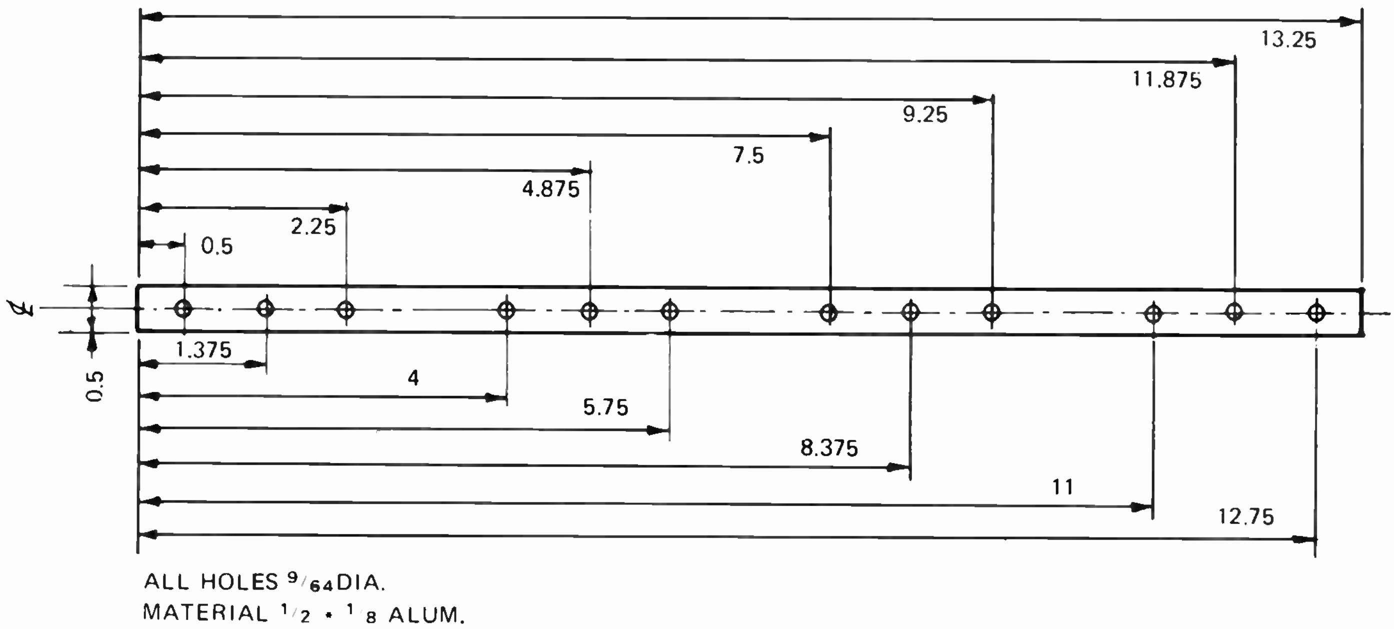


Fig. 13. Slide potentiometer support bars (two required)

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 ETI413 amplifier this level should be approx 1V rms.

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

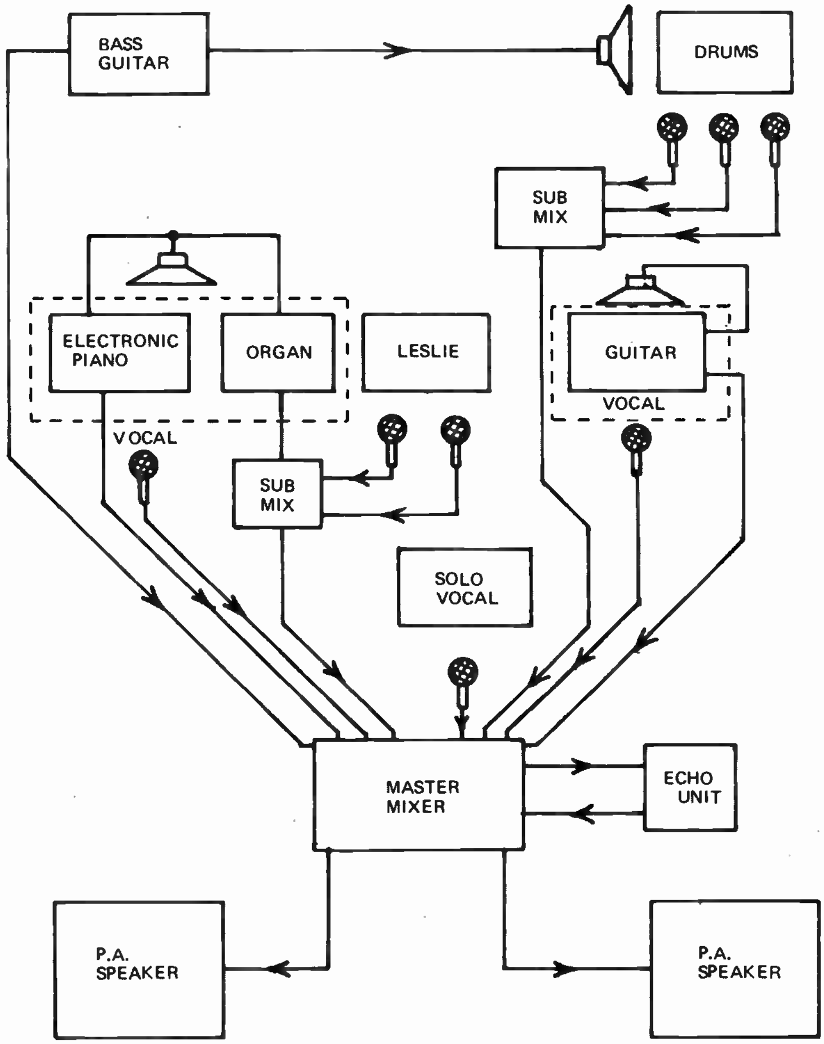


Fig. 1. Arrangement of average-sized group

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

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 72, we see that the input amplfiier ICI 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 71 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 mu t 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 74) 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

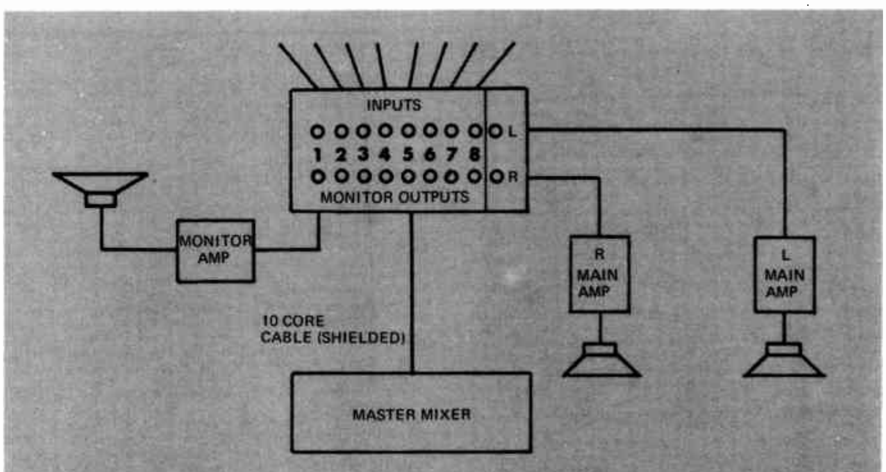


Fig. 2. A separate monitor amplifier may be used if desired.

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 70. 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 70) 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 70) 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! ●

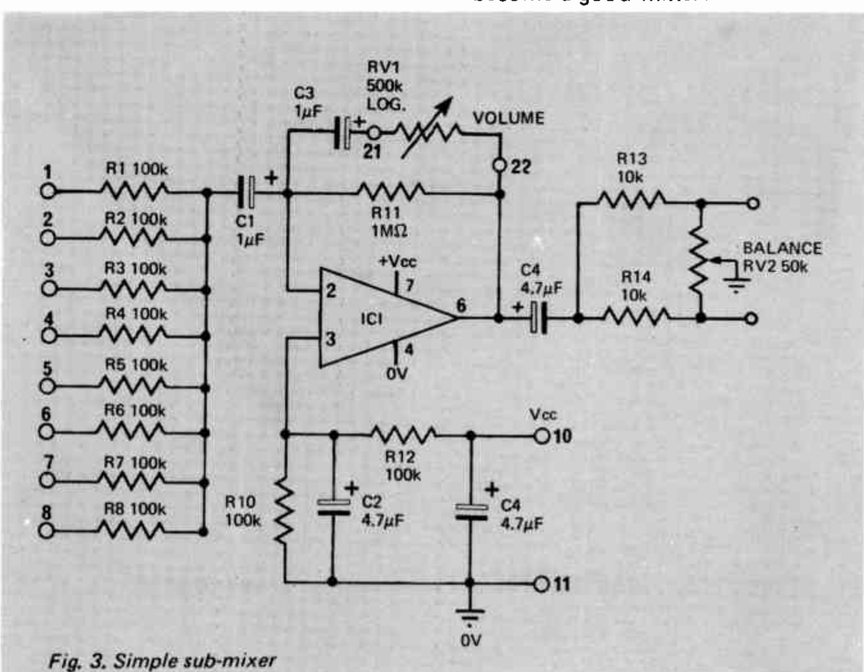


Fig. 3. Simple sub-mixer

(continued from page 65).

The transmission line speaker project was originally published in ETI August '77 — and proved tremendously popular with readers — virtually all who built them were very pleased indeed with the results.

The original article also triggered off a flood of enquiries from readers seeking advice on various aspects of performance etc, the most typical of these enquiries are answered here.

Non-polarised capacitors can be obtained from Plessey Ducon and also in values sufficient to make up the total values required — from Audioson, Winbourne Rd, Brookvale, NSW — who also supply KEF drivers.

It's been found advisable to connect small value polyester capacitors across the main non-polarised units to allow a 'passage' for small musical details. A microfarad or so will suffice for C2 and a few picofarads for C4 and C6.

The HF1300 and HF2000 drivers were readily obtainable from M & G Hoskins of Kent St, Sydney at the time of writing.

The crossover network shown has turnover frequencies of approximately 400 Hz, 3.5 kHz and 12 kHz. The extra network associated with the midrange driver reinforces output below 600 Hz to compensate for losses below this.

Component values should be adhered to as closely as possible although the 1.8 ohm resistor is not critical — but keep it within 1.5 - 2.2 ohms. This resistor can readily be made from a short length of jug element wound around a former — such as another higher value resistor. All resistors should preferably be ten watt rating although five watt types will do at a pinch for the tweeters.

As for the actual construction and stuffing, it is essential that all panels are accurately cut and fitted together to give an airtight seal the length of the 'line' and between the main enclosure and midrange sub-enclosure. As we mentioned in the previous article, it is very difficult to quantify the precise amount of acoustic fibreglass needed to fill the line, but the best way to estimate requirements will be to calculate the volume required to fill the enclosure without compressing the material, then add another 15% so that a greater density of stuffing can be placed near folds in the tube.

With care and attention in construction these speakers can provide excellent results, with truly musical bass. But if you like a lot of bass (most commercial speakers provide this; it's pretty boomy too) the design may disappoint you. On the other hand, once accustomed to the very accurate bass these speakers provide you'll probably find it difficult to listen to most others unless they happen to be very good infinite baffles or, of course, good transmission lines.

HOWL-ROUND STABILIZER

Feedback problem in halls can be corrected by the use of this clever gadget.

ANYONE WHO HAS USED a microphone in public address work has come across problems with feedback. These are caused by the level of sound reaching the microphone from the speaker approaching or exceeding that from the person originating the sound. As the reflected sound approaches the level of the original signal, the sound becomes distorted or 'coloured', then audible ringing occurs and finally complete oscillation or howl-round occurs as the reflected sound exceeds the level of the original signal.

The most effective method of eliminating this problem in most cases is to use the correct location for the speakers and the correct choice of microphone. Also the use of the microphone is important so if you are in charge of a sound system don't be afraid to tell the singer or speaker how to use the microphone as a good performer will take advice.

However in certain environments the most effective use and selection of microphone/speakers does not help the problem of feedback. These are the halls and rooms which have little sound-absorbing material on the walls and are very 'live'. If a frequency response curve is drawn for such a room it will be found that there are many peaks and troughs, normally only 4 or 5 Hz apart, along with perhaps major resonances.

The printed circuit board layout for this project is on page 90.

Solutions

There are various electronic devices which have been developed to deal with this problem, the main ones being the graphic equalizer, the variable notch filter and the frequency shifter. The first two (especially the notch filter) are ideal for eliminating major resonances. These however also alter the frequency response of the original sound. They can also help if the offending 'echo' is actually a direct path and not dependent on the room (i.e. if the speakers are behind the microphone). The other method, frequency shifting, is described here.

With a frequency shifter the echo signal is of slightly different frequency on each path round the loop and cannot directly reinforce itself so that while on the first echo it may strike a room resonance the second time it will probably be in a null. This tends to even out the frequency response of the room and allows 5 to 8 dB higher levels to be used in the average room. Also the onset of howl-round is not as dramatic as with the conventional system and the distortion which normally occurs below the howl-round level is not as noticeable. The system does not however do a great deal for howl-round not associated with room resonances.

Only a small shift is normally required and it does not matter if it is an increase or a decrease. We chose to increase the frequency by about 5 Hz as it is easier to tell if a vocalist is flat rather than sharp. As the frequency response of the unit is good it is suitable for vocal work as well as general public address use. The frequency shift and the slight amplitude modulation cannot be detected by most people.

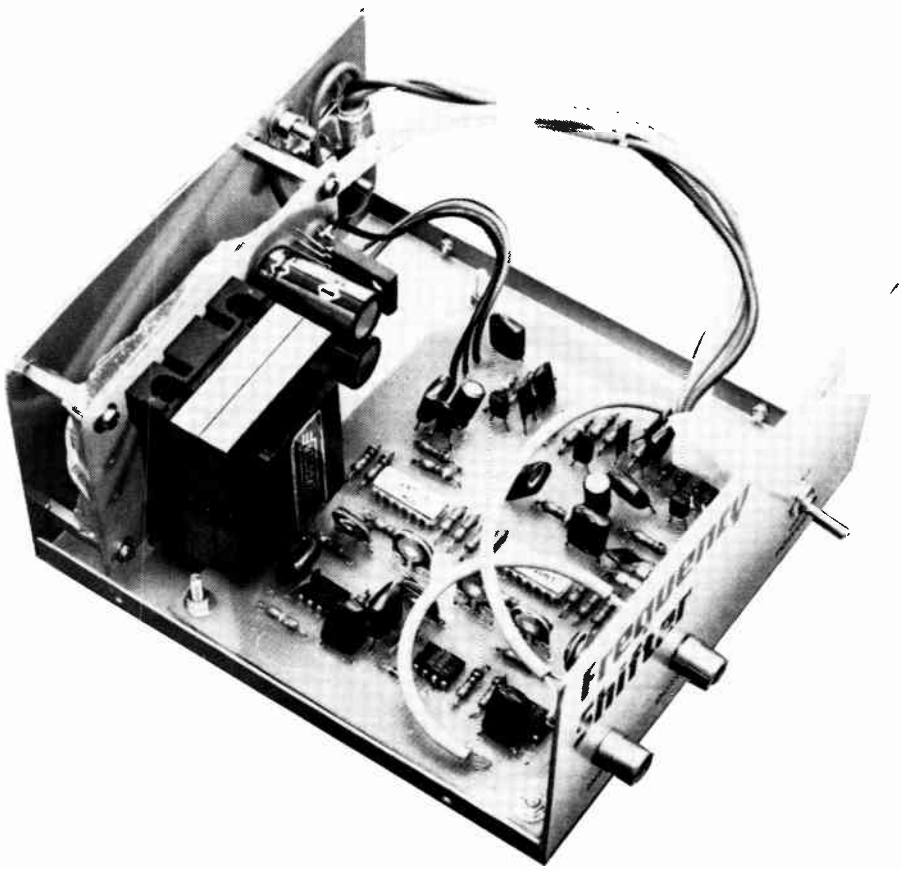
Alignment

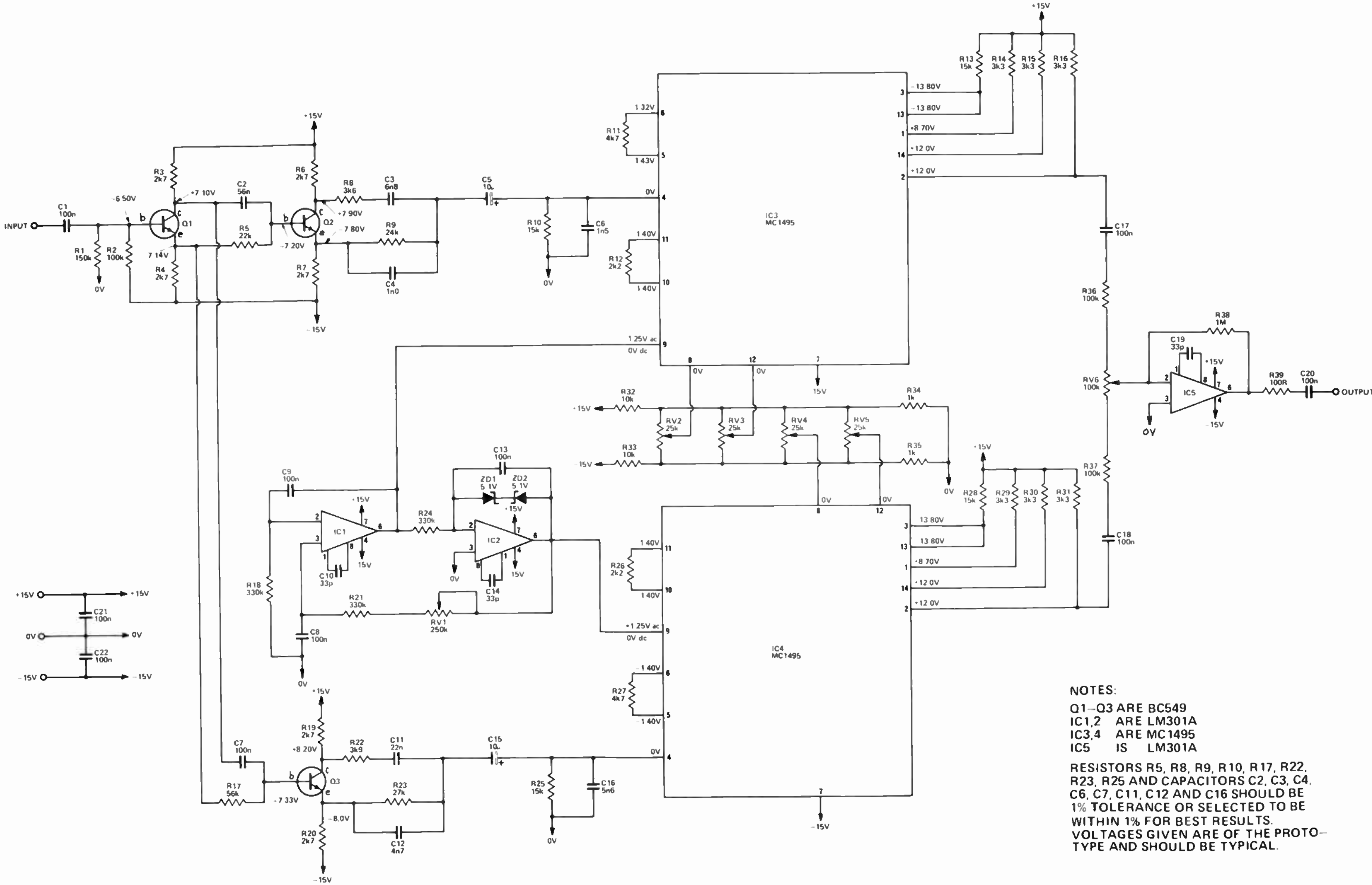
Equipment needed — a sensitive AC voltmeter (100 mV or less) or preferably an oscilloscope and an audio oscillator.

1. Check the output of the 5 Hz oscillator and adjust RV1 until it stops. If it cannot be completely stopped, try a link across C9.
2. Apply a signal of about 1 – 2 V amplitude at about 1 kHz to the input and measure the output of IC3 at pin 2. (If your meter does not reject DC, measure at the junction of C17 and R36). Adjust RV3 to give the minimum output.
3. Measure the output of IC4, pin 2 (or the junction of C18 and R37) and adjust RV5 for minimum output.
4. Measure the output of the 5 Hz oscillator on pin 6 of IC1 and adjust RV1 until it starts, then adjust to give about 1.25 V RMS.
5. With no input signal, measure the output of IC3 (or the junction...) and adjust RV2 for minimum output.
6. Measure the output of IC4 (or...) and adjust RV4 for minimum output.
7. If an oscilloscope is available, monitor the output with a 1 – 2 V input signal and adjust RV6 to give the minimum amplitude modulation. Alternatively, by using an amplifier and speaker, RV5 can be adjusted by ear. The unit is now set up.

SPECIFICATION — ETI 486

Frequency shift	5kHz upwards
Maximium input voltage	3V
Frequency response +½ dB, −3dB	30Hz — 20kHz
Signal to noise ration re 3V output	70 dB
Distortion @ 1kHz, 2V out	0.25%
Amplitude modulation	100Hz — 10kHz < 1dB
Phase shift network 50Hz — 20kHz	$90^{o} \pm 5^{o}$





NOTES:

Q1–Q3 ARE BC549
IC1,2 ARE LM301A
IC3,4 ARE MC1495
IC5 IS LM301A

RESISTORS R5, R8, R9, R10, R17, R22, R23, R25 AND CAPACITORS C2, C3, C4, C6, C7, C11, C12 AND C16 SHOULD BE 1% TOLERANCE OR SELECTED TO BE WITHIN 1% FOR BEST RESULTS.
VOLTAGES GIVEN ARE OF THE PROTOTYPE AND SHOULD BE TYPICAL.

Fig. 1. The circuit diagram of the phase shifter. For the power supply see ETI Project 581 – page 89.

HOW IT WORKS – ETI 486

There are numerous methods of generating a frequency shift in an audio signal. Most however require coils and precise tuning which rules them out for a project. With this method only resistors and capacitors have to be accurate, yet it gives a result adequate for the purpose.

The audio input is split into two circuits which provide a frequency-related phase shift as shown in Fig. 4. The amplitude however remains constant. Due to the different component values in the two networks the phase shifts are not the same but differ by 90° at all frequencies (50 Hz – 20 kHz +/– 5°).

IC1 and IC2 form a quadrature sine wave oscillator with the frequency set by R18, R21, R24, C8, C9 and C13. Amplitude stability is provided by ZD1 and ZD2 along with RV1 (see adjustment section). The outputs from these two op amps are the same amplitude but 90° phase shifted.

We now multiply (the MC1495 is a four-quadrant multiplier) one of the audio signals by one of the 5 Hz outputs and the second audio input by the second 5 Hz signal. When we multiply two waveforms together the output consists of the sum of the two frequencies and their difference. This means that if the audio signal is 100 Hz the output will contain a 95 Hz signal and a 105 Hz signal. These will beat with each other to produce a 10 Hz beat note as shown in Fig. 2. Due to the phase shift between the inputs of the multipliers the 105 Hz components of the outputs are in phase, while the 95 Hz components are 180° out of phase. Therefore by adding the outputs of the two multipliers in IC5 the 95 Hz components cancel out, leaving only the 105 Hz signal. Provided the multiplier inputs have the 90° phase relationship there will always be a 5 Hz shift, independent of frequency.

Due to the inability to maintain exactly the 90° phase relationship, the 95 Hz, or lower sideband, will not completely cancel and the result is a slight beat giving rise to an amplitude modulation effect (we had about 1 dB). This is not normally noticeable on speech or music.

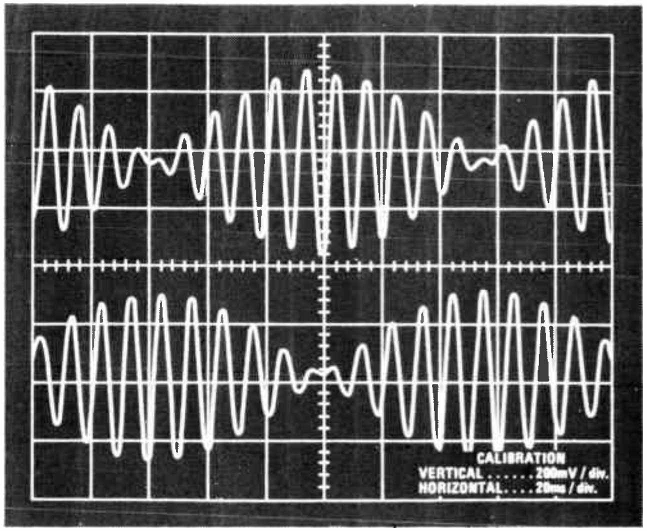


Fig. 2. The output of IC3 (top) and IC4 (lower) with a 100 Hz input signal. Note the phase difference.

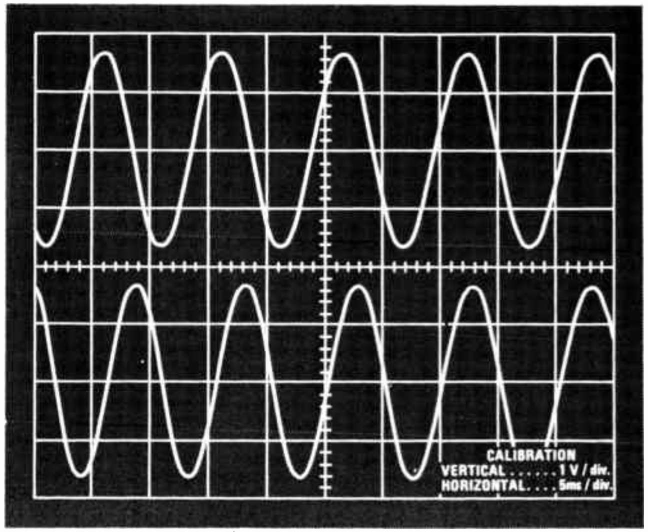


Fig. 3. The input signal (top) and the output (lower). Note the difference in frequency.

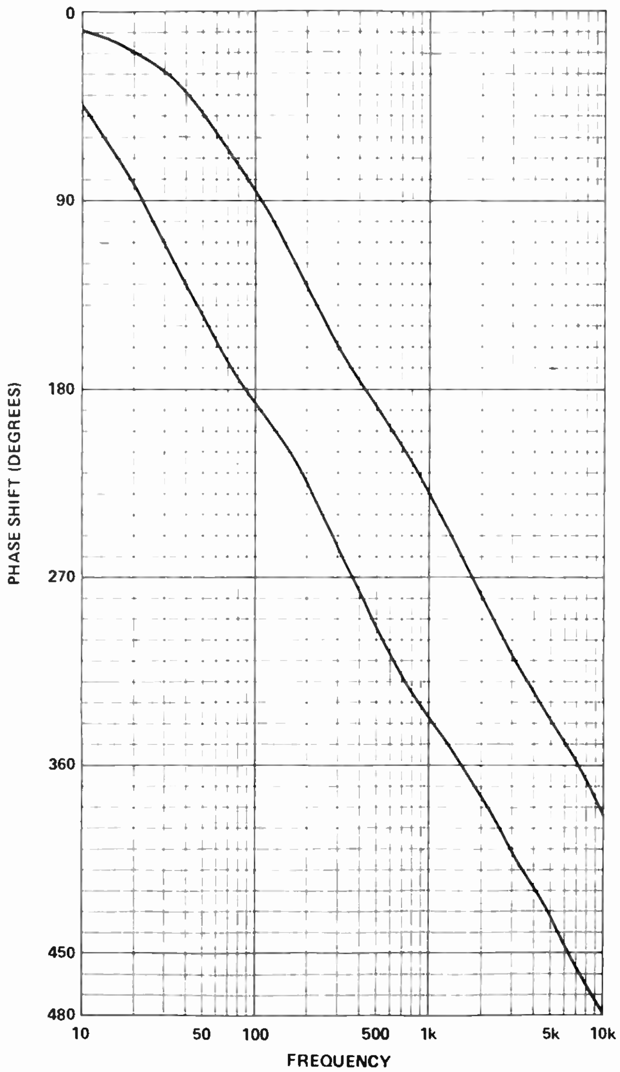


Fig. 4. The phase response of the two filters.

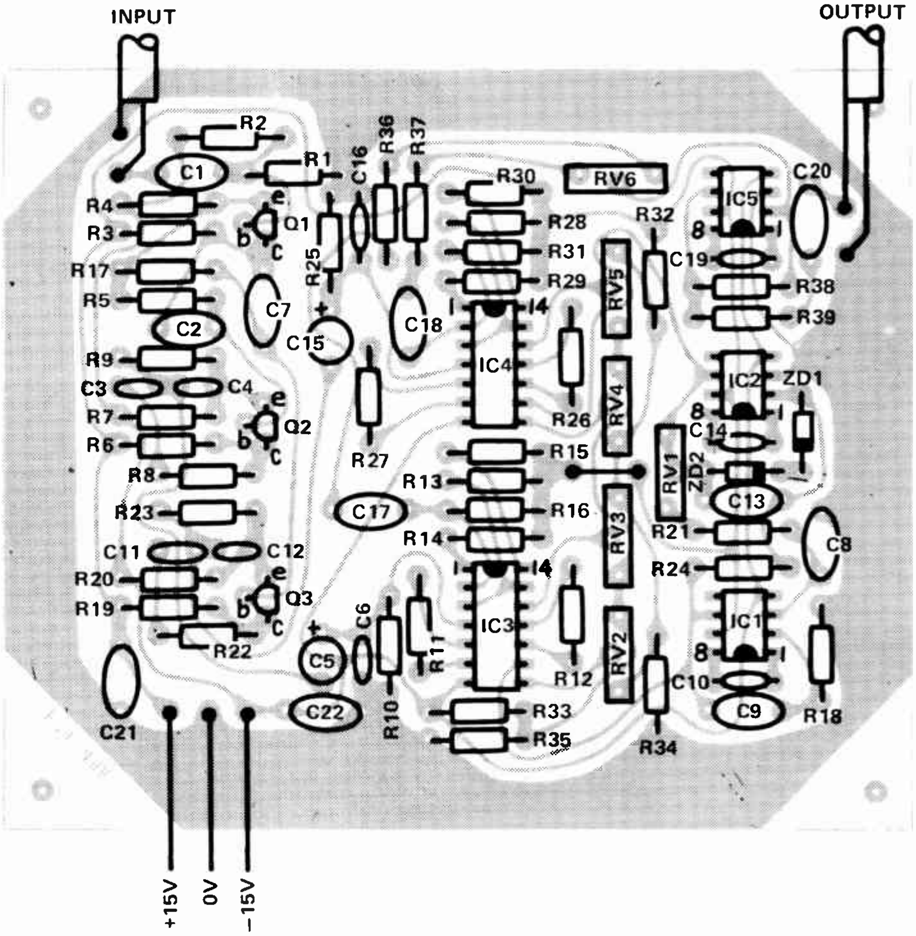


Fig. 5. The component overlay.

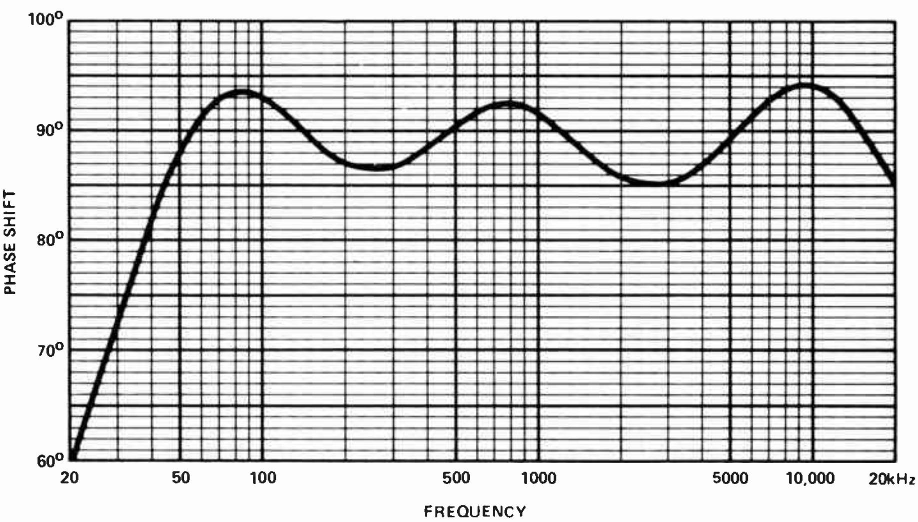


Fig. 6. The phase difference between the two filter networks.

PARTS LIST – ETI 486

Resistors all ½ W 5%

R1	150k
R2	100k
R3,4	2k7
* R5	22k
R6,7	2k7
* R8	3k6
* R9	24k
* R10	15k
R11	4k7
R12	2k2
R13	15k
R14–R16	3k3
* R17	56k
R18	330k
R19,20	2k7
R21	330k
* R22	3k9
* R23	27k
R24	330k
* R25	15k
R26	2k2
R27	4k7
R28	15k
R29–R31	3k3
R32,33	10k
R34,35	1k
R36,37	100k
R38	1M
R39	100R

Potentiometers

RV1	250k trim
RV2–RV5	25k trim
RV6	100k trim

Capacitors

C1	100n polyester
* C2	56n polyester
* C3	6n8 polyester
* C4	1n0 polyester
C5	10µ 25V electro
* C6	1n5 polyester
* C7	100n polyester
C8,9	100n polyester
C10	33p ceramic
* C11	22n polyester
* C12	4n7 polyester
C13	100n polyester
C14	33p ceramic
C15	10µ 25V electro
* C16	5n6 polyester
C17,18	100n polyester
C19	33p ceramic
C20–C22	100n polyester

Semiconductors

IC1,2	LM301A
IC3,4	MC1495
IC5	LM301A
Q1–Q3	BC549
ZD1,2	5.1V 300mW

Miscellaneous

PC board ETI 486
Power supply ± 15V 40mA (ETI 581)

* For best results the components should be as accurate as possible, preferably 1% tolerance or selected to be within 1%.

DUAL POWER SUPPLY

This simple regulated supply is suitable for most projects requiring a dual voltage.

WITH THE PRICE of operational amplifiers being so low today, their use is becoming very popular among home constructors. These devices, however, normally need a dual power supply voltage, usually +15 and −15 volts.

A simple rectified and filtered supply suffers from the drawback that if it is designed to supply the correct voltage at a reasonable current, when a light load is connected the output may rise to an over-voltage condition. This problem is aggravated by variations in mains voltage. The regulated supply takes care of this problem, and also offers better hum rejection as the ripple voltage is also 'regulated'.

Most of the projects undertaken do not use more than 10 or so ICs and a high powered supply is not required. This simple supply has all the comp–onents mounted on the PC board including the transformer. Either of two regulators can be used giving either 40mA or 150mA outputs.

If a different output voltage is required regulators of the desired voltage can be used along with a different voltage transformer. If only a single output is required the unwanted components can be deleted.

To drive a power indicator LED we have provided a current limiting resistor R1. This comes from the unregulated supply so as not to load the regulator.

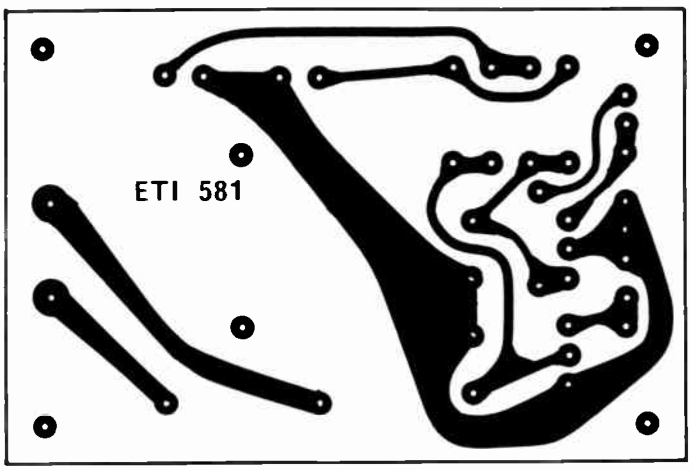


Fig. 1. Printed circuit layout. Full size 90 x 60 mm.

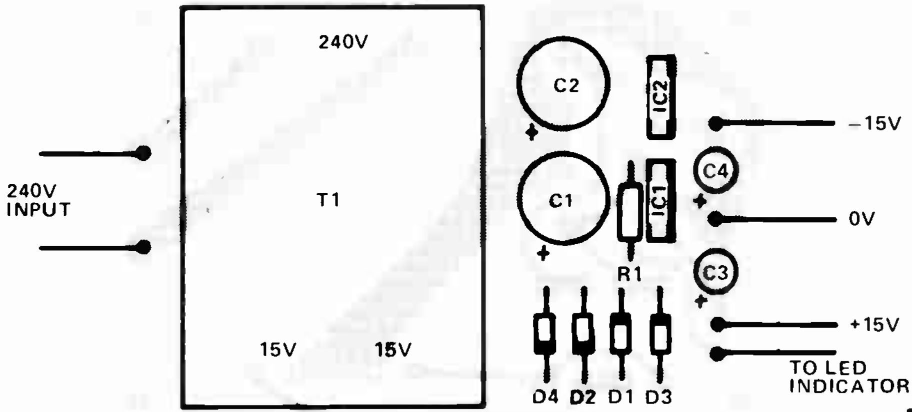
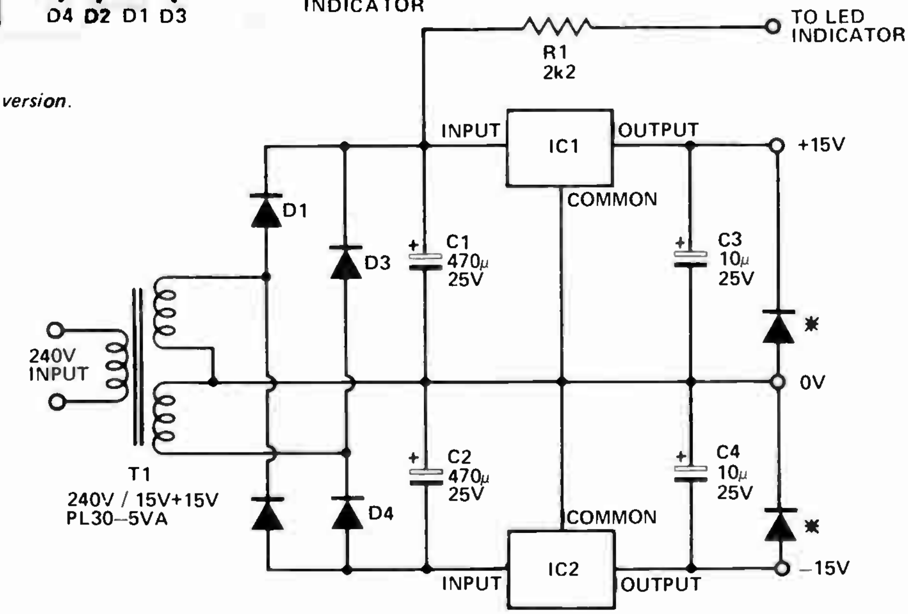


Fig. 3. Component overlay of the higher powered version.

Fig. 2. Circuit diagram. It has been found advantageous to some applications to connect diodes (1N914, 1N4001 etc) across each regulator to prevent the output from being reversed biased. These diodes are shown in the circuit (below) and in Fig. 4 but are not included in the parts list.



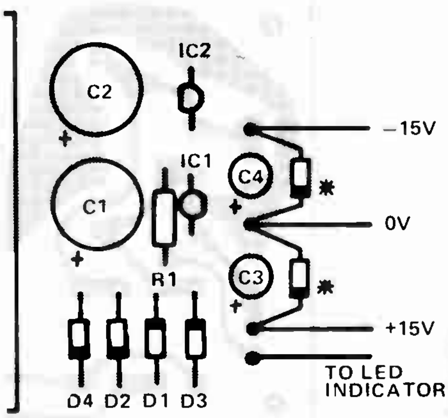


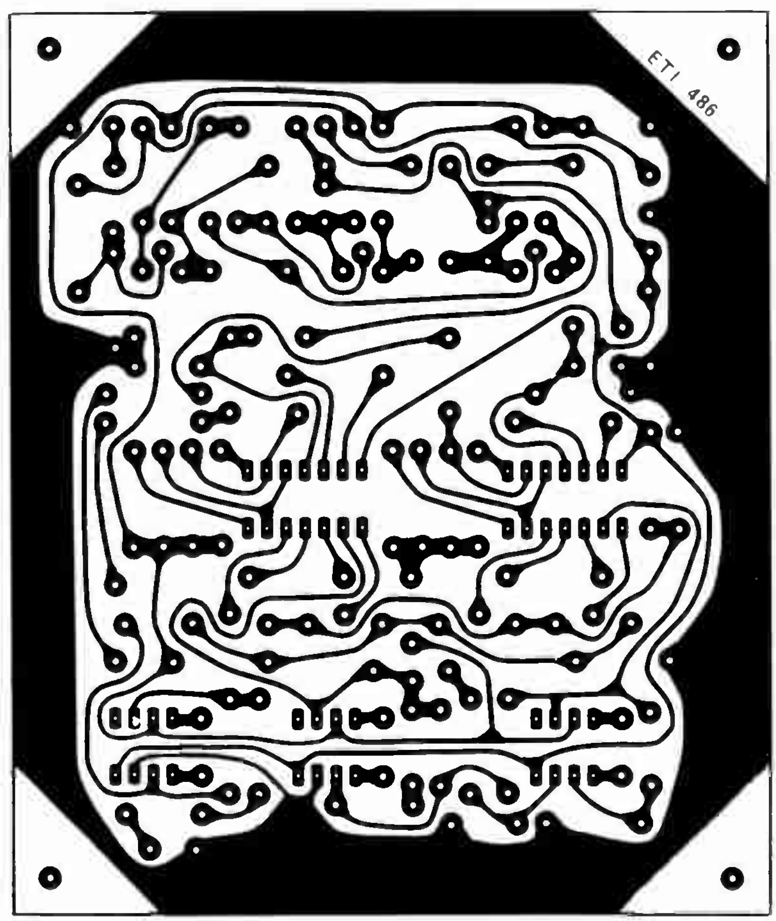
Fig. 4. Changes in the overlay for the low power version.

PARTS LIST – ETI 581

R1	Resistor	2k2 ½W 5%
C1,2	Capacitor	470µ 35V electro
C3,4	"	10µ 25 V "
D1–D4	Diodes	1N4001
LED1	Indicator	
IC1	Regulator	7815 *
IC2	"	7915 *
T1	Transformer	PL30–5VA

* 78L15 and 79L15 can be used if less than 40mA is required

(Continued from page 88).



AUTO-AMP

Boost portable radio output in your car.

eti PROJECT 314

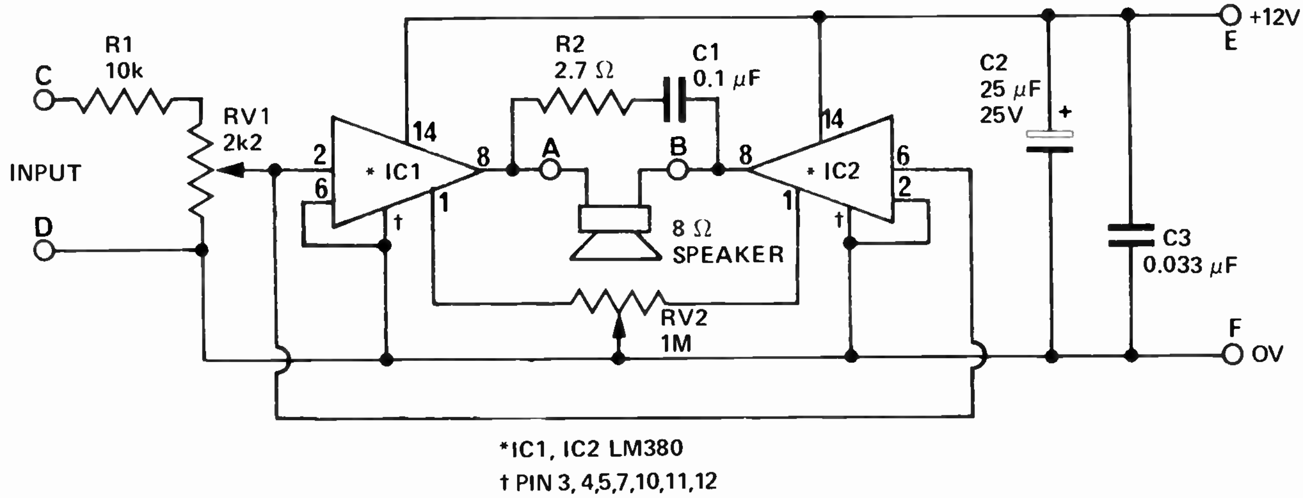


Fig. 1. Circuit diagram of the booster amplifier.

MOST portable radios and cassette players have a power output which seldom exceeds 100 milliwatts. Whilst this is entirely adequate for normal listening, many people find that it is entirely inadequate when such equipment is used in a car. There the extremely high noise level effectively drowns out such radios and one is left with the choice of buying a proper (and quite expensive) car radio, or, of forgetting about the whole deal.

However this problem can be overcome by using a small booster-amplifier to provide the additional power required. Such an amplifier should be powered from the 12 volt car supply and should accept an input from the earphone, or external speaker socket of the radio or cassette player.

The ETI booster amplifier has been designed to suit such applications and uses the inexpensive LM380 ICs. Two ICs are connected in a bridge arrangement which provides an output of around five watts RMS (12 volt supply and 8 ohm speaker). The amplifier may be used to drive an eight-ohm speaker permanently mounted in a suitable position in the car.

CONSTRUCTION

The components should all be mounted on a small printed circuit board (or Veroboard etc) as shown in the component overlay diagram. If Veroboard construction is used it is preferable to mount the ICs, in line, such that a common heatsink may be attached to both ICs on each side. Each heatsink should be at least 25x50mm and be constructed from copper or tin plate.

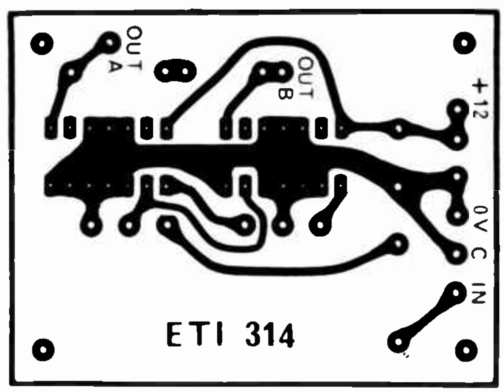


Fig. 2. Printed circuit board. Full size 50 x 65 mm.

Two preset potentiometers are provided for setting up the amplifier. The preset-volume potentiometer, RV1 should be adjusted to suit the output voltage available from the radio or cassette. Sensitivity of the booster is such that 5 watts output will be obtained (with RV1 at maximum sensitivity) with an input of 50 mV. This should be entirely adequate as most radios will provide in excess of 200 millivolts.

The balance potentiometer should be set for minimum dc through the speaker as detailed in the 'How It Works' section.

The compactness and simplicity of the amplifier enable it to be mounted in any convenient position, eg, even on the rear of the speaker itself! However, care should be taken to position it such that mechanical damage is unlikely to occur, and that adequate ventilation of the heatsink is obtained.

PARTS LIST ETI 314

* R1	Resistor	10k ½W 5%
R2	"	2.7 ohm ½W 5%
* RV1	Potentiometer	2k2 Trim
RV2	"	1M Trim
C1	Capacitor	0.1μF polyester
C2	"	25μF 25V electro
C3	"	0.033μF polyester

IC1, IC2 Integrated Circuit LM380
PC Board ETI 314

* The value of these components may vary for different input requirements.

HOW IT WORKS – ETI 314

The LM380 is an integrated audio amplifier which has a fixed gain of 50 dB and can be connected in either inverting or non-inverting mode (ie output 'out of phase' or 'in phase' with the input respectively).

Two of these ICs have been used in a bridge arrangement which allows a higher power output to be obtained with the low supply voltage (12 volts) available from the car. To do this we drive both amplifiers with the same signal, but connect one for inverting, and the other for non-inverting mode. The speaker is now connected between them and thus receives twice the output voltage that would be available from a single IC.

The input required for full power output is about 50 millivolts. Hence we have provided an input attenuator to increase the input requirement to about one volt which will enable preset adjustment to suit most radios or cassettes.

We used a trim potentiometer on the board to adjust sensitivity such that full volume is obtained with the volume control of the source about half way up. If desired, a separate potentiometer may be used in place of the preset as a volume control.

Output voltage of the ICs is about half of the supply. However since the speaker is direct coupled, any slight difference in amplifier outputs will result in a dc current flow through the speaker. Potentiometer RV2 should be adjusted, with the aid of a multimeter, for zero volts across the speaker (or minimum current from the supply). Alternatively, if a multimeter is not available, make and break one speaker connection and adjust RV2 for minimum 'clicking' sound from the speaker.

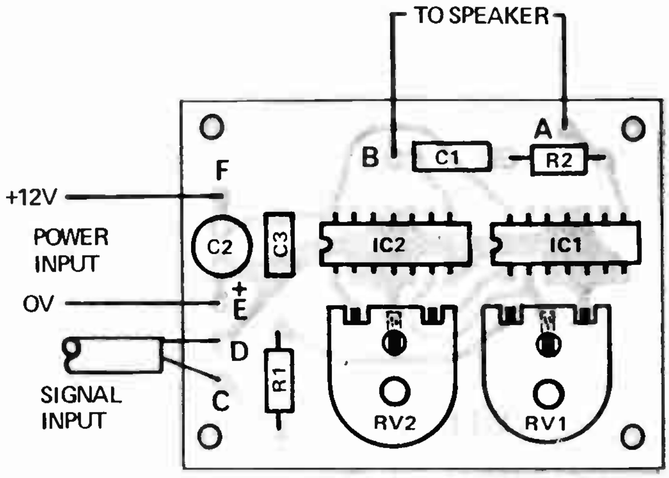


Fig. 3. Component overlay.

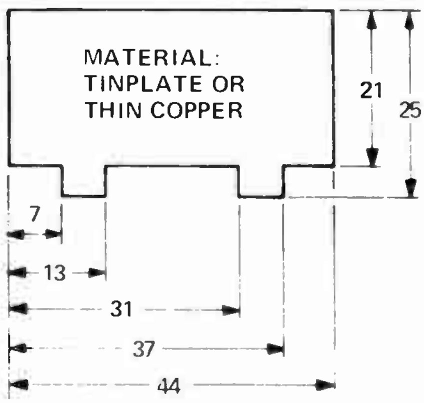
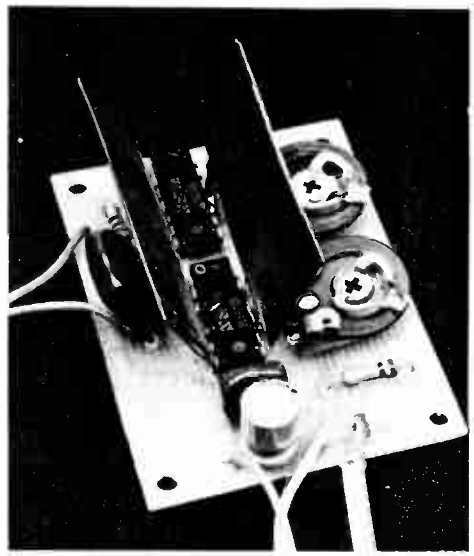


Fig. 4. Heatsink (two required) to be attached to either side of both IC's as shown in main picture.



BATTERY TRAP

When soldering leads to AA size cells (the size used in pen torches) it is most important to remove the metal disc at the bottom of the cell, and solder directly on to the zinc case. These cells are designed to be used under compression, by the spring in a torch, and they depend on the compression to ensure good contact between the metal disc and the cell itself. Without the pressure – such as when a cell is soldered into circuit, it's easy to get a very poor contact or an actual open circuit

A.J. Lowe

SPECIFICATION

POWER OUTPUT		
12.6 volt supply 8 ohm load		5 watts
DISTORTION		
12.6 volt, 8 ohm, 1 kHz		
at 5 watts		3%
at 3 watts		0.5%
SUPPLY VOLTAGE		
Nominal		12 volts
MAX SUPPLY VOLTS	Speaker load	
LM380	8	15 volts
	16	22 volts
SPEAKER IMPEDANCE		> 7 ohms
FREQUENCY RESPONSE		
10 Hz – 100 kHz		$^{+0}_{-3}$ dB
SENSITIVITY		
Maximum (no input attenuator) into 75 k ohm		50 mV

Project 449

BALANCED MICROPHONE AMPLIFIER

High-quality transformers for matching balanced microphones into unbalanced inputs are quite expensive. This simple preamplifier will accept balanced inputs directly.

THIS IS THE third of a series of pre-amplifiers we have published lately and is one which has been requested by several readers. It will be especially useful to users of the Master Mixer project.

Because there is no provision for balanced inputs, transformers have to be used, but they become expensive if good performance is required.

This unit provides good common mode rejection, a gain of 40 dB and low noise.

Construction

A pc board is not necessary but it does make the assembly much easier, and it reduces the risk of mistakes. Assemble the components to the board according to the overlay in Fig. 3. Watch the orientation of the BC 549 transistors as there are two different pin-outs depending on the manufacturer. The Philips is shown on the overlay. Note also that the IC is marked on one end and this should be orientated as shown.

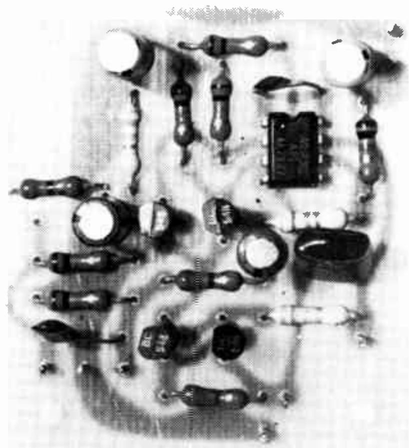
As the unit is to be used with low level inputs it will be necessary to use shielded cables on the input and it may also be necessary to shield the complete unit to eliminate unwanted pick up from the mains.

Although the unit is designed for 600 ohm input and 40 dB gain (i.e., a gain of 100) other input impedances and gains can be handled —
R1 = R4 = input impedance divided by two.
R5 = R13 = voltage gain times the value of R3.

The first equation works for impedances up to about 5k. Above this value R2 + R3 must be included in the calculation.

The connection of the Cannon socket is shown in Table 1.

The Common mode rejection ratio is determined by the accuracy of R1, R4, R5 and R13. If high rejection is needed this can be done in either of two ways: The resistors can be selected to be the same value (R1 = R4, R5 = R13) or 1% resistors can be used.



SPECIFICATION* ETI 449

Frequency Response	10 Hz – 20 kHz (<5 V output) $^{+0}_{-3}$ dB
	10 Hz – 10 kHz (10 V output) $^{+0}_{-3}$ dB
Gain	40 dB
Equivalent Input Noise	−123 dB (0.5 μV)
Distortion	0.05% 300 mV – 5 V output 100 Hz – 10 kHz
Max Input Voltage	100 mV
Common Mode Rejection Ratio	60 dB
Maximum Common Mode Signal	3 V
Supply Voltage	±8 to ±16 V

*Measured performance of prototype with supply rails of ± 15 V— the figures should be typical.

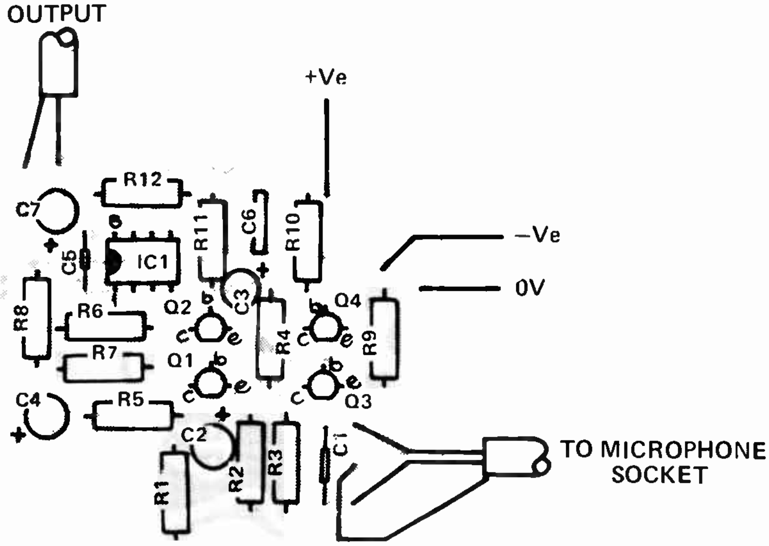


Fig. 3. Component overlay

PARTS LIST ETI 449

Resistors all 1 W 5%		Capacitors	
R1	330	C1	1n0 polyester
R2,3	10k	C2,3	33µ 10v electro
R4	330	C4	10µ 16v electro
R5	33k	C5	33p ceramic
		C6	100n polyester
R6,7,8	10k	C7	10µ 16v electro
R9	3k3		
R10,11	33k	Q1-Q4	Transistors BC 549
R12	1k	IC1	LM301A
		PC Board	ETI 449

TABLE 1
Connection of Cannon plug for microphones

Pin 1	EARTH
Pin 2	BLACK INPUT connect to R1
Pin 3	RED INPUT connect to R4

FOR UNBALANCED INPUT CONNECT PIN 1 AND 2 TOGETHER ON MICROPHONE PLUG.

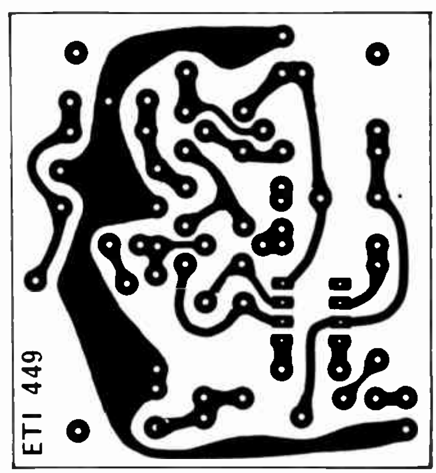


Fig. 4. Printed Circuit board layout. Full size 60 x 55 mm

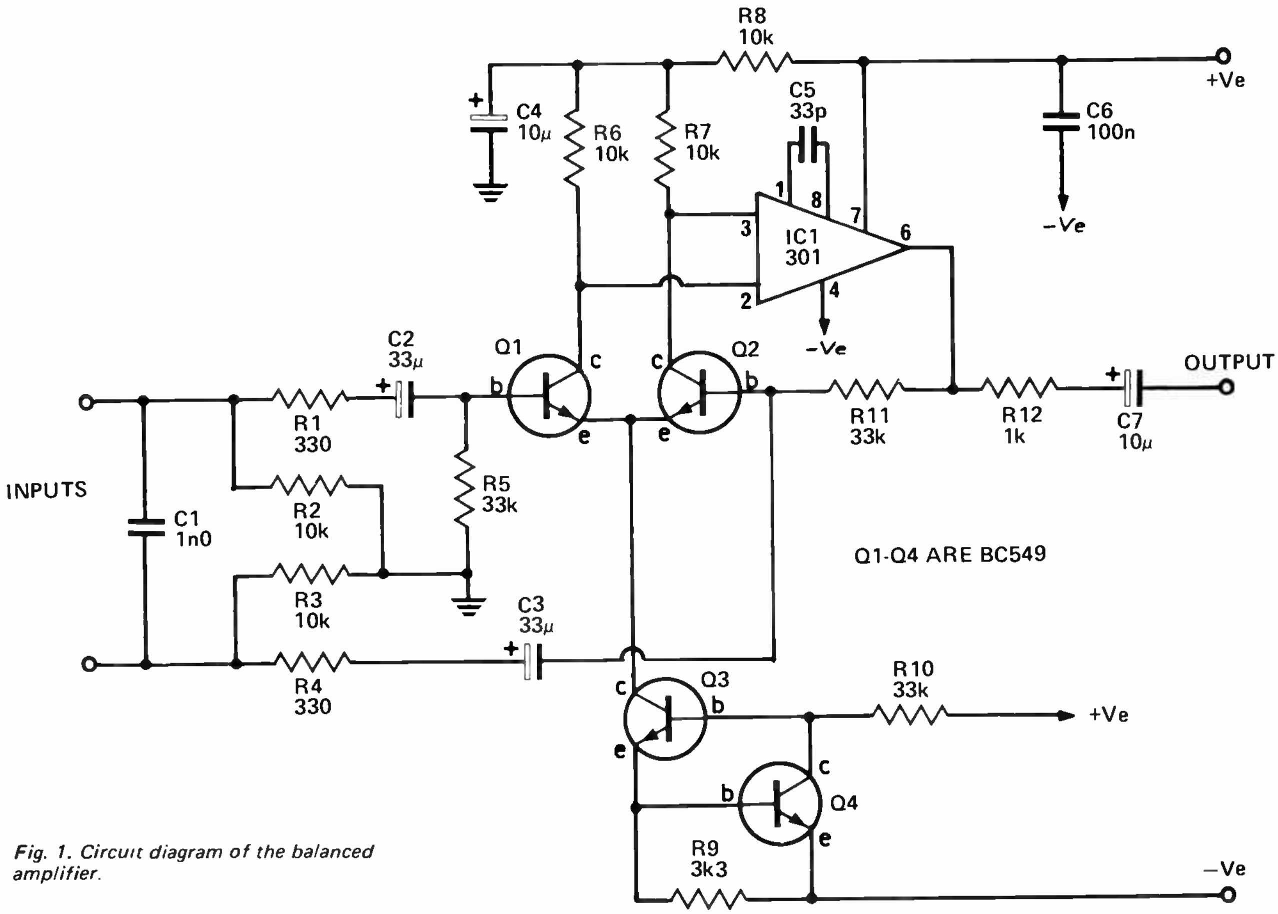


Fig. 1. Circuit diagram of the balanced amplifier.

HOW IT WORKS ETI 449

A "balanced" amplifier or differential amplifier has two separate inputs and only the difference between these inputs is amplified. To explain how this works refer to Fig. 2, which is a simplified version of the actual circuit. To make tne maths easier we will reduce the gain to nine by making R1 = R4 = 1 and R5 = R13 = 9. The actual units are not important, only the ratio.

We will start the explanation by looking at the case where point B is at 0V and A is at +100 mV. An ideal amplifier does two things — it does not take any current into the input terminals and it adjusts the output to maintain no voltage difference between the input terminals. We therefore must have 100 mV across R4 and consequently a voltage of 900 mV across R13 (it has 9 times the resistance and the same current as R4). This gives a gain of nine. The output is therefore –900 mV.

In the case when point A is at 0V and point B is at +100 mV, point D will be at $\left(V_B \times \frac{R5}{R1 + R9}\right) = 90$ mV

Therefore point C will also be at +90 mV. The voltage across R4 will be 90 mV and voltage across R13 will be 810 mV (9 x 90 mV)

This means the output voltage must be +900 mV. This is also a gain of nine. Notice, however, that the polarity (or phase) is different.

Now suppose both inputs are at, say, +1 V, point D will be at +900 mV and so will point C. The voltage across R14 is 100 mV and R13 900 mV. This gives an output voltage of 0 V. The common signal is not amplified in any way. If, however, one input (B) is at 1 V and the other (A) is at 1.01 V the difference is amplified and the output will be –1 V.

Getting back to the actual circuit, we have used an LM301A with two low-noise transistors in the front stage. These transistors are supplied with a constant current by Q3 and Q4. A constant current is needed as this allows the inputs to move up and down without changing the voltage across R6 or R7.

We have decoupled the supply to Q1 and Q2 by C4 to improve supply rail rejection.

The input resistors are decoupled by C2 and C3 to minimise the offset voltage of Q1 and Q2 being amplified by 100 (which could be excessive if matched transistors are not used). The capacitor C1 is to ensure the inputs are terminated at high frequencies. This ensures stability which would otherwise be a problem when the microphone is unplugged.

The resistors R2 and R3 refer the inputs to 0 V but are high enough not to affect the operation in any way.

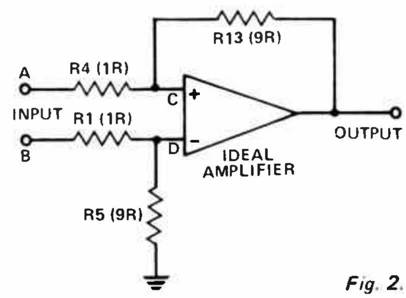


Fig. 2.

GRAPHIC EQUALIZER

This revised version of our earlier equalizer now uses gyrators to replace the inductors making construction easier.

GRAPHIC EQUALIZERS are popular with both the professional and domestic user alike. However until the presentation of our earlier equalizer (ETI 427) the cost of such a device was very high and this limited its wide use. We have now redesigned the equalizer to simplify the construction and it now has no coils and one additional filter has also been added.

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 deficiencies 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 have filters 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.

The equalizer described here has 10 octave spaced filters but if desired it could be modified to give 1/2 or 1/3 octave spacing as large values of inductance are easily obtained with gyrators (active inductors).

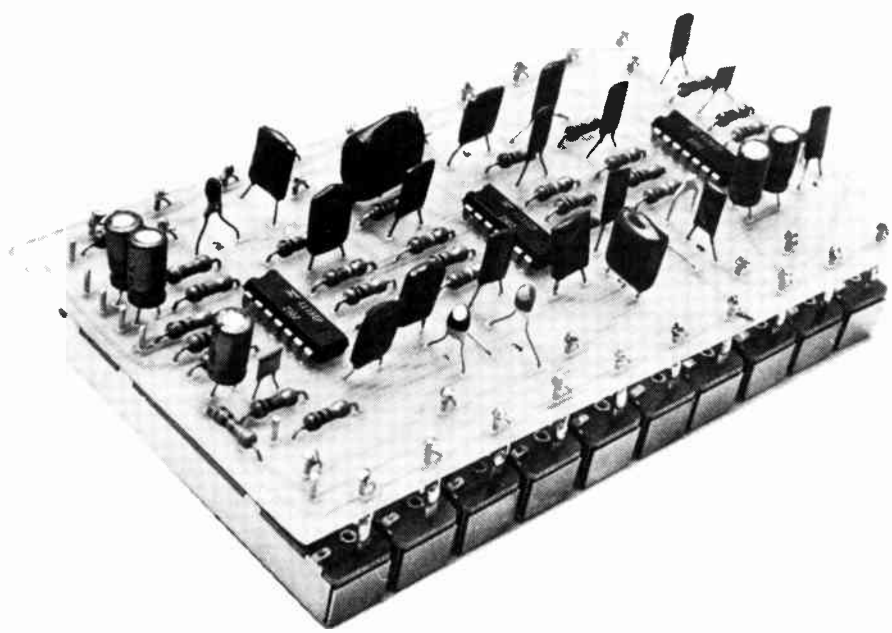
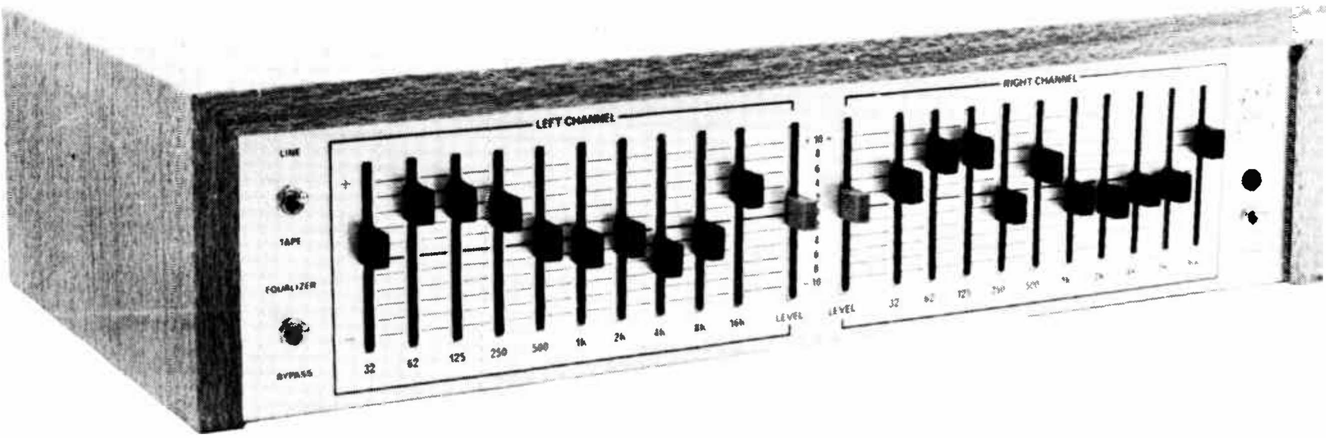


Photo showing one complete channel, less the volume control, removed from the chassis.



Construction

Assemble the PC board(s) with the aid of the overlay in fig 3 initially leaving off the potentiometers. Add pc pins for the external wires and the potentiometers connection points. Now double check the PC board soldering, the positioning and polarity of the components as once the potentiometers are in position changing components is difficult.

Now solder lengths, about 40mm, of tinned copper wire onto the end terminals of the potentiometers, and also onto one of the wiper contacts. Note that half the potentiometers use one wiper connection and the others use the other end. Now slide the wires through the holes provided such that the potentiometers are on the copper side of the board. Before soldering mount the potentiometers onto the support rails, space the board back about 10mm then twist the wire around the PC pins and solder the connections.

The volume controls can now be mounted and connected and the complete assembly mounted into the chassis using 12mm spacers. The power supply can be added along with the other components in the box and finally wire as shown in fig6.

Third octave filters

While we have not built up a third octave unit we see no reason why it will not work. Additional stages can simply be added except that the Q of the circuits must be changed to narrow the band. At the moment the impedance of the capacitor and inductor (gyrator) is about 3000 ohms at the centre frequency and this should be increased to about 8000 ohms for the third octave unit. The capacitors and inductors can be calculated by

$$C = \frac{1}{2\pi f X_C} \qquad L = \frac{X_L}{2\pi f}$$

where $X_C = X_L = 8000\Omega$
and f = centre frequency

It is recommended to reduce loading IC1/2 that the potentiometers be increased to 10k.

SPECIFICATION ETI 485

Frequency response			
Equalizer out			Flat
Equalizer in and all controls at zero	10Hz – 20kHz		± ½dB
Range of controls			
Individual filters			± 13dB
Level control			+ 14dB – 9dB
Maximum output signal at <0.1% distortion			6 volts
Maximum input voltage			10 volts
Distortion			
at 2 volts out, controls flat	100Hz	1kHz	6.3kHz
	0.02%	0.02%	0.04%
Signal to noise ratio re 2 volts out, controls flat			82 dB
Input impedance			47 k
Output impedance			100 ohms

GRAPHIC EQUALIZER

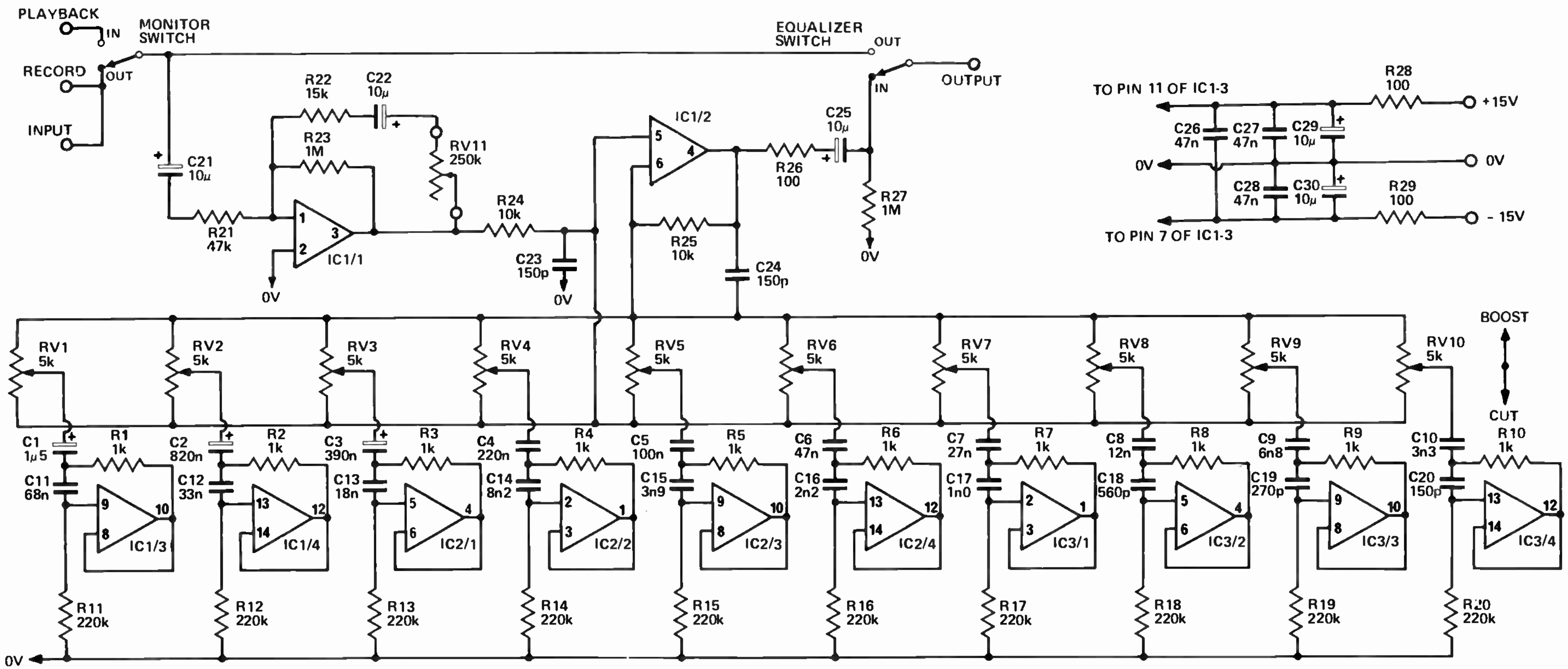


Fig. 1. Circuit diagram of one channel of the equalizer.

How It Works – ETI 485

This equalizer is basically similar to that used in the previous unit with the addition of an extra filter in each channel. The previous unit also used coils (inductors) – these have been replaced by gyrators to simplify construction. We will explain more about gyrators later but at the moment just assume that they are an inductor.

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 impedance of the LCR network will be 1 k ohms at the resonant frequency circuit.

With the slider of the potentiometer at the top end (Fig. A) we have 1 k ohms to the OV line from the negative input of the amplifier, and 5 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.

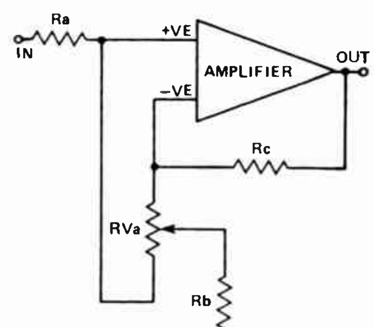
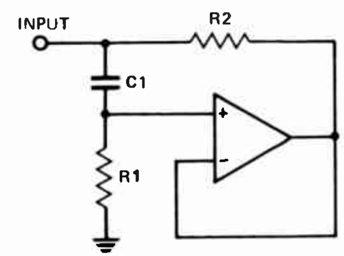


Fig. C

of the amplifier. The use of a second amplifier will increase the resistance to many megohms while the same formula holds for inductance.



of the network. At either side of resonance the impedance will rise (with a slope dependant on the Q of the network which is 3) 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.

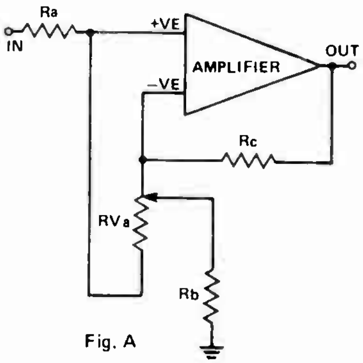


Fig. A

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

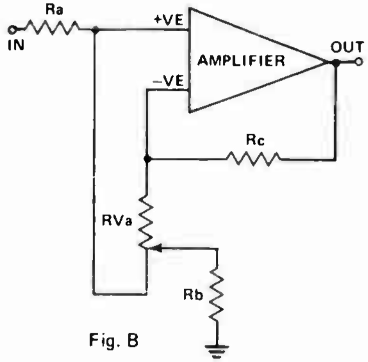


Fig. B

The output of the amplifier in this case is approximately the input signal times (10 000 + 1000)/100 giving a gain of 20 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 0.1 (1000/(10 000 + 1000)) of the input. There will still be no current of the potentiometer and in RC, thus the output will be 0.1 of the input. That is, there will be a loss of 20 dB.

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

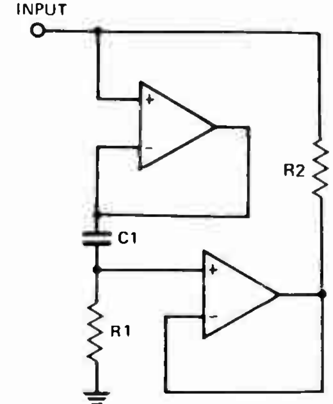
With all filter sections in circuit the maximum cut and boost available is reduced, but ± 14 dB is still available.

In the actual circuit we have used the first op-amp (IC1/1) as a buffer for the input and also as the overall gain control stage. With the values shown the gain is adjustable over a range of −9 to +14 dB. By replacing R22 by a link RV11 will act like a normal volume control. Now to the gyrator.

The only difference between an inductor and a capacitor – electrically, that is, not mechanically – is the phase relationship between the current and voltage. In the gyrator we use an op-amp to reverse the phase relationship of a capacitor and make it appear like an inductor. In the circuit below the inductance is given by the formula

L = R1 x R2 x C1 H where C is in Farads

Like a real inductor there is a series resistance (winding resistance) or R2 and a parallel resistance R1 (in a coil this is due to winding capacitance). The lowest value of R2 depends on the amplifier used but for standard op-amps it would be about 100 ohms. At the high end the value of R1 is limited by input current.



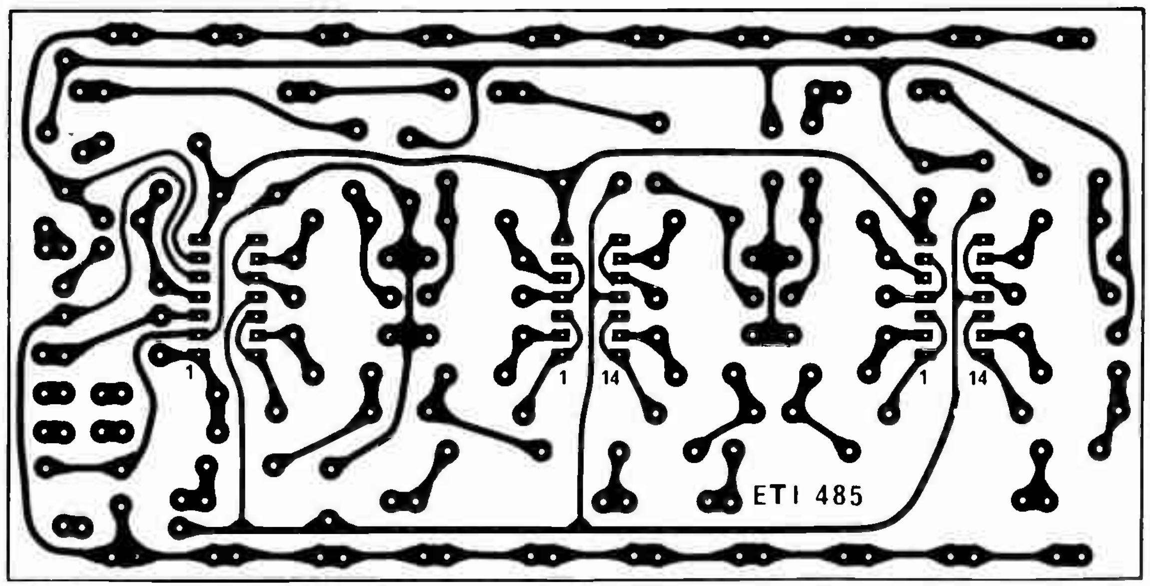


Fig. 2. Printed circuit layout. Full size 150 x 75 mm.

PARTS LIST – ETI 485

Resistors all ½W 5%

R1–R10	1k
R11–R20	220k
R21	47k
R22	15k
R23	1M
R24,25	10k
R26	100
R27	1M
R28,29	100

Potentiometers

RV1–RV10	5k lin 45mm slide
RV11	250k log 45mm slide

Capacitors

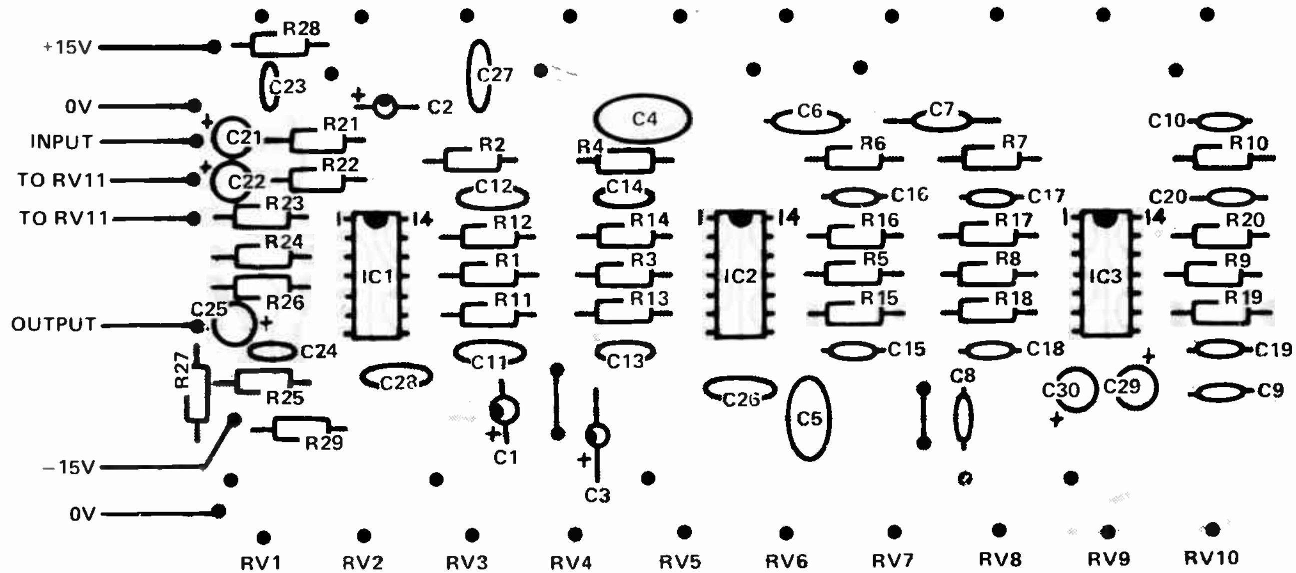
C1	1µ5 tantalum
C2	820n "
C3	390n "
C4	220n polyester
C5	100n "
C6	47n "
C7	27n "
C8	12n "
C9	6n8 "
C10	3n3 "
C11	68n "
C12	33n "
C13	18n "
C14	8n2 "
C15	3n9 "
C16	2n2 "
C17	1n0 "
C18	560p ceramic
C19	270p "
C20	150p "
C21,22	10µ 25V electro
C23,24	150p ceramic
C25	10µ 25V electro
C26–C28	47n polyester
C29,30	10µ 25V electro

Semiconductors

IC1–IC3	4136 Quad op-amp

Miscellaneous

PC board ETI 485
11 knobs McMurdo P/N 4093
For stereo operation twice the above components are required
Power supply ETI 581 (15V 40mA)
Box to suit (see diagram)
Three DPDT toggle switches
Two 4 way RCA sockets
Three core flex and plug
Cable clamp, terminal block



NOTE: RV1–RV10 ARE ON THE COPPER SIDE OF THE BOARD

Fig. 3. Component overlay of the equalizer board.

Note

The 4136 amplifier is manufactured by Raytheon and is distributed in Australia by Soanar Electronics (who supplied us with samples). It is also supplied by Tecnico Electronics.

Tantalum capacitors of 390 nF and 820 nF may be difficult to obtain. 470 nF and 1µF may be substituted causing only a small shift in centre frequency.

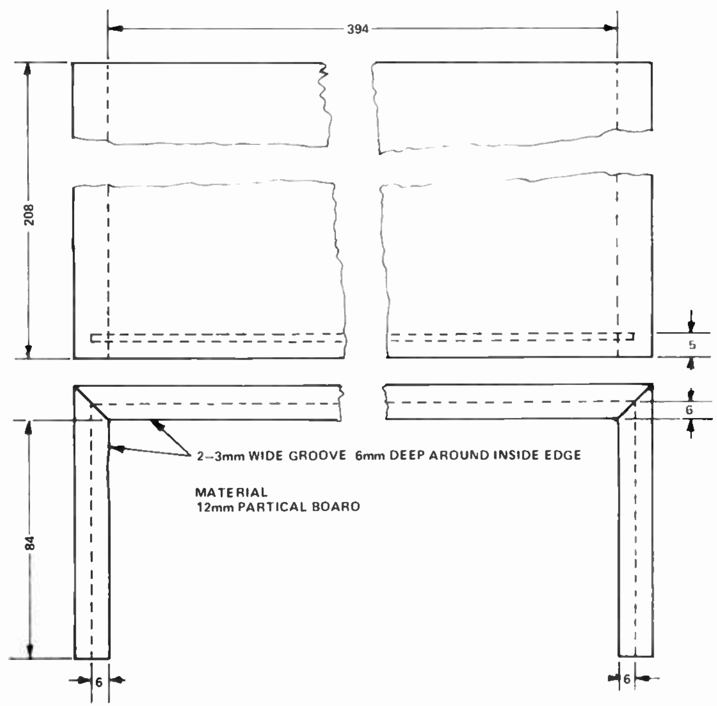


Fig. 4. Constructional details of the cover.

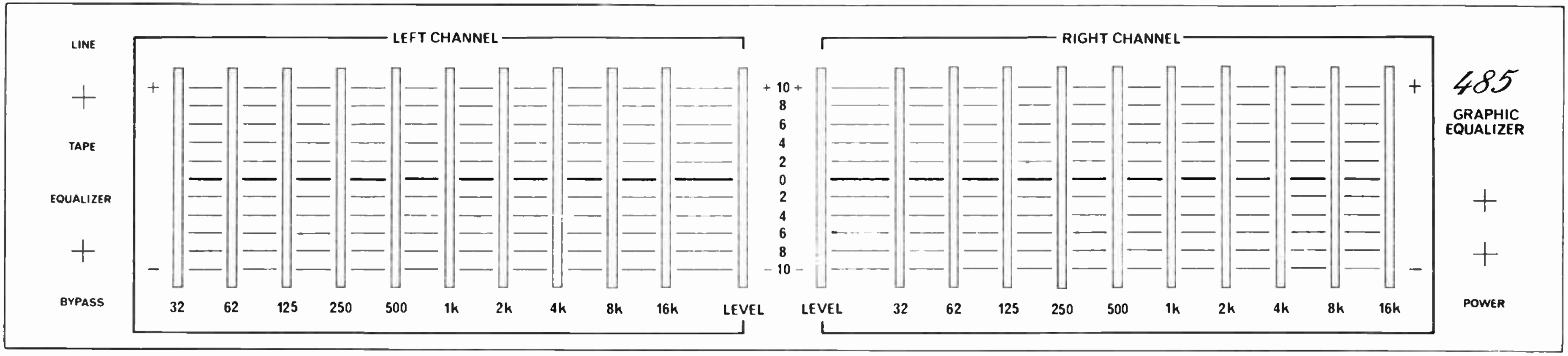


Fig. 5. Front panel artwork for the equalizer. See fig. 8 for dimensions.

Metal work drawing for this project is on page 119.

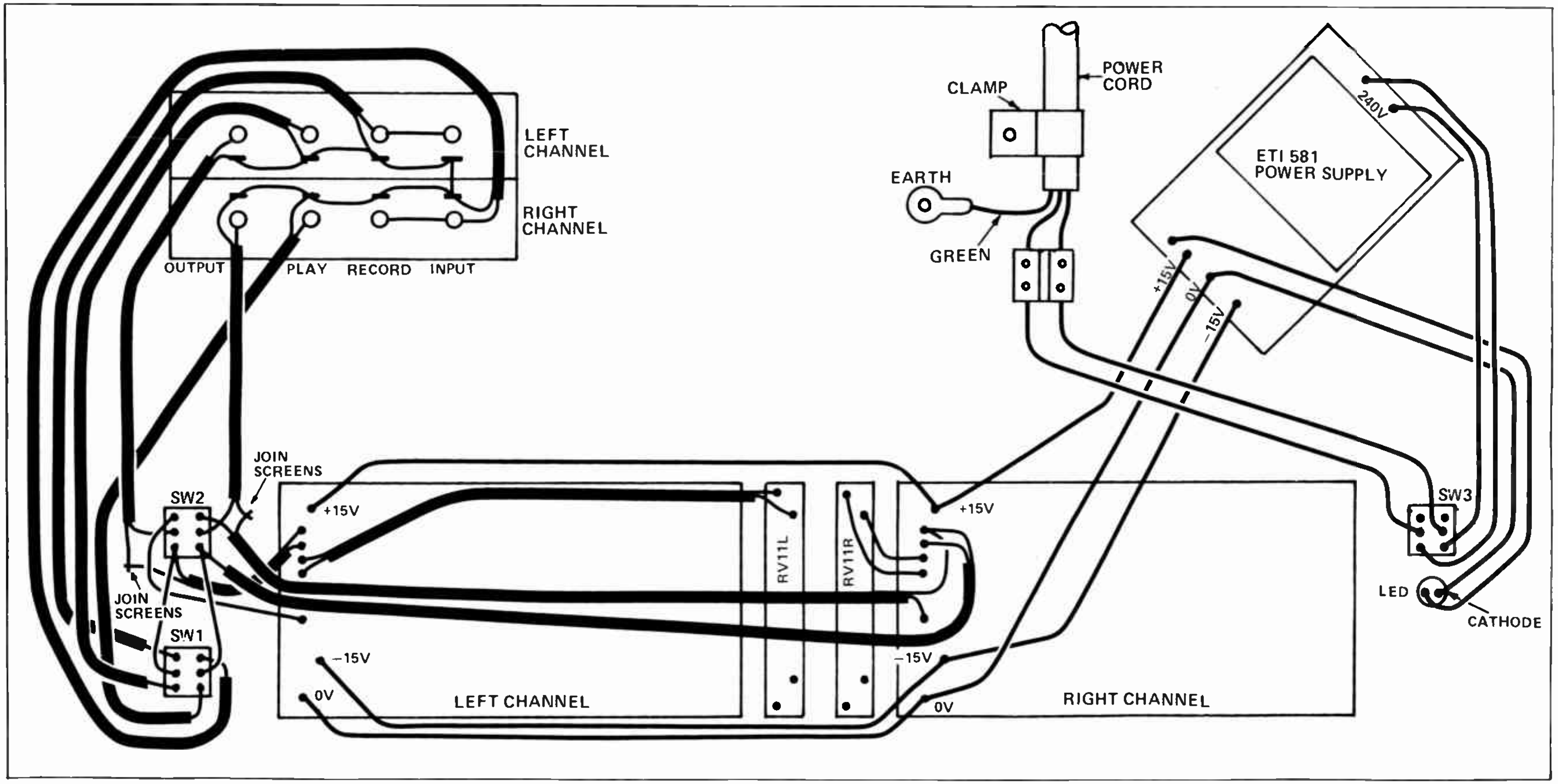


Fig. 6. Interconnection diagram for the unit.

50 – 100 WATT AMPLIFIER MODULES

This is our very reliable 422 amplifier redesigned for simplicity in construction.

THE MOST POPULAR AMPLIFIERS we have ever published are the 100 W guitar amplifier (ETI 413) and the 50 W stereo amplifier (ETI 422). These amplifiers have proved very reliable for the many hundreds of readers who have built them.

Both of the amplifiers are, however, a bit fiddly to build (as are most power amplifiers) because the power transistors must be mounted on a heat sink which therefore needs wiring to the control board. Whilst this module has the same electrical design as the 422 the layout has been greatly simplified.

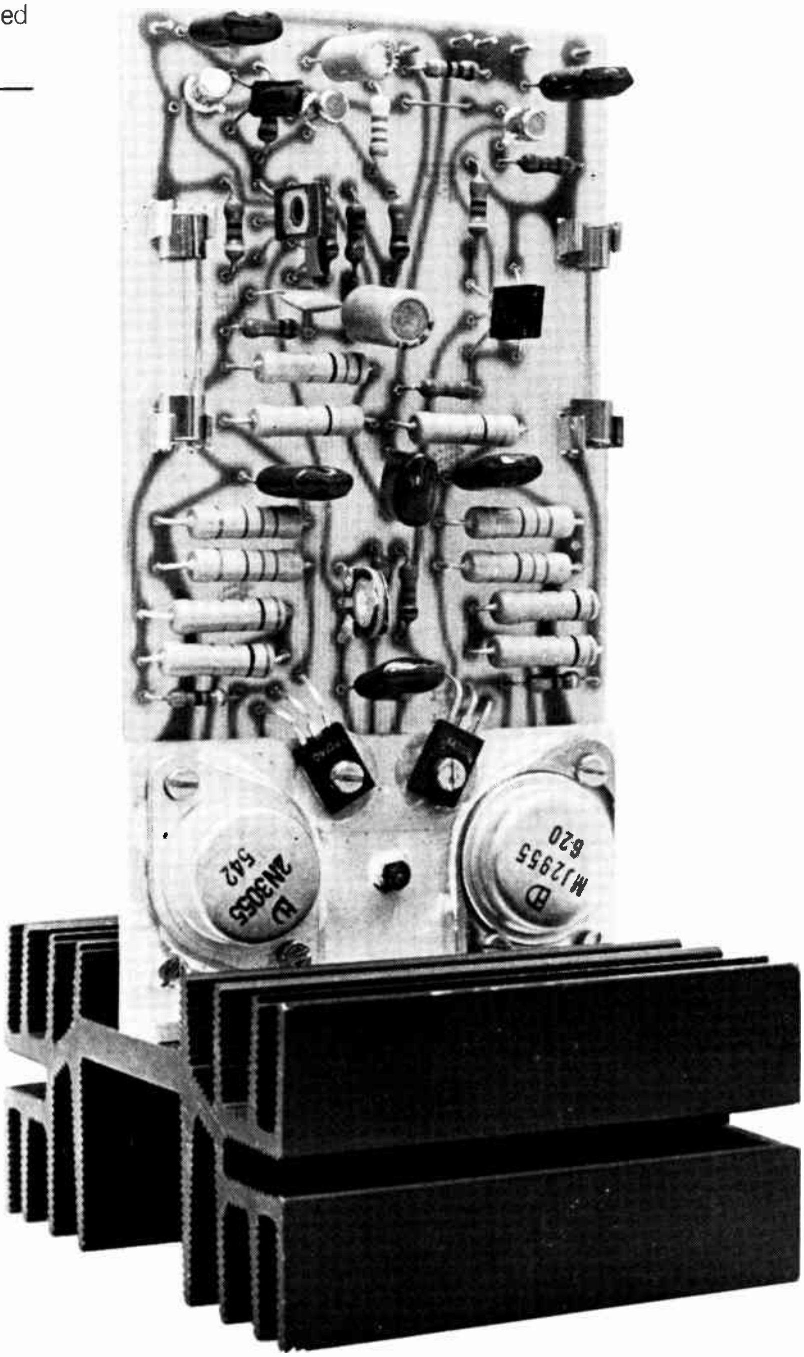
The new design was originally a replacement amplifier for the 422. However we soon realised that by adding two transistors we had a replacement for the 100 watt amplifier as well.

Both versions are very easy to build and set up with all the components, including the power transistors, on the pc board (eliminating another source of possible errors).

Construction

Assemble the module, less the heatsink components, with the aid of the overlay in Fig. 5. Now mount the heatsink bracket on the component side of the board with two 6 BA screws making sure the other holes line up with those in the pc board.

Mount the power transistors and the BD139/140 using insulating washers and silicon grease. If the amplifier is to be run continuously at full power we recommend you use beryllium oxide washers rather than mica ones. This will lower the junction temperature about 10°C.



SPECIFICATION*	ETI 480	
	50 W version	100 W version
Output power	50 W into 8 ohms	100 W into 4 ohms
Frequency response	5 Hz – 50 kHz	5 Hz – 50 kHz
at rated power	$^{+0}_{-3}$ dB	$^{+0}_{-3}$ dB
Input Sensitivity	500 mV	1 V
Distortion	see Fig. 1	
Signal to noise ratio	100 dB	100 dB
Protection	1.5 A fuses	3 A fuses
Damping Factor	25	20
Power Requirement	33 V @ 1.2 A	33 V @ 2.4 A

** Measured performance of prototype*

The screws holding the 2N3055/MJ 2955 should also be insulated where they pass through the heatsink bracket. The BD 139 and BD 140 do not need any insulation other than the mica, provided 6 BA (or 3 mm) screws are used. In the 100 W version the addition transistors are mounted on the heatsink bracket outside the pc board area.

The heat sensing transistor Q6 should be inserted into the bracket using silicon grease, bend the lead flat against the pc board and solder to the pads provided When installed, the transistor should be in the centre of the heatsink.

The recommended power supply is shown in Fig. 3. This supply gives about 40 V on no load, dropping to about 32 V on full output. This allows reproduction of transients beyond 50 W (or 100 W) whilst providing a degree of protection for the output transistors. If a regulated supply is used it should not be higher than ± 35 V.

If no preamp is to be used, a couple of chassis-mounting capacitors (4700 µF) with the diodes wired across the terminals will suffice. If the pc board is used there is facility for building the preamplifier regulator and fitting a dethump relay (if required). The power amplifier itself does not produce any thump.

Alignment

The only adjustment you have to make is to set the current using RV1. The bias current for the 50 W version should be 20–25 mA and for the 100 W version it should be 30–35 mA. The figures are for the amplifier running cold. These currents increase about 50% when the amp gets hot.

To measure the current we recommend soldering a 100 ohm ½ W resistor across each fuse holder and removing the fuses. With no load connected and no input adjust RV1 until there is about 2.5 V (3.5 V for 100 W version) across the resistors. There may be a slight voltage difference between the two resistors, so just take an average. It's not that critical. This method of measuring current is much easier on your testmeter should there be a fault in the amplifier.

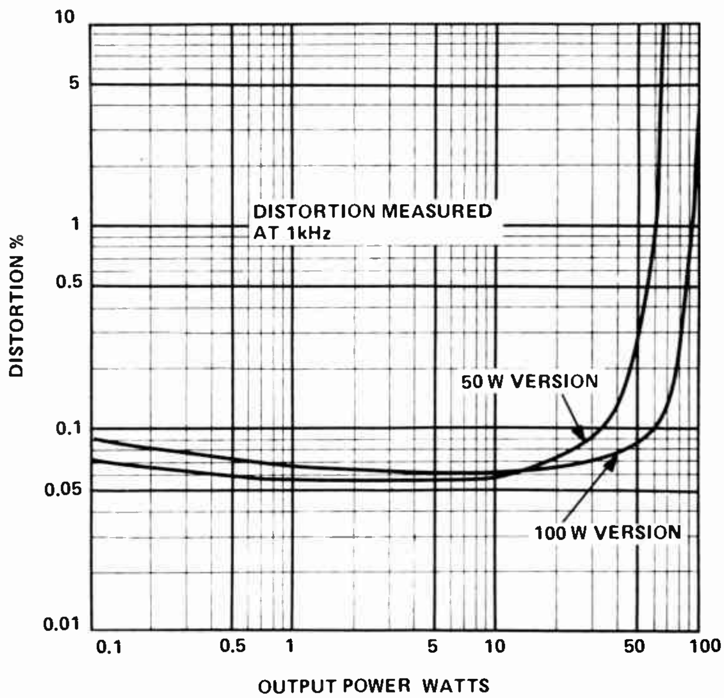
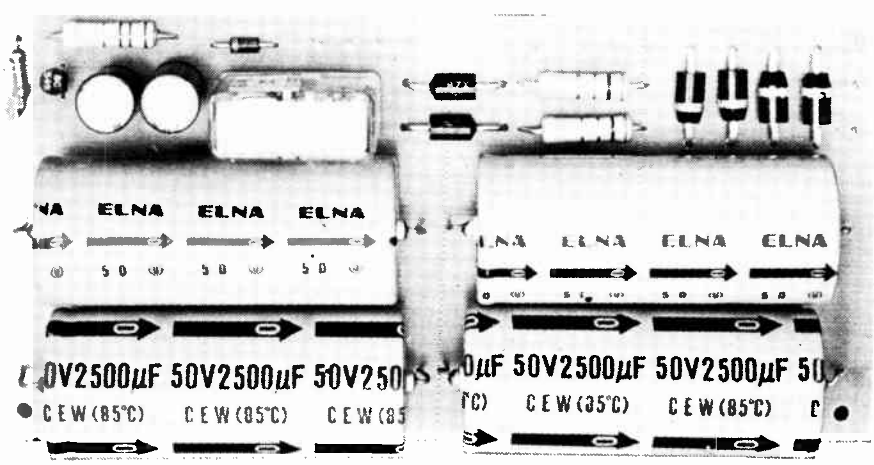


Fig. 1. Graph showing relationship between output power and distortion.



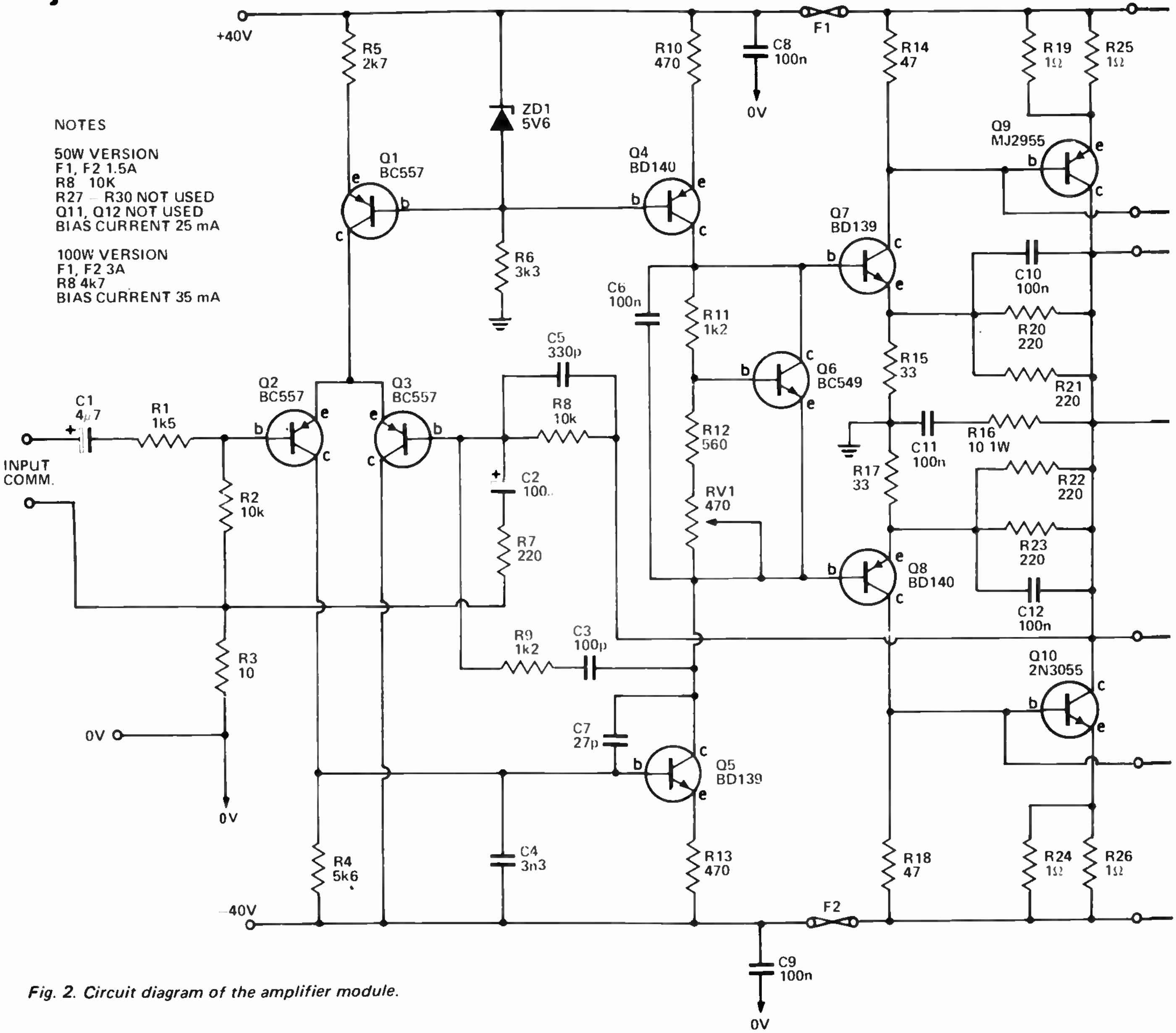


Fig. 2. Circuit diagram of the amplifier module.

How it Works ETI 480

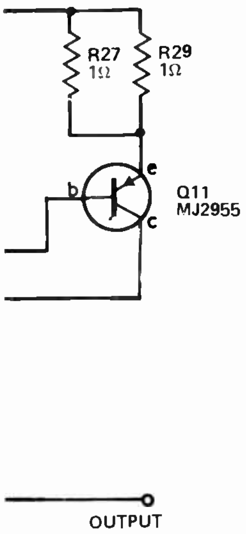
The input signal is fed via C1 and R1 to the base of Q2 which, with Q3, forms a differential pair. Transistor Q1 is a constant-current source supplying about 2 mA. This current is shared by Q2 and Q3. Transistor Q4 is also a constant-current source supplying about 10 mA which, if no input signal exists, flows through Q5 and Q6. The differential pair controls Q5 and thus the voltage at its collector.

The resistors R11 and R12 together with potentiometer RV1; control the voltage across Q6 and maintains it at about 1.9 V. But as Q6 is mounted on the heatsink, this voltage will vary with heatsink temperature. Assuming that the voltage on the bases of Q7 and Q8 is equally spaced about zero volts (i.e., 0.95 volts) the current will be set at about 12 mA through Q7 and Q8. The voltage drop across the 47 ohm resistors (R14, R18) will be enough to bias the output transistors Q9 and Q10, on slightly to give about 10 mA quiescent current in these transistors. This quiescent current is adjustable by means of potentiometer RV1.

Local feedback is applied to the output stage by the network R20–R23, giving the output stage a voltage gain of about four. The overall feedback resistor, R8, 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 attained by mounting Q6 on the heatsink and this transistor automatically



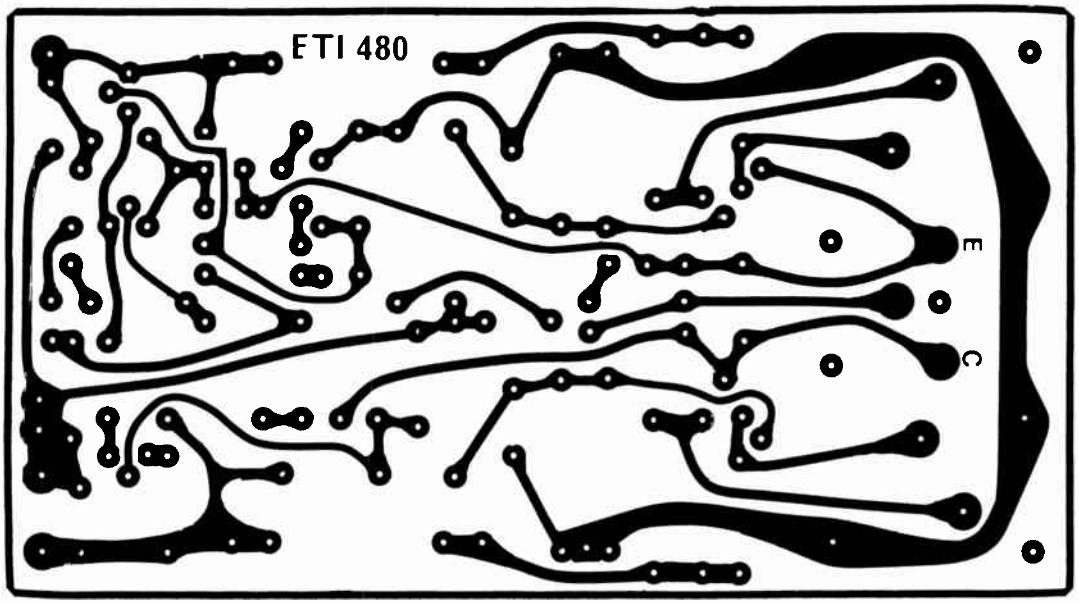
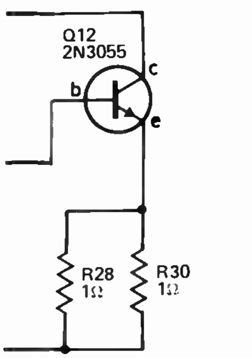
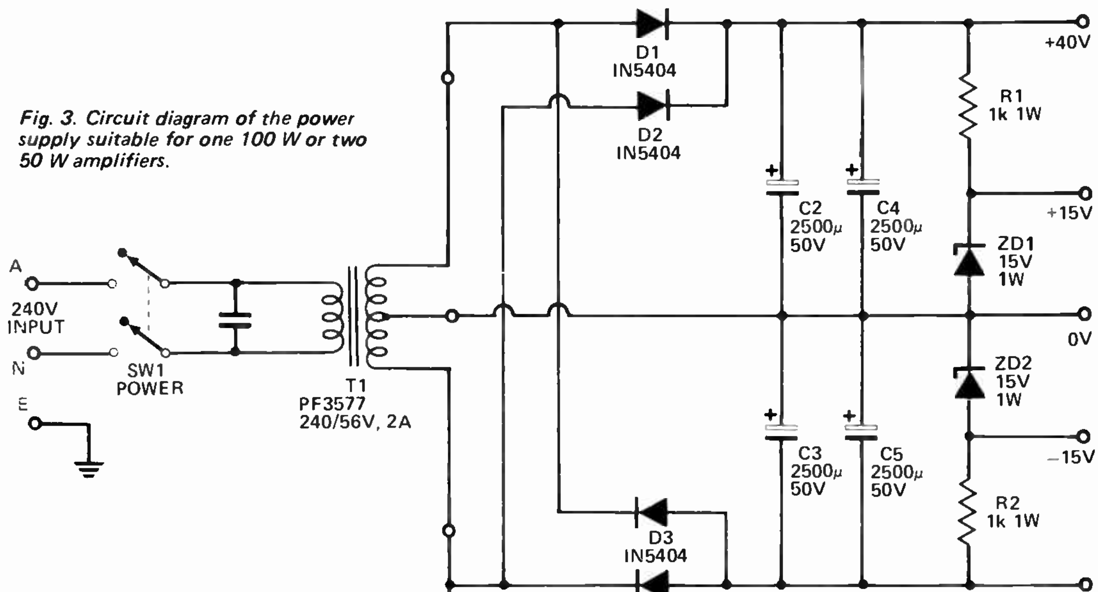


Fig. 4. Printed circuit layout of the amplifier. Full size 140mm x 76mm.





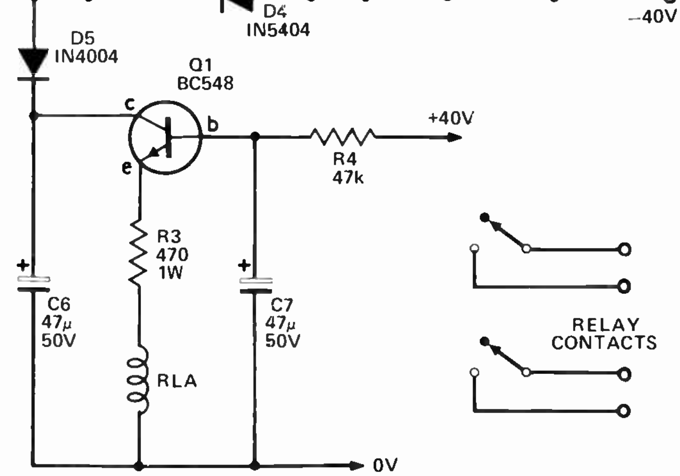


Fig. 3. Circuit diagram of the power supply suitable for one 100 W or two 50 W amplifiers.

adjusts the bias voltage.

The power supply uses a full wave rectifier and a centre tap to derive ± 40 V dc. Dropping resistors and zener diodes are also provided for a preamplifier (if required).

As some preamplifiers cause the main amplifier to give a thump on switch-on, a relay is provided to overcome this. R4 and C7 cause a delay of about 3 sec on switch-on. The relay can be used to switch the output leads from the main amplifier.

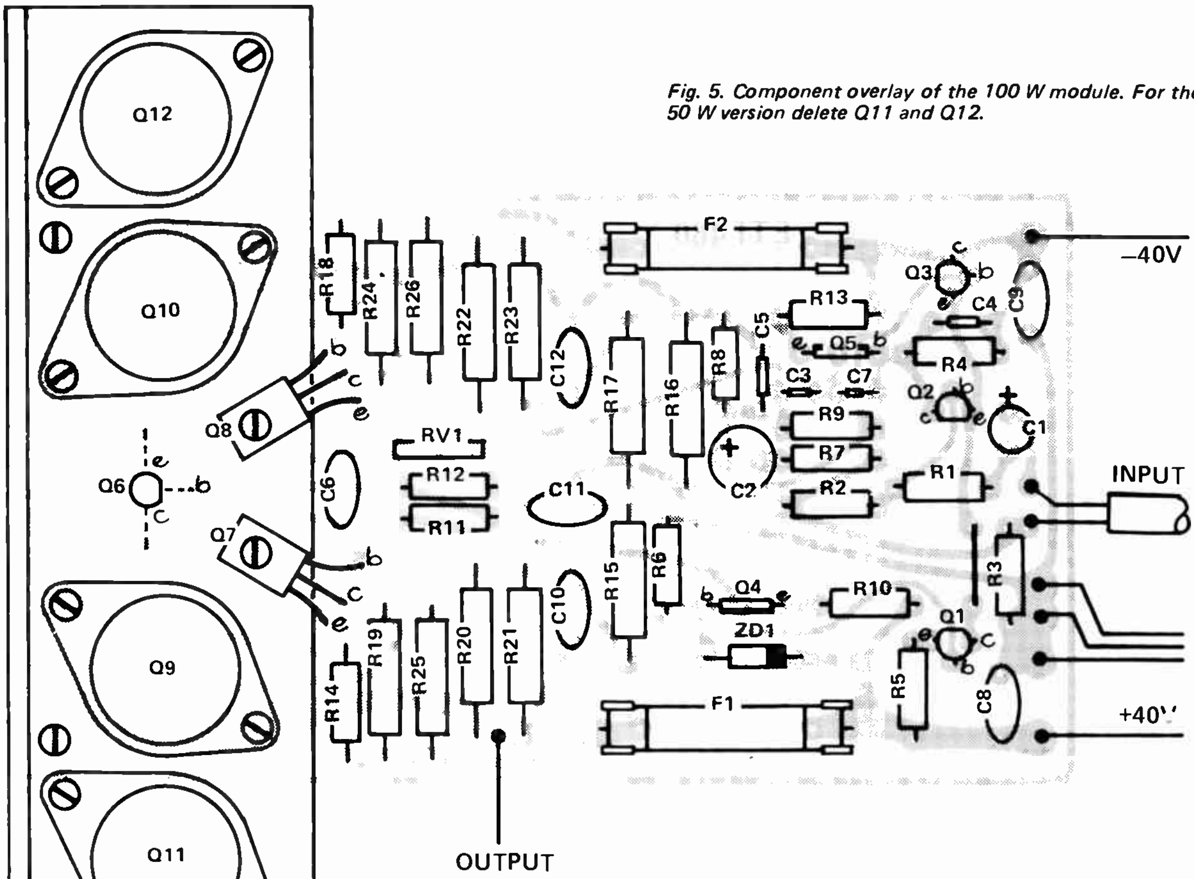


Fig. 5. Component overlay of the 100 W module. For the 50 W version delete Q11 and Q12.

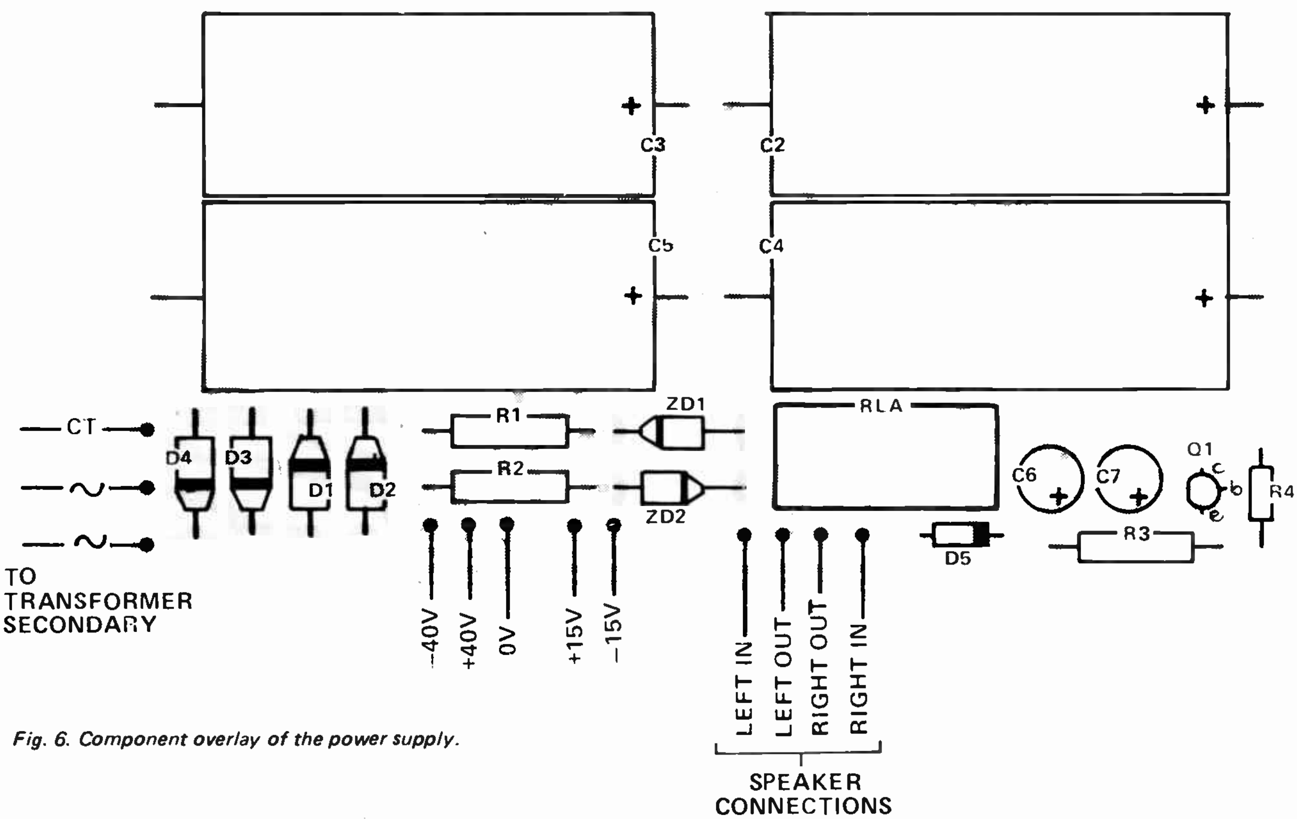


Fig. 6. Component overlay of the power supply.

PARTS LIST – ETI 480

Resistors *all ½ W 5% unless noted*

R1	1k5 ½ W 5%
R2	10 k
R3	10
R4	5k6
R5	2k7
R6	3k3
R7	220
R8*	10 k
R9	1k2
R10	470
R11	1k2
R12	560
R13	470
R14	47
R15	33 1 W
R16	10 1 W
R17	33 1 W
R18	47 ½ W
R19	1 Ω 1 W
R20–R23	220 1 W
R24–R26	1 Ω 1 W
R27 R30*	

Potentiometer

RV1	470 trim type

Capacitors

C1	4 µ7 25 V electro
C2	100 µ 16 V electro
C3	100 p ceramic
C4	3n3 polyester
C5	330 p ceramic
C6	100 n polyester
C7	27 p ceramic
C8 -C12	100 n polyester

Transistors

Q1–Q3	BC557
Q4	BD140
Q5	BD139
Q6	BC549
Q7	BD139
Q8	BD140
Q9	MJ2955
Q10	2N3055
Q11*	
Q12*	

Zener diode

ZD1	5.6 V 400 mW

Miscellaneous

PC board ETI 480
Four PC mounting fuse clips (FC1)
Two fuses 1.5 A*
Heatsink bracket to Fig. 9 *
Insulation kits for Q7–Q12.

*** For 100 W version**
R8 is 4k7 ½ W
R27–R30 are 1 Ω 1 W
Q11 is MJ2955
Q12 is 2N3055
Fuses are 3A
Bracket is to Fig. 8.

PARTS LIST – ETI 480 PS

Resistors

R1, 2	1 k	1 W	5%
R3	470	1 W	5%
R4	47 k	½ W	5%

Capacitors

C1	33 n 250 V ac
C2–C5	2500 µF 50 V electro
C6,7	47 µF 50 V electro

Diodes

D1-D4	1N5404
D5	1N4004

Transistor

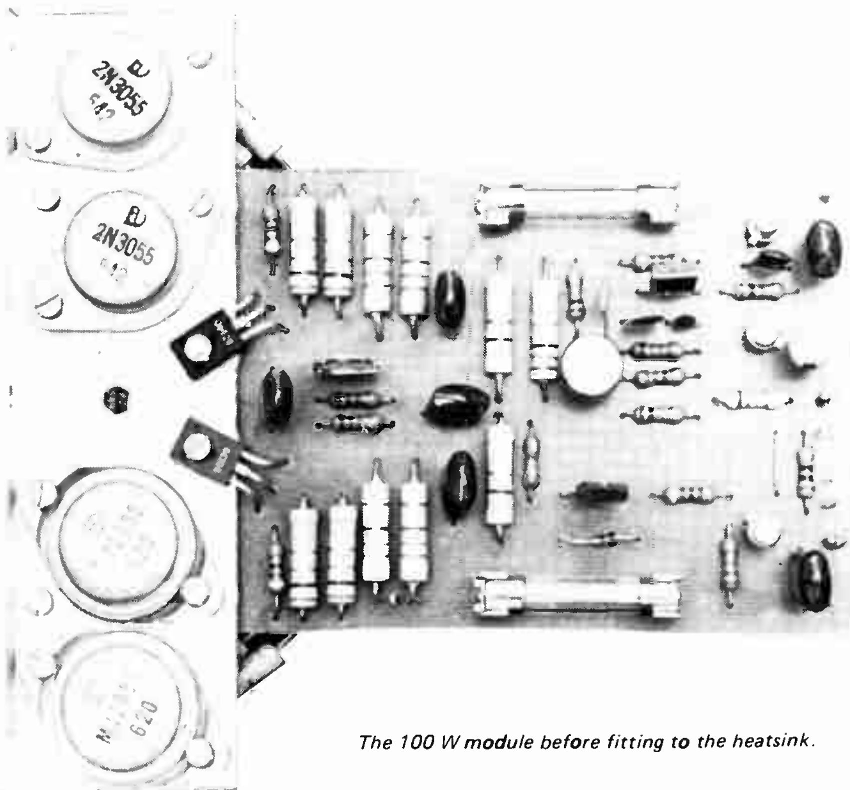
Q1	BC 548

Zener diodes

ZD1, 2	15 V 1 W

Miscellaneous

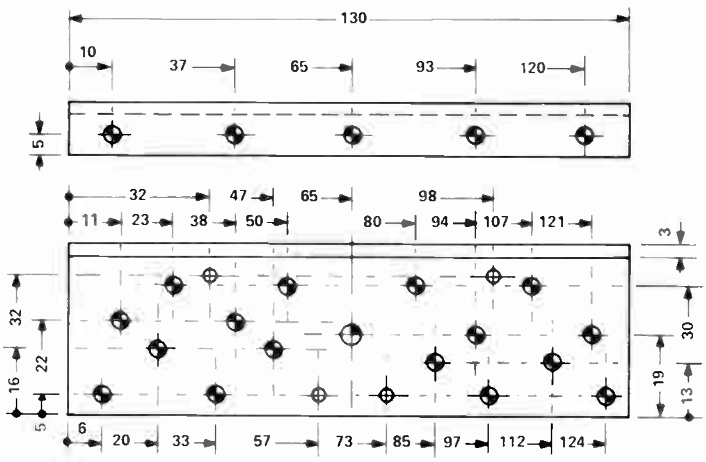
PC board – ETI 480 PS
Relay 2 pole 280 ohm coil
Transformer PF3577 or similar



The 100 W module before fitting to the heatsink.

Rear view of the 100 W module showing the links and resistor which are external to the pc board.

50 – 100 WATT AMPLIFIER MODULES



4 HOLES 3mm DIA
21 HOLES 4mm DIA
1 HOLE 5mm DIA

MATERIAL: ALUMINIUM
ALL DIMENSIONS ARE IN MILLIMETRES

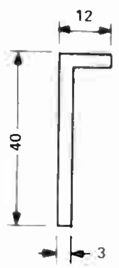


Fig. 8. Heatsink bracket for the 100 W module.

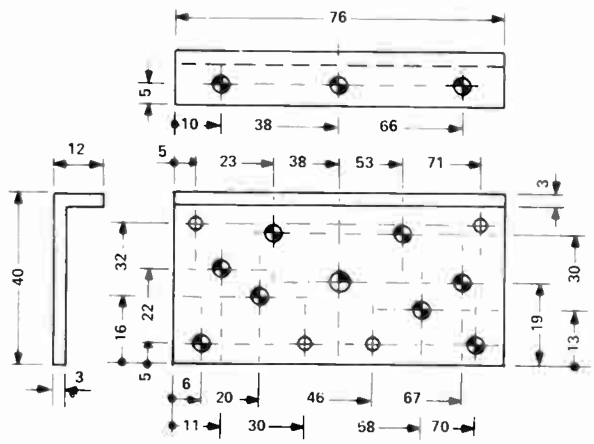


Fig. 9. Heatsink bracket for the 50 W module.

4 HOLES 3 mm DIA
11 HOLES 4 mm DIA
1 HOLE 5 mm DIA

MATERIAL: ALUMINIUM

NOTE
ALL DIMENSIONS
ARE IN MILLIMETRES

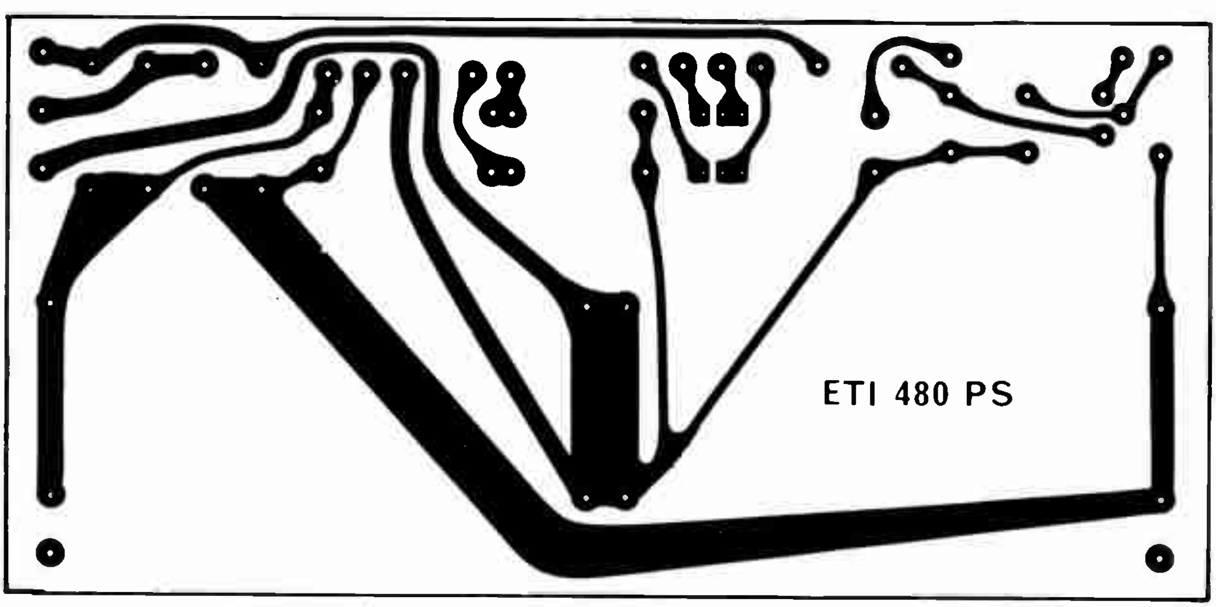


Fig. 7. Printed circuit layout of the power supply. Full size 160mm x 76mm.

12 VOLT 100 WATT AUDIO AMPLIFIER

Here's how to adapt ETI's 50/100 watt power amplifier modules for 12 volt operation. This article shows how to build the necessary power supply.

MANY POP GROUPS and a surprising number of car and van owners have asked us to design a high power amplifier energised from a 12 volt supply.

We have based the design on our 480 power modules (described on pages 102 to 108 in this book). This article describes the power supply and the following one shows how to combine all the bits and how to construct a suitable mixer input stage.

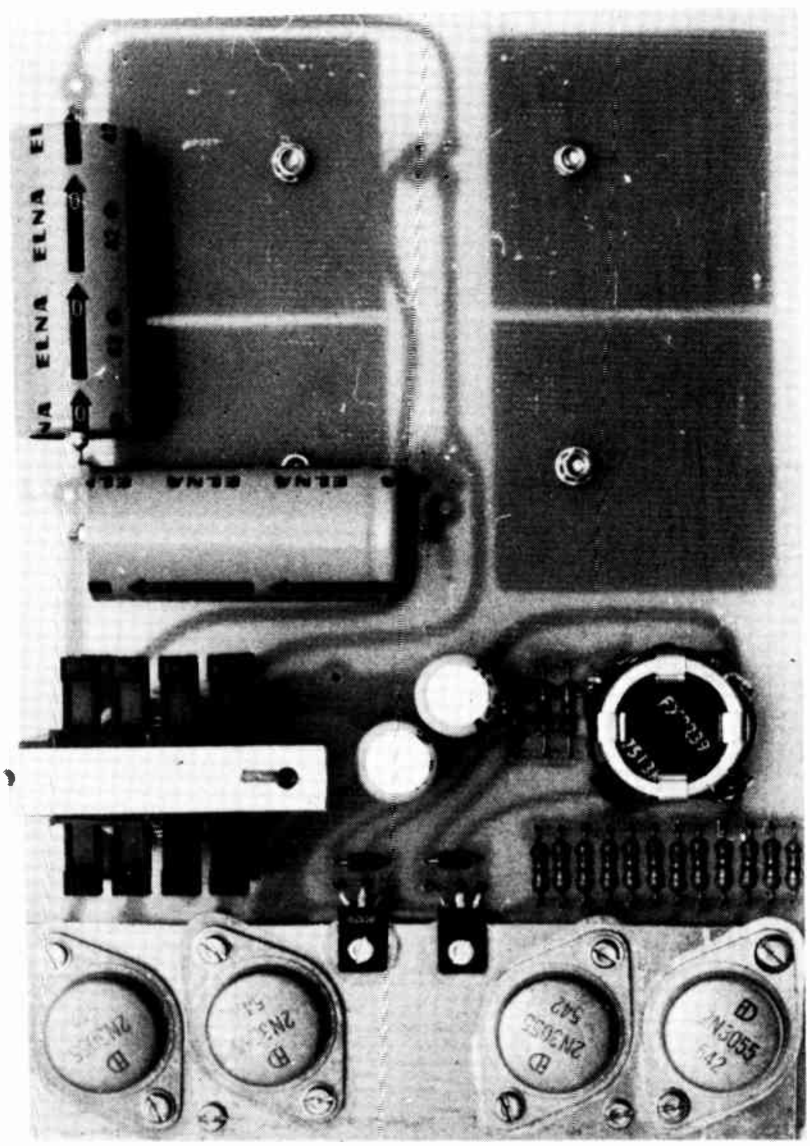
The converter described here supplies ∓40 volts required by the ETI 480 module — it can handle either a single 100 watt module or two 50 watt modules.

We have kept the converter's oscillator frequency above the audio band, and providing the unit is shielded (to prevent RF noise) there is no noise or interference from the converter.

When operating any amplifier off a battery supply, it must be remembered that the power consumption can be high. For 100 watts sinewave, the amplifier needs 150 watts input (assuming 66% efficiency) and with a converter efficiency of 80%. This gives an input power of about 180 watts. That's 15 A from a 12 V supply. With music waveforms this does, however, drop to 8-10 A but even this can quickly flatten a normal car battery.

Design Features

With an inverter of this power there are a number of problems to be solved at the design stage. The simplest type of inverter is the self-oscillating feedback type which also has the advantage that it is short-circuit proof. However, it usually operates at about 2 kHz and the noise generated can be annoying for audio use. This type can be made to oscillate at over 20 kHz but high speed transistors must be used (which are expensive).



12 VOLT 100 WATT AUDIO AMPLIFIER

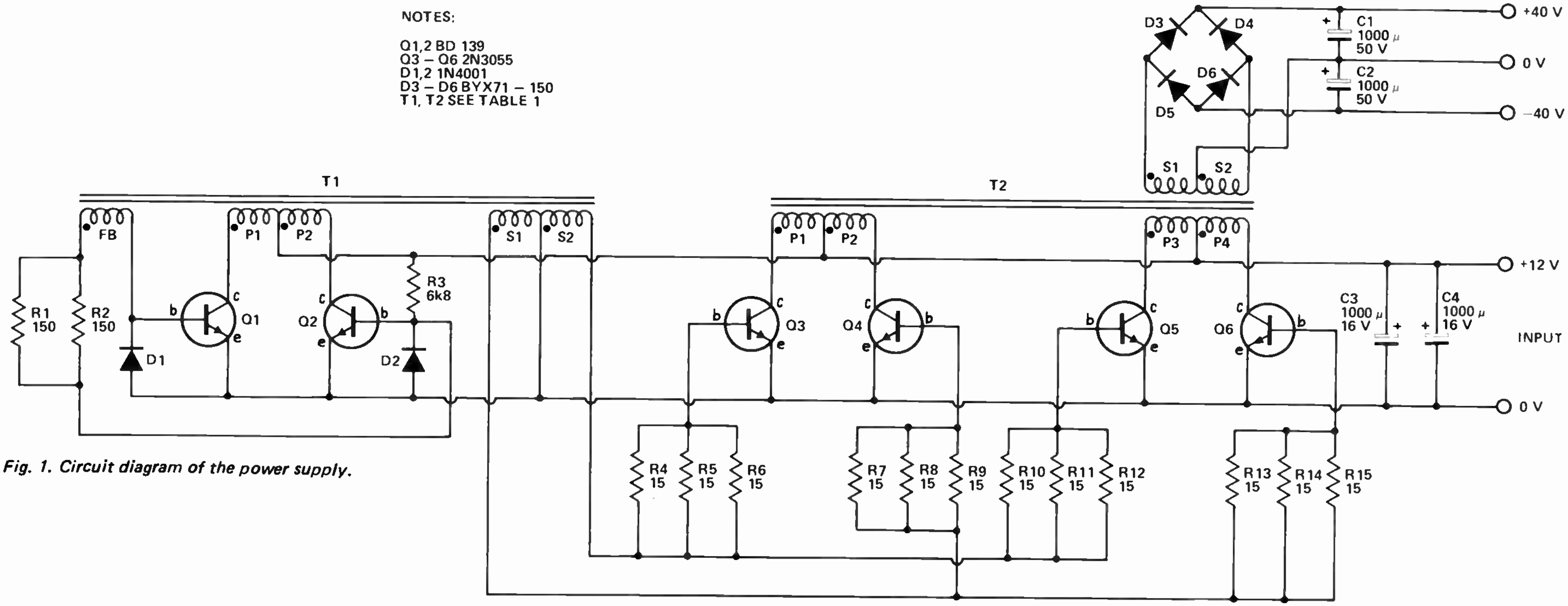
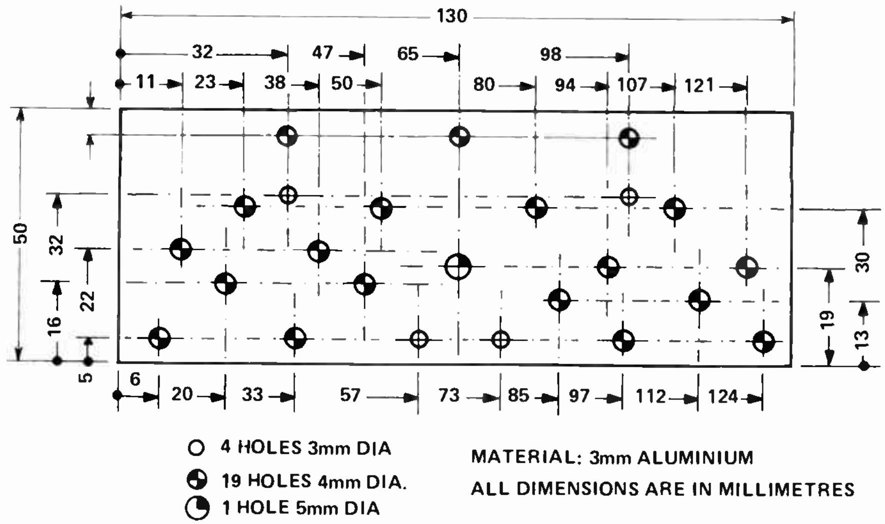


Fig. 1. Circuit diagram of the power supply.

Fig. 2. Heatsink bracket. The unit must be bolted onto a heatsink before use.



How it Works – ETI 481 PS

The transformer T1 and transistors Q1 and Q2 form a self-oscillating oscillator at about 25 kHz (with a 14V input). The secondary of this transformer is used to drive the base circuit of the main transistors, Q3-Q6. Resistors R4-R15 limit the current into the base to about 0.5 A. The transformer T2 has two primaries, each connected to separate transistors, providing the necessary current sharing. The secondary of T2 is rectified by D3-D6, which are high speed diodes, and filtered by C1 and C2.

When measuring the output voltage a minimum load of about 30 mA is needed. When driving an amplifier the quiescent current is enough to provide this minimum current.

PARTS LIST – ETI 481 PS

Resistors all ½ W 5%

R1,2	150 ohm
R3	6k8
R4-R15	15 ohm

Capacitors

C1,2	1000 µ 50 V electro
C3,4	1000 µ 16 V electro

Semi-conductors

Q1,2	Transistors BD139
Q3-Q6	Transistors 2N3055
D1,2	Diodes 1N4001
D3-D6	Diodes BYX71-150

Miscellaneous

T1	Transformer see Table 1.
T2	Transformer see Table 1.
PCB	ETI 481 PS

Heatsink bracket to Fig. 2.

We looked at the special ICs designed for use in switching-mode power supplies (a similar application) as voltage regulation. However, if normal transistors (2N3055) are to be used a complex control circuit is required to ensure the transistors turn off quickly enough to prevent overheating. The main requirement in turning the transistor off is that the base must be reverse biased. To do this up to one amp must be taken out of the base (only for about 2 µs) to reduce the storage time of the transistor. If this is done the 3055 will turn off in about 2 µs (where simply removing base current means a 5 µs turn off time).

The design finally settled on is a dual transformer type. This uses a low-power self-oscillating inverter driving the main inverter, using 3055s in parallel to handle the current. As high-speed low-power transistors are available (BD139) there was no problem in obtaining 20-25 kHz operation.

The other problem of operating at over 20 kHz occurs when you rectify the output. Normal rectifier diodes conduct for about 5 µs after being reverse-biased — this causes high power dissipation. We tried 1N4004 diodes at 1 A and they lasted about 30 seconds! High-speed diodes are available but they are generally expensive and/or on long delivery.

We finally settled on some Philips types (BYX71) which are reasonably priced and available (at the time of writing anyway). They are 7 A, 150 V devices, mounted on suitable heatsink, and they turn off in less than 0.5 µs, greatly reducing power dissipation.

To filter this frequency only small capacitors are needed, but due to the variable load of the audio amplifier (peak currents of about 8 amps) we used 1000 µF (to smooth the load and stop the high peaks overloading the converter).

TABLE 1 Transformer Winding Details

Transformer T1

Core 2 x FX2239 21 mm A5
Former 1 x DT2204
Clip 4 x DT2362
Ring 1 x DT2361
Base 1 x DT2364

Windings	Start	Finish	Turns	Gauge	Notes
Sec 1	4	6	2	0.8 mm	Bifilar
Sec 2	6	5	2	0.8 mm	wound
Prim 1	2	1	8	0.5 mm	Bifilar
Prim 2	1	3	8	0.5 mm	wound
Feedback	8	7	5	0.5 mm	

Transformer T2

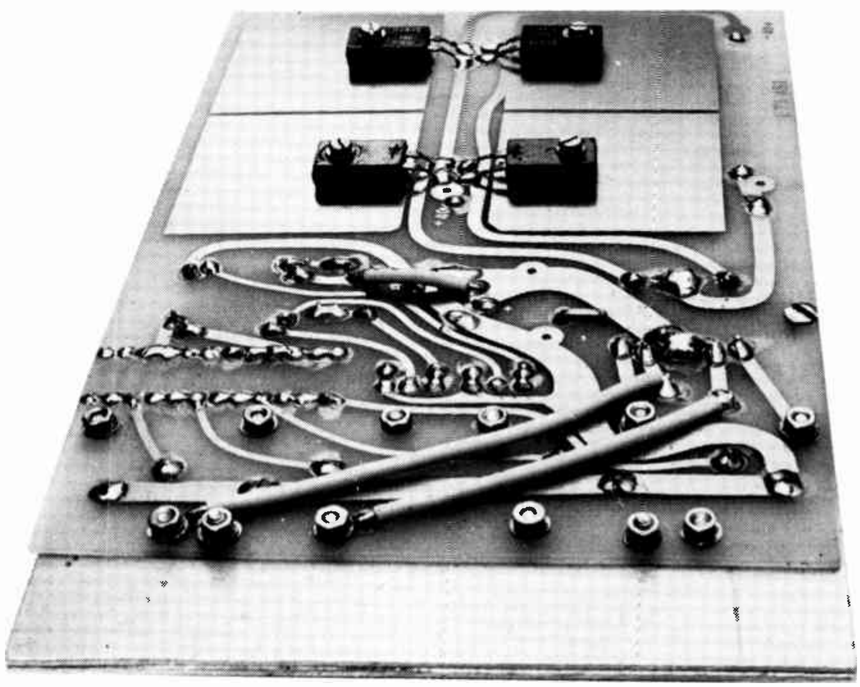
Core 2 x FX3730
Former 1 x DT2730

Winding	Start	Finish	Turns	Gauge	Notes
Prim 1			5	1.25 mm	
Prim 2	SEE OVERLAY		5	1.25 mm	QUADFILAR
Prim 3			5	1.25 mm	wound
Prim 4			5	1.25 mm	
SEC 1	SEE OVERLAY		16	1.0 mm	Bifilar
SEC 2	*		16	1.0 mm	wound

** Secondary is on opposite side to primary. All windings come out the one end. To allow the wires to come out at the one end the lugs marked 2,3,6 and 7 should be broken off on the former.*

TABLE 2

ALL RESULTS WITH 12.6V INPUT		
OUTPUT POWER WATTS	OUTPUT VOLTAGE VOLTS	INPUT CURRENT AMPS
0	80	1
1.5	68	1.1
25	67	3
50	65	5
75	63	7.2
100	61	10
125	59	12.5



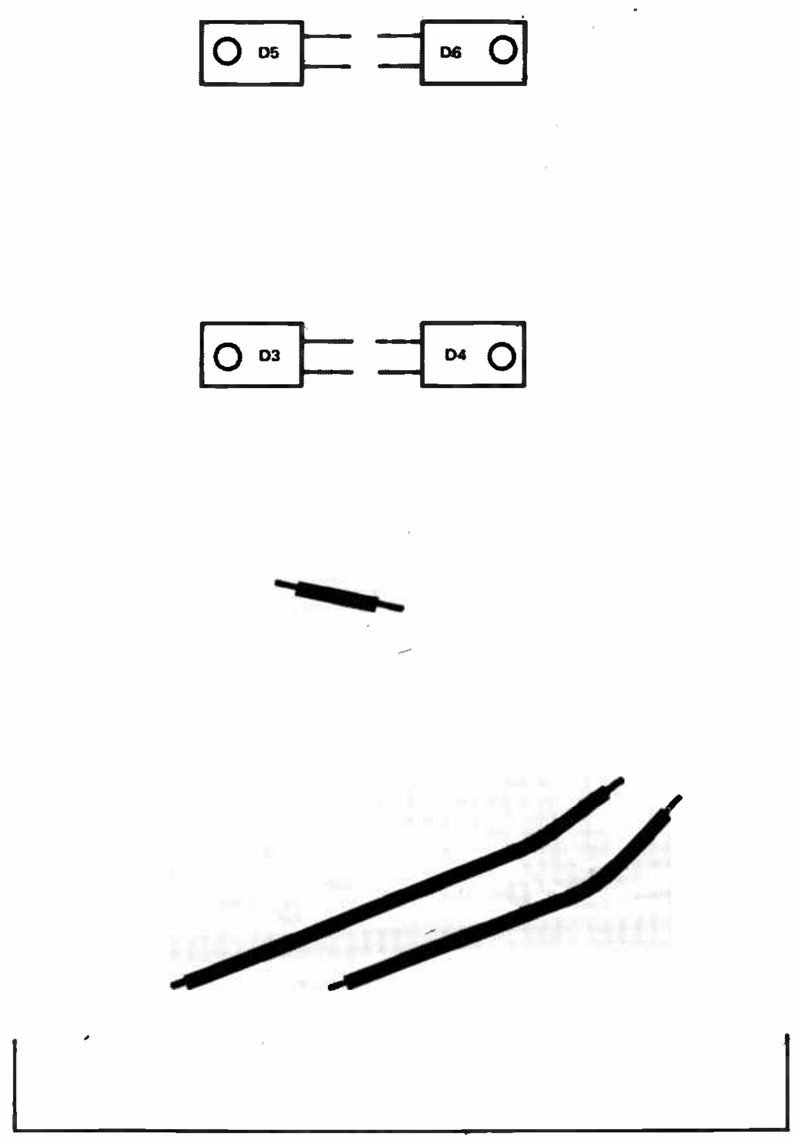


Fig. 3. Diagram showing position of the links and diodes D3-D6.

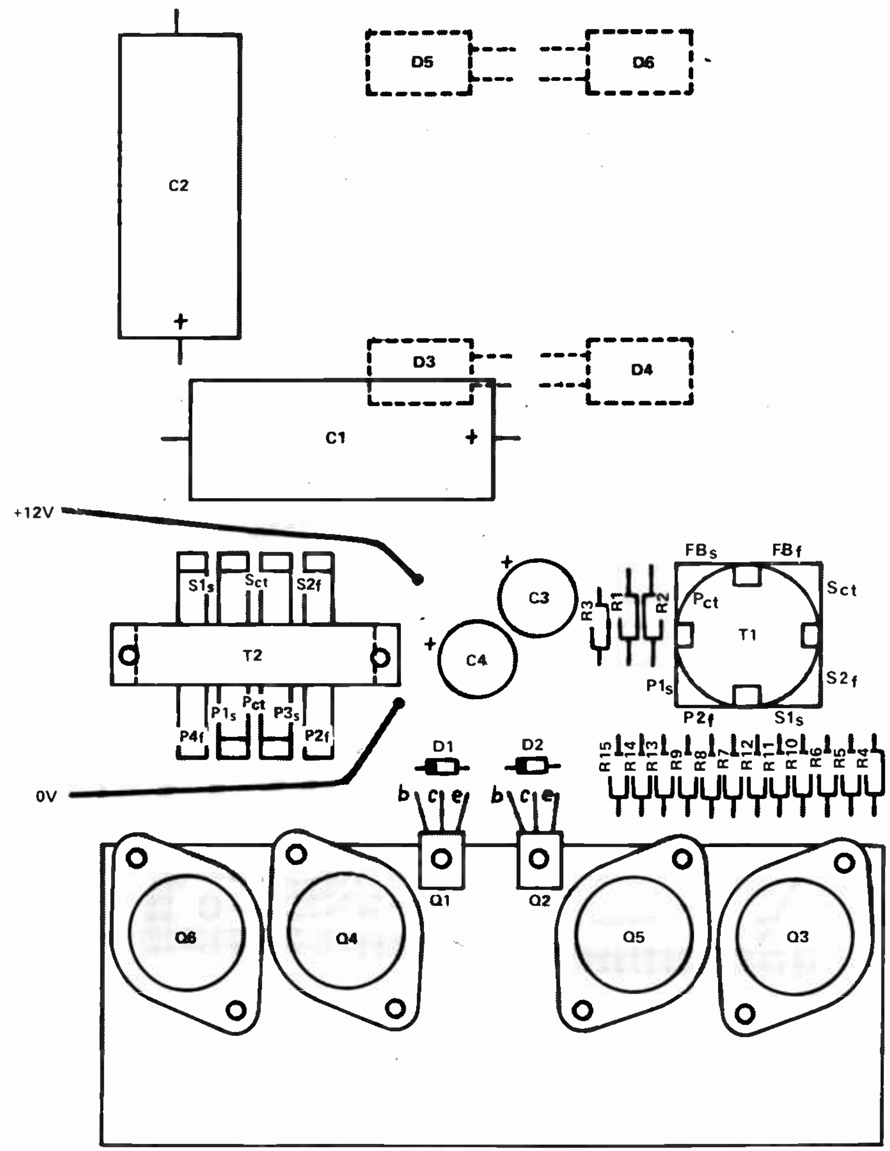


Fig. 4. Component overlay.

Construction

Except for the winding of the transformers, construction is simply assembling a pc board. If the transformers are not available ready-wound they can be made from the details in Table 1. Note that there are not many turns but they are wound using a reasonably heavy gauge of wire. One turn short or too many on one winding could damage the main transistors.

When assembling the pc board note that D3-D6 are on the copper side and the metal surfaces are bolted in contact with the pc board which then acts as a heatsink. Also there are a few links on the copper side of the board as it was not possible to get the tracks on the pc board wide enough to carry the currents.

The unit, when assembled onto the heatsink bracket, must be mounted onto a heatsink similar to the one used in the amplifier module.

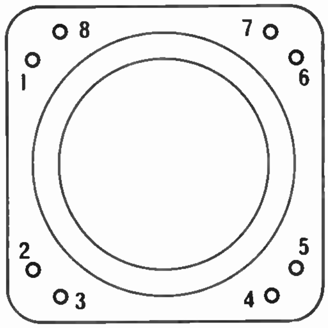


Fig. 5. The pin numbering sequence for T1.

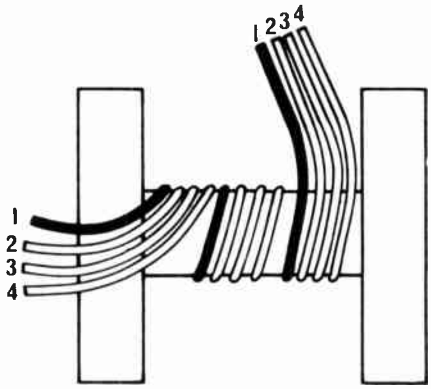
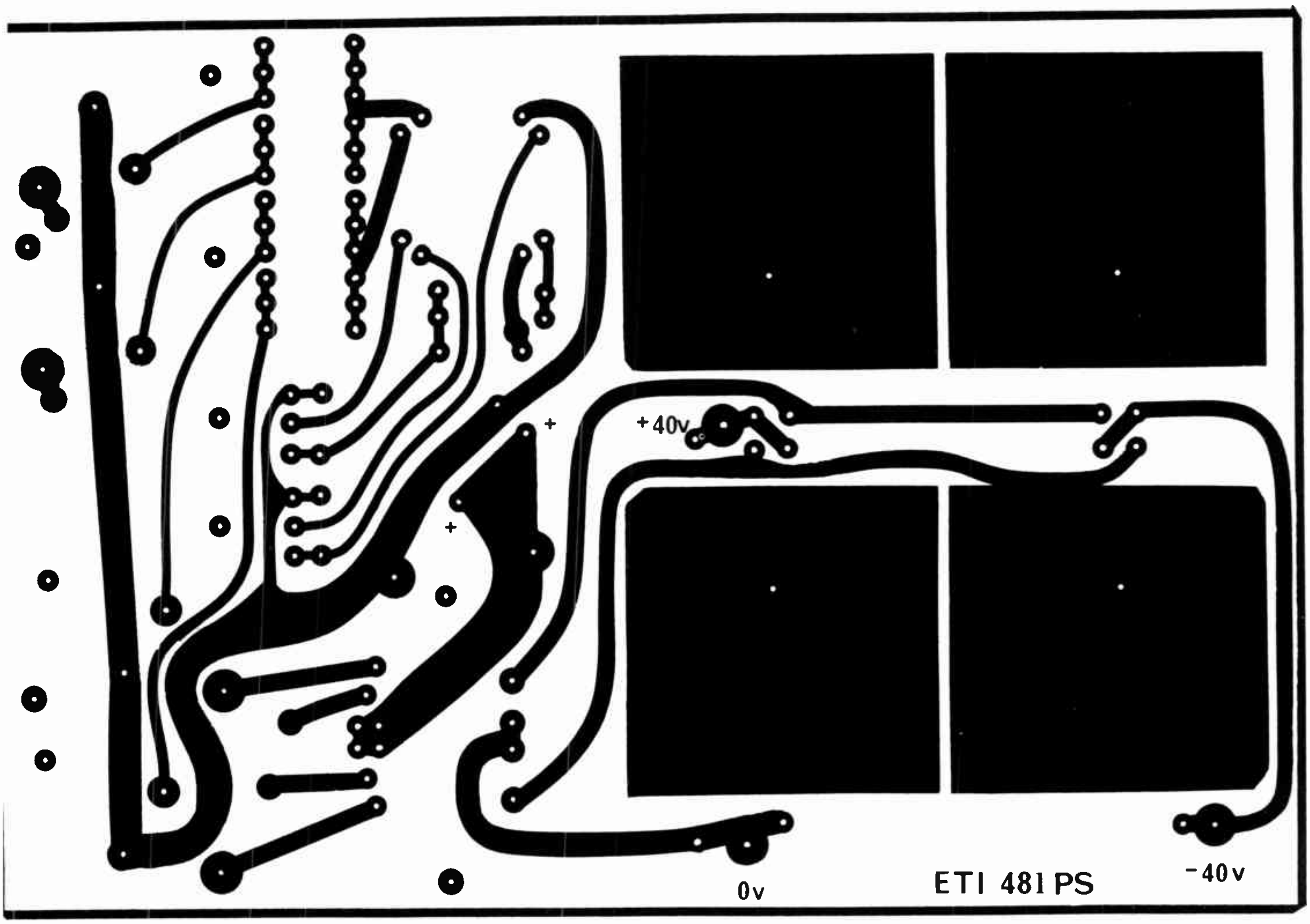


Fig. 6. Diagram showing the start of the quadfilar primary winding of T2. Note that it must be wound this way to reduce power consumption and improve regulation.

Fig. 7. Printed circuit layout. Full size 188 x 129 mm.



HIGH POWER PA / GUITAR AMPLIFIER

Revised version of the ETI 413 guitar amplifier uses our new ETI 480 power amplifier with a new tone control/mixer board and any of our existing preamplifier modules. It can be powered either by the mains or 12 volt dc.

AS THE ETI 480 amplifier module is so much easier to build than the 413 guitar amplifier and has the same power rating we have had many requests to describe it as a complete amplifier.

Although this article does not specifically describe that unit, it shows how to interconnect modules which are all described in this book, to make an amplifier to suit your particular need. The new mixer board has two out-of-phase outputs to allow two 100 watt modules to be connected in a bridge configuration to give 200 watts into eight ohm loads.
give 200 watts into eight ohm loads.
Note however that loads less than eight ohms cannot be used while in bridge and if the load can be broken into two separate four ohm loads this should be done and the amplifiers used separately.

SPECIFICATION ETI 481

Unit built using two 100 watt modules connected in bridge with ETI 449 balanced microphone preamplifiers.

Output power	
into 8 ohms at < 1% distortion	200 watts
Frequency response	
controls flat, +0dB, –3dB	15Hz – 30kHz
Tone control range	
bass at 100Hz	+10dB –11dB
treble at 10kHz	+11dB –12dB
Sensitivity	3mV
Maximum input voltage *	500mV

* limit due to preamplifier clipping.

Constructional Hints

Due to the various requirements of different groups we are not giving formal constructional details but will just give general procedures and things to watch for.

The first thing to decide is what arrangment of modules is needed. The various modules available are as follows;

Power modules

ETI 480/50W	50W into 8 ohms
ETI 480/100W	100W into 4 ohms
2xETI480/100W	200W into 8 ohms

Power supply

1xTransformer	240V in 100W out
2xTransformer	240V in 200W out
1x ETI 480PS	12Vdc in 100W out
1x ETI 480PS	
+1 Transformer	12V/240V input 100W output

Mixer / tone controls
ETI 481M

Preamplifiers

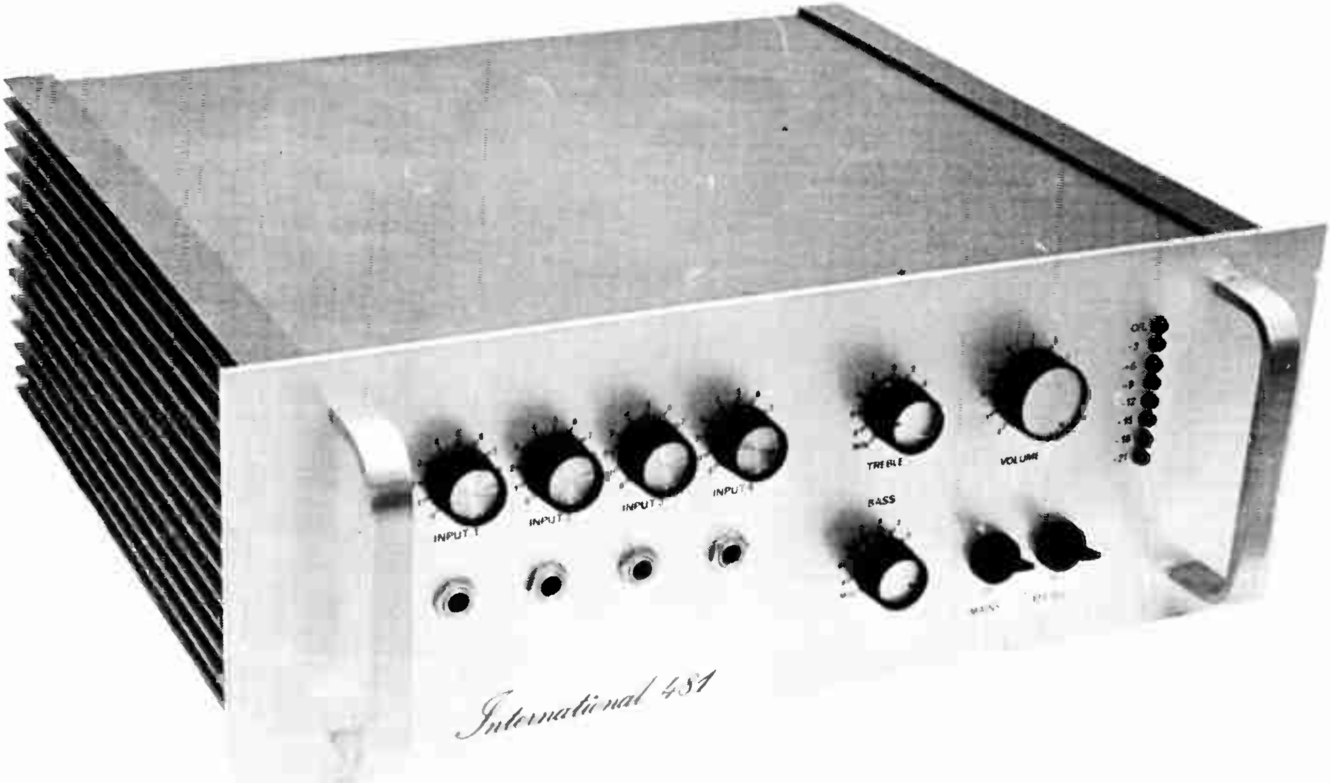
ETI 445	two channel preamp. for magnetic pickups.
ETI 446	two channel limiter suitable for high level microphones.
ETI 449	balanced mic. amplifier for low level microphones up to 500mV max.output

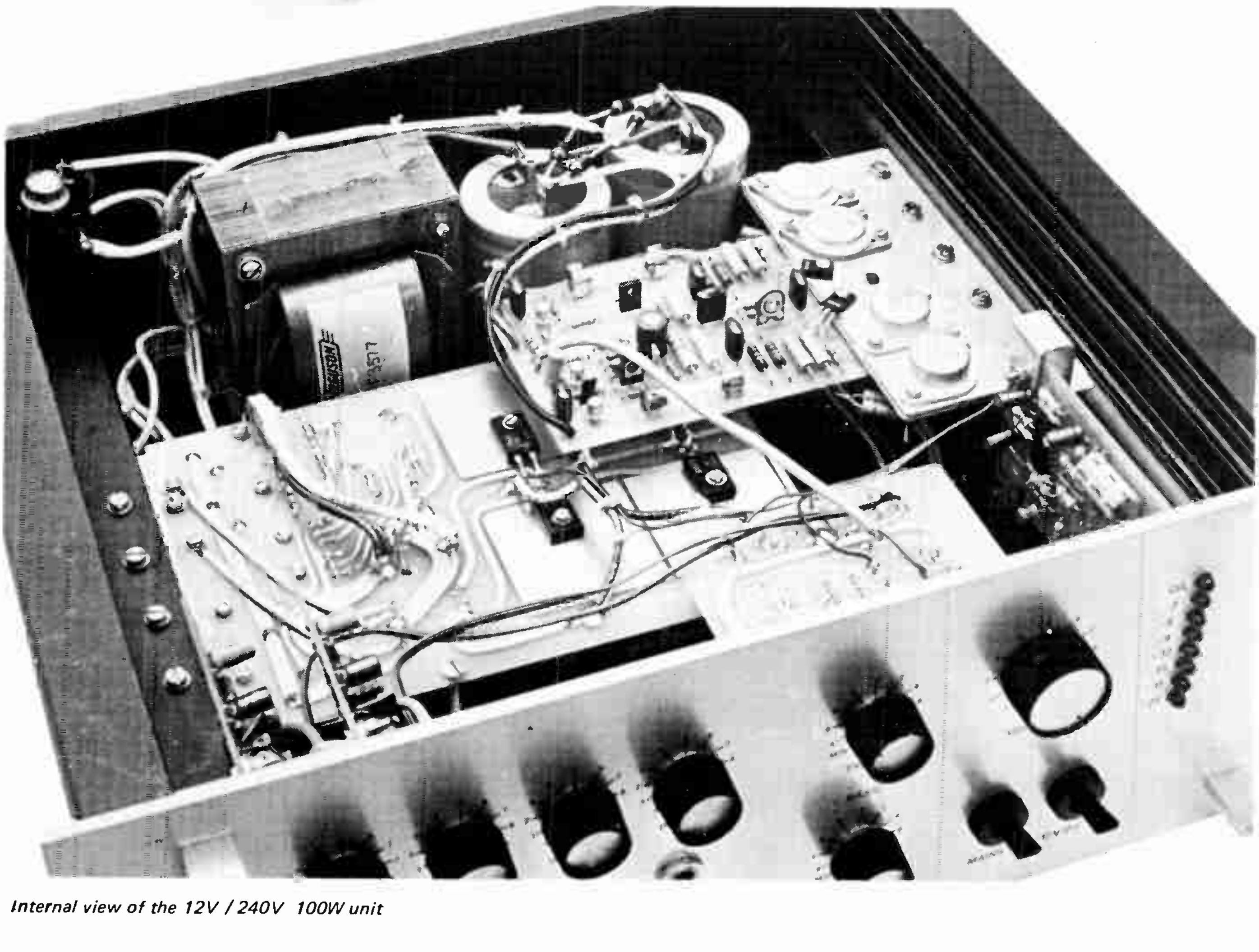
Options
ETI 438 audio level meter

The box we used was supplied to us by Applied Technology and is suitable for rack mounting. If this box is used the heatsink bracket of the amplifier(s) should be the same as described for the 12V power supply to allow it to clamp directly onto the heatsink/ sides of the box. While this box looks large, when two power modules, power supplies and the preamps are fitted it becomes more crowded. Therefore layout all the boards before drilling any holes.

With the 12V supply we used a 15A

(Text continued on page 116)





Internal view of the 12V / 240V 100W unit

HIGH POWER PA / GUITAR AMPLIFIER

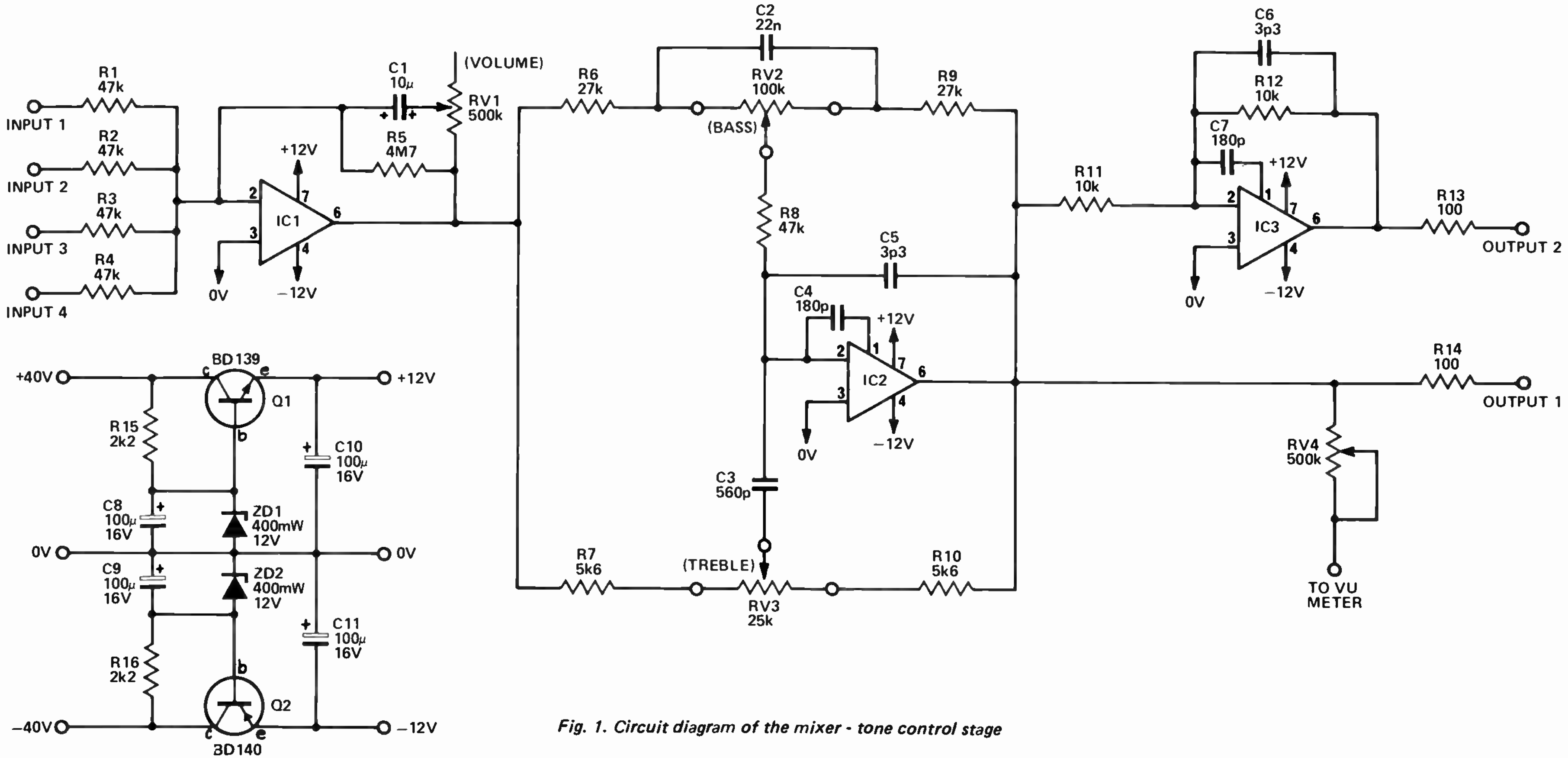


Fig. 1. Circuit diagram of the mixer - tone control stage

Continued from page 114

fuse in series and a large automotive type diode (cathode to +Ve) across the 12V after the fuse so that the fuse will blow if the supply is the wrong polarity. The output of the 12V converter and the output of the 240V rectifier can simply be paralleled allowing the two supplies to be used separately or together without damage. The wiring on the 12V side carries 15 amps and should be a reasonable gauge,

For connecting in bridge two separate transformers (or a larger one) are needed, separate rectifiers should be used and larger filter capacitors or paralleled capacitors (we used 6800µ 50V) should be used. When connecting the power modules the +40V, 0V and −40V connections between the modules should be short.

Most of the preamplifier modules need a dual 12V supply and this is available from the 481M board. In the audio level meter the diode D3 should be replaced by a 680 ohm resistor to lower the +40V to less than the 32V allowed. It should be connected as a VU meter with R5 as 220k. The unit can be calibrated on the mixer board.

We mounted the preamplifiers, mixer and VU meter on small brackets which were held on by the potentiometers. For the balanced microphone inputs we used stereo phone jacks as they are economical and also allow an unbalanced microphone to be used. The level control potentiometers on the microphone inputs were 10k log and are on the output of the preamplifier.

We had some hum pickup into the mixer board from the wiring to the power switch and we have moved it more to the right since the photos were taken to help this problem.

How It Works — ETI 481M

The mixing is done by IC1 where the gain is set by RV1 and is variable from zero to 10 (20dB) and for dc biasing R5 is used. Due to the high value of R5 we have used a FET op-amp (CA3140) for IC1. The tone controls are a standard network around IC2. The output of IC2 is inverted by IC3 to give the second output needed. The frequency compensation used on IC2 and IC3 is called 'feed forward' and extends the frequency response of the IC.

The power supply reduces the ± 40V to ± 12V and has enough power to drive the preamplifiers as well.

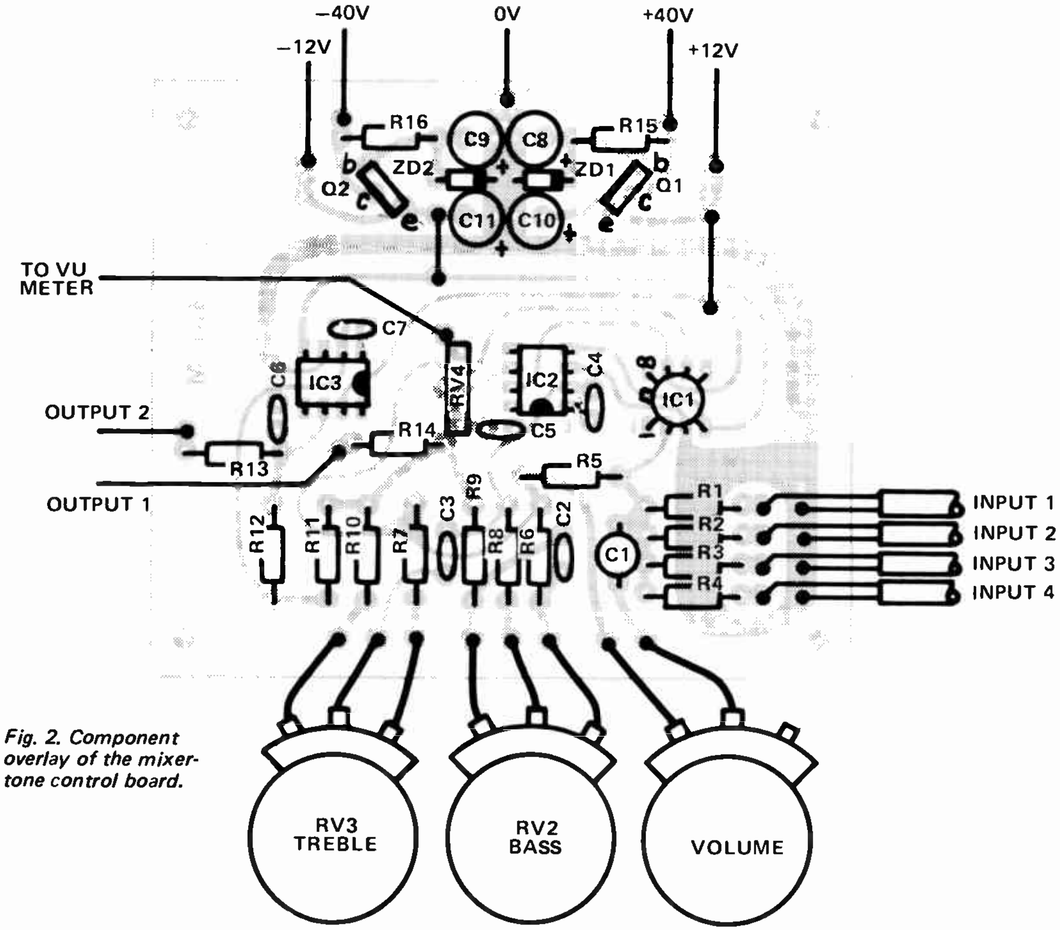


Fig. 2. Component overlay of the mixer-tone control board.

PARTS LIST – ETI 481M

Resistors all ½W 5%		Potentiometers		Semiconductors	
R1–R4	47k	RV1	500k log rotary	IC1	CA3140
R5	4M7	RV2	100k lin "	IC2,3	LM301A
R6	27k	RV3	25k lin "		
R7	5k6	RV4	500k trim	Q1	BD139
R8	47k			Q2	BD140
		Capacitors			
R9	27k	C1	10µ non polarised electro	ZD1,2	12V 400mW
R10	5k6	C2	22n polyester		
R11,12	10k	C3	560p ceramic	**Miscellaneous**	
R13,14	100	C4	180p "	PC board ETI 481M	
R15,16	2k2	C5,6	3p3 "		
		C7	180p "		
		C8–C11	100µ 16V electro		

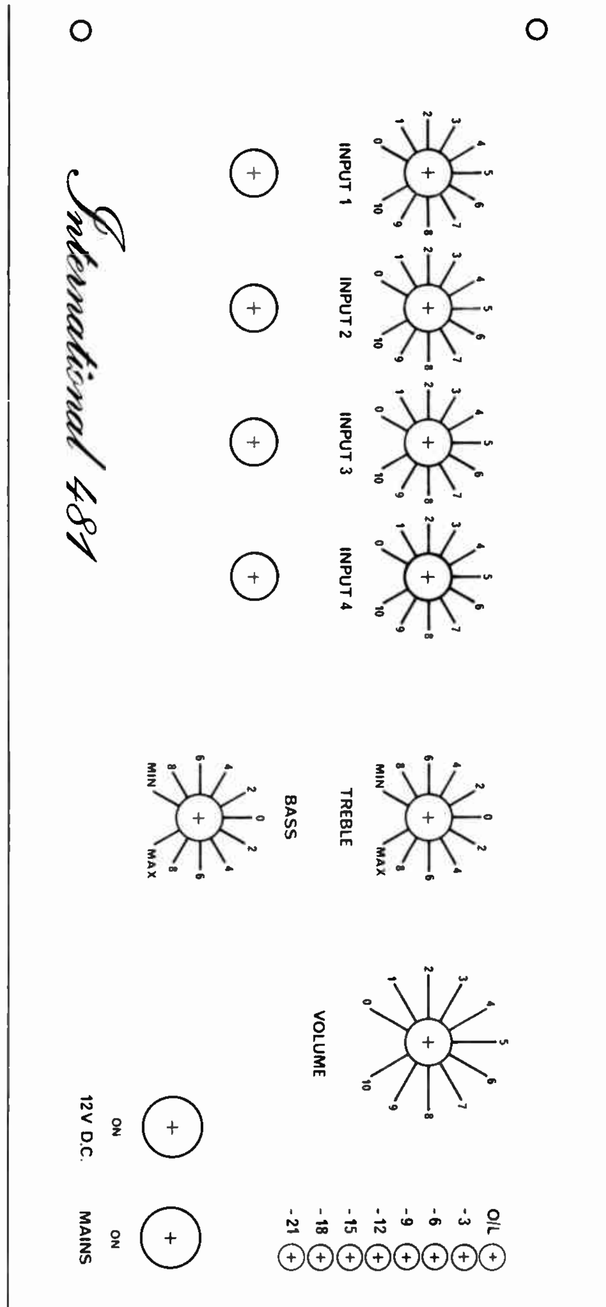


Fig. 3. Front panel artwork. Full size 375 x 135 mm. Note that the full length is not shown here.

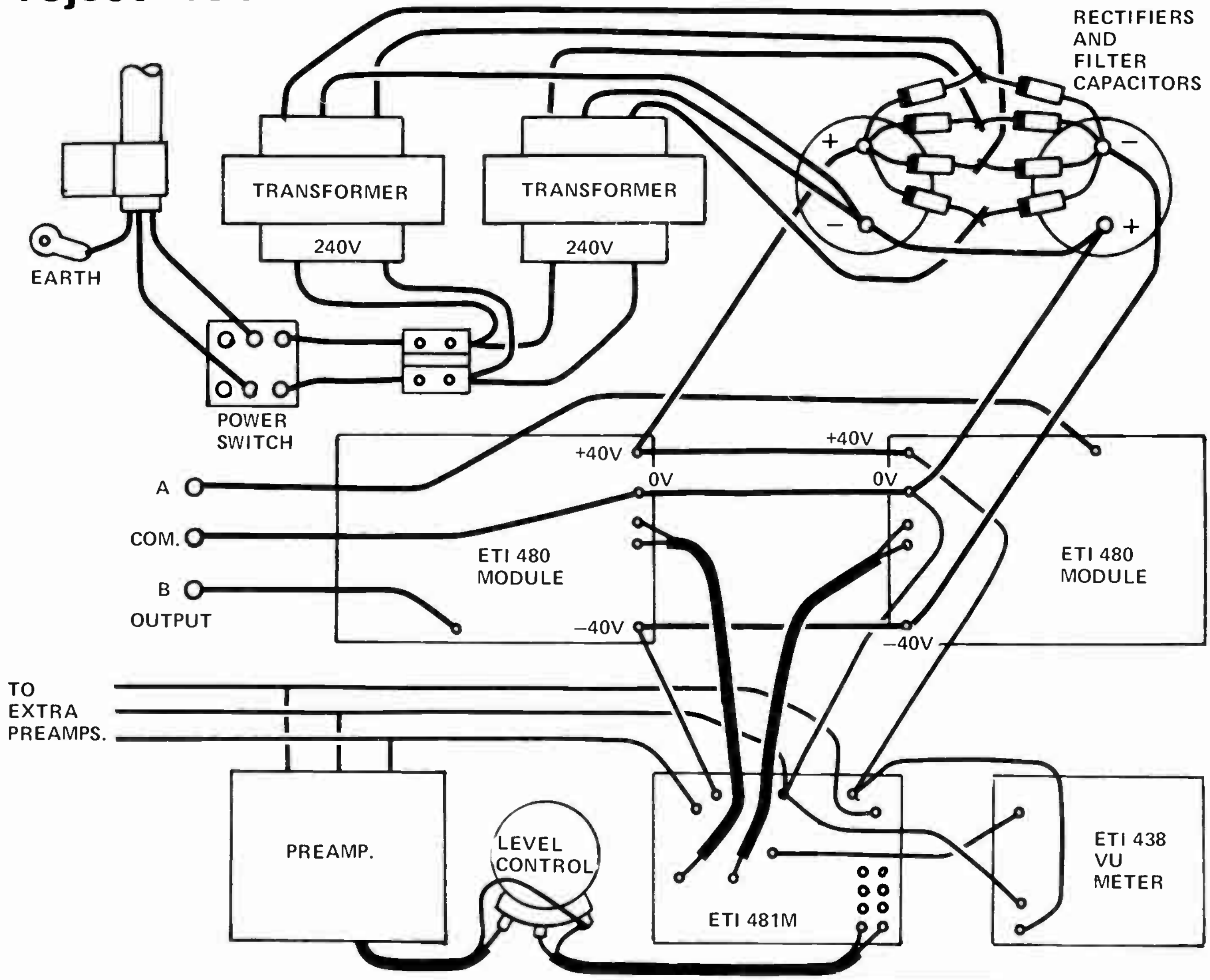


Fig. 4. Wiring diagram for the 200W bridge amplifier.

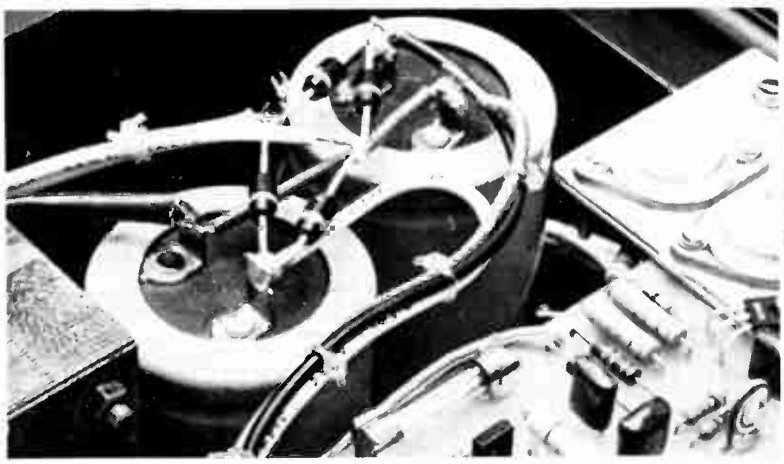


Photo showing how the rectifier diodes are connected. This is on the 12V / 240V version where a single transformer and rectifier are used.

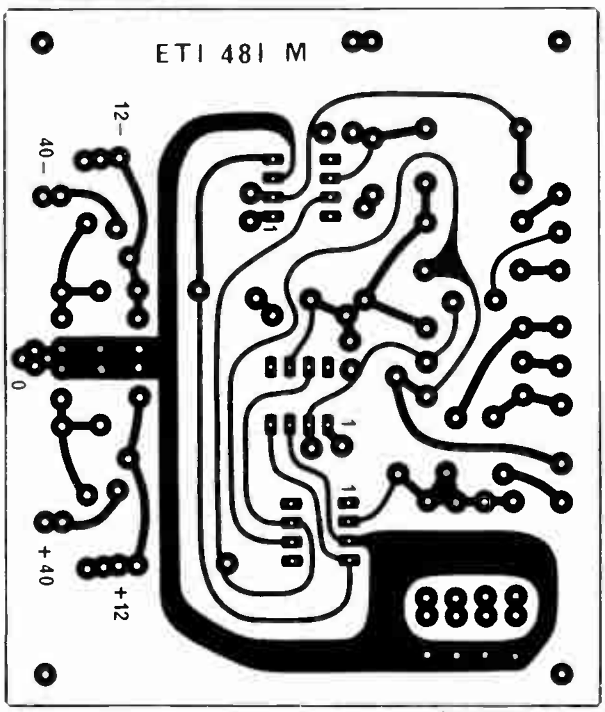


Fig. 5. Printed circuit layout. Full size 92 x 78 mm.

GRAPHIC EQUALIZER METALWORK (continued from page 101).

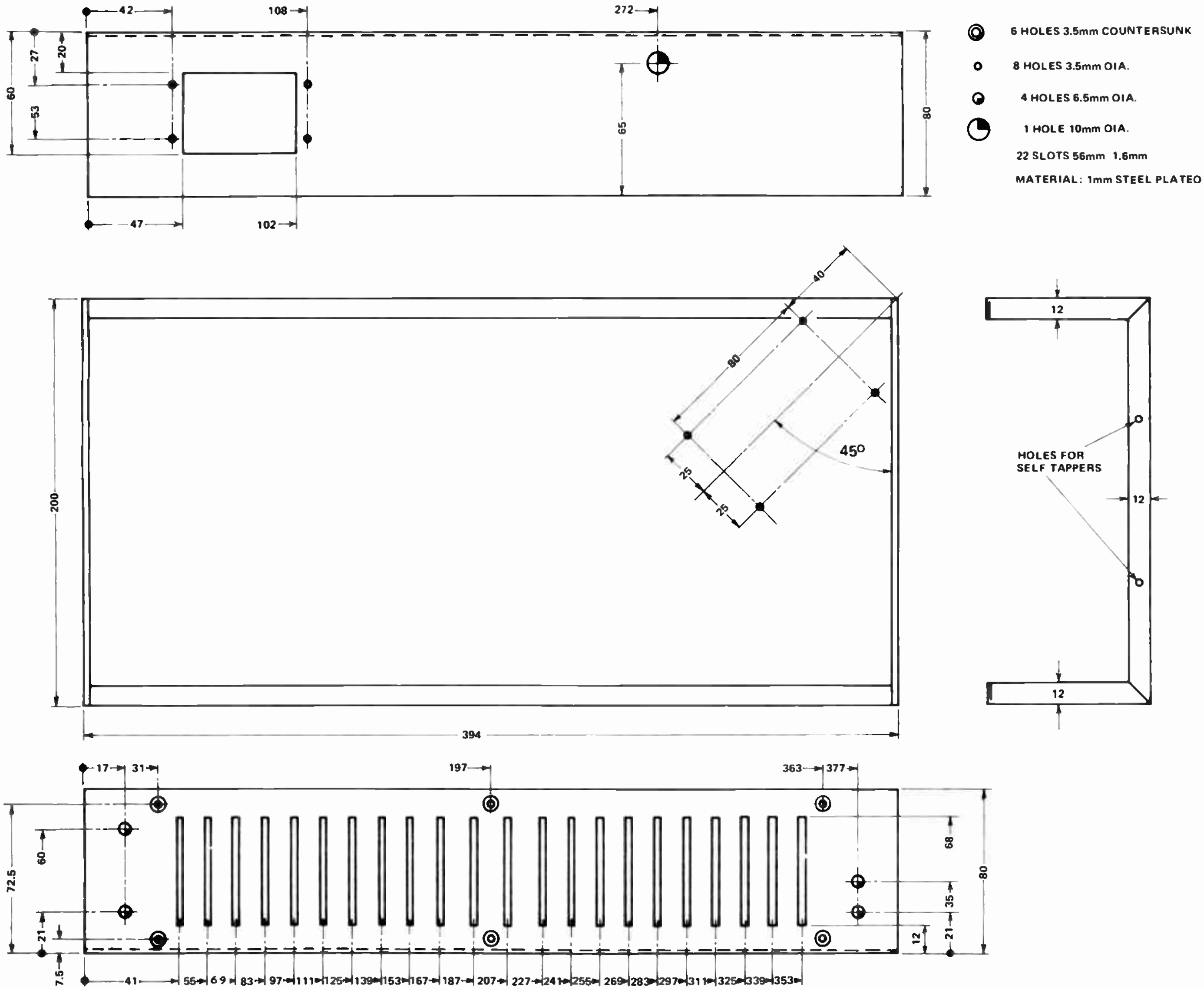


Fig. 7. Details of the chassis.

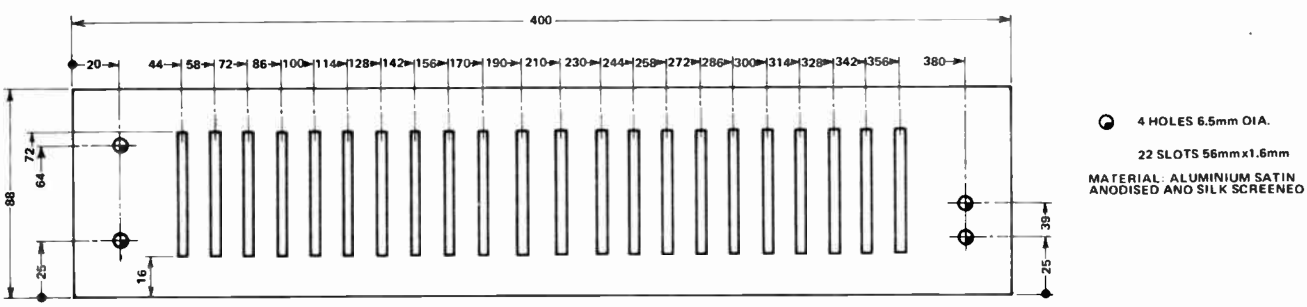


Fig. 8. Metalwork details of the front panel.

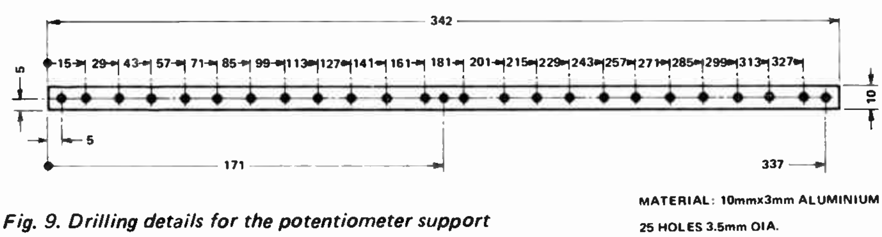


Fig. 9. Drilling details for the potentiometer support

LINE AMPLIFIER

PROJECT 430

Boost microphone output with this low noise amplifier.

The completed line amplifier. Note the use of Cannon plugs and the gain switch on the side.

MODERN high quality microphones are low impedance units having a very low output voltage. To minimize noise, picked up on long leads, it is usually necessary to use special balanced and screened leads together with balancing transformers. An alternative approach is to use a low noise amplifier to boost the signal *before* passing it down the cable. The ETI 430 line amplifier, described here, is intended for this purpose.

Such a unit, when used with the ETI Master Mixer provides either 20 or 40 dB of gain prior to the mixer. This allows the mixer to be used on the low-sensitivity range. Thus the larger signal now available, effectively over-rides the inherent noise of the first amplifier in the mixer.

The overall effect of using such an amplifier is to vastly improve the signal-to-noise ratio of the particular microphone channel and to eliminate the need for an expensive balanced and screened cable and balancing transformer.

To reduce the possibility of mains – hum pickup we have used a small nine volt battery to power the unit. Since the current drawn is a mere 0.5 mA, the battery should last about three to

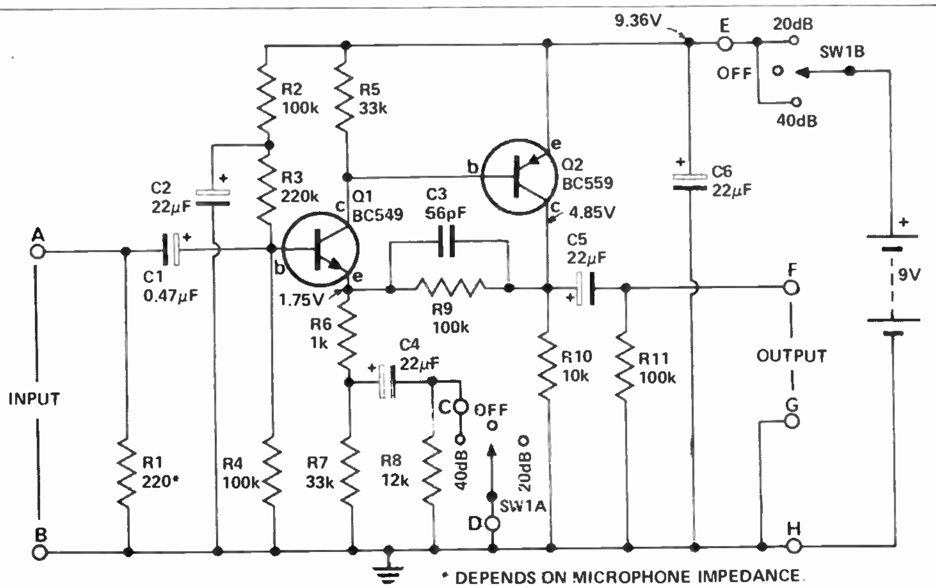
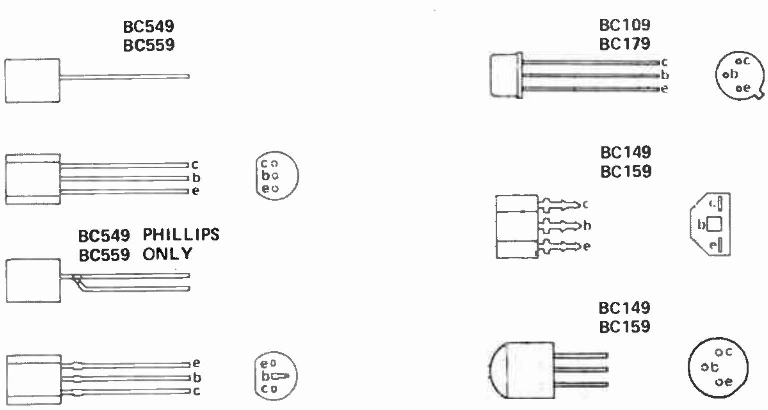
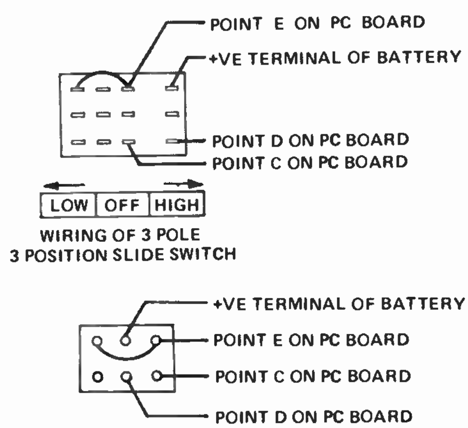


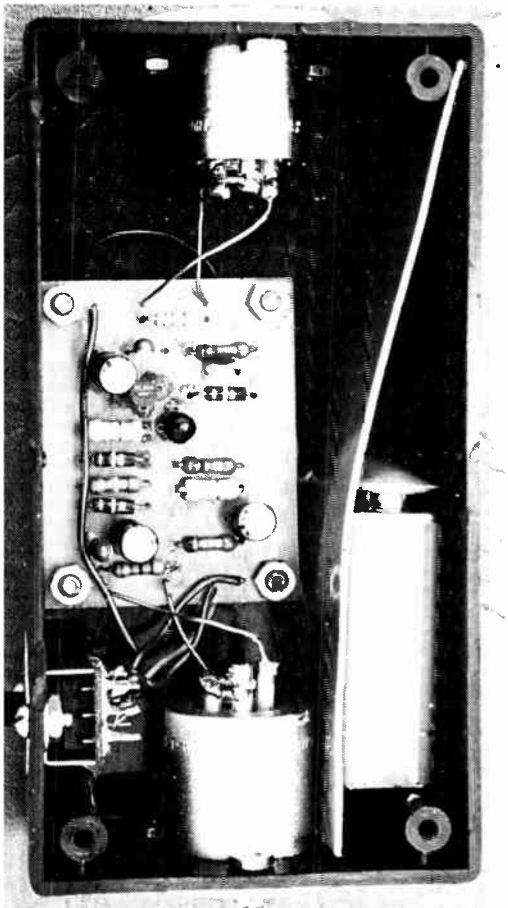
Fig.1. Circuit diagram of the microphone line amplifier. Voltages shown are typical and as measured on our prototype.





WIRING OF 3 POLE
3 POSITION SLIDE SWITCH

WIRING OF DPDT TOGGLE SWITCH
WITH CENTRE OFF POSITION



Internal layout of the line amplifier.

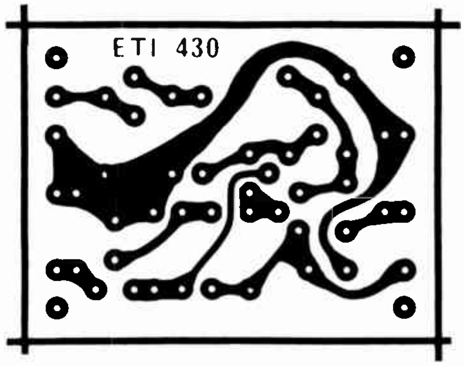


Fig.2. Printed circuit board layout for the amplifier. Full size 55 x 42 mm.

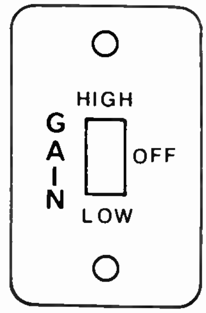


Fig.3. Artwork for the gain switch label. (Shown full size).

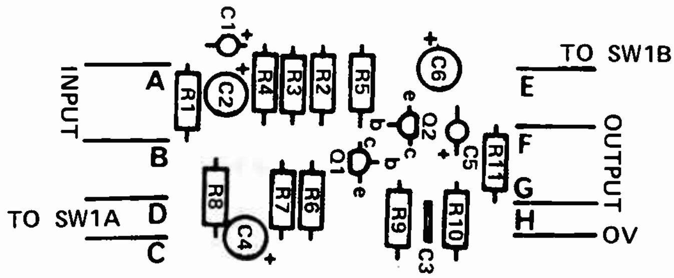


Fig.3. Component overlay. Note particularly the orientation of the transistors and electrolytic capacitors.

four hundred hours before replacement is required.

The ETI line amplifier can of course be used to great advantage with any recording equipment where low noise operation is necessary. When used with the Master Mixer the low impedance input should be used but the terminating resistor (fitted across the mixer input socket) should be removed so that a 4.7 k input impedance is obtained.

CONSTRUCTION

The circuit is not critical in any way hence, practically any construction method may be used. However, the use of the printed circuit board specified will considerably simplify construction.

We used an unbreakable plastic box (polycarbonate) to house our unit but if the unit is to be used in the proximity of power cables etc it would be advisable to mount the unit in a metal box (diecast or similar). This is especially so if an input impedance above 1 k is to be used as the higher the impedance the more likely is hum pickup.

If Cannon plugs are used, as in our prototype, pins 1 and 2 should be linked and used as the earth line. Pin 3 is then used as the active line.

(Continued next page)

MEASURED PERFORMANCE

IMPEDANCE			
Input	selectable up to 68k max		
Output	≈ 1.5k		
GAIN			
High	40 dB		
Low	20 dB		
OUTPUT VOLTAGE			
Maximum	3 volts		
INPUT VOLTAGE			
Maximum (high range)	30 mV		
Maximum (low range)	300 mV		
FREQUENCY RESPONSE			
10 Hz – 30 kHz	+0 – 3 dB		
EQUIVALENT INPUT NOISE			
(referred to 1 mW into 600Ω)			
High Range	–110 dBm		
Low Range	– 102 dBm		
DISTORTION			
Output Voltage	100 Hz	1 kHz	6.3 kHz
300 mV	<0.1%	<0.1%	<0.1%
1 V	0.17%	0.2%	0.17%
2 V	0.5%	0.5%	0.5%
3 V	1.75%	1.8%	1.7%

HOW IT WORKS – ETI 430

The line amplifier is basically a two transistor amplifier having a selectable gain of either 20 dB (x10) or 40 dB (x 100).

The input impedance of the amplifier (referring to Fig. 1) is determined by the combined values of R1, R3 and R4 – all in parallel. The parallel impedance of R3 and R4 is 68 k and this is therefore the upper limit of input impedance ($R = \infty$).

For impedances less than 5 k the values of R3 and R4 may be ignored and R1 is set to the same value as the desired input impedance. Hence the circuit as shown matches microphones having 200 ohm output impedance.

Resistor R2, in conjunction with R3 and R4 determines the dc bias for transistor Q3 whilst capacitor C2 decouples the input bias network from any supply rail noise. The output of Q2 is fed back to the emitter of Q1 thus providing negative feedback which as well as supplying a dc bias, sets the ac gain of the stage.

The gain of the amplifier may be calculated using the following formula (assuming ideal transistors).

$$\text{Gain} = \frac{R9 + R6 + (R7//R8)}{R6 + (R7//R8)}$$

Thus for R8 = 12 k the gain is 11.2 or 21 dB. For R8 = 0 the gain is 101 or 40 dB. The actual gain obtained is slightly lower than this due to the finite betas of the transistors used.

The value of capacitor C3 determines the upper 3 dB point of 30 kHz whilst capacitors C1, C4 and C5 all give individual break points at the low end of 5 Hz, 7 Hz and 1.5 Hz respectively.

PARTS LIST – ETI 430

R1	resistor	selected to suit input impedance
R2,4, 9,11	"	100 k ¼W 5%
R3	"	220 k " "
R5,R7	"	33 k " "
R6	"	1 k " "
R8	"	12 k " "
R10	"	10 k " "
C1	Capacitor	0.47µF 25V TAG
C2,4,5,6	"	22µF 16V electrolytic
C3	"	56pF ceramic
Q1	Transistor	Bc109, BC149,BC549,
Q2	"	BC179, BC159, BC559
SW1	Switch	2 pole 3 position slide or 2 pole centre off toggle

PC board ETI-430

Cannon sockets = XLP-3-13 female
= XLP-3-14 male.

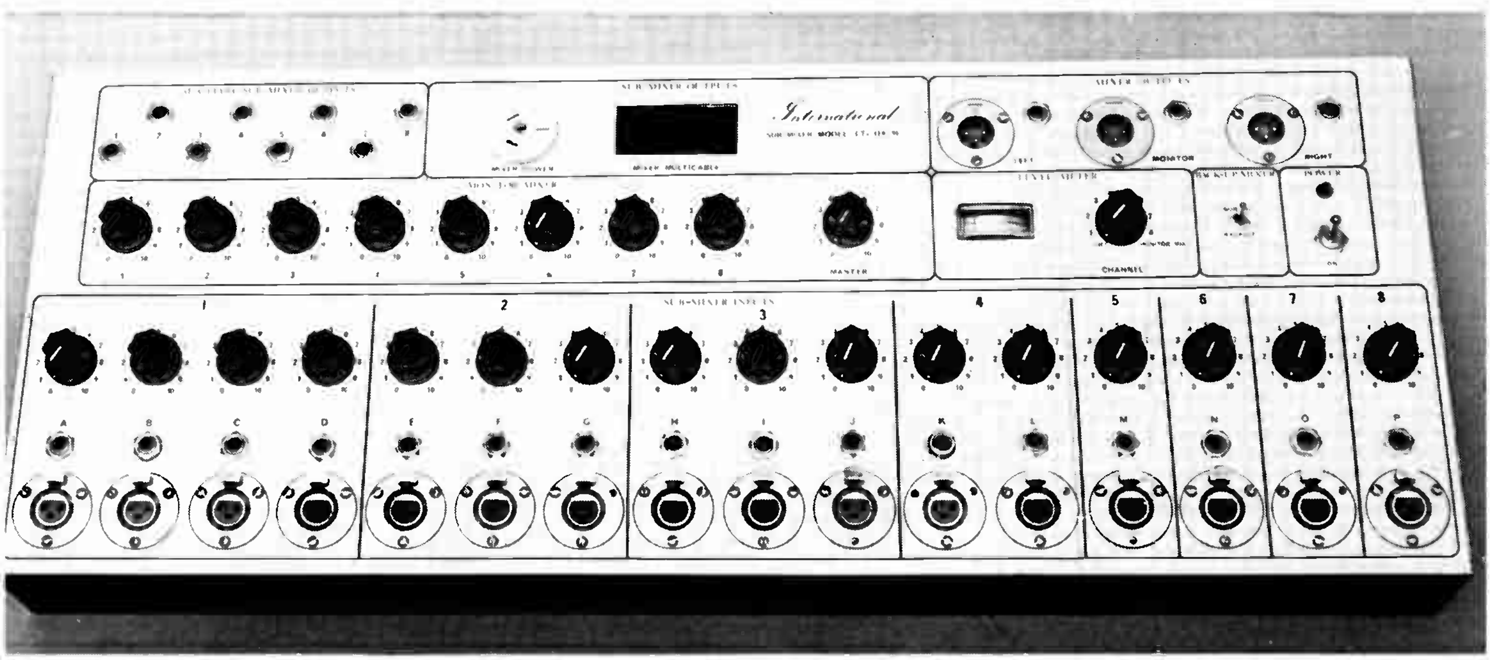
Cord plugs = XLP-3-11 female
= XLP-3-12 male.

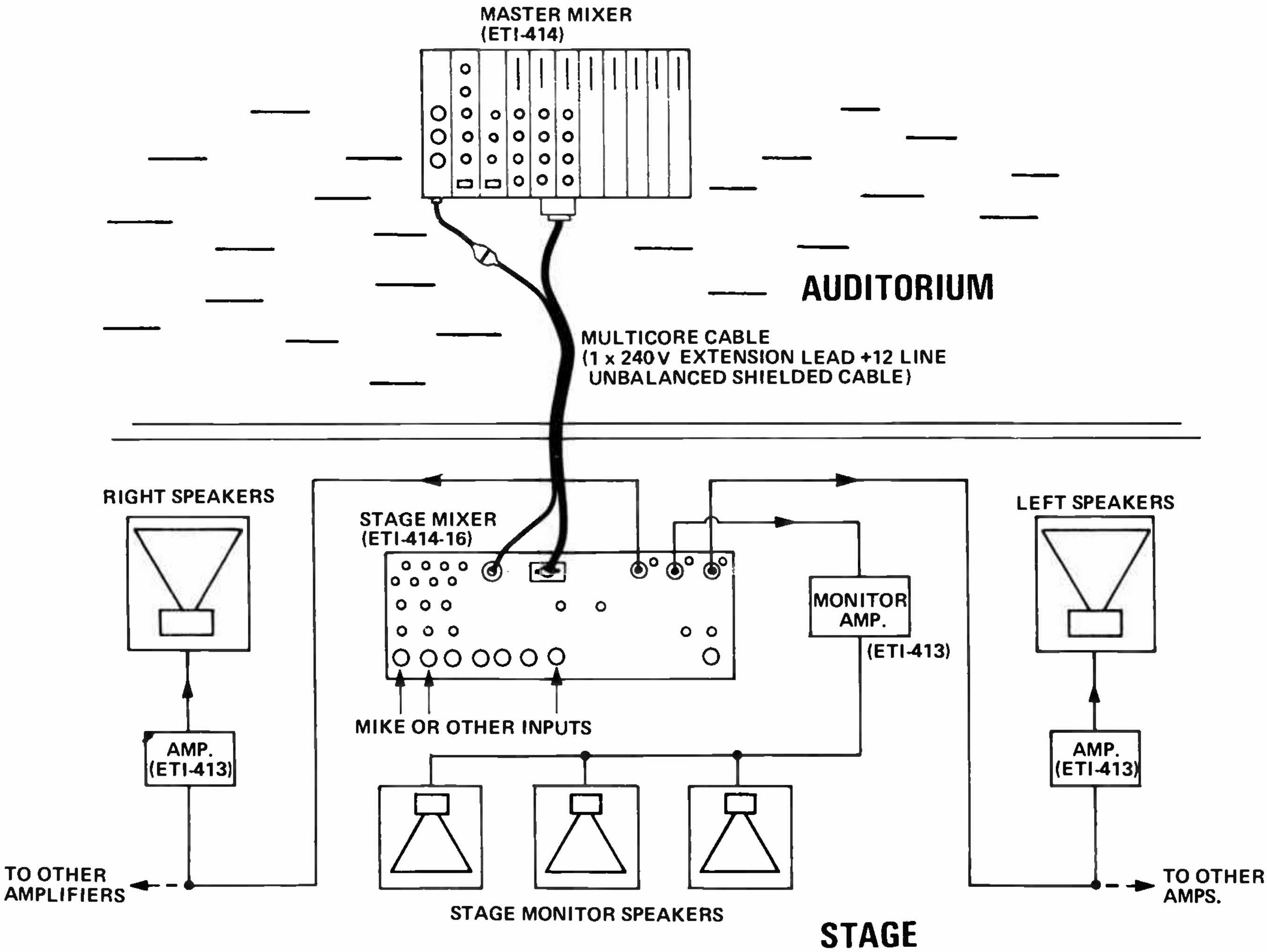
Box to suit (preferable metal), 9 V battery and clip input and output sockets etc.

PROJECT 414 STAGE MIXER

Sixteen amplifiers sub-mixed to eight channels — plus monitor

SEVERAL hundred of our master mixers have been built and are in use by groups and recording studios throughout Australia. Whilst this mixer has been enormously successful, there are several areas in which improvements can be made which will still further improve the flexibility and usefulness of this instrument especially for on-stage performances.





This is the way that the ETI Stage Mixer would be used for a live performance.

LONG-LINE WORKING

For most live performances the master mixer is best located in the listening area so that the mix can be continuously monitored, and controlled, for best effect. Whilst such operation is possible with the ETI master mixer, the inputs are not designed for long line work, especially with low-output, or unbalanced high impedance microphones. This deficiency may be overcome by using a line amplifier for each input.

THE NEED FOR SUB MIXERS

The next obvious deficiency in stage applications is that several microphones are often needed to mike

SPECIFICATION

NO OF INPUTS	16
NO OF OUTPUTS	8 normal + 1 monitor
NOMINAL INPUT	
maximum gain	10 mV
NOMINAL OUTPUT	
maximum	8 volts
nominal	3 volts
INPUT IMPEDANCE	
selectable	< 68 k
SIGNAL TO NOISE	
re 10 mV single channel input	74 dB
MAXIMUM INPUT	
on maximum gain	30 mV
on minimum gain	1 V
GAIN	
maximum	50 dB
variation possible	36 dB

Any number of inputs can be connected to any submixer. However no input may be connected to more than one sub-mixer. The VU metering is switchable to any one output channel.

TABLE 1
Selection value of R11 (or 21, 31 etc)

Input Impedance	R11
200Ω	220Ω
600Ω	680Ω
47 k	150 k

STAGE MIXER

the drums, or the several speakers of an organ etc. This requires the use of separate mixers, in front of the main mixer, to avoid wasting the 8-channel master mixer's capability. To overcome both these disadvantages we have incorporated 16 line amplifiers and eight sub-mixers into a common unit such that the 16 channels may be grouped in any desired combination to the eight master mixer channels. The grouping shown for our prototype stage mixer (in the block diagram Fig. 1) is 4,3,3,2 plus 4 individual channels. This may of course be varied to suit individual requirements.

THE STAGE MIXER

Thus the unit described here is a 16 channel to eight channel sub-mixer which is specifically designed for use on stage. It accepts high or low impedance microphone inputs, which may be balanced or unbalanced. The unit provides eight high-level outputs for transmission to the master mixer.

The inputs may be made by either Cannon connectors or by standard tip-and-sleeve jacks. We strongly recommend that Cannon connectors be used for on-stage work because of their ruggedness. The input impedance of each channel may be tailored to suit the individual microphone (or other source) by selecting one resistor.

The gain of each line amplifier is adjustable from unity to 63 (36 dB) and the sub-mixer adds a further (14 dB), that is, a total of 50 dB gain is available.

The output level of each channel (even from a low output microphone) will be of the order of 1 volt and may be as high as 22 volts peak-to-peak without overload distortion occuring. Thus an extremely wide dynamic range may be accommodated by this mixer and the same dynamic range will also be accommodated by the master mixer. The master mixer, when used with the stage mixer may be used switched to the low sensitivity input position and such operation greatly improves the signal-to-noise ratio.

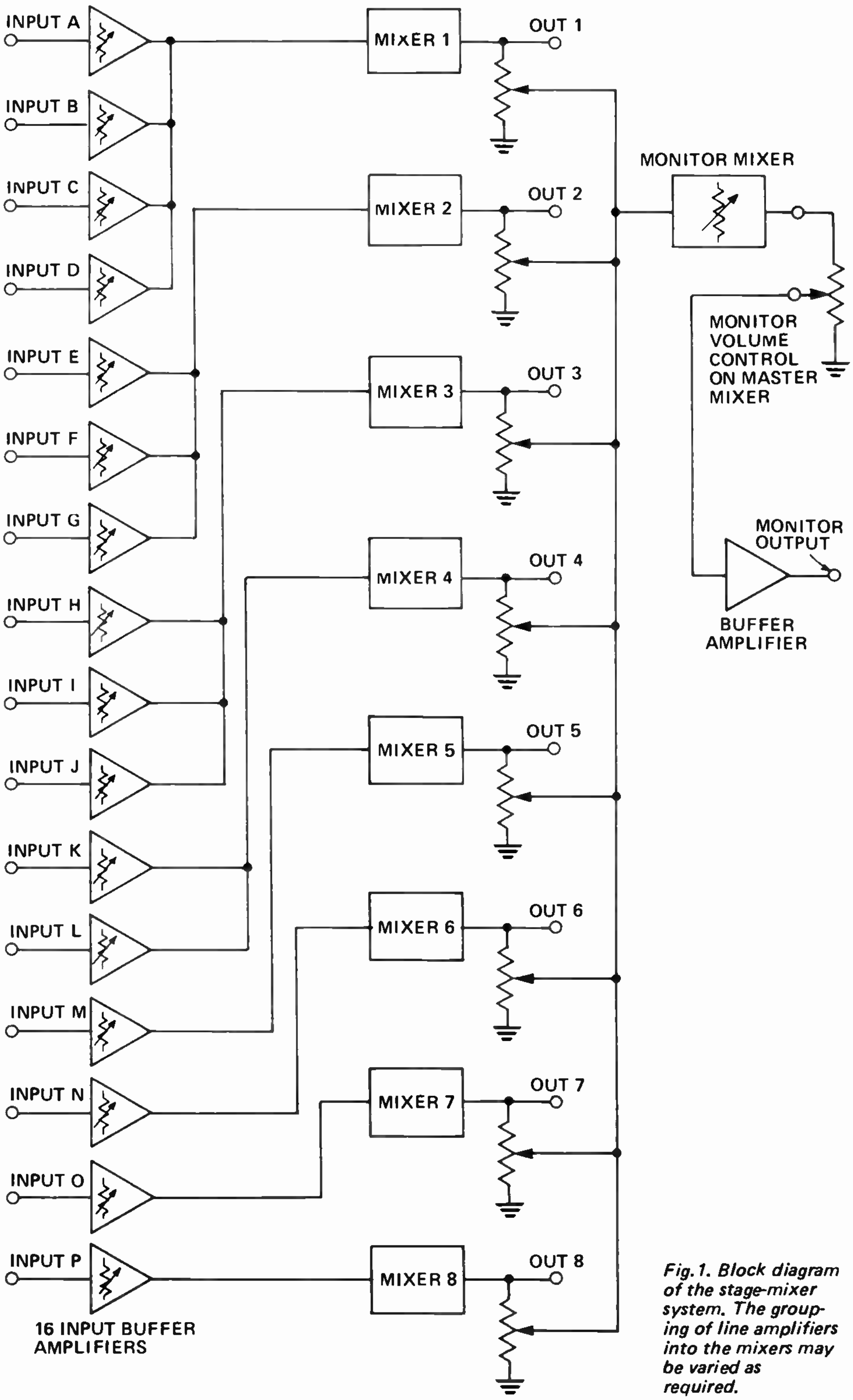


Fig. 1. Block diagram of the stage-mixer system. The grouping of line amplifiers into the mixers may be varied as required.

MONITOR FACILITIES

The original master mixer does not incorporate any monitor facilities. It is possible to use the echo-mix channel for monitoring but the level controls for each channel will also affect the monitor output. This is undesirable as if a louder level is required in the auditorium the monitor will also become louder – introducing a danger of acoustic feedback occuring.

Within the stage mixer we have incorporated a special monitor mixer which has its own level control followed by a buffer amplifier. A second 'Master' monitor volume control is physically located on the main mixer so that it can be adjusted should acoustic feedback occur.

BACK UP MONITOR

Facilities are provided such that should the master mixer fail, or the cables between the two mixers be damaged etc, the stage mixer may be switched to provide an output direct to the PA system.

In this mode a 'Back up' switch takes the output from the monitor mixer and transmits it direct to both channels of the PA system. The monitor signal is still transmitted to the monitor amplifier when the mixer is in this mode. In normal use the 'back up' switch must be at 'normal'.

When the stage mixer is in 'back up' mode the master monitor level control, located on the master mixer, is by-passed (full volume) regardless of whether the master mixer is connected or not.

FINAL OUTPUTS

The master mixer outputs (i.e. left and right stereo plus monitor mix) are returned to the stage as part of the multicore cable and terminated on the 'stage mixer' with both 'Cannon' and standard 'Jack' type connectors.

METERING

A VU meter is provided on the stage mixer which can be used to monitor the output of any of the eight (sub) mixers or the stage monitor output. This meter will be useful for initial level settings on each sub-mixer.

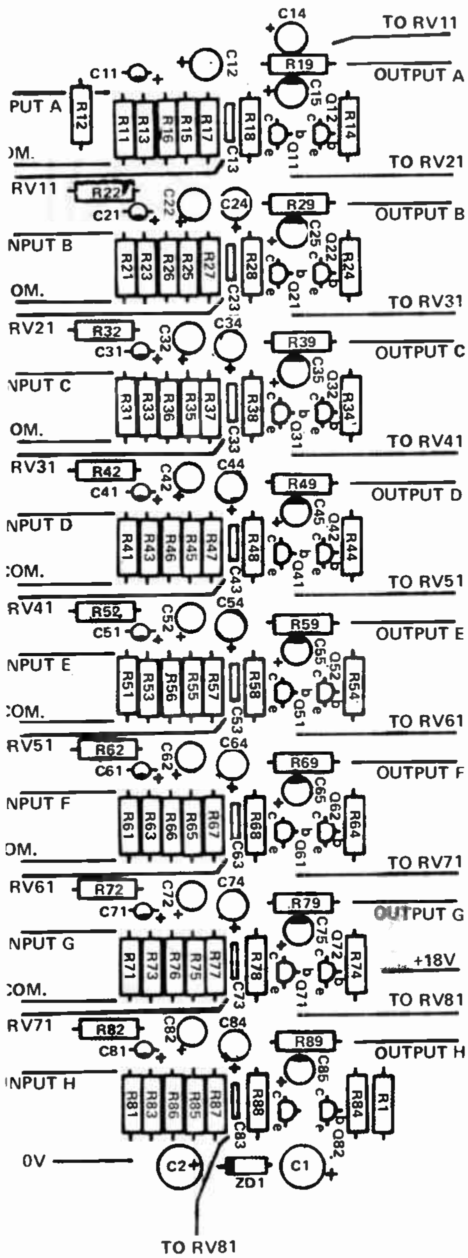


Fig.2. Component overlay for the preamplifier board.

POWER OUTLET

A switched, 240 volt power outlet is provided on the stage mixer. This is intended to provide power for the master mixer via an extension cable. Thus the power cable and the multicore cable are the only ones required between the two mixers.

HOW IT WORKS – ETI 414

LINE AMPLIFIER

The line amplifier used is similar to the ETI 430 line amplifier except that the gain is variable from unity to 40 dB (actually 36 dB in a practical circuit).

The input impedance of the amplifier (referring to Fig. 2) is determined by the combined value of R11, R12 and R13 – all in parallel. The parallel impedance of R12 and R13 is 68 k and this is therefore the upper limit of input impedance (R = ∝).

For impedances less than 5 k the values of R12 and R13 may be ignored and R11 is set to the same value as the desired input impedance. Hence the circuit as shown matches microphones having 200 ohm output impedance.

The output of Q12 is fed back to the emitter of Q11. This path via R17 in parallel with RV11 and C14 provides negative feedback as well as supplying a dc bias which sets the overall gain of the stage.

The gain of the amplifier may be calculated using the following formula (assuming ideal transistors).

$$\text{Gain} = \frac{(R17//RV11) + R15}{R15}$$

When the gain control is at maximum the gain is 102 or 40 dB (in practice 36 dB), and when the gain control is at minimum R17//RV11 is zero and the gain is therefore unity.

The signals from any number of line amplifiers may be summed by one of the sub mixers (eight per board IC1-IC8) the output from each mixer is taken directly to output socket to the master mixer, and via a 22 k level. control to the monitor mixer, IC9.

The output of the monitor mixer is taken to the master-monitor, level control on the master mixer and then returned to a buffer amplifier in the stage mixer, IC10.

In an emergency (main mixer faulty) SW2 disconnects the outputs from the master mixer and connects the output of the monitor amplifier to the PA channels.

Power for the Stage mixer is provided by a conventional supply which provides plus and minus 15 volts for the mixer amplifiers and plus 19.6 volts for the line amplifiers.

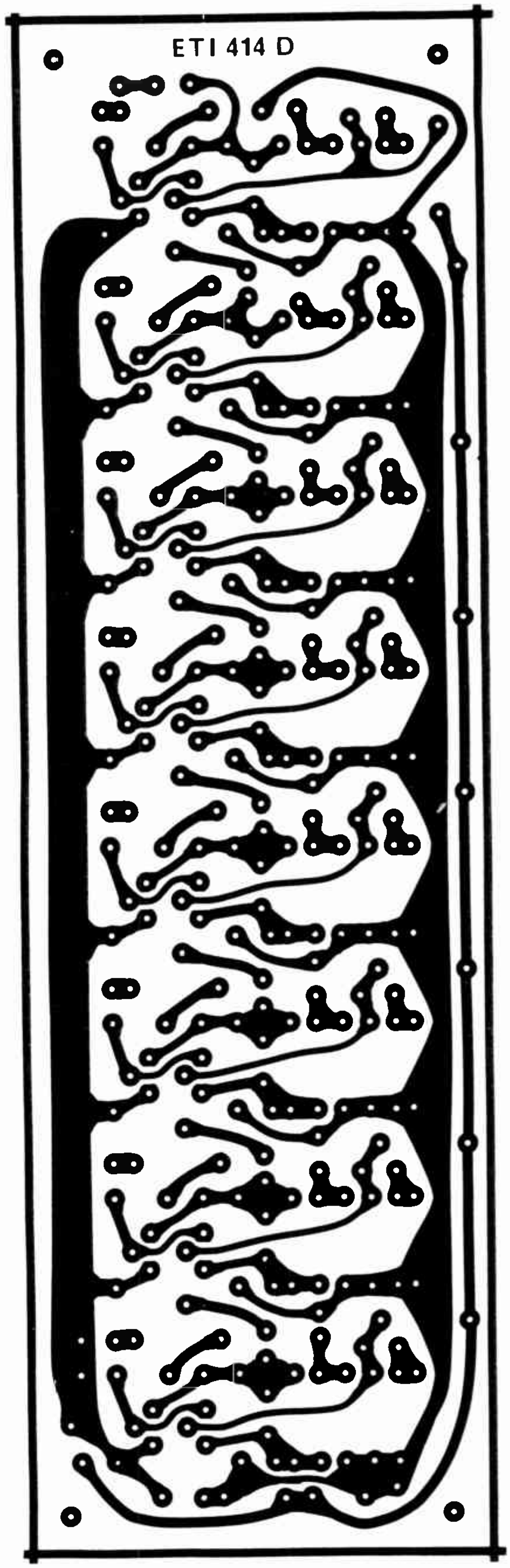
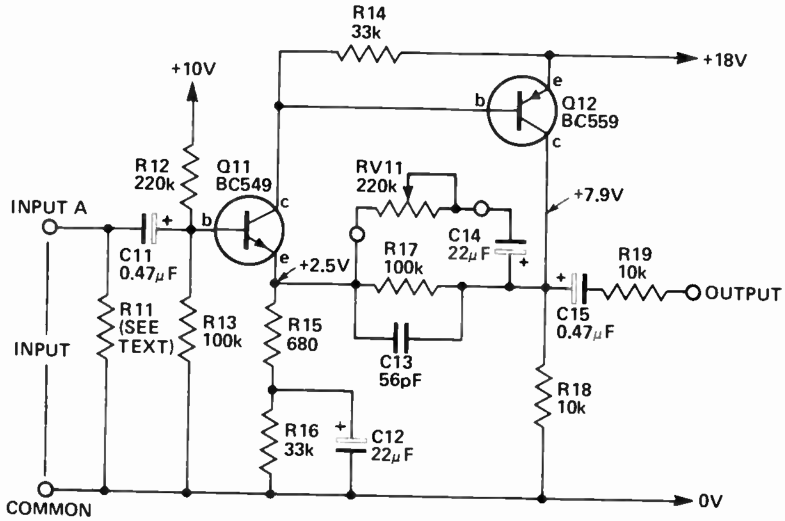
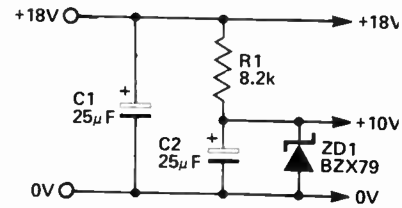


Fig.3. Printed circuit board layout for the preamplifiers (two required for 16 channels). Full size 223 x 63 mm.

STAGE MIXER





NOTE
ONE AMPLIFIER ONLY SHOWN. PC BOARD ETI-414D CONTAINS 8 AMPLIFIERS. COMPONENT NUMBERING OF SECOND AMPLIFIER STARTS AT 21, THE THIRD AMPLIFIE AT 31, ETC. COMPONENTS R1, C1, C2 AND ZD1 ARE COMMON TO ALL AMPLIFIERS.

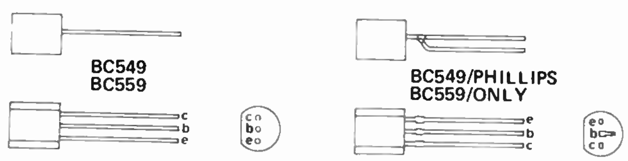


Fig.4. Circuit diagram of the line amplifier (eight per board).

CONSTRUCTION

The mixer board (ETI 414E) should be assembled with the aid of the circuit diagram, Fig. 5, and the component overlay, Fig. 7. The parts list for this board was given in last month's issue. It is advisable to use terminal posts or pins for the eight input lines, the O V line and the +19.6 volt line. This makes later interconnection considerably easier.

Our prototype was constructed in a simple pan shaped chassis and cover. We suggest that the sides of the front panel be bent up (rather than the ends as shown in the photographs).This will strengthen the front panel and allow the transformer to be mounted on it rather than in the case as shown in our prototype unit.

Mount the spacers for the printed circuit boards, the multi-cable socket, VU meter and power outlet socket to the front panel with countersunk screws. It is suggested that the wires to the three-pin socket be attached before mounting – it is difficult later. All other front-panel components can now be mounted along with the escutcheon.

Since the mixer may be subject to rough handling it is recommended that all screws be sealed in position with LOCTITE or similar compound.

Commence interconnection wiring by connecting the input sockets and potentiometers as shown in Fig. 8. This diagram shows connections to channel 1 of the preamplifiers – all other channels being similar. For neatness, we terminated these wires by soldering to the appropriate places on the underside of the board. Attach wires to the preamplifier outputs, on both boards, long enough to reach the appropriate mixer inputs. Similarly attach wires for the O volt and +18 volt supply lines.

The +18 volt supply comes from the negative side of the LED, the positive side being fed from the 19.6 volts of the power supply (1.6 volts drop across LED). When all these leads are attached, both boards may be mounted in position on the chassis.

The mixer/power-supply board may now be interconnected with the aid of Fig. 9. Figure 10 shows the wiring to output sockets and VU meter.

The selector switch and VU meter wiring is as shown in Fig.10 and 11.

STAGE MIXER PARTS LIST

PARTS LIST GENERAL

Chassis
Box
Escutcheon
16 Cannon sockets XLP-3-13
3 Cannon plugs XLP-3-14
27 Phone jacks — mono— 6.4mm
1 LED and panel holder
1 11 position 1 pole rotary switch
1 VU meter
1 240V power outlet HPM type 55 or similar
1 21 pin socket John Carr KA/213
26 Knobs
12 1" spacers
nuts, bolts, 3 core flex & plug etc.

SUB-MIXERS,POWER SUPPLY

R2,5,8,11 resistor 100Ω ¼w 5%
R14,17,20 " 100Ω ¼w 5%
R23,25,28 " 100Ω " "
R29,30,31 " 390Ω ½w "
R1,4,7,10 " 47 k ¼w "
R13,16,19,22 " 47 k " "
R3,6,9,12 " 100 k " "
R15,18,21 " 100 k " "
R24,26,27 " 100 k " "

RV1,2,3,4 potentiometer 22k rotary log
RV5,6,7,8 potentiometer 22k rotary log
RV9 potentiometer 470k rotary log

C4,5,6 capacitor 0.1μF polyester
C1,2,3 capacitor 470μF 25V electrolytic

IC1-IC10 integrated circuit μA741C
Mini dip or TO5

D1-D4 diode EM401 or similar
ZD1,2 Zener diode BZX79C15

T1 transformer 240V/15-0-15V PL30/20VA
PC Board ETI-414E
SW1 switch DPDT toggle 240V rated
SW2 switch 4PDT toggle

INPUT AMPLIFIERS

16 off are required for all components below

R11 resistor see text
R15 resistor 680Ω ¼w 5%
R18, 19 resistor 10k ¼w 5%
R14,16 resistor 33k ¼w 5%
R13 resistor 100k ¼w 5%
R12 resistor 220k ¼w 5%

RV11 potentiometer 220 k rotary log.

C13 capacitor 56pF ceramic
C11, 15 capacitor 0.47μF TAG Tantalum
C12 capacitor 22μF 16V electrolytic

Q11 transistor BC549 or similar
Q12 transistor BC559 or similar

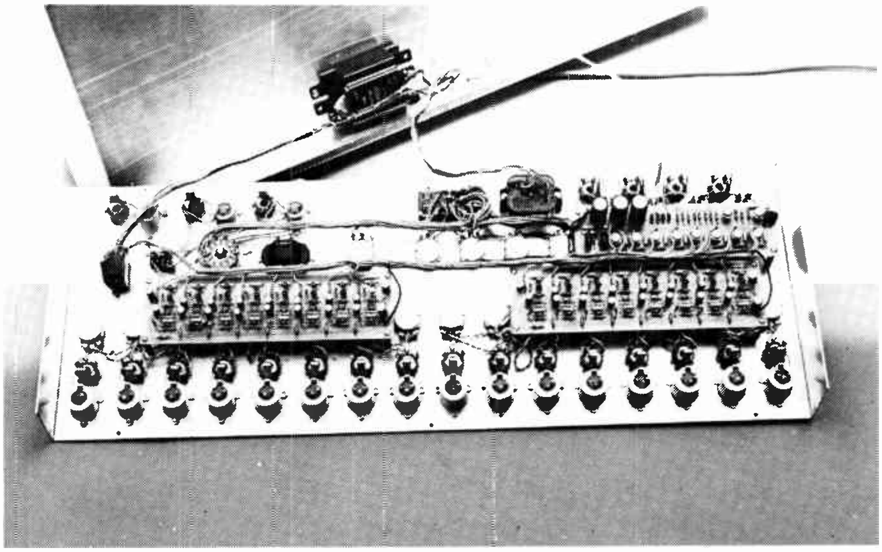
2 off are required for all components below —

R1 resistor 8k2 ¼w 5%

C1, 2 capacitor 25μF 25V electrolytic

ZD1 Zener diode BZX79C10

PC Board ETI-414D



Internal view of the completed mixer.

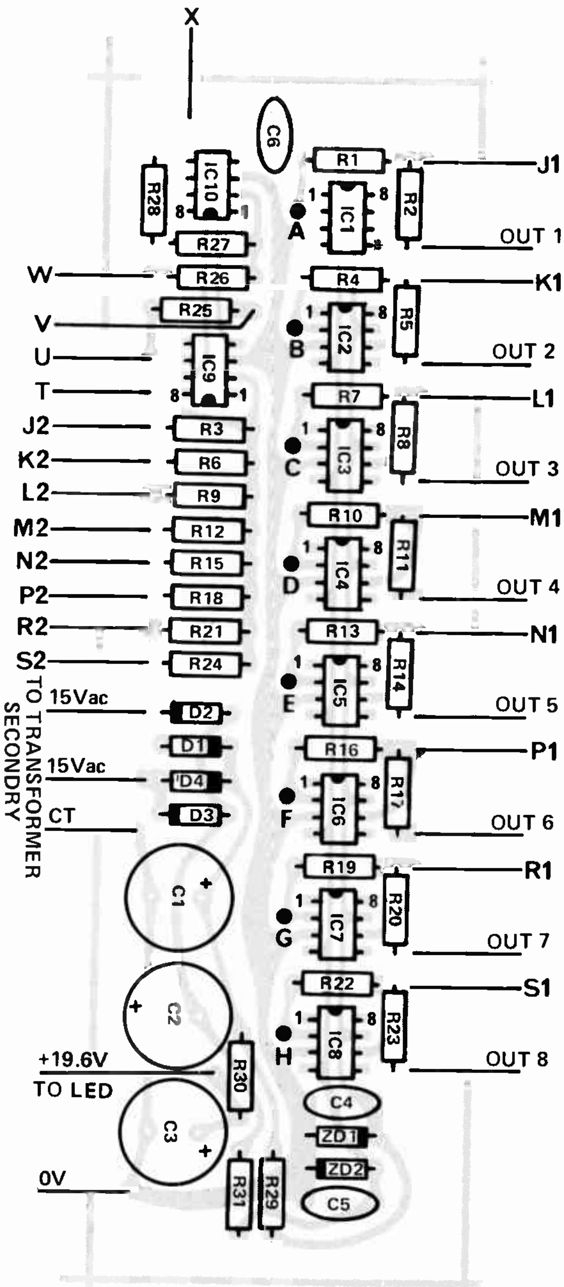
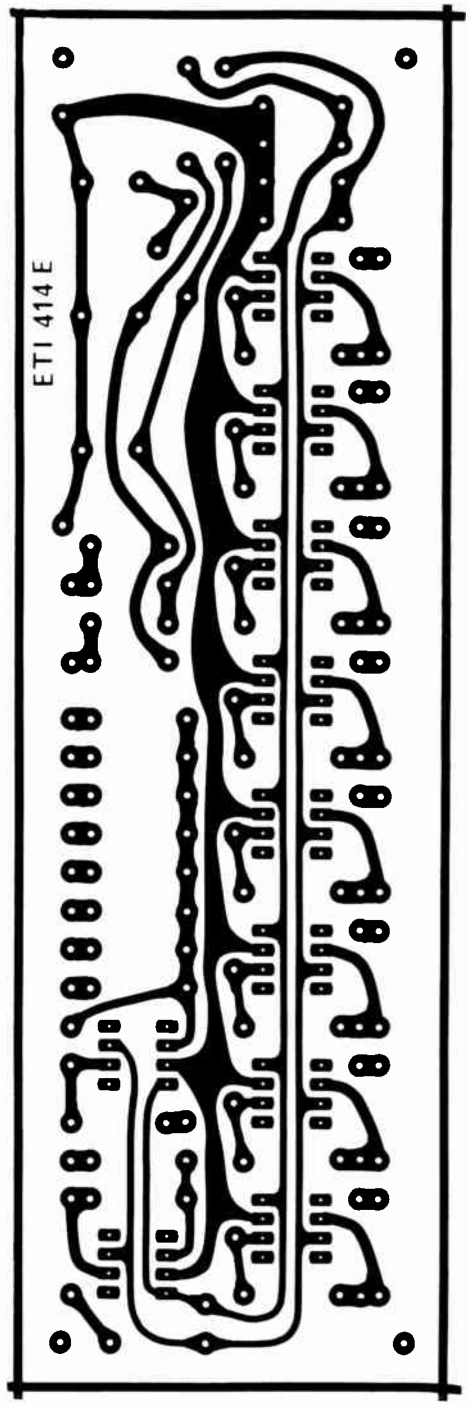


Fig. 7. Component overlay for the mixer/power-supply board.

Note that pins 1 to 9 of the multi-cable socket will have 2 sets of leads, one set from the mixer outputs and one set from the VU meter selector switch. ●

Circuit diagram, How it Works and wiring diagrams — next page.

Fig. 6. Printed circuit layout for the mixer/power-supply board. Full size 182 x 57 mm.



STAGE MIXER

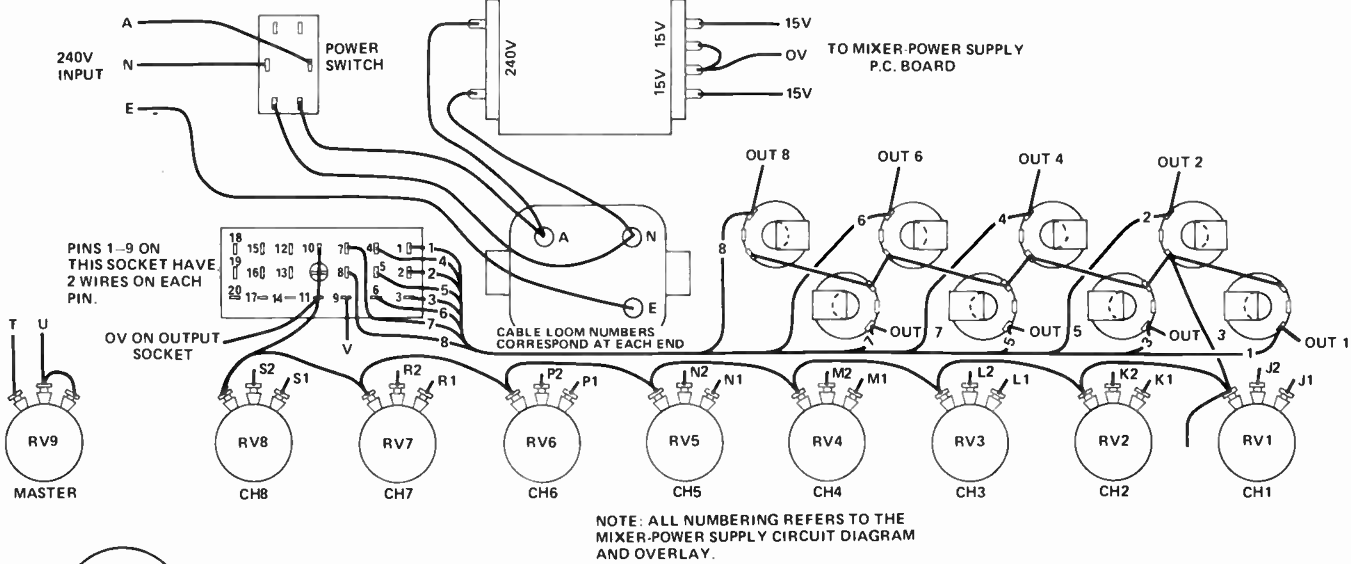


Fig. 9. Connection diagram for the mixer/ power-supply board.

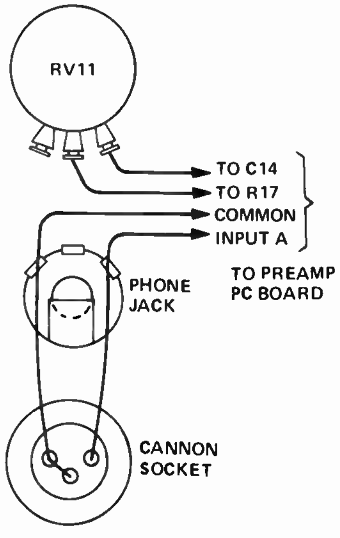


Fig. 8. Wiring to input sockets.

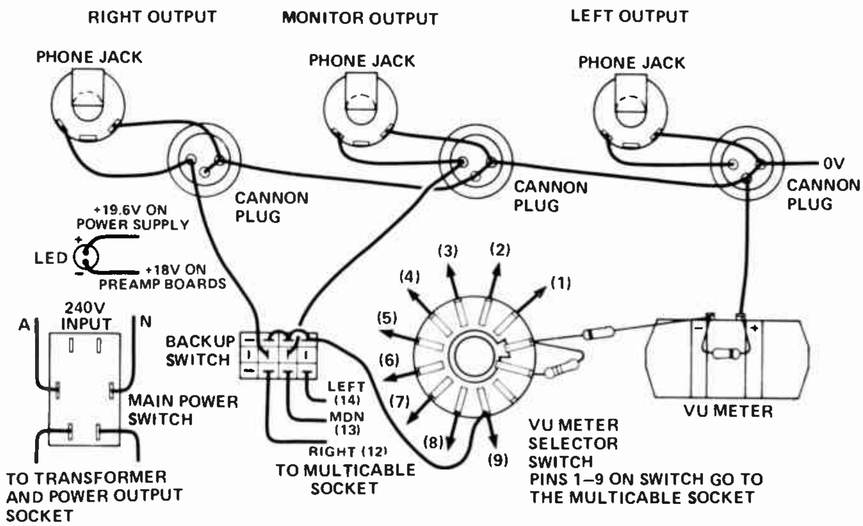
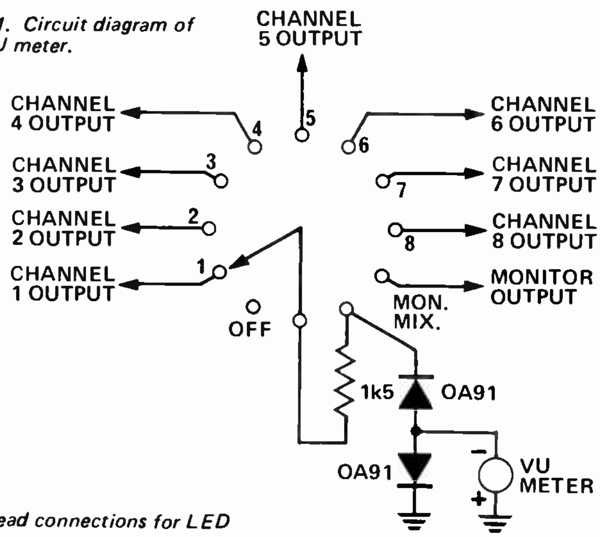
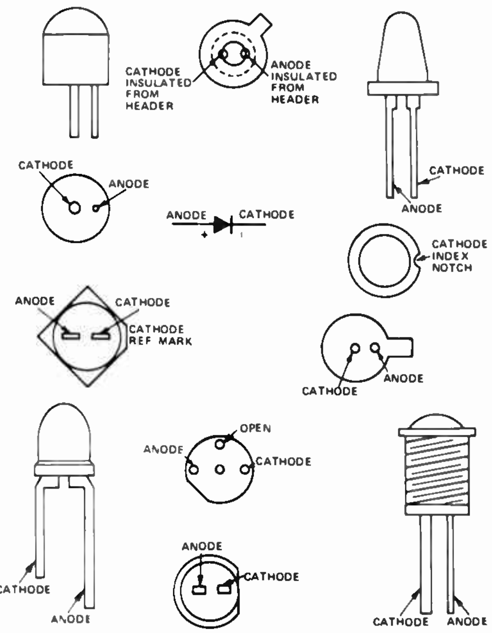


Fig. 10. Interconnection of output sockets VU meter and switch and backup switch.

Fig. 11. Circuit diagram of the VU meter. ▶





◀ *Fig. 12. Common lead connections for LED indicator lamps.*

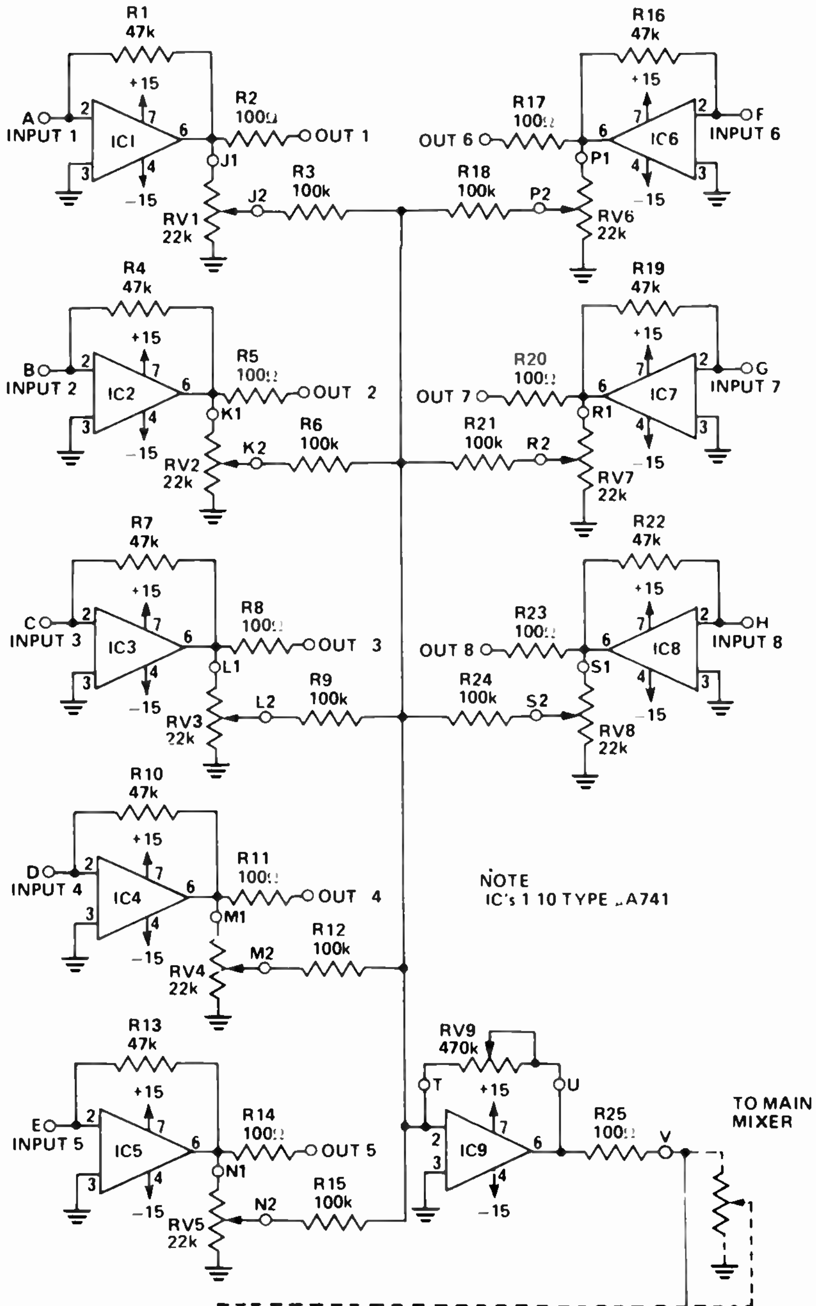
HOW IT WORKS – ETI 414 MIXER/POWER SUPPLY

The signals from any number of line amplifiers may be summed by one of the sub mixers (eight per board IC1-IC8) the output from each mixer is taken directly via the output socket to the master mixer, and via a 22 k level control to the monitor mixer, IC9.

The output of the monitor mixer is taken to the master-monitor level control on the master mixer and then returned to a buffer amplifier in the stage mixer, IC10.

In an emergency (main mixer faulty) SW2 disconnects the outputs from the master mixer and connects the output of the monitor amplifier to the PA channels.

Power for the stage mixer is provided by a conventional supply which provides plus and minus 15 volts for the mixer amplifiers and plus 19.6 volts for the line amplifiers.



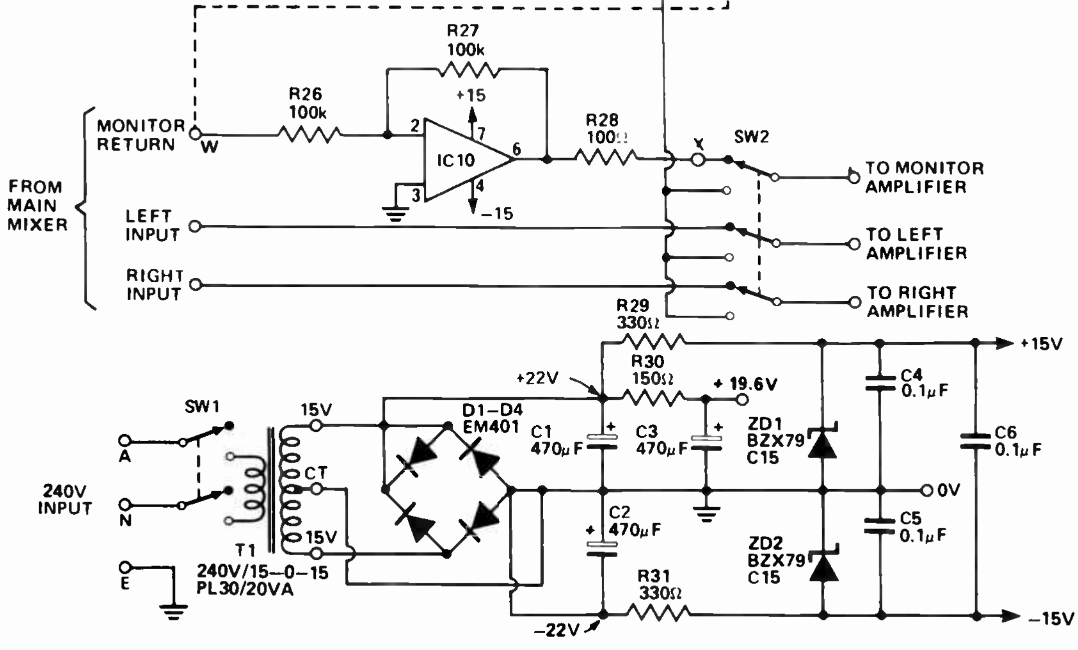
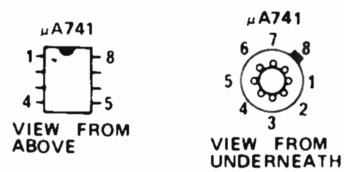
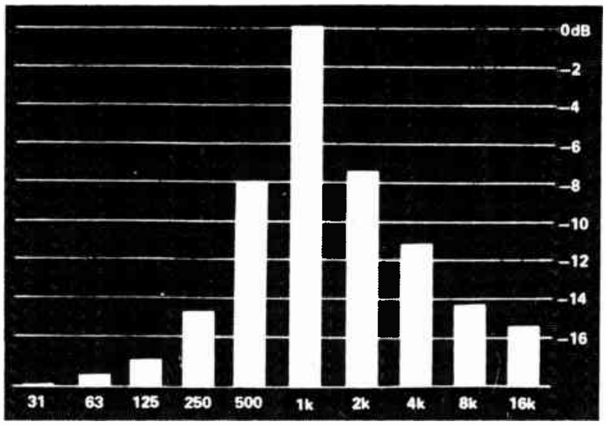


Fig. 5. Circuit diagram of the mixer board.



Audio Spectrum Analyser



Equalise systems for room acoustics accurately using this neat piece of 'test' gear.

AUDIO SPECTRUM ANALYSERS can be a valuable tool used in the setting up of a room acoustically with a graphic equalizer such as the ETI 485; to monitor programme material or just as a gimmick to please yourself and friends.

When setting up rooms pink noise is pumped into the room using an amplifier. A microphone is then used to monitor the sound and its output is the input to the analyser. Now by adjusting the graphic equalizer a flat response can (hopefully) be obtained.

Design Features

Spectrum analysis can be done by two main methods. The first is to have a tuneable filter which is swept across the band of interest. The output of the filter, when displayed on an oscilloscope, will be a frequency/amplitude graph of the input. While this gives a well-formatted and accurate display it is not "real time" in that if an event occurs at one frequency while the filter is sweeping elsewhere it will not be recorded. For this reason this method is used normally where the spectral content is constant and the sweep is only over a small percentage of total frequency (such as the output of a radio transmitter).

For real time analysis the frequency spectrum is broken into bands using bandpass filters and the output of each rectified. The output from these rectifiers can be displayed on a CRO as in this project or by columns of LEDs or similar methods. The number of bands and the dynamic range required determine the filters used. In this project where only about 20dB is required a single LC network is sufficient. Another unit we have built (not for a project) uses a 6 pole high pass filter followed by a 6 pole low pass one. This gives a flat response (∓ 1dB) over ∓ ½ octave and is 36dB down 1 octave away. However, it uses 6 op amps and 2% capacitors and resistors in each filter!

If there are sufficient requests for it we will publish a LED version of this unit.

SPECIFICATION – ETI 487

No. of bands	10
Frequencies	31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16k
Filter characteristics	−12dB, one octave from nominal centre frequency
Display	CRO in XY mode
Input level	50mV – 10V
Input impedance	200k
Pink noise output	200mV
X output	±4 V approx
Y output	0V to 10V

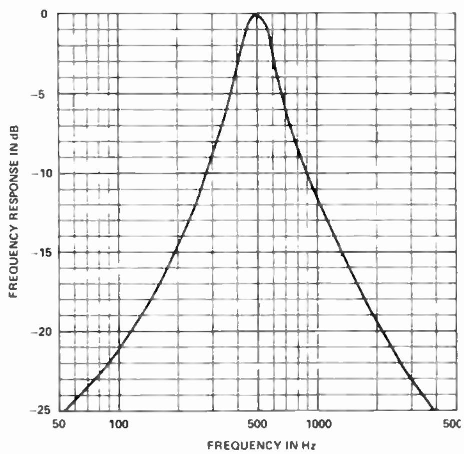


Fig. 1. The frequency response of the 500Hz filter. All other filters follow a similar curve.

HOW IT WORKS – ETI 487

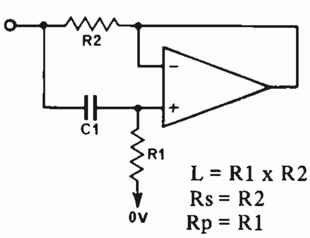
The unit can be broken into eight sections to help the explanation of how it works.

(a) Input amplifier
(b) Ten individual filters and rectifiers.
(c) Ten way analogue switch with decade counter.
(d) Staircase generator controlled by "c". (X output).
(e) Log converter.
(f) Ramp generator and comparator. (Y output)
(g) A pink noise generator.
(h) Power supply.

(a) The input amplifier has an input impedance of 220 k (set by R1) and a gain of 101 ((R3 + R2)/R2). The output of the amplifier drives all ten filters and Q1 and Q2 are used to buffer IC1 to give the drive capability required.

(b) The ten filter-rectifiers are identical except for component values and a bias resistor in the three lowest frequency filters, where tantalum capacitors are used in series. The filter is a parallel LC network which, with a series resistor, gives a band-pass filter.

As large value inductors are expensive we have used an active one using an operational amplifier, two resistors and a capacitor. The value of such a network is as follows:



The frequency response of the networks is given in fig. 1.

The rectifier is a half wave type where the gain is variable from about 4 to 12. A diode from the output back to pin 2 keeps the op-amp in the linear region on the negative half cycle allowing operation up to the 16kHz of the top filter.

(c) The analogue switches IC23/1 – IC25/2 are controlled by IC22. This is a decade counter with 10 decoded outputs, each of which is high only for one clock period. As the analogue switches need a high to switch them on, only one will be selected at any one time.

(d) The output of the decade counter also controls the staircase generator IC28 with the weighting networks R58 – R72 giving equal steps of about 0.9 volts. Resistor R89 provides a bias current and the output of IC28 starts at about +4 volts and steps down in 0.9V steps to about -4.2 volts when the output switches back to +4 volts. This is used to drive the X input of the CRO. To add some width to the vertical lines, IC29/1 and IC29/2 form an oscillator of about 300 kHz and after filtering by R90 and C69 is coupled into the input of IC28 by R91.

(e) The output of the analogue switch is fed to the diode-resistor network (D21 - D26, R73 - R77) which gives a simple log conversion. This method is simple, needs no adjustments and is adequate for the purpose. As there is some loss in this network IC26 is used to provide a gain of three to recover this loss.

(f) The ramp generator is formed by the constant current (12μA) source and capacitor C71. The capacitor can be discharged by IC25/4 and the current source is controlled by IC24/3. The voltage out of the log converter (IC26) can vary between zero and +10 volts and this is compared to the ramp voltage by IC30. The output of IC30 controls the oscillator formed by IC29/3 and IC29/4. When the ramp voltage exceeds the voltage from IC26 the output of IC30 goes high allowing the oscillator to start. This immediately discharges C71 and switches off the current source which causes the output of IC30 to go low again after only about 2μs. Diode D27 ensures however that the oscillator acts as a monostable giving an output of about 6μs to ensure the capacitor C71 is completely discharged. The output of IC29/4 also clocks IC22 which selects the next inut. If the input from IC26 is ever negative and C71 cannot be discharged to less than this voltage, IC29/3 and IC29/4 will oscillate continuously at about 100kHz clocking IC22 until it finds an input higher. This prevents possibility of lockup if the offset voltages of the op-amps all go the wrong way.

(g) White noise is generated by the zener action of Q3 which is reversed biased. It is amplified by Q4 to give 200 mV of white noise on its collector. White noise however has equal energy per unit bandwidth and what we need is pink noise which selects the next input. If the input bandwidth (i.e., equal energy per octave). To convert white to pink we need a filter at 3 db/octave. This is performed by IC27 with the RC networks providing the necessary curve.

(h) The power supply is a simple rectifier type with IC regulators to give stable supply voltages.

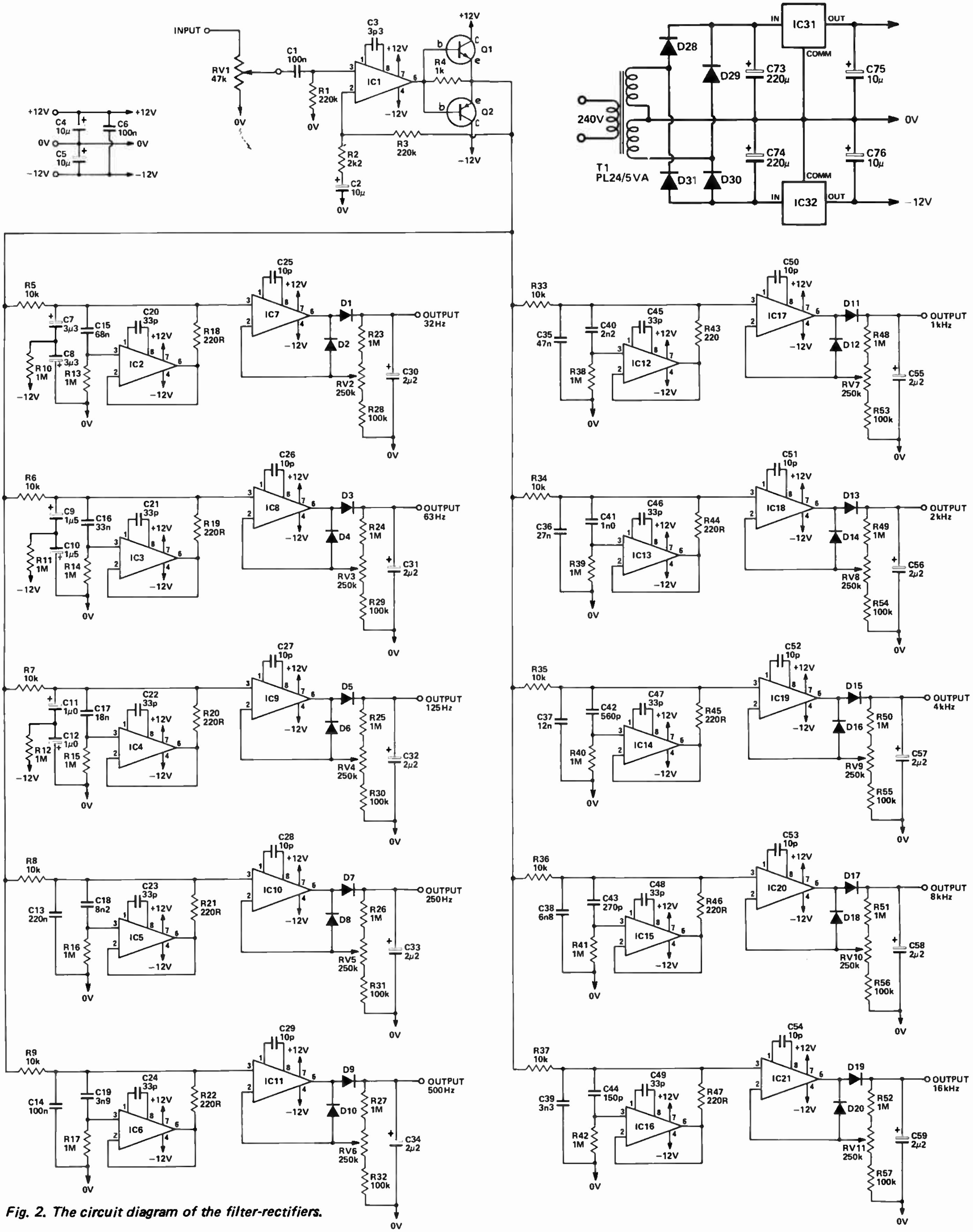


Fig. 2. The circuit diagram of the filter-rectifiers.

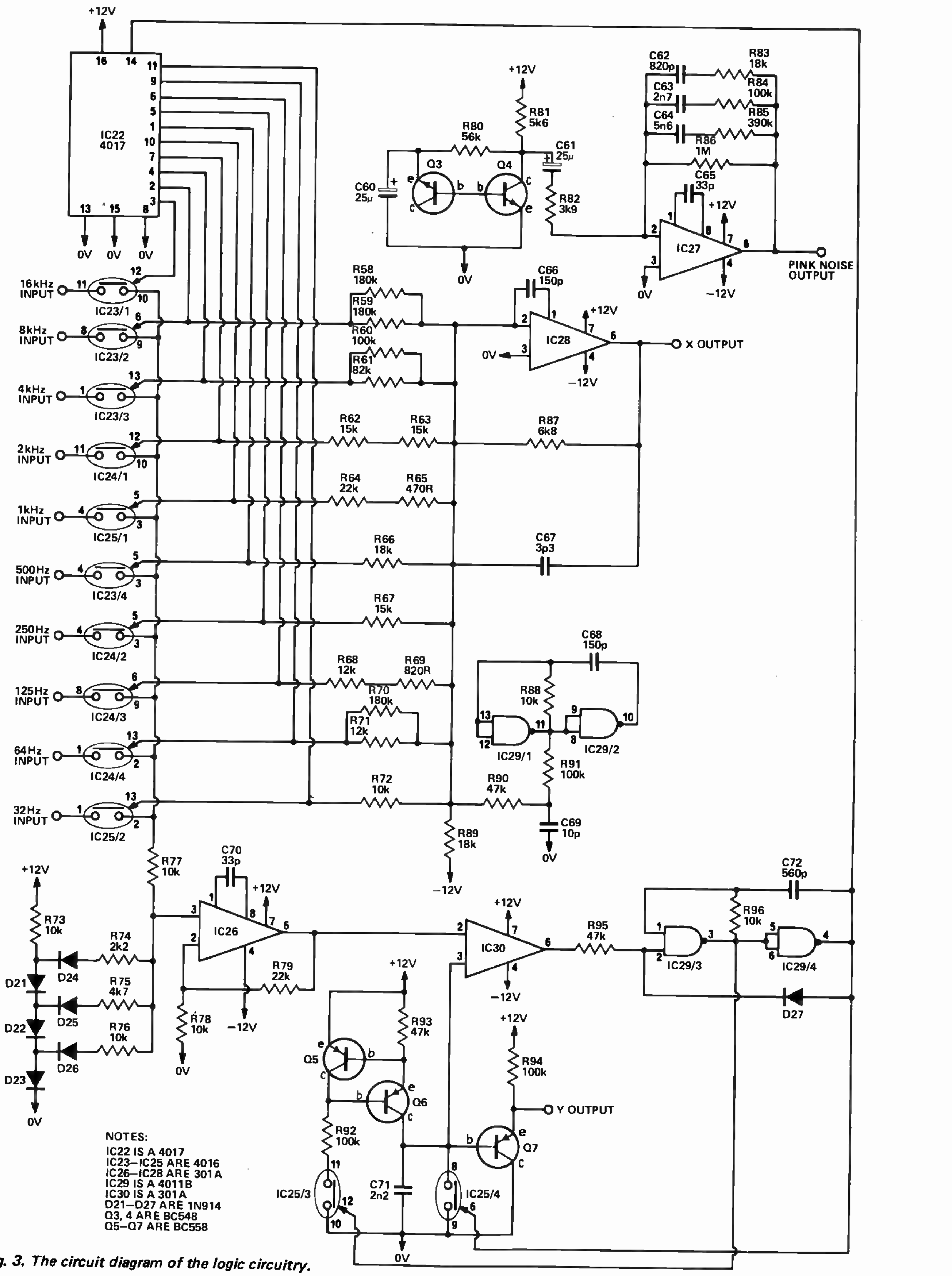
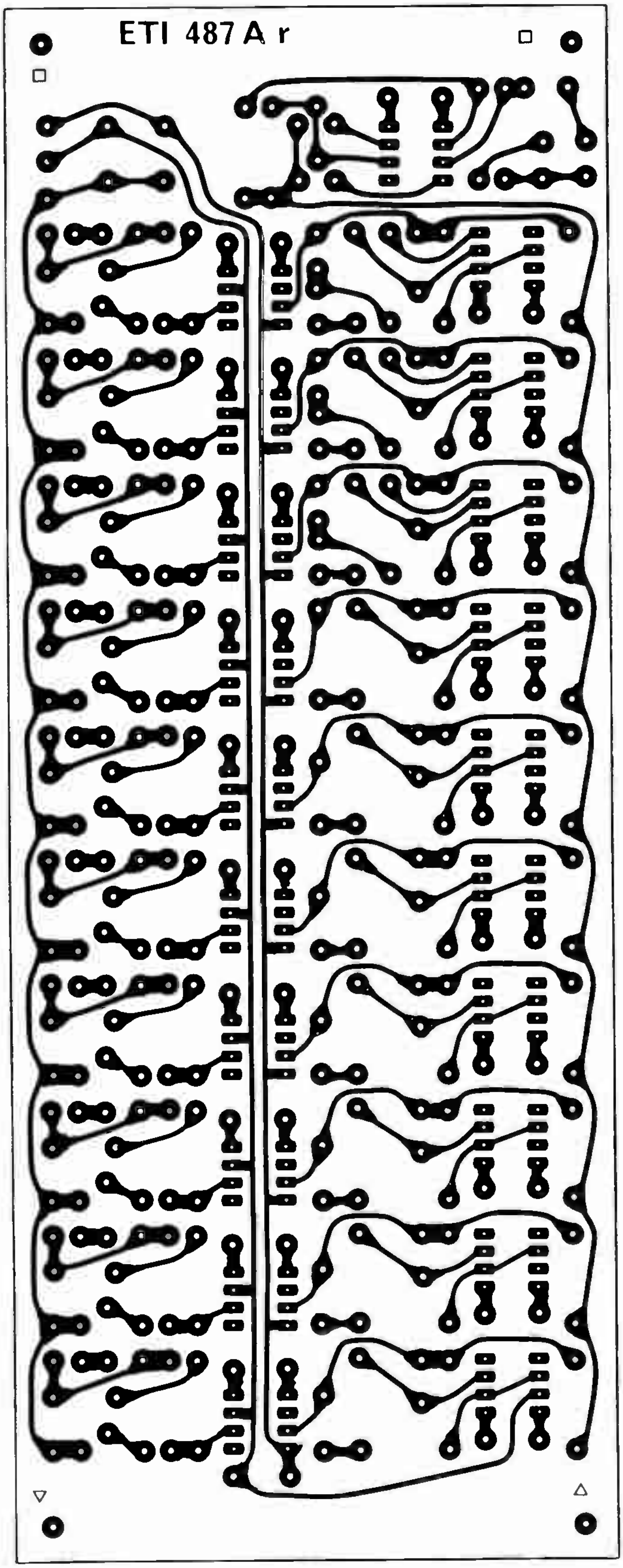


Fig. 3. The circuit diagram of the logic circuitry.

(description continues on page 136)



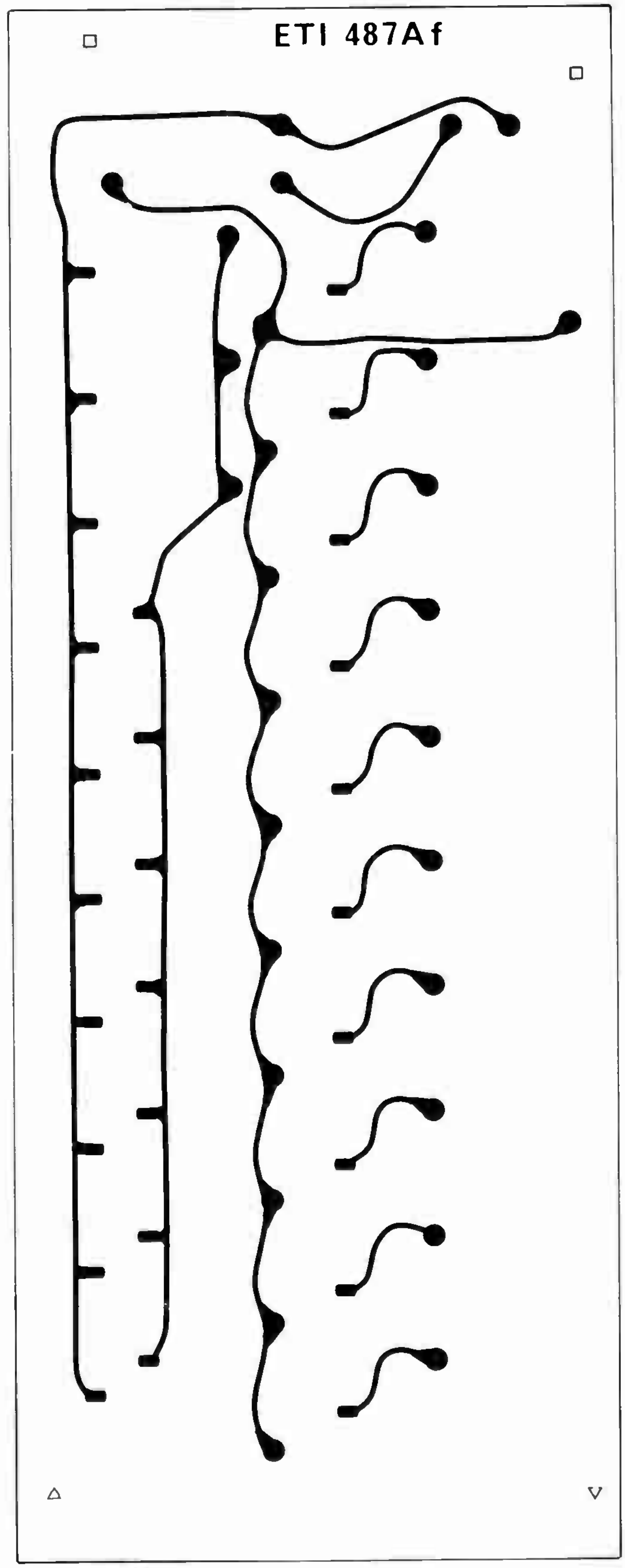
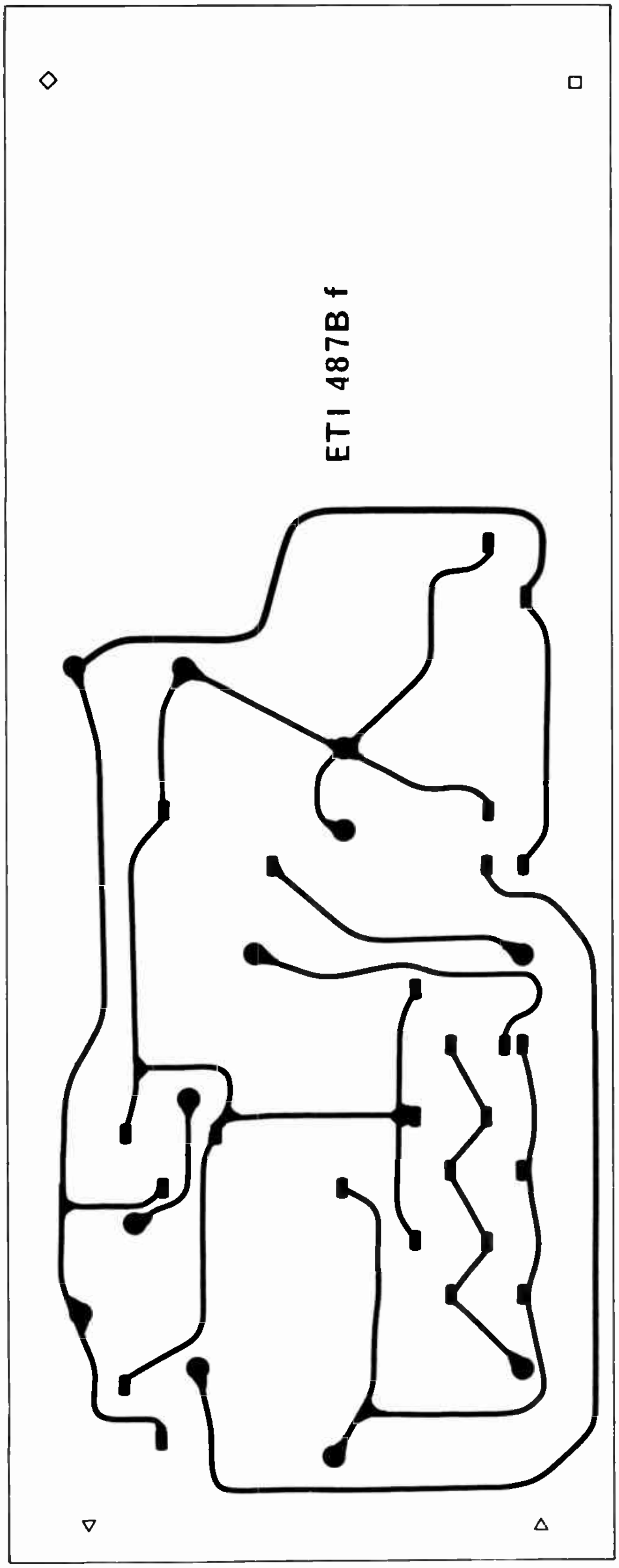


Fig. 4. Both sides of the ETI 487A board shown full size.



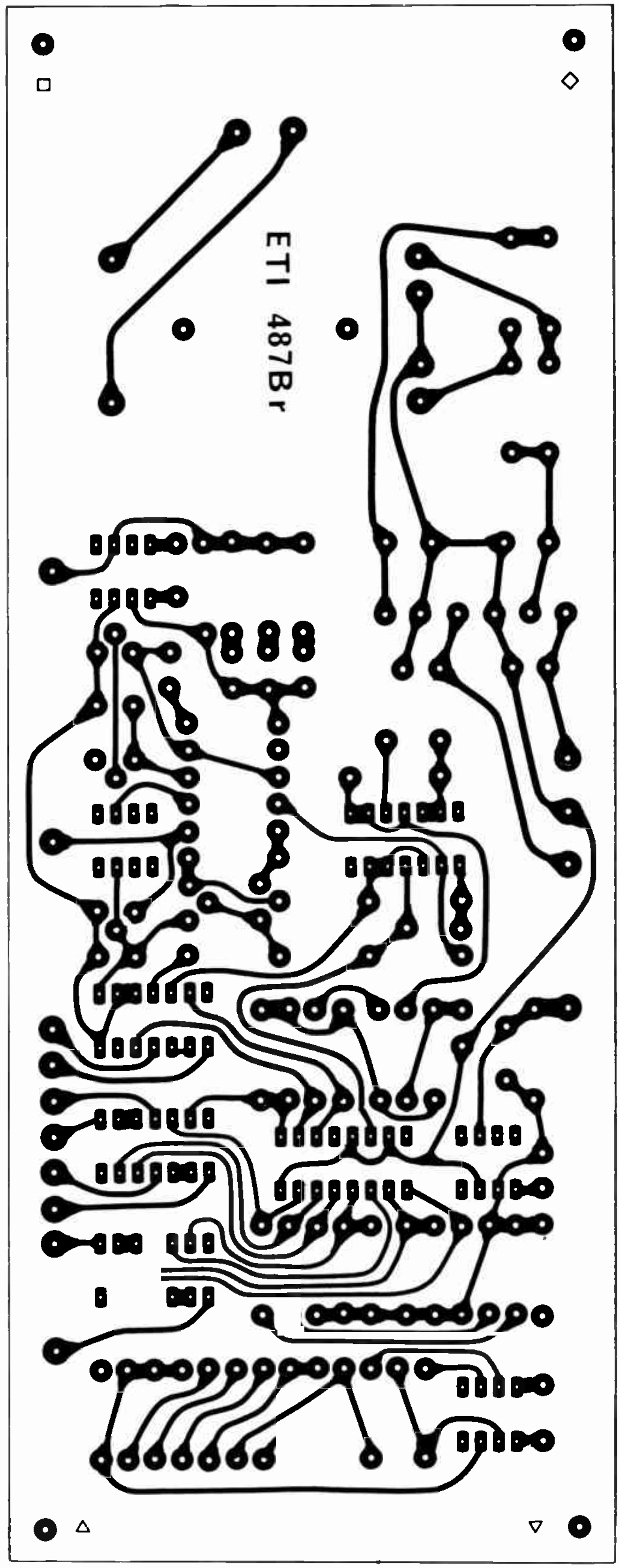


Fig. 9. Both sides of the ETI 487B board shown full size.

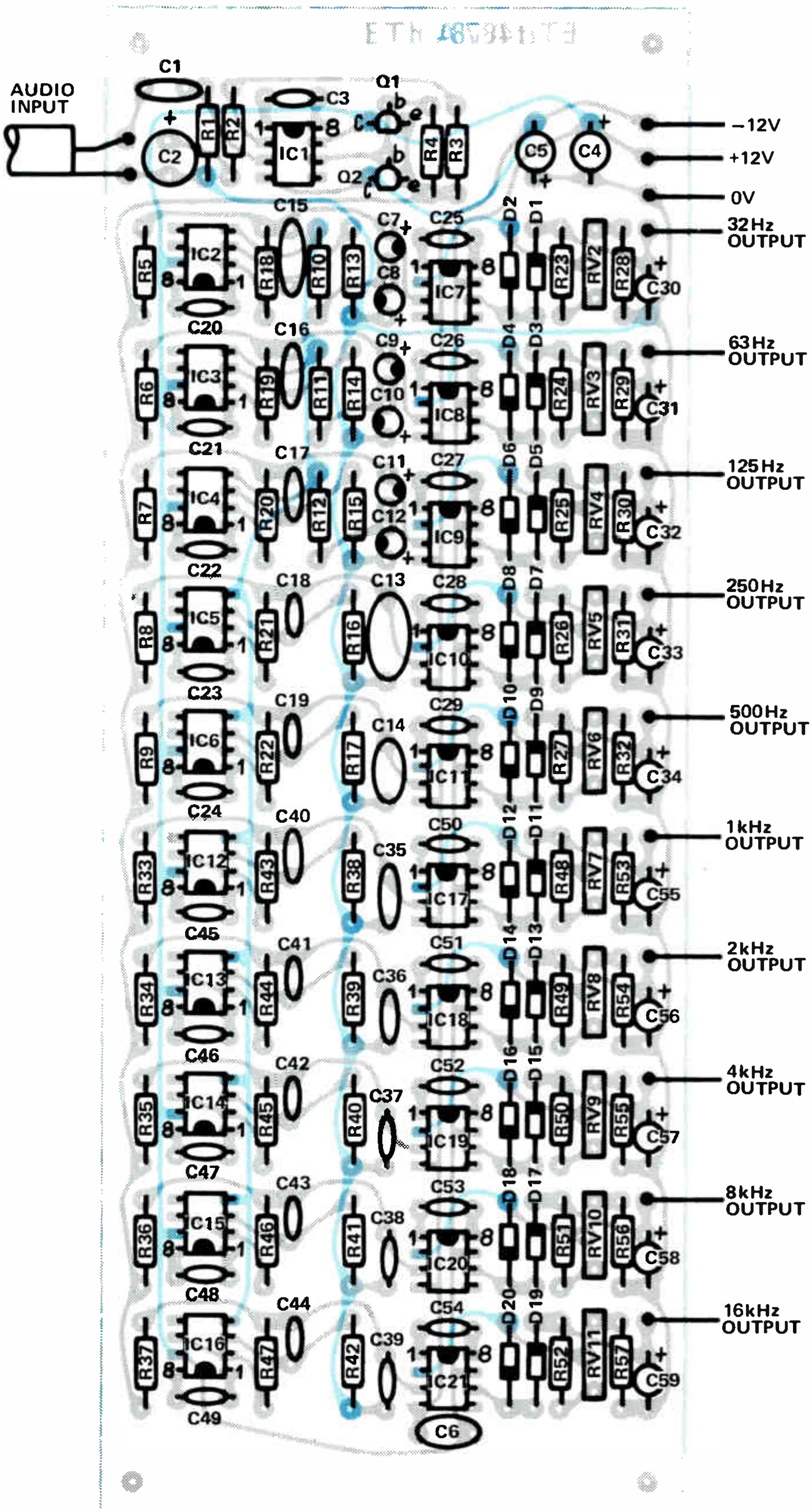


Fig. 5. The component overlay of the filter-rectifier board.

PARTS LIST – ETI 487

Resistors	all ½W 5%
R1	220k
R2	2k2
R3	220k
R4	1k
R5–R9	10k
R10–R17	1M
R18–R22	220R
R23–R27	1M
R28–R32	100k
R33–R37	10k
R38–R42	1M
R43–R47	220R
R48–R52	1M
R53–R57	100k
R58,59	180k
R60	100k
R61	82k
R62,63	15k
R64	22k
R65	470R
R66	18k
R67	15k
R68	12k
R69	820R
R70	180k
R71	12k
R72,73	10k
R74	2k2
R75	4k7
R76–R78	10k
R79	22k
R80	56k
R81	5k6
R82	3k9
R83	18k
R84	100k
R85	390k
R86	1M
R87	6k8
R88	10k
R89	18k
R90	47k
R91,92	100k
R93	47k
R94	100k
R95	47k
R96	10k
Potentiometers	
RV1	47k log rotary
RV2–RV11	250k trim
Capacitors	
C1	100n polyester
C2	10µ 25V electro
C3	3p3 ceramic
C4,5	10µ 25V electro
C6	100n polyester
C7,8	3µ3 16V tantalum
C9,10	1µ5 16V "
C11,12	1µ0 16V "

Construction

Due to the complexity of the unit it is recommended that PC boards are used. These boards are assembled as per the overlay diagrams. Watch the orientation of all the ICs, diodes, capacitors, etc., when installing them. Note that as the board is not a plated through type that the tracks on the top side of the board must also be soldered to the components. This prevents the use of sockets for the ICs but they are not really worth the cost for low priced ICs

Capacitors continued

C13 . . . . 220n polyester
C14 . . . . 100n "
C15 . . . . 68n "
C16 . . . . 33n "
C17 . . . . 18n "
C18 . . . . 8n2 "
C19 . . . . 3n9 "
C20–C24 . . . 33p ceramic
C25–C29 . . . 10p "
C30–C34 . . . 2µ2 25 V electro
C35 . . . . 47n polyester
C36 . . . . 27n "
C37 . . . . 12n "
C38 . . . . 6n8 "
C39 . . . . 3n3 "
C40 . . . . 2n2 "
C41 . . . . 1n0 "
C42 . . . . 560p ceramic
C43 . . . . 270p "
C44 . . . . 150p "
C45–C49 . . . 33p "
C50–C54 . . . 10p "
C55–C59 . . . 2µ2 16 V electro
C60,61 . . . . . 25µ 16 V "
C62 . . . . 820p ceramic
C63 . . . . 2n7 polyester
C64 . . . . 5n6 "
C65 . . . . 33p ceramic
C66 . . . . 150p "
C67 . . . . 3p3 "
C68 . . . . 150p "
C69 . . . . 10p "
C70 . . . . 33p "
C71 . . . . 2n2 polyester
C72 . . . . 560p ceramic
C73,74 . . . . . 220µ 25 V electro
C75,76 . . . . . 10µ 25 V "

Semiconductors

IC1–IC21 . . . LM301A
IC22 . . . . . . 4017 (CMOS)
IC23–IC25 . . 4016 (CMOS)
IC26–IC28 . . LM301A
IC29 . . . . . . 4011B (CMOS)
IC30 . . . . . . LM301A
IC31 7812
IC32 7912
Q1 . . . . BC548
Q2 . . . . BC558
Q3,4 . . . . . . BC548
Q5–Q7 . . . . . BC558
D1–D27 . . . . 1N914
D28–D31 . . . 1N4001

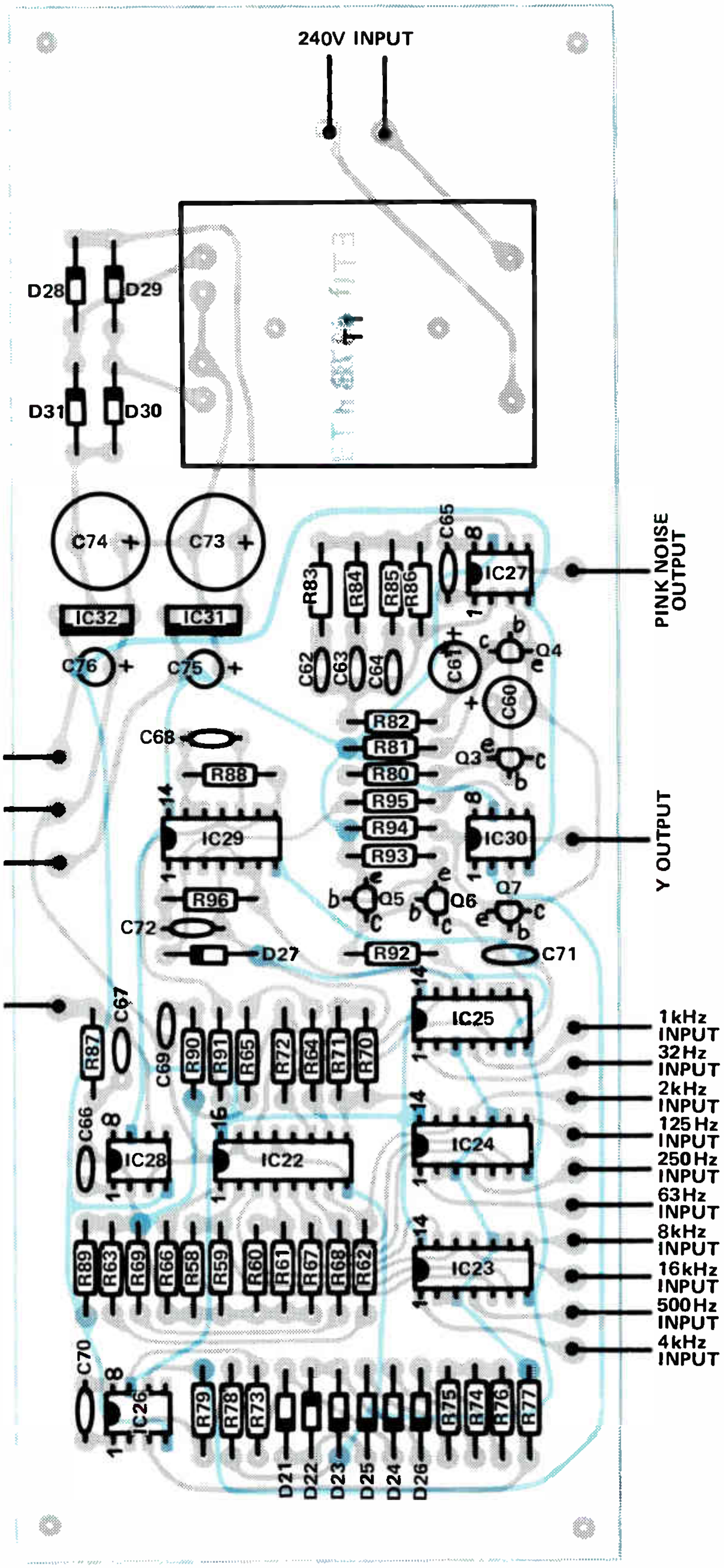
Miscellaneous

PC boards ETI 487A, 487B
Transformer PL24–5VA
Case to suit
3 core flex and plug
240V power switch
Input / output terminals to suit

as used.

With the board 487A be very careful as there is 240V on the board. It is recommended that the wires be terminated directly to the board, without PC board pins, and that the 240V tracks on the underside of the board be coated with epoxy to prevent contact.

We mounted the unit into a home-made box as we did not have a commercial one on hand to suit.



Erratum: Capacitor C75 is shown positioned incorrectly. It should be turned through 180°. A link must also be made between the collector of Q2 and the end of R10 where the –12V track begins on the underside on the board.

Number Two ETI CIRCUITS

Contents

Alignment

This can be done using the pink noise generator or preferably with a sine wave oscillator.

Connect the unit to the oscilloscope switched into the X Y mode. With the unit switched on and a signal connected, adjust the X gain and shift to obtain a series of ten vertical bars across the screen. Increase the input signal until the columns will not get any higher. Adjust the Y gain and shift until the column is the height of the screen. Note that the scope should be dc coupled.

Now by sweeping the oscillator frequency it will be found that each column will come up in sequence. Adjust the frequency to peak the 16 kHz column. Now adjust RV11 to about 75% of its travel (wiper towards RV10) and then adjust the overall sensitivity control to give a column height of about 80%.

Now using the same amplitude adjust the signal generator frequency until the 8kHz column peaks and adjust RV10 to give the same height. Each of the filters should be adjusted in the same way. Note that due to component variations the actual peak of a filter may not exactly coincide with its nominal frequency. Also the 16kHz filter has the greatest loss which is the reason for starting with it near its maximum gain.

By taking the pink noise output to the input each column should be approximately the same height. Due to the nature of noise the top of the columns will jump up and down a little and this should be averaged out by the eye.

If an oscillator is not available the noise generator can be used and the potentiometers adjusted to give an even response. Also, if desired, a vertical dB scale can be made.

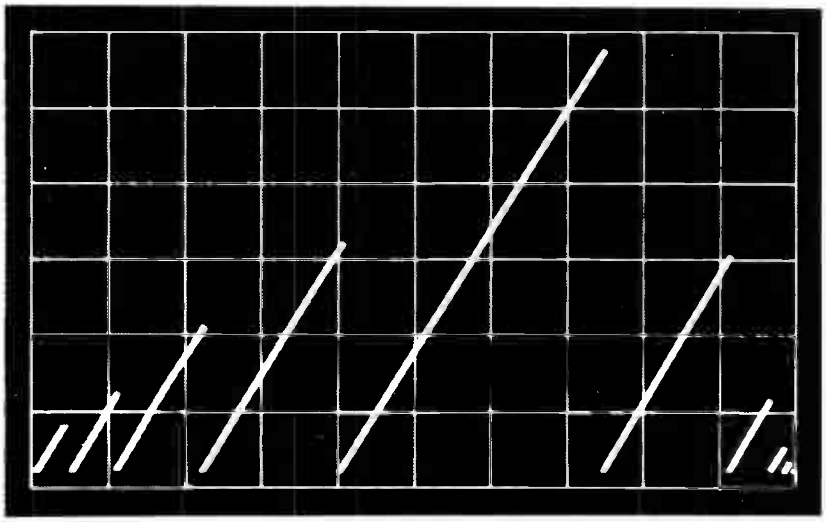


Fig.7. The waveform on the Y output (vertical) with a 1 kHz tone input. See page 38 for the X-Y display. Note that the time between cycles varies with the height.

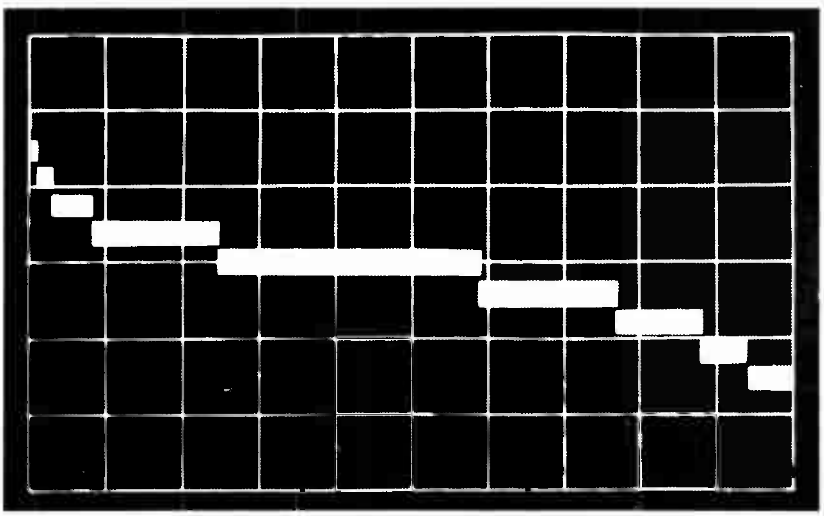
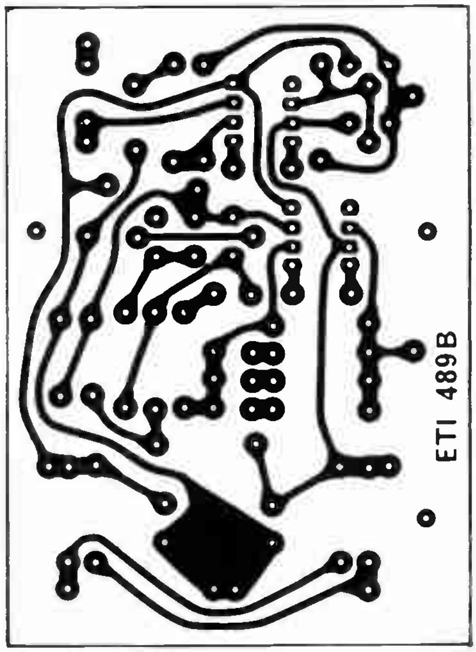
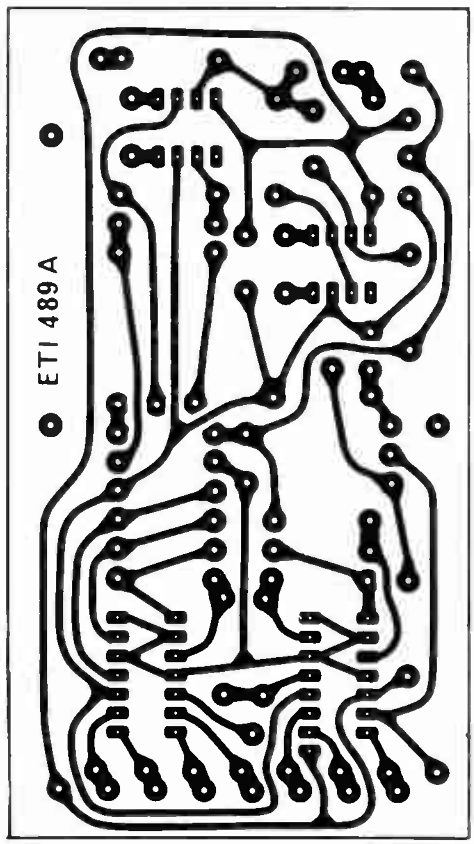


Fig.8. The waveform on the X (horizontal) output. As this starts at +4V which is the right hand side of the screen, the 16 kHz output is sampled first. Note that the time between steps corresponds to that in fig. 7.

AUDIO SPECTRUM ANALYSER 2 (see page 140)
PC Boards 489A & B.



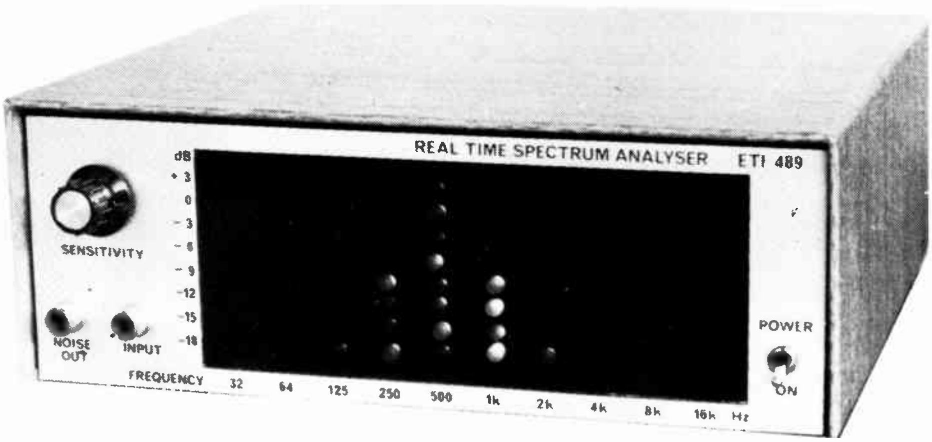


Audio Spectrum Analyser 2

LED display for compact, easy-to-build unit.

OUR PREVIOUS Real Time Audio Analyser design produced beautiful displays on the screen of an oscilloscope but this means, of course, that to use the device one has to have a scope. Not everyone has, and with this in mind, we contemplated the design of a more conventional analyser with LED bar-graph display. Urged on by reader response to our hint that this design was on the cards, we have gone ahead and produced the project in double-quick time.

This version has the great advantage of portability over the previous design, and also looks better than a scope sitting next to your brand new, 21st century styled hi-fi! It is also easier to set up and trouble-shoot.



Design Features

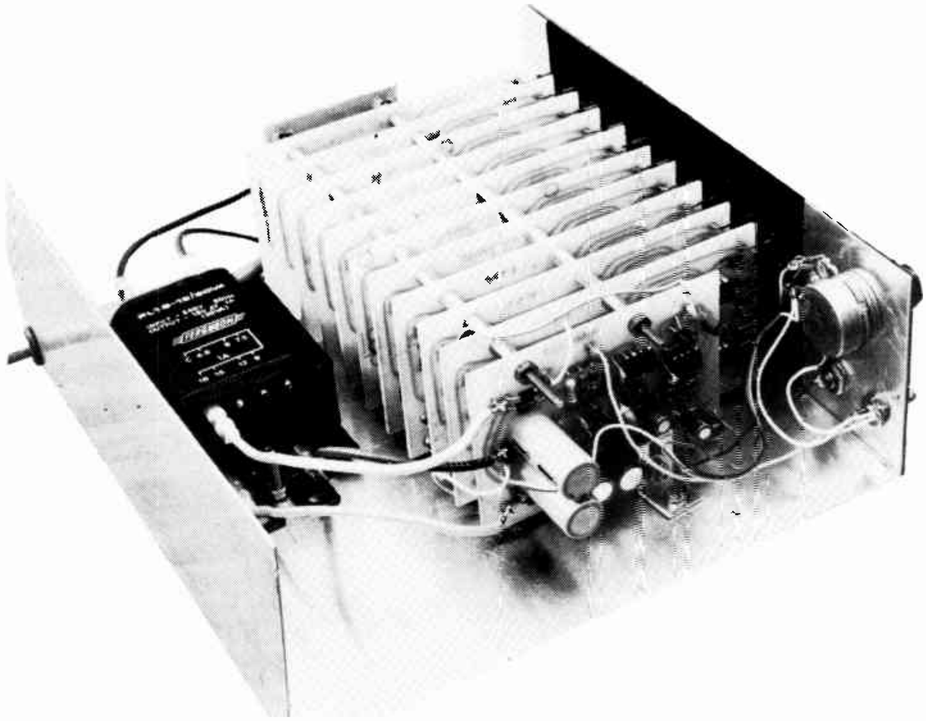
When we proposed a LED version of the spectrum analyser we initially were going to use the original filter board and design a new logic board which multiplexed the LED display. The only question at that time was whether to multiplex the LEDs as columns or as individual LEDs. The column method is easier on the power supply as the peak current is only 10 times the average current while singly the peak current is 80 times the average. This is not quite accurate because a multiplexed LED requires less average current for the same output than one continuously on. However the column method also requires one extra diode per LED to give the isolation required between columns.

SPECIFICATION – ETI 489

No. of bands	10
Frequencies	31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16k
Filter characteristics	−12dB, one octave from nominal centre frequency
Display	LED display 3dB spacing
Input level	50mV – 10V
Input impedance	200k
Pink noise output	200mV

After struggling with the PC board layout which was developing into a double sided board similar to the filter board of the previous analyser, we decided there must be an easier way to make a living! The question was then raised of whether it was worthwhile to multiplex the display at all and the answer was the project as it appears here.

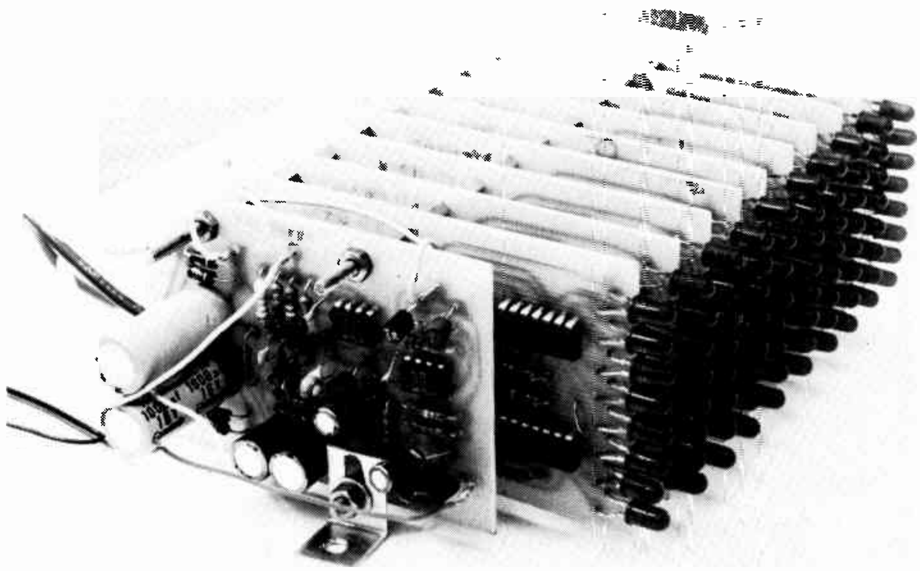
The individual board approach not only makes fault finding easier and less likely, it also allows single sided PC boards to be used throughout. The system can also be expanded (or cut down) as desired simply by changing the filter components and the number of display boards. The power supply is capable of supplying up to 20 display boards without increasing the filter capacitors.



Construction

Assemble the power supply board and the ten filter display boards with the aid of the overlays. The filter components can be selected from Table 1 noting that when the tantalum capacitors are used in the three lower octaves a bias resistor R15 is needed. The LEDs should be installed as evenly as possible with the polarity correct.

We assembled the units on 1/8" brooker rod with 12.5mm spacers between the boards. Metal brackets are used at each end to support the assembly. On the filter display boards the power rails and the input are all common and for the power supply we used long lengths of tinned copper wire threaded through the holes. The input lead should be done with separate links to allow the units to be serviced later if needed.



Before assembling the unit however each board should be checked with an oscillator to check it for the correct frequency and to adjust the calibration potentiometer. This is best done by measuring the sensitivity of the 16 kHz board with RV2 set for maximum sensitivity and adjusting all the others till they are the same.

We made a metal box with a piece of red perspex for a window to house the unit. If it is to be used with an equaliser (such as the ETI 484) it could be built into the same box.

It will be found with the economical LED available that there will be a difference in brilliance between them. If desired matched LEDs are available but not for 20 cents each!

TABLE 1

Centre frequency	R15	C14,C15 tantalum	C16 polyester	C17 polyester
32	1 M	$3\mu3$	–	68n
63	1 M	$1\mu5$	–	33n
125	1 M	$1\mu0$	–	18n
250	–	–	220n	8n2
500	–	–	100n	3n9
1 k	–	–	47n	2n2
2 k	–	–	27n	1n0
4 k	–	–	12n	560p
8 k	–	–	6n8	270p
16 k	–	–	3n3	150p

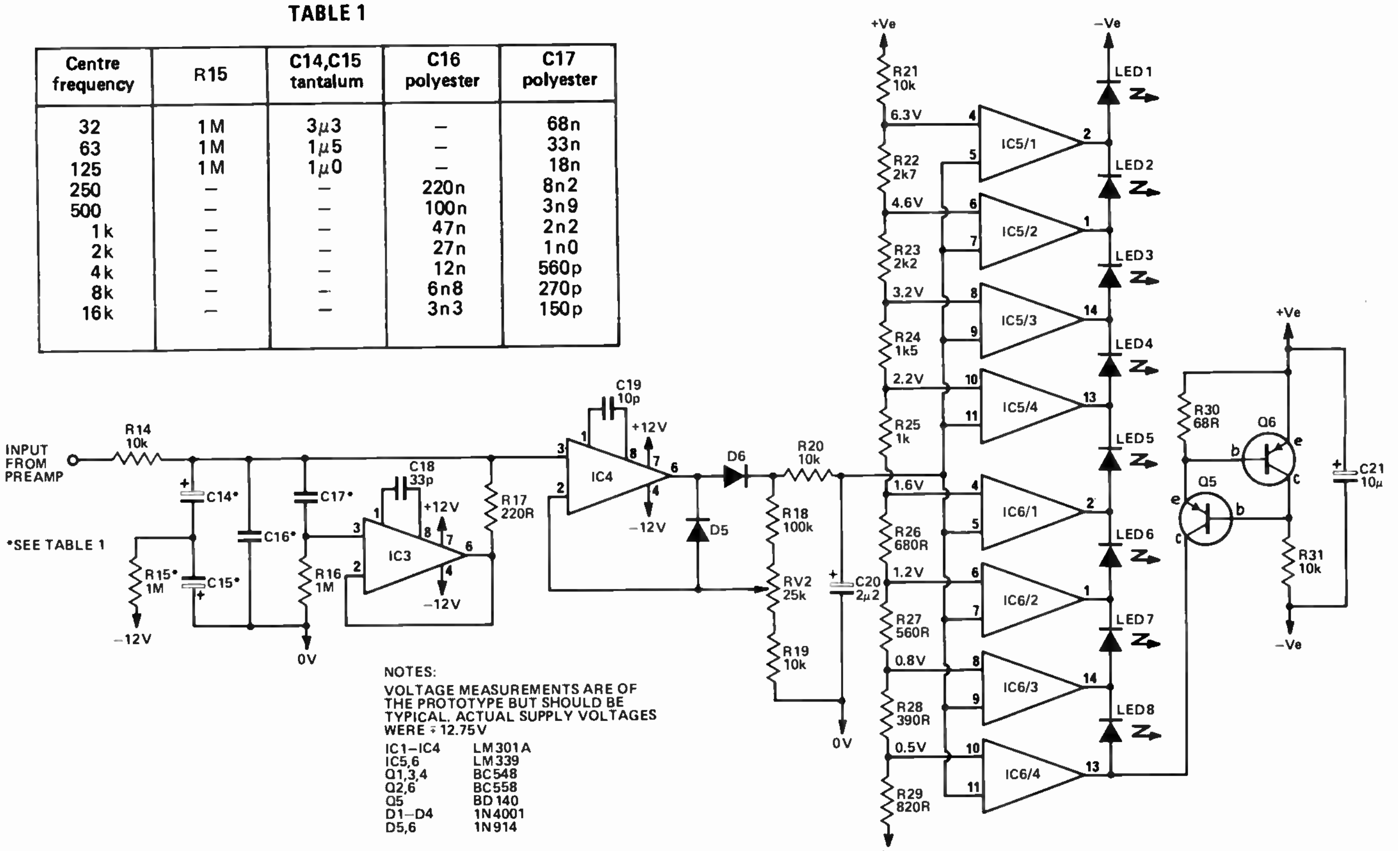


Fig. 1. The circuit diagram of the filter display board.

HOW IT WORKS – ETI 489

The input signal to the unit is initially buffered and amplified by IC1 and is then split into octave bands, rectified and displayed by a "bar" of LEDs. We have used 10 separate boards for the rectifier-display as only the component valves in the filter are different.

The filter is a parallel LC network where the inductive part is a gyrator formed by IC3, CI7, R16 and R17. The valve of such an "inductor" is R16 x R17 x C17 Henrys (C17 in Farads). This, with the parallel capacitor C16 and the series resistor R14, form a band pass filter.

The output of the filter is half wave rectified by IC4 which also provides a gain of about 5 before the signal is smoothed by C20.

The eight LEDs in each individual display are connected in a series chain which is supplied with 10 mA by the constant current source Q5, Q6. Control of how many LEDs will be on is done by IC5 and IC6. These are quad voltage comparators which have as the output stage an open collector NPN transistor with its emitter connected to the negative supply rail. These compare the output of the rectifier with the voltage set on the resistive divider R21-R29 and "short" out the unwanted LEDs.

The power supply is a simple fullwave rectified with a centre tap giving ∓12 V dc. Due to the load (about 150 mA) there is about one volt ripple on the supply rail but this does not affect the operation of the unit. As the current drawn by the filter display boards does not change with the number of LEDs on the supply voltage remains reasonably constant.

The 100 Hz ripple does however affect the noise generator and this has been changed from the 487 analyser to accommodate this. The noise generator consists of Q3 which is used as a zener diode where the noise current is amplified by Q4. The output of Q4 is white noise and to give pink noise a 3 dB/octave filter is needed. IC2 and the associated capacitors and resistors provide this filter.

PARTS LIST – ETI 489A

Filter–Display boards 10 required

Resistors all ½W 5%
R14 . . . . . . 10 k
R15 . . . . . . see table 1
R16 . . . . . . 1 M
R17 . . . . . . 220 R
R18 . . . . . . 100 k
R19–R21 . . 10 k
R22 . . . . . . 2 k7
R23 . . . . . . 2 k2
R24 . . . . . . 1 k5
R25 . . . . . . 1 k
R26 . . . . . . 680 R
R27 . . . . . . 560 R
R28 . . . . . . 390 R
R29 . . . . . . 820 R
R30 . . . . . . 68 R
R31 . . . . . . 10 k

Potentiometers
RV2 . . . . . 25 k trim

Capacitors
C14–C17 . . see table 1
C18 . . . . . . 33p ceramic
C19 . . . . . . 10p ceramic
C20 . . . . . . $2\mu2$ 25 V electro*
C21 . . . . . . 10μ 25 V electro*

Semiconductors
IC3,4 . . . . . LM301 A
IC5,6 . . . . . LM339
Q5 . . . . . . BD 140
Q6 . . . . . . BC558
D5,6 . . . . . 1N914
LED1–LED8

Miscellaneous
PC board ETI 489 A

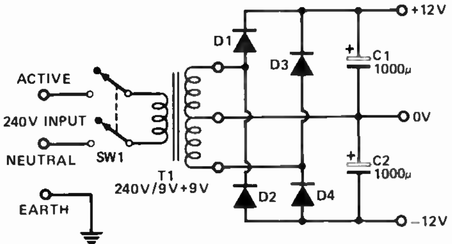


Fig. 2. The power supply circuit.

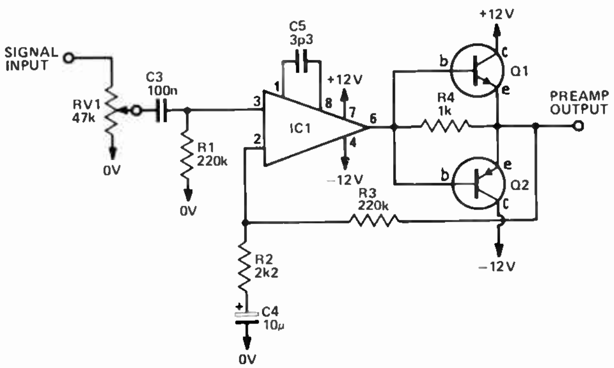


Fig. 3. The circuit of the preamplifier—buffer.

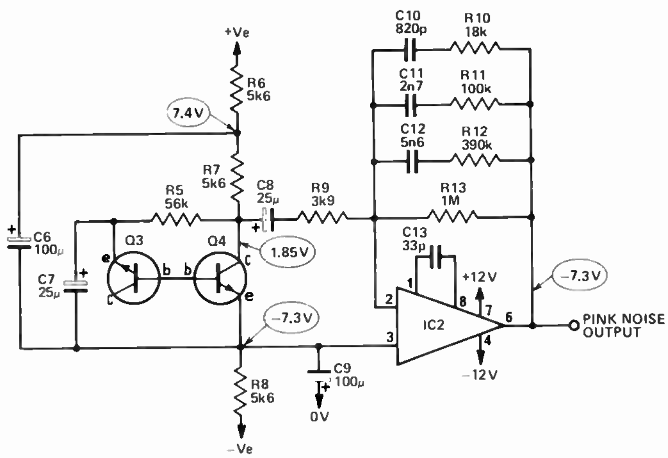


Fig. 4. The circuit diagram of the pink noise generator.

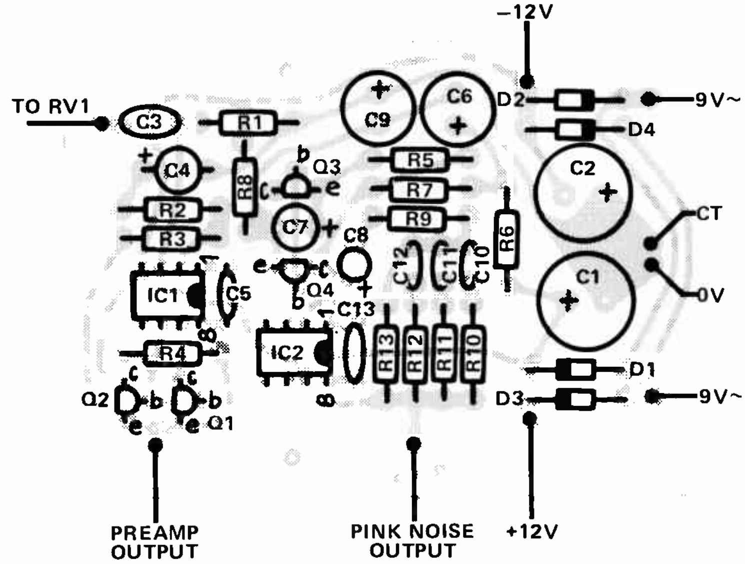
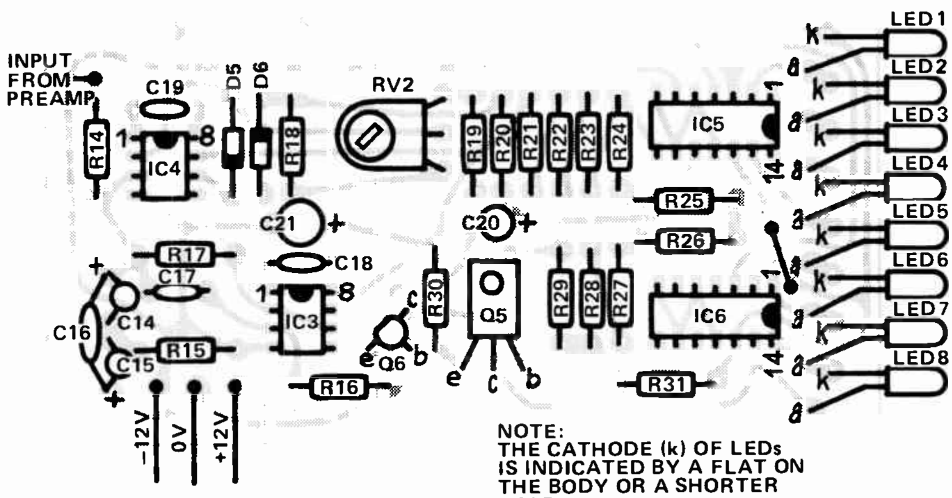


Fig. 5. The component overlay of board B.

PC boards for this project are on page 139.



NOTE:
THE CATHODE (k) OF LEDs IS INDICATED BY A FLAT ON THE BODY OR A SHORTER LEAD.

Fig. 6. The component overlay of board A.

PARTS LIST – ETI 489B

Power Supply board

Resistors all ½W 5%

R1 . . . . . . 220k
R2 . . . . . . 2k2
R3 . . . . . . 220k
R4 . . . . . . 1k
R5 . . . . . . 56k
R6–R8. . . . 5k6
R9 . . . . . . 3k9
R10. . . . . . 18k
R11. . . . . . 100k
R12. . . . . . 390k
R13. . . . . . 1M

Potentiometers

RV1 . . . . . 47k log rotary

Capacitors

C1,2 . . . . . 1000µ 16V electro*
C3. . . . . . . 100n polyester
C4. . . . . . . 10µ 25V electro*
C5. . . . . . . 3p3 ceramic
C6. . . . . . . 100µ 25V electro*
C7,8 . . . . . 25µ 25V electro*
C9. . . . . . . 100µ 25V electro*
C10 . . . . . . 820p ceramic
C11 . . . . . . 2n7 polyester
C12 . . . . . . 5n6 polyester
C13 . . . . . . 33p ceramic

Semiconductors

IC1,2 . . . . . LM301A
Q1 . . . . . . BC548
Q2 . . . . . . BC558
Q3,4 . . . . . BC548
D1–D4. . . . 1N4001

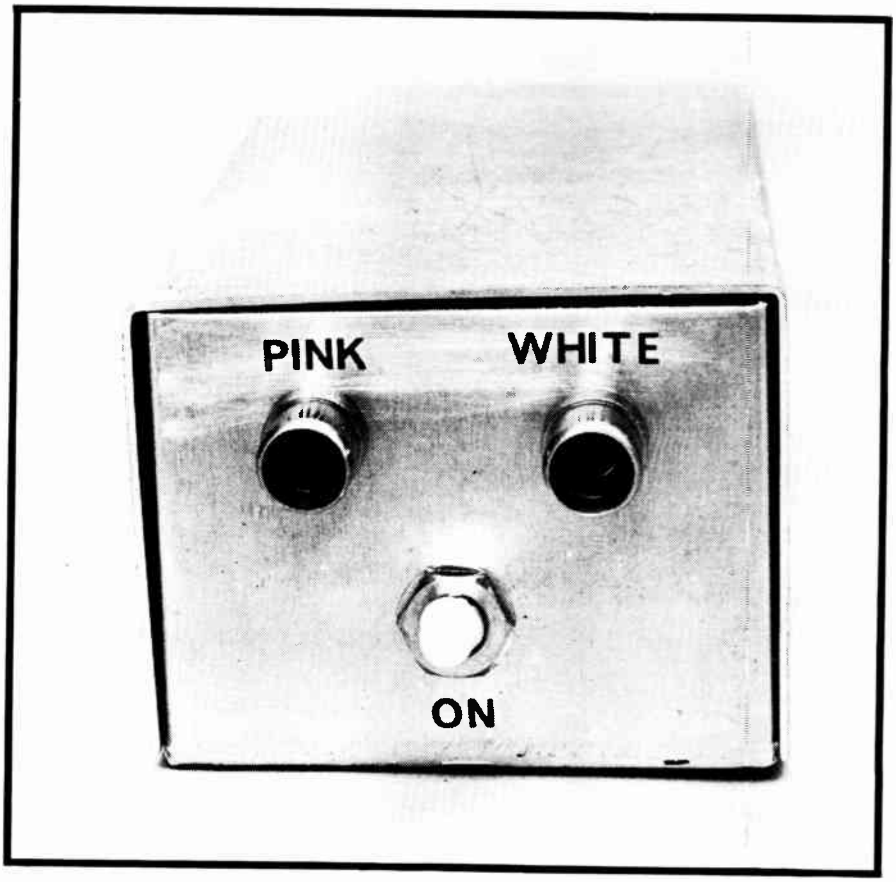
Miscellaneous

PC board ETI 489B
Transformer 240V/9V+9V PL18/20VA
SW1 DPDT 240V toggle switch
Case to suit

*all electrolytic capacitors PC board or single ended type.

PROJECT 441

AUDIO NOISE GENERATOR



Simple circuit generates both white and pink noise.

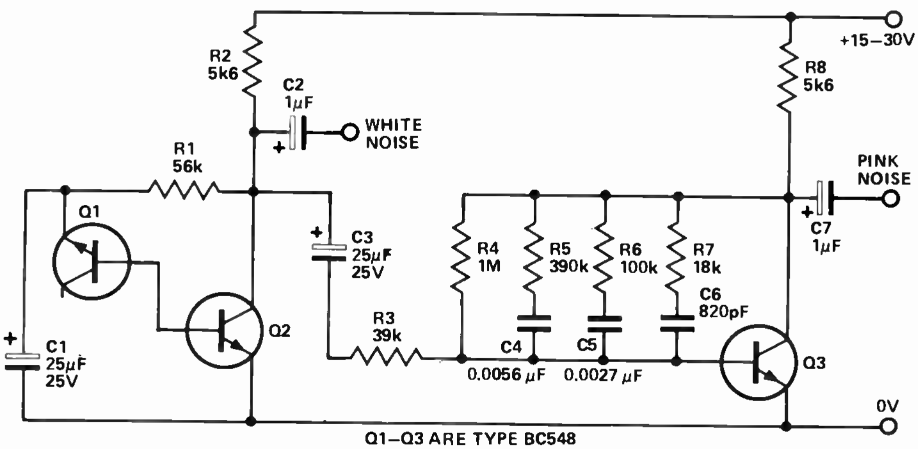


Fig. 1. Circuit diagram of the noise generator.

NOISE is generally an undesirable phenomena that degrades the performance of many measurement and instrumentation systems. It therefore seems strange that anyone should want to generate noise, but this is often the case.

Noise generators are often used to inject noise into radio-frequency amplifiers in order to evaluate their small signal performance. They are also used to test audio systems, and as random signal sources for wind-like effects in electronic music.

There are two commonly used noise source characteristics, 'pink' and 'white'. White noise is so called because it has equal noise energy in equal bandwidths over the total frequency range of interest. Thus, for example, a white noise source would have equal energy in the band 100 to 200 Hz to that in the band 5000 to 5100 Hz.

HOW IT WORKS — ETI 441

In the days when vacuum tubes were in common use the most commonly used form of noise generator was a vacuum-tube diode operated in the current saturation mode. Nowadays noise generators may be very complex indeed. Highly complex digital generators which produce psuedo-random digital noise may cost many thousands of dollars. An example of a simpler type of digital noise source may be found in our synthesizer design (see International Music Synthesizer 4600 ETI December 1973). However for audio work of a general nature the most commonly used, and the simplest, method is to use a zener diode as a noise generator.

Transistor Q1 is in fact used as a zener diode. The normal base-emitter junction is reverse-biased and goes into zener break-down at about 7 to 8 volts. The zener noise current from Q1 flows into the base of Q2 such that an output of about 150 millivolts of white noise is available.

The 'zener', besides being the noise source, also biases Q2 correctly, and the noise output of Q2 is fed directly to the White Noise output.

To convert the white noise to pink a filter is required which provides a 3 dB cut per octave as the frequency increases. A conventional RC network is not suitable as a single RC stage gives a cut of 6 dB per octave. Hence a special network of Rs and Cs is required in order to approximate the 3 dB-per-octave slope required. Since such a filter attenuates the noise considerably an amplifier is used to restore the output level. Transistor Q3 is this amplifier and the pink noise filter is connected as a feedback network between collector and base in order to obtain the required characteristic by controlling the gain-versus-frequency of the transistor. The output of transistor Q3 is thus the pink-noise required and is fed to the relevant output socket.

PARTS LIST — ETI 441

R1	Resistor	56k	½W	5%
R2	"	5k6	½W	5%
R3	"	39k	½W	5%
R4	"	1M	½W	5%
R5	"	390k	½W	5%
R6	"	100k	½W	5%
R7	"	18k	½W	5%
R8	"	5k6	½W	5%
C1	Capacitor	25μF	25V	electro
C2	"	1μF	25V	electro
C3	"	25μF	25V	electro
C4	"	0.0056μF		polyester
C5	"	0.0027μF		polyester
C6	"	820pF		ceramic
C7	"	1μF	25V	electro

Q1-Q3 Transistor BC548, BC108 or similar

PC board ETI 441

CASE

BATTERIES

OUTPUT SOCKETS

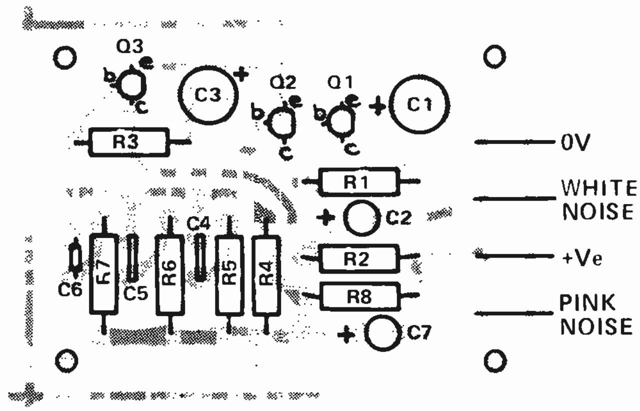
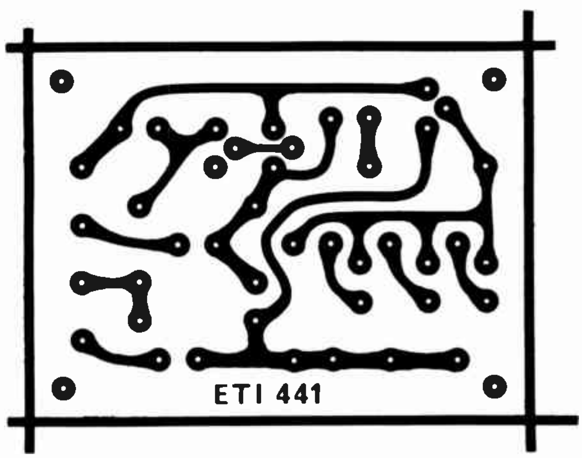


Fig. 2. Component overlay.



Printed circuit layout. Full size 67 x 49 mm.

If white noise is filtered or modified in any way it is referred to as coloured noise or, often more specifically, as 'pink' or 'grey' noise. The term pink noise should be restricted to the noise characteristic that has equal energy per percentage change in bandwidth. For example with true pink noise the energy between 100 Hz and 200 Hz should equal that between 5000 Hz and 10 000 Hz (100% change in both cases).

Pink noise therefore appears to have more bass content than does white noise, and it appears to the ear to have a more uniform output level in audio testing. To change white noise to pink noise a filter is required that reduces the output level by 3 dB per octave (10 dB per decade) as the frequency is increased. The ETI 441 Noise Generator is designed to provide both white and pink noise as required.

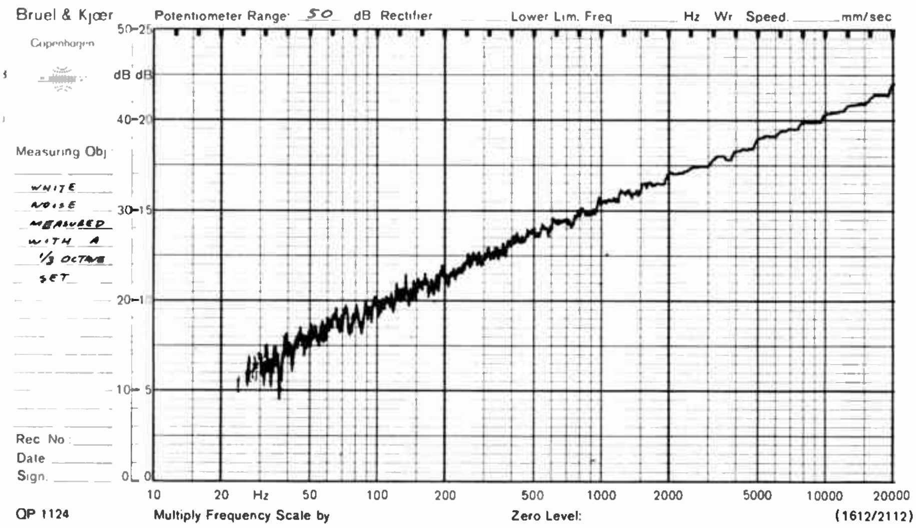
CONSTRUCTION

Construction is relatively simple and almost any of the common methods, such as Veroboard or Matrix board, may be used if desired. For neatness and ease of assembly it is hard to beat a proper printed-circuit board and for this reason we have provided details of a suitable board.

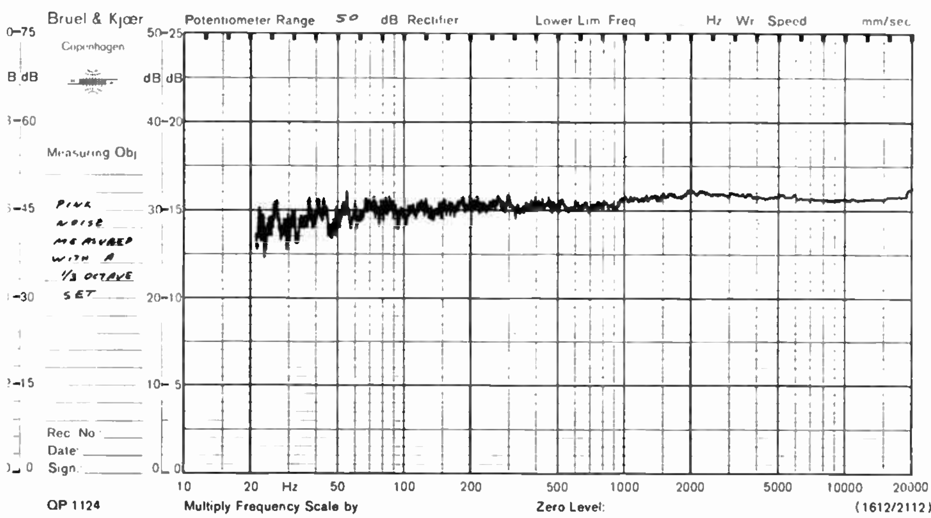
Almost any type of NPN transistor will do for the generator provided that the one used for Q3 has a gain of 100 or more. If BC548 type are used watch out for the two different pin connections used by different manufacturers.

For use as a separate instrument in general experimentation the unit will need to be powered by a pair of nine-volt batteries. However if the unit is to be built into some other piece of equipment, as is often the case, any supply within the equipment which has an output of between 15 and 30 volts dc will be suitable.

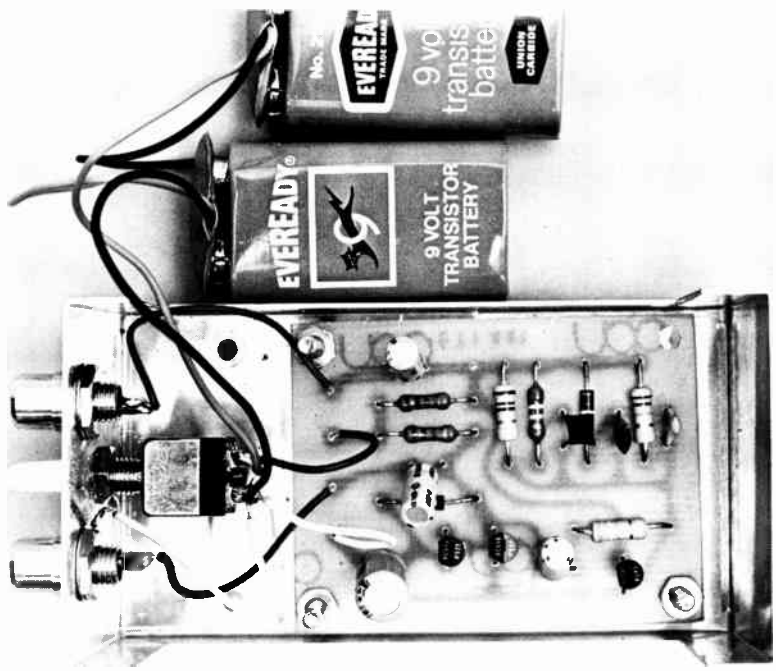
AUDIO NOISE GENERATOR



Amplitude of white noise versus frequency as measured with a one-third octave filter set.

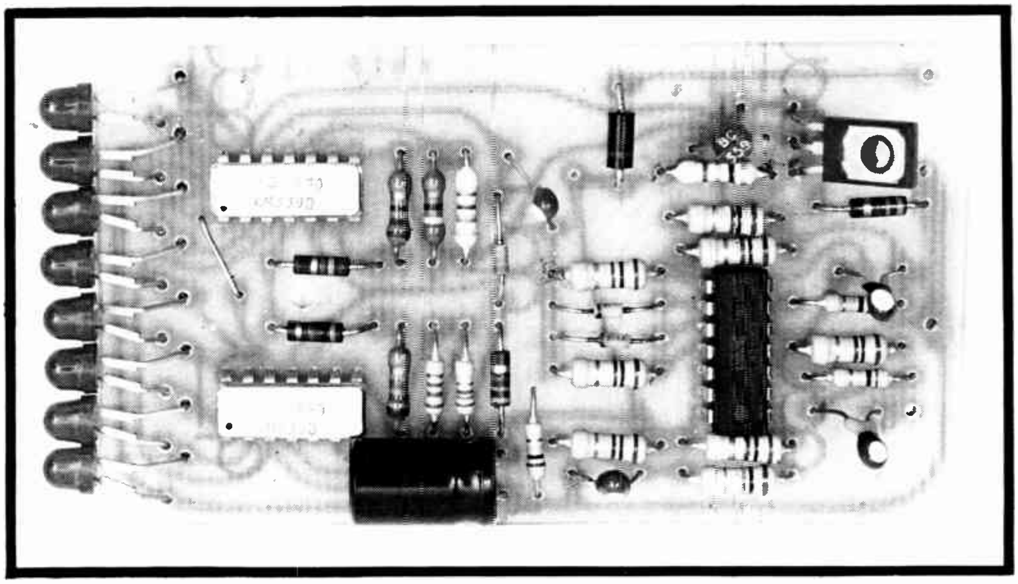


Amplitude of pink noise versus frequency as measured with a one-third octave filter set.



Internal layout of the generator.

AUDIO LEVEL METER



PROJECT 438

Peak and average audio levels are indicated by a bar of light.

HIGH-POWER amplifiers usually incorporate meters to indicate the output-power levels in each channel. These meters are often called VU meters but in most cases they resemble proper VU meters only in the way they are scaled.

A professional VU meter is the industry standard for measuring the levels of complex music waveforms. It has a scale marked from −20 to +3 VU (on a steady state signal VU correspond to dB) where '0' VU corresponds to a level of one milliwatt into 600 ohms. The meter has a carefully controlled time constant such that if a reference tone level is applied the pointer of the meter will take 0.3 seconds to reach 99% of the reference level, and will then overshoot by not more than 1.5% and not less than 1.0%.

The professional VU meter is thus an instrument that has been designed to give a reasonable compromise between indicating the fast peaks and the average levels of a complex music waveform.

In contrast the meters fitted to some amplifiers have scales calibrated in VU but usually relying on the inertia of the meter movement to provide meter averaging. Apart from this the 0 VU point corresponds to the rated power output of the amplifier — not to 1 mW into 600 ohms (equivalent to 75 mW in 8 ohms). Strictly speaking therefore such meters should be called level or power meters, not VU meters.

Even the best of such meters are not fast enough to indicate accurately the peak levels which occur in music and hence are useless for detecting the onset of amplifier clipping. This is vital as at clipping amplifier distortion rises rapidly.

One alternative is to use in addition to the level meter a clipping indicator that detects fast peaks which exceed a preset level. The ETI 417 OVER-LED project was such an instrument — it flashed an LED when a music transient exceeded clipping level.

The circuit described in this project is best described as a 'level meter'. It uses an array of LED diodes set to illuminate at successively higher increments in music level. With this type of display an estimate can quite easily be made of channel balance, and all transients, no matter how fast, are detected and indicated.

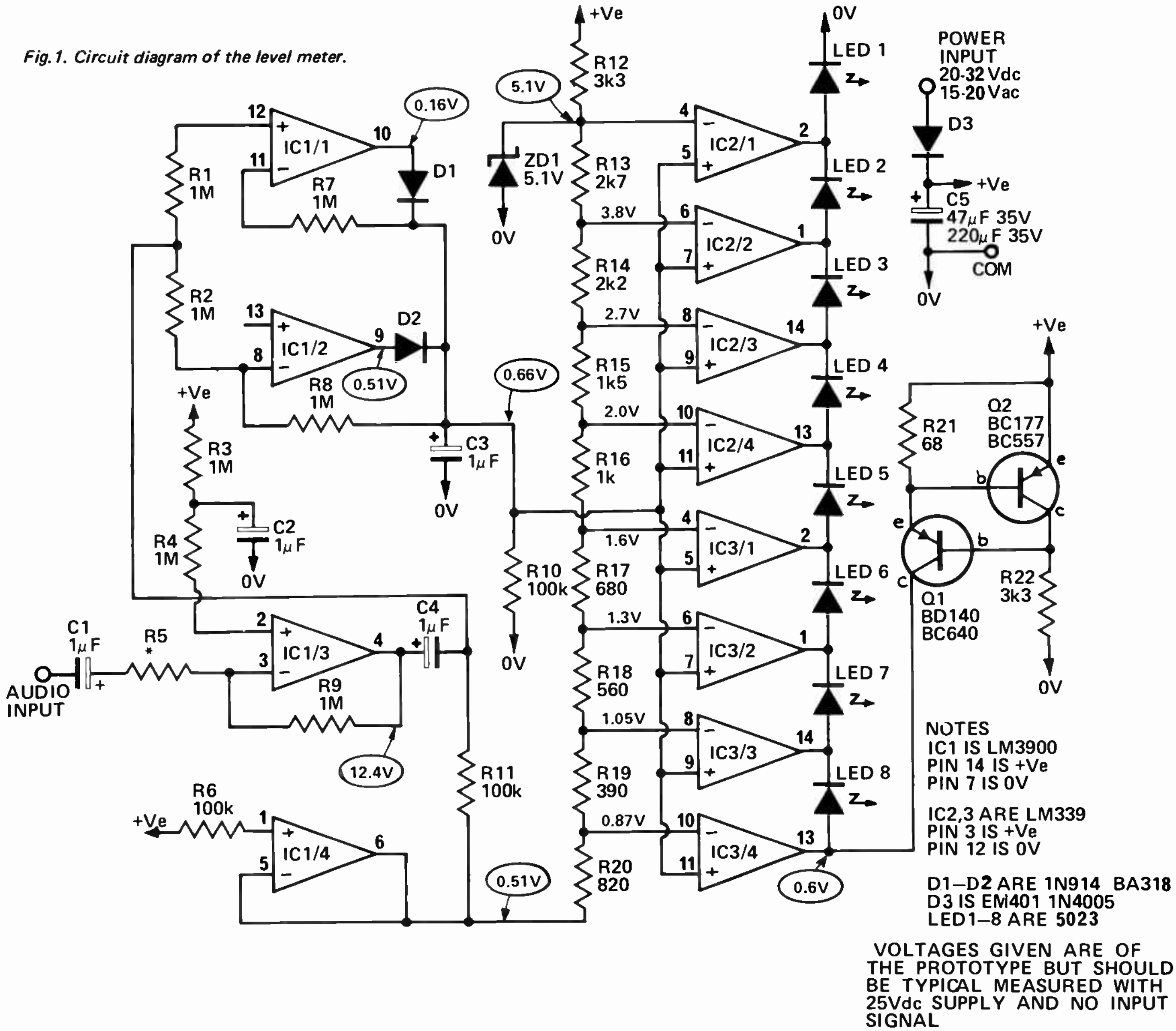
SPECIFICATION

Supply voltage	20 to 32 volts dc 15 to 20 volts dc
Supply current	16 mA dc approx,
Input sensitivity (VU meter)	500 k/v
Indication	8 LEDs 3 dB apart
Attack time	1 ms
Release time	0.5 sec.

DESIGN FEATURES

The ETI 438 Level Meter can be arranged to indicate levels either in 'VU meter' format or in output power format. In the 'VU-meter' format the eight diodes light at 3 dB intervals from −18 to +3 VU where 0 VU corresponds to the nominal voltage required. Alternately as a power meter (remember that an amplifier cannot be driven beyond the clipping point) the top LED indicates maximum power and each lower LED indicates half the power of the one above it. The LEDs of the meter could thus be labelled,

Fig. 1. Circuit diagram of the level meter.



HOW IT WORKS – ETI 438

Although the circuitry of the level meter looks complicated the complete instrument only uses three ICs. These are an LM3900 which is a quad amplifier and two LM339s which are quad voltage comparators.

The input signal is amplified and buffered by IC1/3 to provide about 2.5 volts out at 0 VU input. The value of R5 is selected to give the sensitivity required for amplifiers of different power outputs. The gain of this amplifier is equal to the ratio of R9/R5.

A positive peak detector, IC1/1, and an inverting negative peak detector, IC1/2, give an output which represents the absolute peak level. Capacitor C3 and resistor R10 provide the peak hold and decay time. IC1/4 provides compensation for the 0.6 volt offsets of the LM3900 inputs.

The eight comparators are connected to a resistor divider chain the top of which is fed from a 5.1 volt supply which is stabilized by a zener. The resistor values are calculated to provide reference voltage steps at 3 dB intervals. The output of the detector is applied to all the non-inverting inputs of the comparators.

The LEDs are all connected in series and supplied with a constant current of 10 mA by the source consisting of Q1 and Q2. The outputs of the comparators are via open collector transistors which are "ON" if the input is lower than the reference voltage at the particular comparator input. With no input signal at all the comparators are all on thus shorting out all the LEDs so that none is on. As the input voltage rises the comparators turn off in sequence allowing the 10 mA to flow through the LEDs. Thus as the voltage increases a bar of light of increasing height is formed by the LEDs.

The current drawn from the power supply is about 16 mA and is independent of the number of LEDs which are on. Supply voltage is not critical and may be anywhere between 20 and 32 volts. Providing the supply is between these limits the unit will also be insensitive to supply ripple. When working from a dc supply a 47 microfarad filter capacitor is required but if an ac supply is used then the capacitor should be increased to 220 microfarad to minimize ripple. A single diode is used to both rectify the ac input and to prevent damage due to accidental reversed polarity if a dc supply is used.

PARTS LIST – ETI 438

R21	Resistor	68 ohm	½W	5%
R19	"	390 ohm	½W	5%
R18	"	560 ohm	½W	5%
R17	"	680 ohm	½W	5%
R20	"	820 ohm	½W	5%
R16	"	1k	½W	5%
R15	"	1k5	½W	5%
R14	"	2k2	½W	5%
R13	"	2k7	½W	5%
R12,22	"	3k3	½W	5%
R6,10,11	Resistor	100k	½W	5%
R1,2,7,8	"	1M	½W	5%
R3,4,9	"	See Table 1	½W	5%
R5	"	See Table 1	½W	5%
C1,2,3,4	Capacitor	1 µF	35V electro	
*C5A	"	47 µF	35V electro	
*C5B	"	220µF	35V electro	

* use 47 µF for dc operation 220 µF for ac operation

IC1 Integrated Circuit LM 3900
IC2,3 Integrated Circuit LM 339

D1,2 Diode IN914, BA318 or similar
D3 " EM401, IN4005 or similar
ZD1 Zener diode 5.1 V 400 mW

Q1 Transistor BD 140, BC640
Q2 " BC177, BC 557

LED 1-8 L.E.D. 5023 or similar

PC board ETI 438

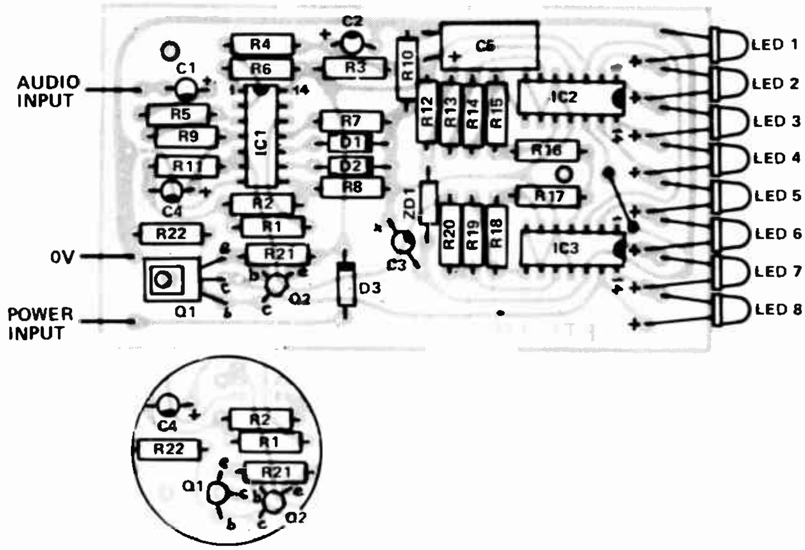


Fig.2. Component overlay using BD140 for Q1. Circled diagram shows use of alternative BC640

for example (for a 100 watt amplifier) 100, 50, 25, 12.5 watts etc.

The fast attack time of the meter (less than one millisecond) ensures that even very short transients are detected, whilst the relatively slow release time (0.5 seconds) provides a reasonably-accurate, average — level indication.

In most previous designs for such meters, discrete transistors were used to build level detectors. Temperature effects and variations in gain led to inaccuracies and to calibration difficulties. These problems have largely been overcome in the ETI 438 meter by using the LM339 IC which contains four accurate level detectors in one package. Additionally the LM339 also has an open-collector output stage which enables a constant current supply for the LEDs to be used. Thus the current and LED brightness are the same no matter how many LEDs are alight.

If required the interval between LEDs may be altered by changing the values of R13 to R20. Thus for example, a 6 dB interval could be used. Additionally the display could be extended to 12 or even 16 diodes by adding comparators and LEDs and by substituting another divider chain for R20 .(values would have to be calculated for the levels required). The positive inputs of the comparators would also be fed from C3 and R10.

A separate current source would be required as there is insufficient supply voltage available to light 16 LEDs in series. If the bottom LED in such a system indicates a level more than 30 dB down it may also be necessary to use a trimpot as the bottom resistor of the second divider chain to adjust for offsets etc.

The LM3900 is a quad differential amplifier which uses a current balancing technique at the input rather than the voltage balancing that is used with conventional operational amplifiers. Both the inputs "look" like the base-emitter junctions of normal transistors and both are at 0.6 volts with respect to ground. The currents into the two inputs must be equal if the output of the amplifier is to be in the linear region. In the case of IC1/3 the current into the positive input is set at about 12 microamps by R3 and R4. Current into the negative input is provided from the output by R9. If the current into the negative input is too low the output voltage will rise thus increasing the current into the negative input until balance is achieved. This self balancing ensures correct static biasing.

Gain is obtained by feeding a signal into R5 which adds or subtracts current into the negative input. For

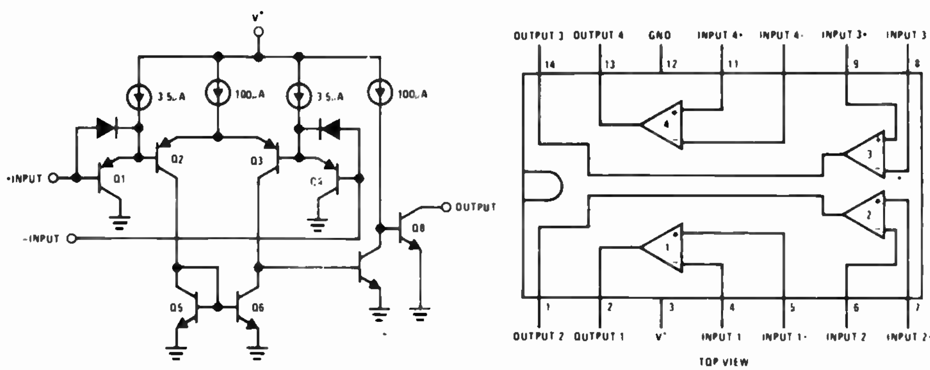


Fig.3. Internal circuitry and pin corrections of the LM339 IC.

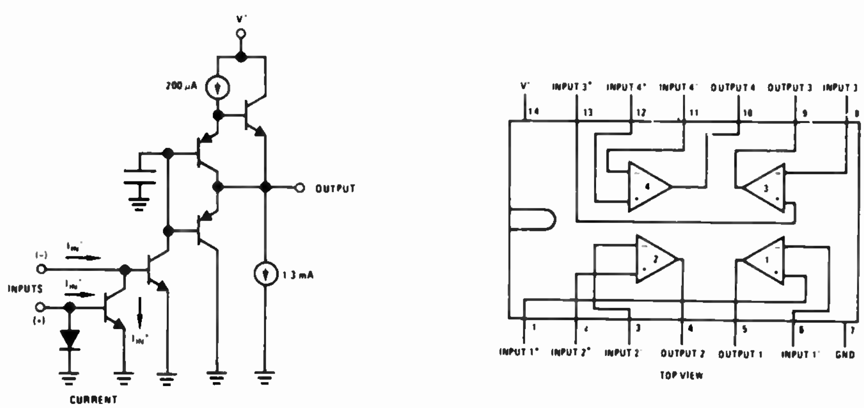


Fig.4. Internal circuitry and pin connections of the LM3900 IC.

AUDIO LEVEL METER

the amplifier to remain balanced there must be a corresponding shift in output voltage. The voltage gain is the ratio of R9 to R5.

SPECIFICATION LM3900

Maximum supply voltage	32 V
Supply current	6 mA typical
Voltage gain	2800 V/V typical
Input current range	1 μA – 1 mA
Current balance	0.9 – 1.1 at 200 μA
Bias current	30 nA typical
Output current capability	18 mA source typical. 1.3 mA sink typical

The LM339 is a quad voltage comparator where the output of each is an NPN transistor which has an unterminated collector and its emitter connected to ground.

SPECIFICATION LM339

Maximum supply voltage	36 V
Supply current	0.8 mA typical
Voltage gain	200 000 V/V typical
Offsett voltage	2 mV typical
Bias current	25 nA typical
Response time	1.3 μS typical
Output sink current	16 mA typical
Input common-mode voltage range	0 to (V^+ – 2 volts)

CONSTRUCTION

The meter will most likely be mounted in an existing amplifier or piece of equipment and for this reason the board construction only is given.

Layout of components is non-critical but, as with any multiple IC device, construction is greatly simplified by using the printed-circuit board specified. The usual precautions with polarities of components, such as capacitors, diodes, ICs and transistors should be observed. Some care must be taken when mounting the LEDs in order to obtain even spacing and good alignment. The long lead of the LED should be inserted in the hole furthest from the edge of the board. Put a slight curvature in the leads so that the LEDs can be aligned against the edge of the board (see photo). Take care not to bend the leads too often or too close to the body of the LED as the leads break very easily.

CALIBRATION

Resistor R5 is selected from Table 1 and this will ensure a result within 10 percent of that required. Greater accuracy may be obtained by using a variable potentiometer in series with R5. To adjust this potentiometer inject a signal (around 1 kHz) equal to 0 VU (VU meter) or maximum power ($E = \sqrt{RP}$, e.g. 4 ohms and 100 watts, E = 20 volts) and adjust such that the second top LED (VU meter) or the top LED (power meter) just lights. ●

TABLE 1A – VU METER

FSD = +3 dB

R3, 4 and 9 are 1 megohm

SENSITIVITY	VALUE OF R5*
50 mV	22 k
100 mV	47 k
250 mV	120 k
500 mV	220 k
1 V	470 k

**R5 = Sensitivity x 500 000 ohms.*

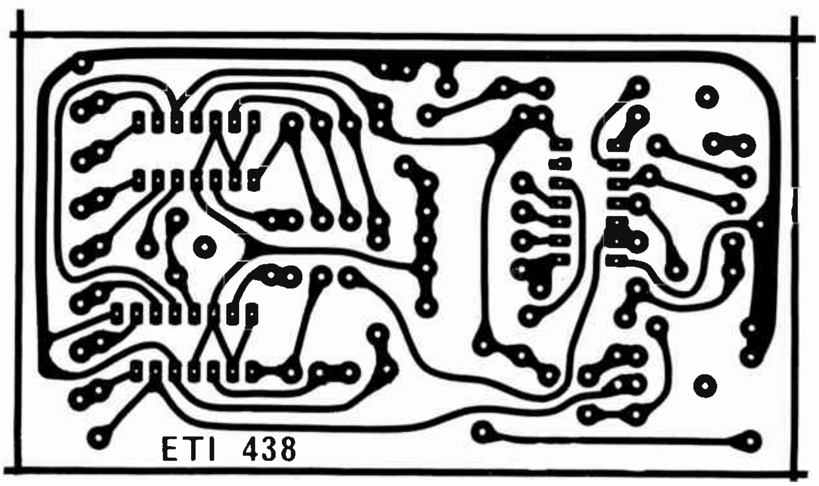
TABLE 1B – POWER METER

FSD = 0 dB

R3, 4 and 9 are 100 k

POWER OUTPUT IN WATTS	VALUE OF R5 4 Ohms	8 Ohms	16 Ohms
5	150 k	200 k	270 k
10	200 k	270 k	390 k
15	240 k	330 k	470 k
20	270 k	390 k	560 k
25	330 k	430 k	620 k
30	360 k	470 k	680 k
40	390 k	560 k	820 k
50	430 k	620 k	910 k
75	560 k	750 k	1.1 M
100	620 k	910 k	1.2 M
150	750 k	1.1 M	1.5 M
200	910 k	1.2 M	1.8 M
250	1 M	1.5 M	2 M

$R5\ k = 32\sqrt{PR}$ Where P = power in watts, R = speaker impedance in Ohms.



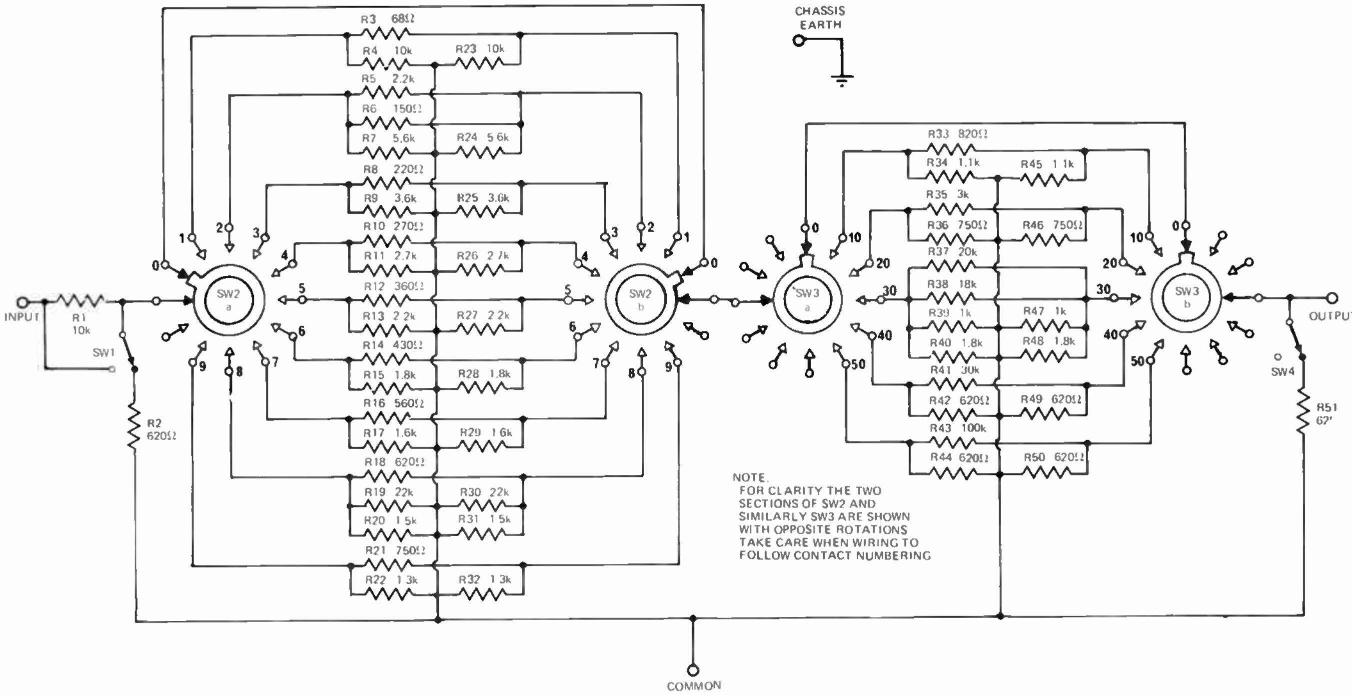
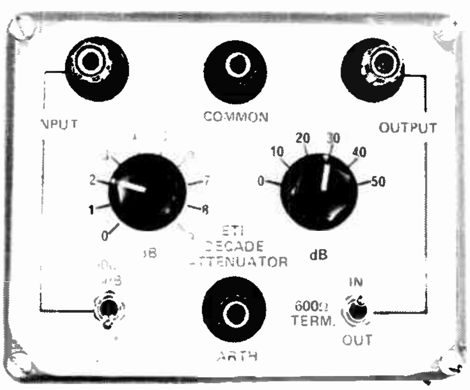


Fig. 1. Circuit diagram of the attenuator.

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

AUDIO ATTENUATOR

PROJECT 112



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.

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.

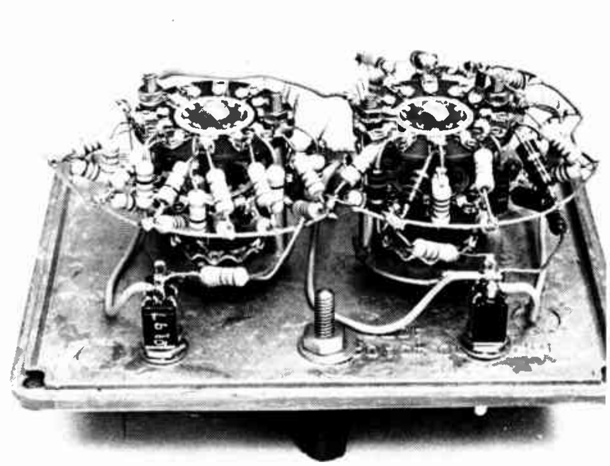
The input and output resistances of the unit remain relatively constant at 600 ohms over the full attenuation range. The input impedance can be changed to 10k by SW1 but an additional 30dB of attenuation is added. The output can also be terminated internally by SW4 when using a high impedance load such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



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 miniature toggle switch
SW2	2 pole 11 position rotary switch
SW3	2 pole 11 position rotary switch
SW4	Single pole change over miniature 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. ●

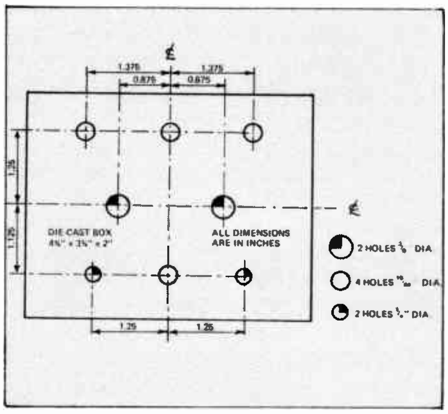


Fig. 2. Drilling details for the die cast box.

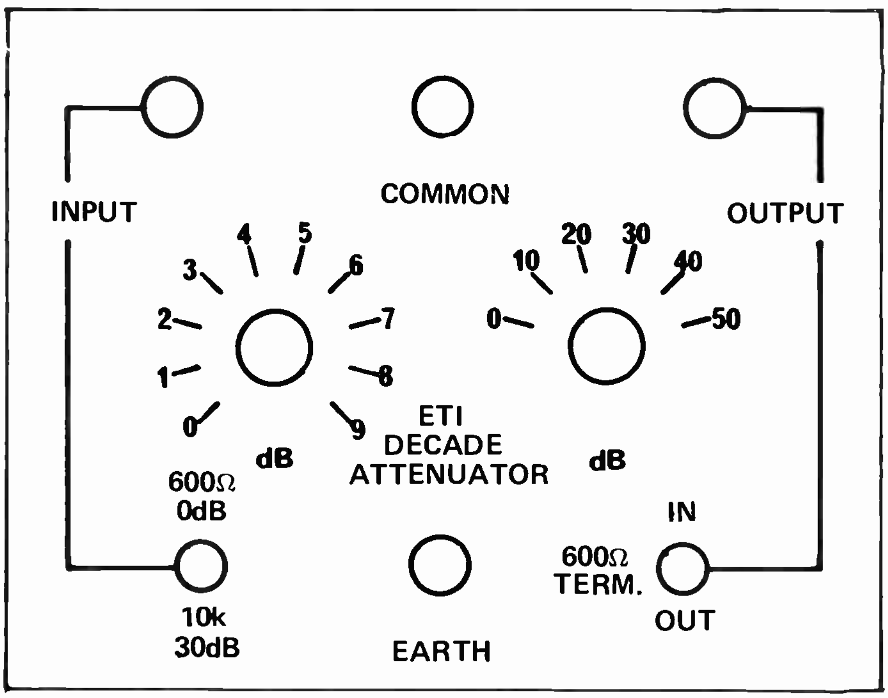


Fig. 3. Lettering and front panel artwork — full size.

TONE BURST testing is a technique which is rapidly gaining acceptance in a wide variety of applications. Typical applications are in testing of hydrophones, signal-to-noise in telephone channels, reverberation chamber testing and in the determination of peak distortion in loudspeakers. With loudspeakers, tone burst testing has the further advantage that the speakers may be tested with their maximum peak power level whilst keeping the average sound output level low enough to not annoy the neighbours — a considerable advantage indeed.

Some time ago our audio consultants, Louis Challis and Associates, asked us to build them a tone-burst generator and the resulting instrument has been used by them ever since with much success. The tone-burst test has been mentioned in several speaker reviews and, as a result, many people have asked for constructional details of this instrument.

PROJECT 124

TONE BURST GENERATOR

A valuable tool for testing loudspeakers.

DESIGN FEATURES

A tone burst must always be an integral number of cycles. If the burst is switched on or off part way through a cycle then undesirable transients will be produced that will mask the test results. Thus the burst must start and end exactly at the zero-crossing point of the sine wave in the burst.

In the original unit, designed for Louis Challis, preset times can be independently selected for the on and off periods of the burst with the exception that the burst time is automatically modified to give an integral number of cycles. The preselected on/off ratio, however, is independant of the burst frequency. To give the required control range, six switched ranges as well as a variable control are provided for both the on and off periods. Other features of the original unit are the ability to start at any point in the cycle as well as the zero crossing point, a phase-inverting switch to select either the positive or the negative half cycle first and an OFF LEVEL control to set a base tone level which is modified when the tone burst occurs. In addition the dc level of the output can be set and a switch is provided to select burst, pure tone or off as required.

When it came to redesigning the unit as a project we decided that many of the features offered by the original design were unnecessary for the user concerned only with testing speakers. Hence the unit has been redesigned in a greatly simplified form.

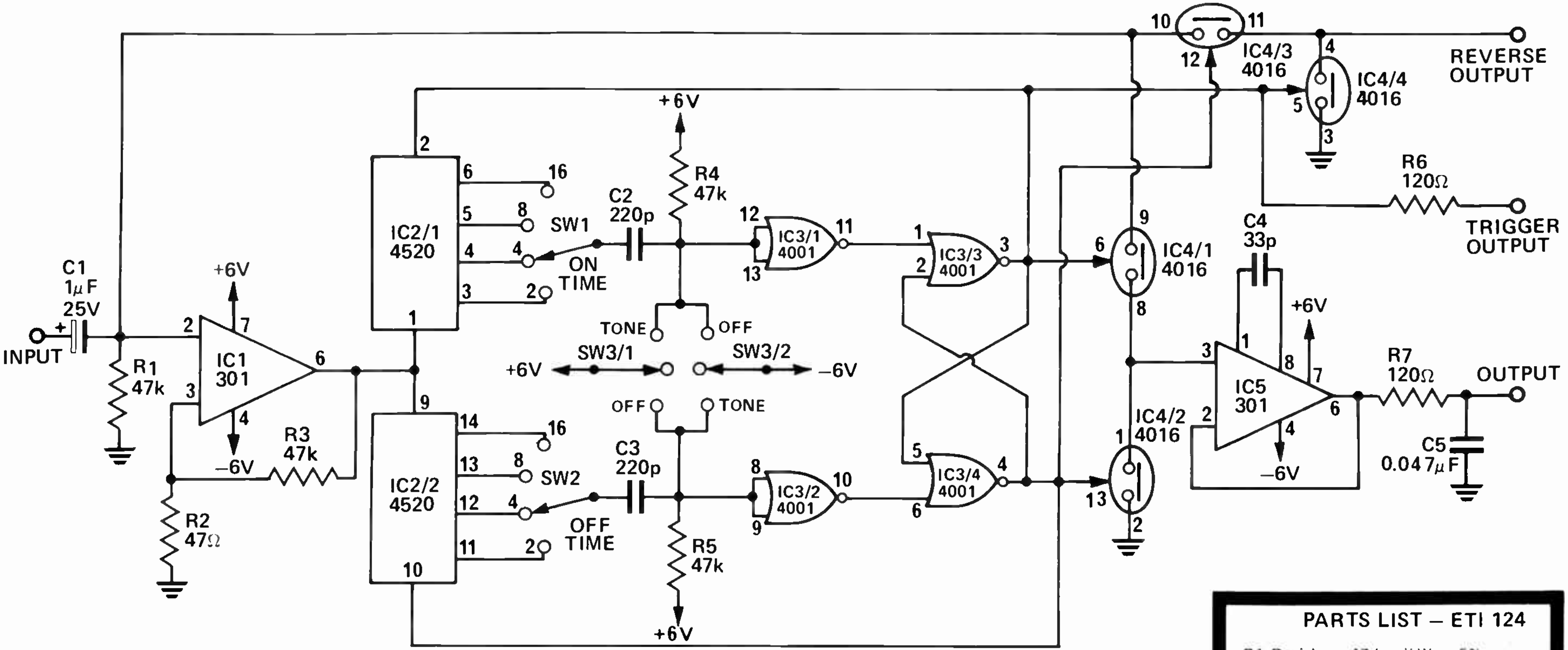
Instead of using monostables to generate variable on/off times we now divide the input with a counter to

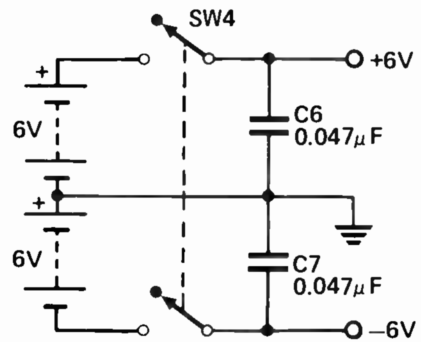
MEASURED PERFORMANCE

TONE BURST GENERATOR.

On Time Cycles.	2,4,8 or 16
Off Time Cycles	2,4,8 or 16
Frequency Response 3 Hz – 300 kHz	+0 –3 dB
Distortion 3 V input at 1 kHz	<0.02%
Input Level Maximum Nominal range	 3 V RMS 100 mV to 1 V
Input Impedance	47 k
Output Noise Voltage with no input	<25 μV
Power Supply Current	4 mA

TONE BURST GENERATOR





POWER RAILS OF IC2, IC3, AND IC4 NOT SHOWN
PIN 16 OF IC2 IS +6V
PIN 8 OF IC2 IS −6V
PIN 14 OF IC3 AND 4 IS +6V
PIN 7 OF IC3 AND 4 IS −6V
PIN 7 AND 15 OF IC2 ARE RESET PINS AND −6V

Fig. 1. Circuit diagram.

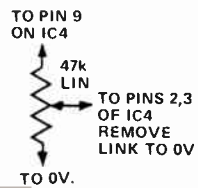


Fig. 4. How to add a potentiometer to the generator for burst-on-tone operation. That is the generator gives a continuous tone level with tone bursts of higher amplitude at intervals.

PARTS LIST — ETI 124

R1	Resistor	47 k	¼W	5%
R2	"	47 Ω	¼W	5%
R3	"	47 k	¼W	5%
R4	"	47 k	¼W	5%
R5	"	47 k	¼W	5%
R6	"	120	¼W	5%
R7	"	120	¼W	5%
C1	Capacitor	1 µF 25V electro		
C2	"	220 pF ceramic		
C3	"	220 pF ceramic		
C4	"	33 pF ceramic		
C5	"	0.047 µF polyester		
C6	"	0.047 µF polyester		
C7	"	0.047 µF polyester		
IC1	Integrated Circuit	LM 301A		
IC2	" "	4520 (CMOS)		
IC3	" "	4001 (CMOS)		
IC4	" "	4016 (CMOS)		
IC5	" "	LM301A		

SW1 Switch 1 pole 4 position rotary
SW2 Switch 1 pole 4 position rotary
SW3 Switch DPDT Toggle with centre off
SW4 Switch DPDT Toggle

PC Board ETI 124
8 AA size batteries
2 4-way battery holders and clips
Plastic case
Escutcheon
3 single RCA sockets
2 knobs

HOW IT WORKS – ETI 124

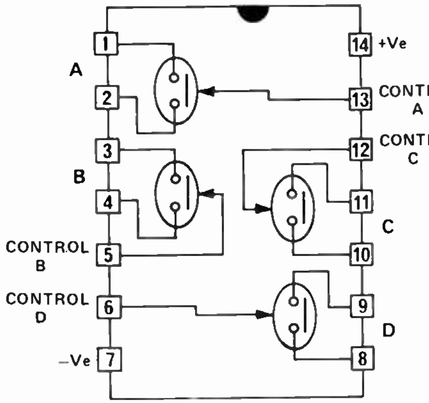
The input signal is squared by comparator IC1 such that the output of the comparator will be high if the input is above +6 mV, and low if the input signal is below −6 mV. Resistors R2 and R3 provide the necessary positive feedback to cause the IC to act as a comparator. The output of the comparator is connected to both clock lines of IC2. If the enable line is high these counters (IC2) will toggle at the input frequency.

IC3/3 and IC3/4 form an RS flip flop where the output must be in either a high or a low state, that is the flip flop has only two stable states. If the output of IC3/3 is high IC2/1 is allowed to clock and, after the number of input pulses selected by SW1 have been counted, the output from SW1 goes low. This low is coupled to the flip flop by C2 toggling the flip flop, disabling IC2/1 and enabling IC3/2. After the number of cycles, as selected by SW2, have been counted the flip flop is again toggled. IC3/1 and IC3/2 are used to square up the pulses generated by C2 and C3 respectively.

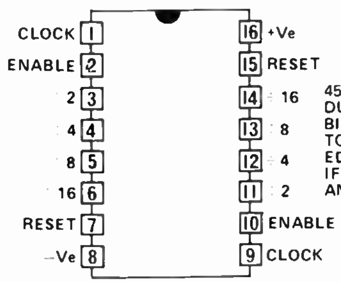
The input signal is also coupled to the output buffer, IC5, by the analogue switch IC4/1. When this switch is closed (control signal high) the output of the buffer will be the same as the input. When switch IC4/1 is open IC4/2 will be closed and the output will be held at zero. Since these switches are controlled by the flip flop the output will be the required tone burst.

A trigger output is taken from the flip flop to synchronize an oscilloscope if required. A second output is also available from pins 4/11 of IC4 which is the reverse of the main output.

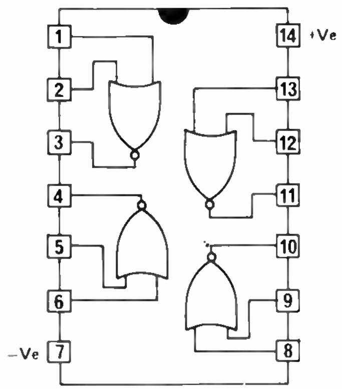
Switch SW3 forces the flip flop into either of its two possible states thus allowing continuous tone or no output to be selected as required. In the centre position the normal tone burst is obtained.



4016
4066
QUAD ANALOGUE SWITCH. ON IF CONTROL IS HIGH
ON RESISTANCE
4016 TYP 280Ω
4066 TYP 80Ω



4520
DUAL 4 BIT
BINARY UP COUNTER
TOGGLE ON POSITIVE
EDGE OF CLOCK
IF ENABLE IS HIGH
AND RESET IS LOW



4001
QUAD 2 INPUT
NOR GATE

INPUTS		OUTPUT
A	B	
0	0	1
0	1	0
1	0	0
1	1	0

Fig. 3. Pin connections of the ICs used in the generator.

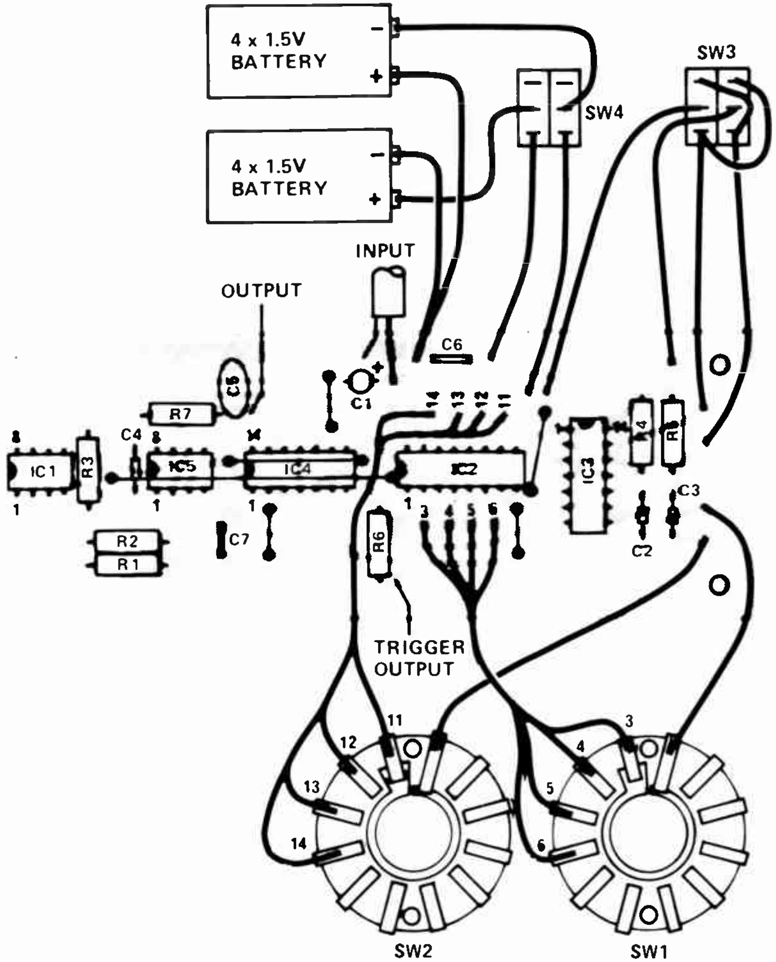


Fig. 2. Component overlay and interconnection diagram. Note that there are six links on the board, including two under IC4, which should be installed first.

TONE BURST GENERATOR

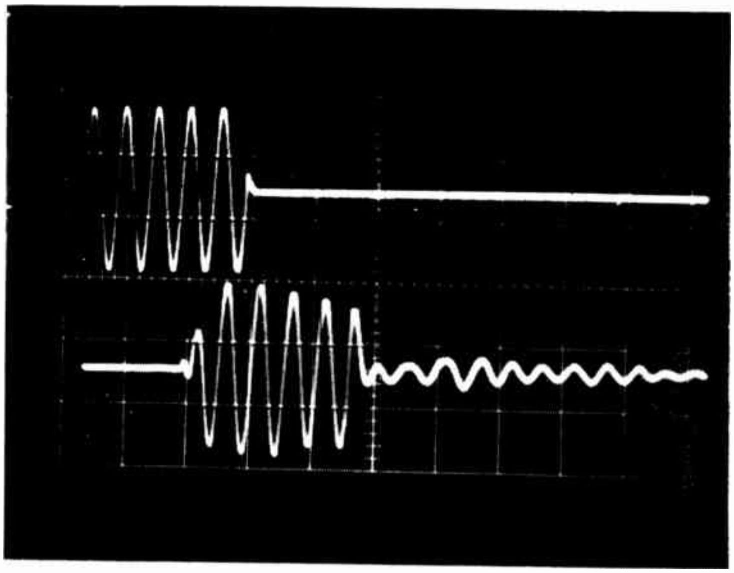


Fig.5 (a) Top trace — the input tone burst of five cycles. (original design).
(b) Bottom trace — the response of a low-cost speaker at 1 kHz. Note the reduced amplitude of the first half cycle and that ringing has added another cycle at the end of the burst. The room reflection can be seen on the trace after the burst.

obtain times that remain in the same ratio regardless of input frequency. We settled for the ability to select 2, 4, 8 and 16 cycles for the duration of either period, as this compromise greatly simplifies the circuitry. We still have the switch to select tone, tone burst or off, but the OFF LEVEL control has been deleted. The latter control may quite easily be added, however, as shown in Fig. 4. . The output dc level control and the starting-point phase change have also been deleted.

Since we only need half of a CMOS 4016 IC, to give the required output, the other half may be used to give an inverse output if required, that is, the reverse output is on when the other is off and vice versa. This output is not buffered or brought out to the front panel. If it is intended to load this output with less than 47 k it is recommended that a 4066 IC be used instead which will handle loads down to 10 k. For loads of lower impedance than this, a buffer such as is on the normal output should be used.

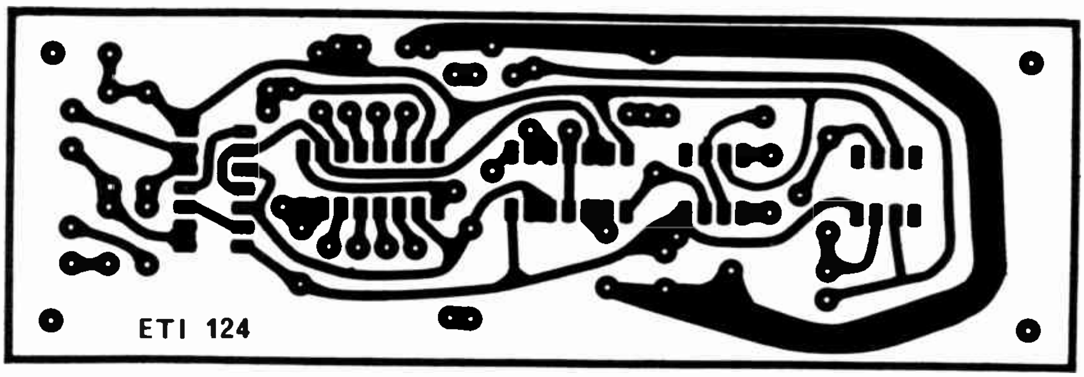


Fig.6. Printed circuit board for the Tone Burst Generator Full size. 142 x 47mm.

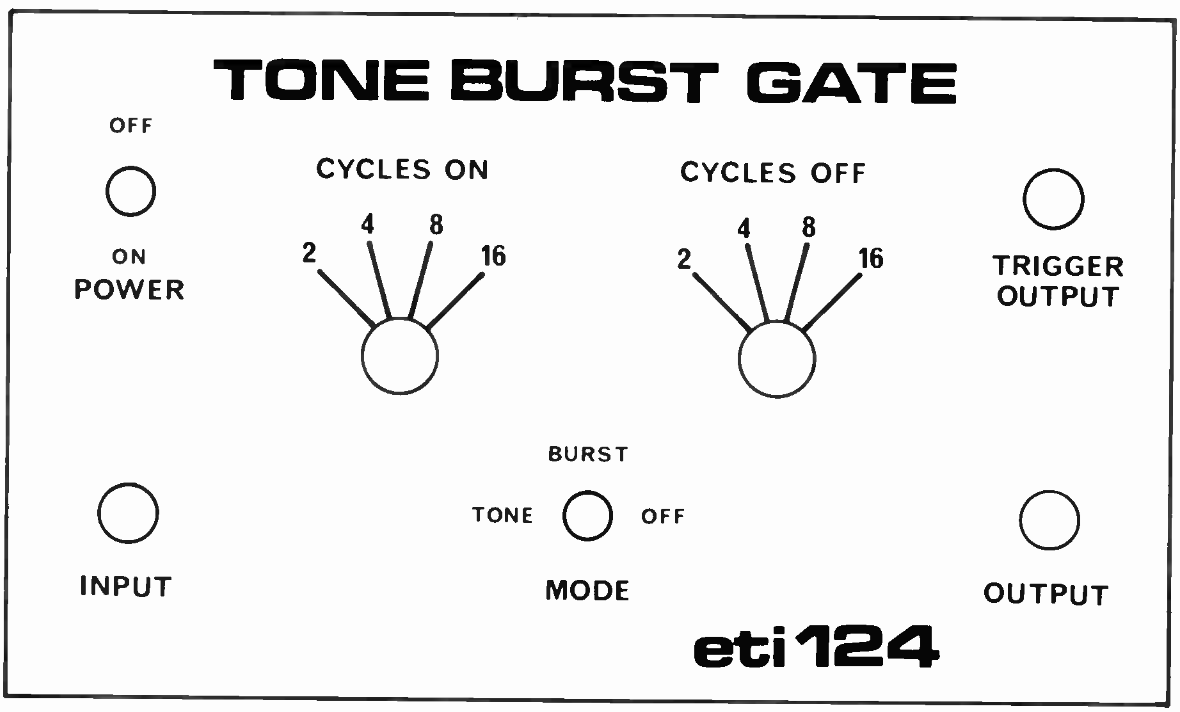


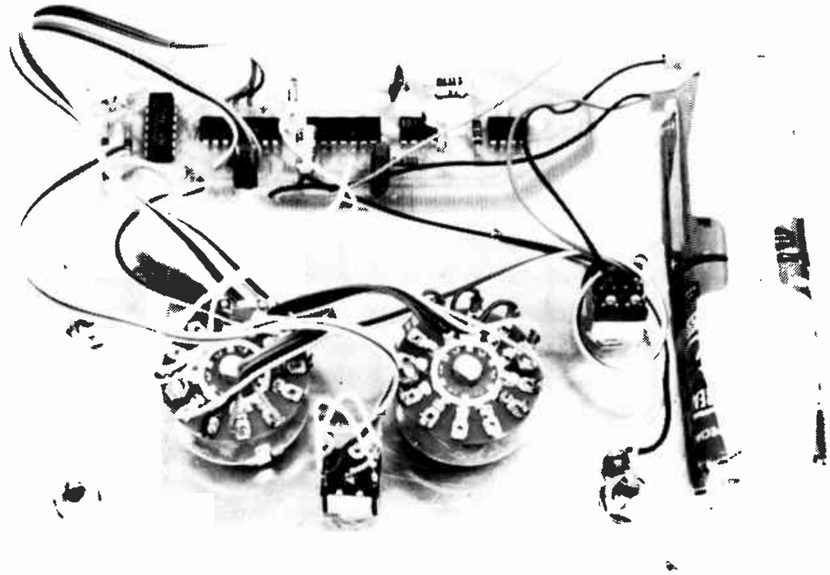
Fig.7. Front panel artwork.

CONSTRUCTION

As with any project construction is greatly simplified if a printed circuit board is used. However the layout of the unit is not critical and any other suitable method, such as Veroboard or Matrix board may be used if desired. We strongly recommend that sockets be used for the CMOS ICs, especially if a printed circuit board is not used, as these devices are quite easily damaged when soldering. The use of IC sockets also facilitates later servicing. Also remember that, unlike TTL, all unused inputs of CMOS must be connected to either the positive or negative supply rail.

The plastic box that we used measured 160 x 95 x 50 mm and is very convenient in that the printed circuit may be held in position by sliding it down behind two of the pillars to which the front panel is screwed. The front-panel overlay on the prototype was made from Scotchcal but, as the amount of lettering required is quite small this may readily be done directly on the panel by hand or with Letraset.

Shielding of the internal wiring is not required providing that the unit is kept away from strong 50 Hz fields. If operation in the vicinity of strong fields cannot be avoided then the unit should be mounted in a diecast box.



How the unit is assembled.

USING THE UNIT

The testing of loudspeakers is very difficult indeed and much effort is still being spent to find test methods which will not only give an accurate understanding of the relative effectiveness of the design, but which will be easy to reproduce.

One of the main problems with speaker testing is that the speaker cannot easily be isolated from its environment. For example, reflections from the walls of a room modify the response, seen by a microphone, no matter where the microphone is placed in the room. If one could eliminate reflections then the situation would be improved considerably, and hence the use of anechoic (echo free) chambers for testing speakers. But such chambers are very expensive to build and consequently not readily accessible to the amateur.

A further problem is in assessing the transient power handling capability of the speaker. Speakers will handle far greater peak transient power than is indicated by their RMS power rating. This is a very important attribute of loudspeakers in handling musical transients. Any attempt to assess this with a sinewave signal may result in the destruction of the speaker due to thermal failure – apart from also being extremely noisy.

The use of a tone-burst generator minimizes both these problems. How this is achieved is better understood by examination of Fig.5. This shows on the upper trace a five cycle 1000 Hz burst that is fed to a loudspeaker. The second trace shows the same burst as picked up by a microphone in front of the speaker. We notice that the burst has been changed by the speaker and an examination of these changes can tell us a lot about the speaker. For example we notice that the first half cycle has not reached full amplitude and this indicates that the speaker would have some difficulty in reproducing high frequency transients. Next we notice that instead of five cycles there are now at least five and a half. This could mean one of two things. Either there is a speaker/room resonance or, the speaker itself is continuing to vibrate after the original excitation has ceased. Which is it? We can determine this by changing the position of the speaker to see if any change occurs in the shape of the burst, if not it is caused by the speaker itself, and if it does then it is a speaker/room resonance. A speaker that lengthens the burst unduly will sound muddy in that region. Of course the speaker must be examined over its whole range to gain a thorough assessment of performance.

It is of course possible to eliminate room reflections simply by performing the tests outside. However unless one lives in a very quiet area, background noise will introduce problems – and your neighbours are unlikely to appreciate the noise that you will generate.

By varying the off period we can also select a ratio where the room reflection, the oscillation seen after the cessation of the burst, does not interfere with the first few cycles of the burst and the response versus frequency of the speaker may then be assessed from the amplitude of the first half cycles that are stable in amplitude. Thus it is possible to gain an appreciation of the frequency response, transient performance and quality in terms of ringing of a speaker by careful use of the tone-burst technique.

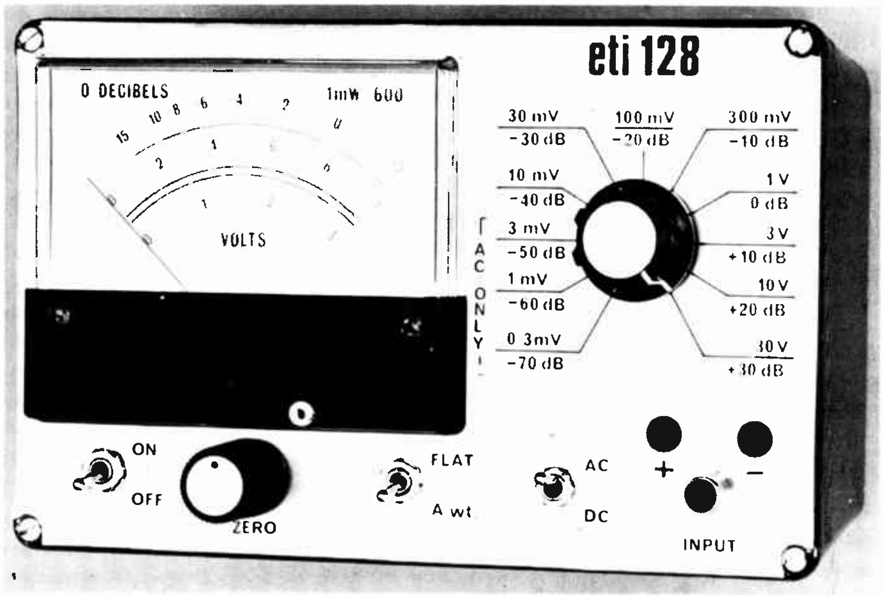
The transient power handling capability of a speaker may be assessed by selecting a fairly long off to on ratio for the burst and by feeding the burst to the speaker via a high-power amplifier. If for example an off to on ratio of 8:1 is used then the peak power will be eight times the average power. Thus the speaker may safely be driven to a peak level where a predetermined amount of distortion occurs. Take care that the amplifier is capable of providing the peak power required.

Of course a tone-burst generator may be used for a wide range of testing. We have mainly concentrated in this article on its application to the testing of loudspeakers.

The circuitry of the tone-burst generator may easily be modified for use as a 'silent switch' for A/B speaker testing. The method of doing this is shown on page 32.

PROJECT 128 AUDIO MILLIVOLTMETER

Sensitive instrument for 'A' weighted audio noise and signal measurements.



AN ACCURATE and sensitive ac voltmeter is needed for many audio equipment measurements.

Whilst for example, maximum power output is readily measurable with a conventional multimeter, more complex instrumentation is required for measuring noise output (a measurement required when checking signal/noise ratio).

Even signal levels as high as 100 mV, typical output of most pre-amplifiers, are not readily measured with accuracy on a conventional multimeter.

The ETI 128 Millivoltmeter is specifically designed for such measurements whilst also being useful as a general purpose ac/dc voltmeter. The lowest range, of 300 microvolts FSD, allows measurements to 80 dB below one volt, whilst other ranges allow measurements up to 30 volts ac or dc. These ranges cover most of the measurement requirements of audio work.

When measuring noise levels account must be taken of the non-linear characteristics of the ear. For this reason a network has been incorporated which tailors the meter response-versus-frequency to match the subjective response of the ear. Such a network is known as an 'A weighting network' and its use provides a measurement which is realistically related to what is heard. When measurements are made using this network the results must be quoted as being 'A weighted'. Typically this is done by quoting dBA rather than just plain dB.

CONSTRUCTION

The meter is a highly sensitive instrument and for this reason the constructional method given should be followed closely if noise and hum pickup are to be minimized.

A diecast box is used to house the meter as this provides excellent shielding against external signals. The front panel label is made from 'Scotchcal'. This is a specially prepared sheet of thin aluminium which is coated with a photo-sensitive emulsion on one side. The reverse side has a self-adhesive coating, protected by waxed paper, which is peeled off when the material is to be stuck down. As Scotchcal is only available in bulk, ETI is making available ready-to-use front panel labels made from this material. Should you require one of these labels send $4.00 and a stamped, self-addressed envelope (minimum size envelope 190 x 127 mm).

The meter used in the prototype was from Dick Smith Electronics. It measures 100 x 82 mm but requires to be rescaled. The scale as published on page 162 should be cut out and glued over the existing scale taking care not to let glue or dirt enter the meter movement. Any similar meter may be

SPECIFICATION

RANGES	
dc (FSD)	10, 30, 100, 300 mV, 1, 3, 30 V. auto-polarity, LED indication.
ac (FSD)	0.3, 1, 3, 10, 30, 100, 300 mV, 1, 3, 10, 30 V 0 dB = 1 mW into 600 ohms (0.775 V) weighting curves, ac only, flat, 'A' weight
ACCURACY	± 3% nominal
MINIMUM READING	
Open circuit	–76 dB
Terminated 47 k	–85 dB
POWER SUPPLY	
Voltage	+6 and –6 volt (batteries)
Current	approximately 12.5 mA
Battery life	approx 100 hours (8 x 1015 cells)

AUDIO MILLIVOLTMETER

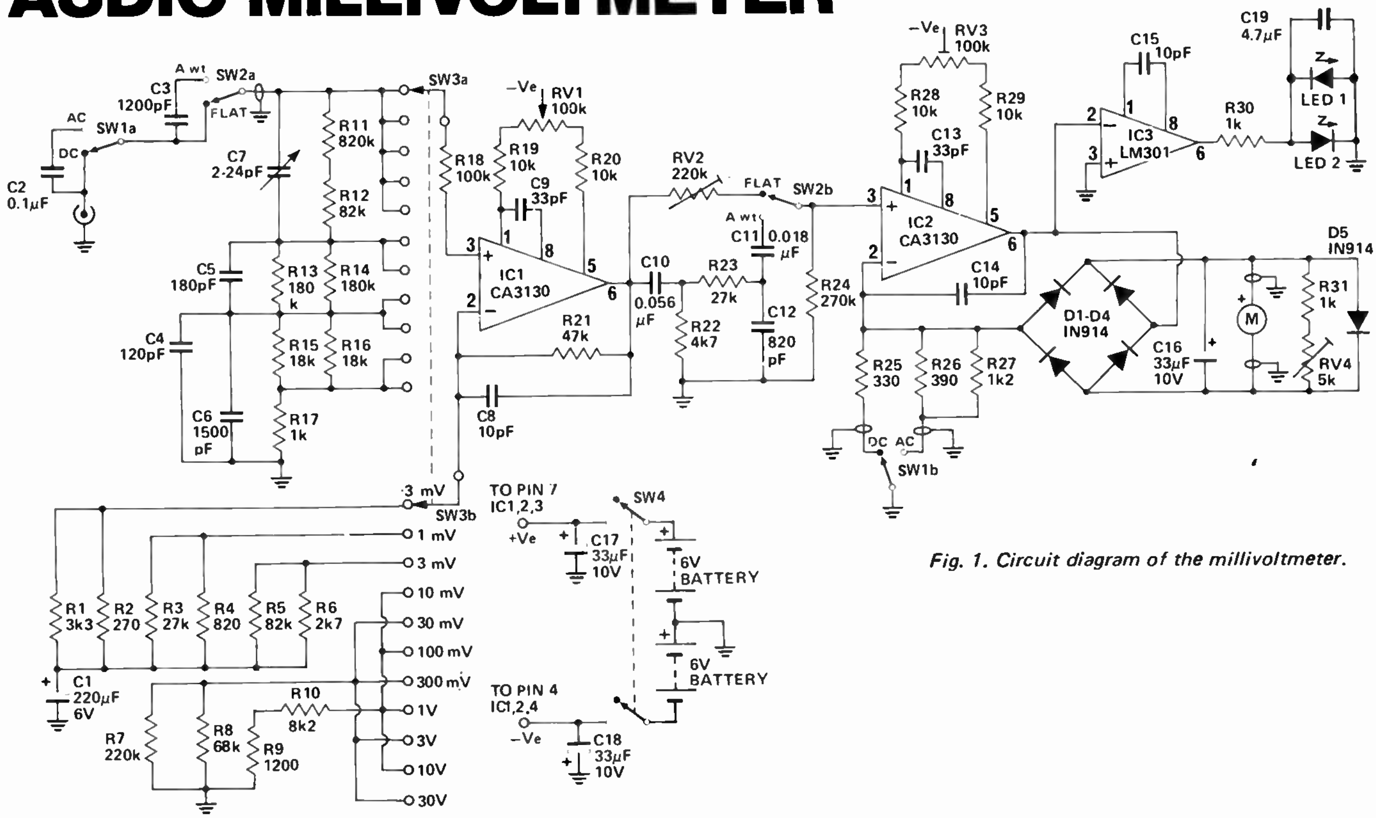


Fig. 1. Circuit diagram of the millivoltmeter.

HOW IT WORKS – ETI 128

The millivoltmeter may be separated into several sections in order to simplify the explanation of its mode of operation. These are:–

(a) Input attenuator.
(b) Input amplifier.
(c) 'A'-weight network.
(d) Meter drive circuitry.
(e) Polarity detector.

The input attenuator consists of resistors R11 to 17 and capacitors C4 to 7, and gives division ratios of 1, 10, 100 and 1000. The capacitors are required to ensure that the division remains accurate at high frequencies.

The input amplifier is a CA3130 operational amplifier where the gain is selected by SW3b. Gains of 190, 60, 19, 6 and 1.9 are available which together with the input divider ratios provide the 11 ranges required. The high gain ranges of 190, 60 and 19 are ac coupled, as the temperature stability of the CA3130 will not allow voltages of less than 10 mV dc to be used. The output of this amplifier is 60 mV when the meter is indicating full scale on any range. A potentiometer, RV1, is provided to adjust the offset voltage on the CA3130 and thus acts as a zero-set control. Since the offset voltage is affected by temperature this control is available externally.

When measuring noise in audio systems a weighting network is often used to give a measurement which is related to the non-linear response of the ear. The most commonly used weighting is known as 'A' weight and this facility is built into the meter. The 'A' weight curve is produced by a network that has a three-pole, high-pass filter and a single-pole, low-pass filter. The main section of this filter is formed by C10, C11, C12 and R22, 23, and R24 (two poles). The third pole is due to C3 and the one megohm combined resistance of R11 to R17. This later section prevents saturation of the input amplifier at low frequencies. Since this filter introduces some loss at 1 kHz, RV2 is incorporated to provide the same loss in the 'flat' mode.

The second IC acts as a meter amplifier. The input signal is rectified by the diode bridge D1 to D4 whilst the amplifier effectively compensates for the diode drops. A preset for offset adjustment, RV3, is provided for this IC. Calibration is performed by adjustment of the shunting resistance, R31 and RV4, across the meter. Due to the full-wave action of the rectifier the meter when on the dc ranges reads uni-directionally regardless of dc polarity. The output of IC2 will however will either be at over one volt positive or one volt negative (voltage drops across the diodes) depending on whether the input voltage is positive or negative. This is compared by IC3 against zero volts and, depending on polarity, either LED 1 or LED 2 will be illuminated. With an ac input both LEDs will be on. These LEDs are therefore the polarity indicators. Capacitor C19 removes any high frequency components which could be coupled into the input, as the LEDs are located next to the input socket.

Due to the difference between the average and the RMS values of a sine-wave a slight change in gain is necessary in the ac mode and, this change is made by SW1b.

used as long as it has 100 microamp sensitivity.

The ac/dc and Flat/'A' weight switches are four-pole types although only the outer two poles are used. The centre two poles are earthed in order to reduce the capacitance between the two outer poles. Such precautions are necessary to prevent any possibility of instability on the most sensitive ranges. The metal bracket which supports the printed-circuit board also acts as a shield between the meter circuitry and the input stages.

Commence construction by assembling components to the printed-circuit board, making absolutely sure that all are mounted in the correct position and with the correct polarity. This should be carefully done — once the meter is

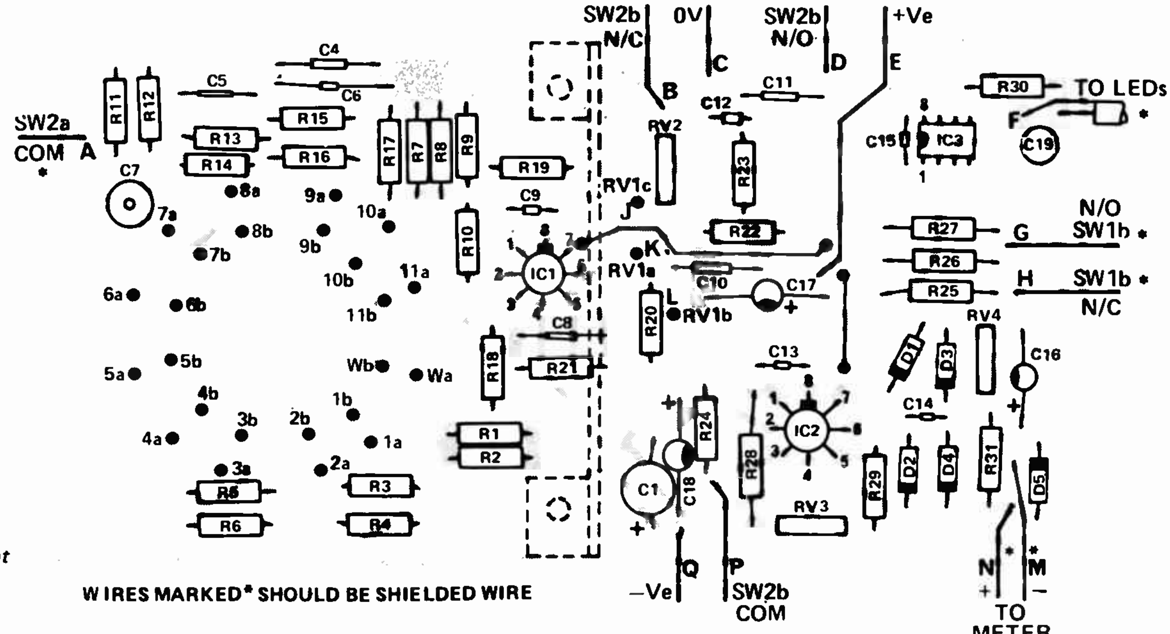


Fig. 2. Component overlay.

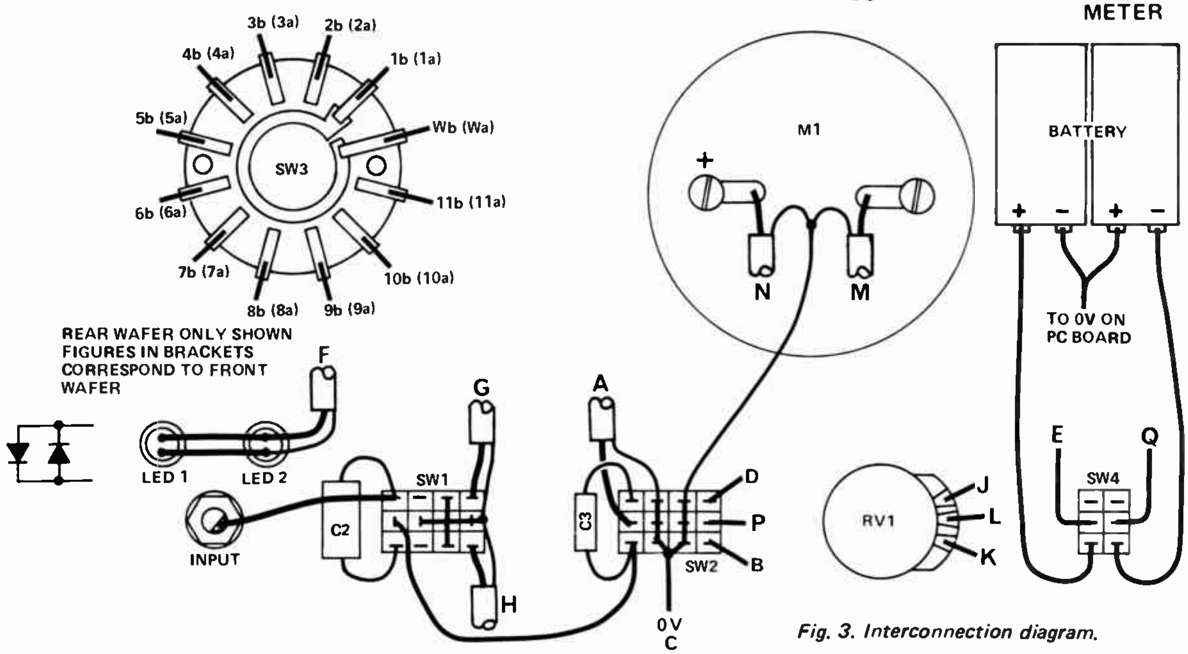


Fig. 3. Interconnection diagram.

fully assembled, it is very difficult to change components.

Assemble the front panel, fitting all switches with the exception of SW3, LEDs, potentiometer, input socket, meter, and the shield. The shield passes between the centre two contacts of the 'A'-weighted switch.

Solder a tinned copper lead to each of the 12 contacts on the rear wafer of switch SW3 (about 25 mm long). Feed these wires through the holes provided in the printed-circuit board (1b to 11b and Wb) making sure that the wiper contact on the switch goes to Wb and that the other wires are inserted in sequence. Do not solder as yet.

Assemble the printed-circuit board onto the shield and the rotary switch to the front panel. We used a 3 mm stack of washers to space the switch back from the front panel so the

PARTS LIST — ETI 128

R2	Resistor	270ohm	2%	¼W
R25	"	330ohm	2%	¼W
R26	"	390ohm	2%	¼W
R4	"	820ohm	2%	¼W
R17	"	1k	2%	¼W
R6	"	2k7	2%	¼W
R10	"	8k2	2%	¼W
R15,16	"	18k	2%	¼W
R21	"	47k	2%	¼W
R8	"	68k	2%	¼W
R13,R14	"	180k	2%	¼W
R11	"	820k	2%	¼W
R30,31	Resistor	1k	5%	¼W
R9,27	"	1k2	5%	¼W
R1	"	3k3	5%	¼W
R22	"	4k7	5%	¼W
R19,20	"	10k	5%	¼W
R25,29	"	10k	5%	¼W
R3,23	"	27k	5%	¼W
R5,12	"	82k	5%	¼W
R18	"	100k	5%	¼W
R7	"	220k	5%	¼W
R24	"	270k	5%	¼W
RV1	Potentio-meter	100k lin rotary		
RV2	"	220k Trim		
RV3	"	100k Trim		
RV4	"	5k Trim		
C7	Capacitor	2-24 pF	Philips 2222 808 00006	
C8,14,15	"	10 pF	Ceramic	
C9,13.	"	33 pF	Ceramic	
C4	"	120 pF	Ceramic	
C5	"	180 pF	Ceramic	
C12	"	820 pF	Ceramic	
C3	"	1200 pF	polyester	
C6	"	1500 pF	polyester	
C11	"	0.018µF	polyester	
C10	"	0.056µF	polyester	
C2	"	0.1µF	polyester	
C19	"	4.7µF	non polarised electro	
C16,17,18	"	33µF	10V electro	
C1	"	220µF	6V electro	
IC1,2	Integrated Circuit		CA3130	
IC3	"	"	LM301	
D1-D5	Diode	IN914, BA318 or similar		
LED 1,2	5023 or similar with panel mounting			
SW1,2	Toggle switch 4 pole 2 positions			
SW3	Rotary switch 2 pole 11 positions			
SW4	Toggle switch 2 pole 2 positions			
M1	Meter	100µA FSD * see text		
PC Board ETI 128				
Die cast Box 6357p				
Two knobs*				
One RCA socket				
Eight AA size batteries				
Two- 4xAA size battery holders				
Shield to Fig. 7				

AUDIO MILLIVOLTMETER

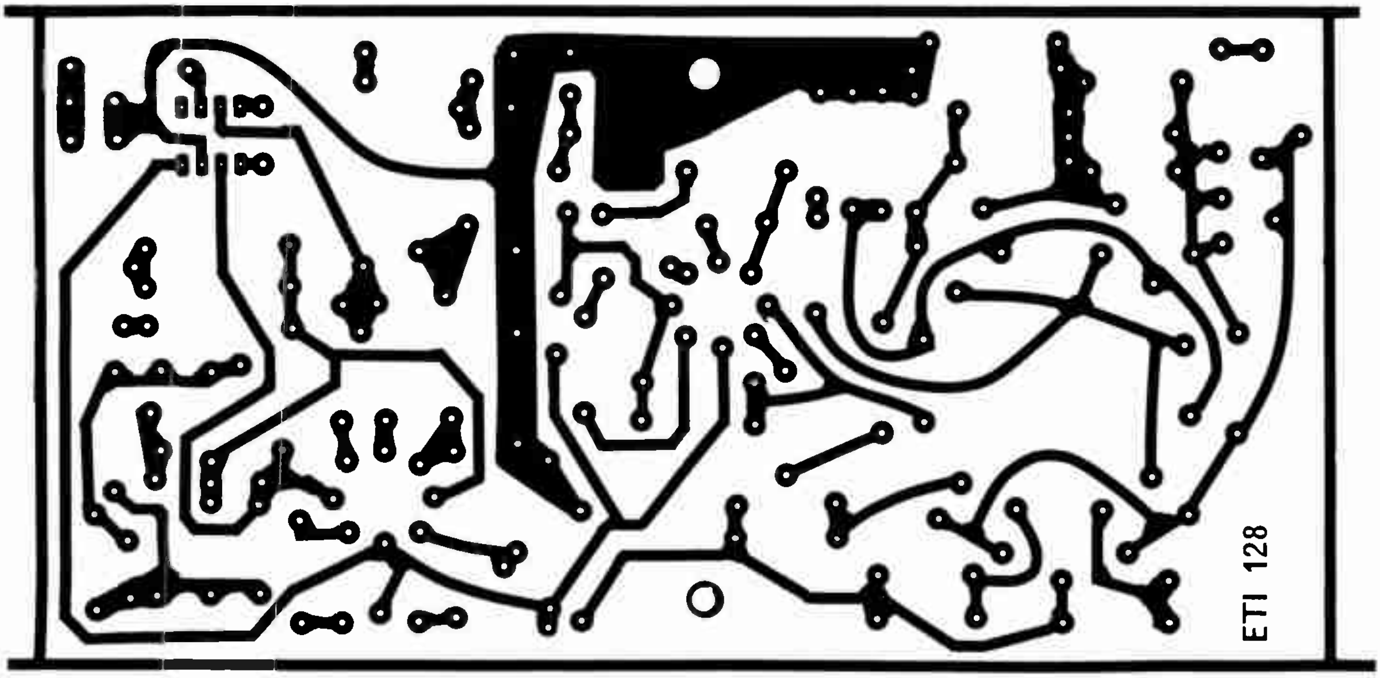


Fig. 5. Printed circuit layout. Full size 170 x 87 mm.

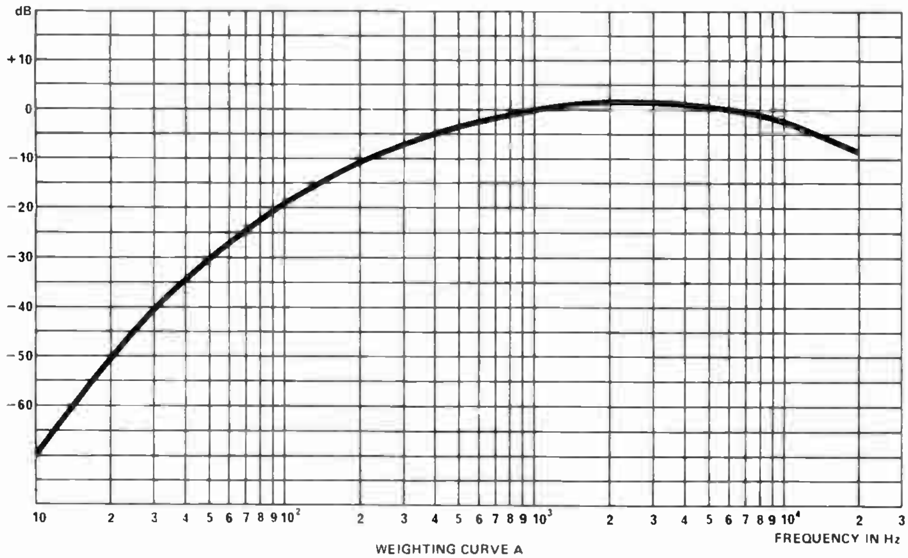


Fig. 4. Curve of 'A' weight response.

control knob would sit down closer to the front panel. Remove any slack in the tinned-copper wires, connecting the switch to the printed-circuit board and ,then solder them to the board. Now remove the printed-circuit board and switch assembly from the front panel. The switch will now be rigidly held onto the board, and the front wafer can now be wired to the board via further tinned-copper links. Make sure that none of these wires is touching.

Add leads to the printed-circuit in the locations shown on the overlay and reassemble the board and switch assembly to the front panel. The components on the front may now be connected to the board by these leads which should be kept as short as possible without placing undue strain on the wires. The only exception to this rule is the wire from SW1a to SW2a which should be kept reasonably well clear of the second pole of SW10. This is best done by running the lead down the front panel along the bottom and then back up to SW2a. Shielded wire should be used where designated on the overlay and wiring diagrams, and this should preferably be of the low capacitance variety.

The LEDs are connected in parallel but in anti-phase, the actual polarities may be determined later if necessary during the calibration procedure.

CALIBRATION

Before commencing calibration, check that the meter performs as it should on all ranges by applying known voltages and checking that a deflection of roughly corresponding magnitude is obtained. Also check that the 'A'-weighted switch appears to work as it should.

1. Short the input, select the 3 mV range and switch on.

2. Allow about 5 minutes for the instrument to stabilize thermally and then adjust RV3 to zero the meter.

3. Select the 10 mV range, dc, and 'flat', and adjust the front panel control RV1 to zero the meter.

4. Remove the short from the input, select the 300 mV range and apply an input having a frequency of less than 500 Hz and a level which gives a convenient indication, eg 0 dB. Change the frequency to somewhere between 10 kHz and 50 kHz making sure that the input level is the same in both cases, and adjust capacitor C7 so that the meter reads the same in both cases.

5. Apply an ac input signal and switch between ac and dc. The reading

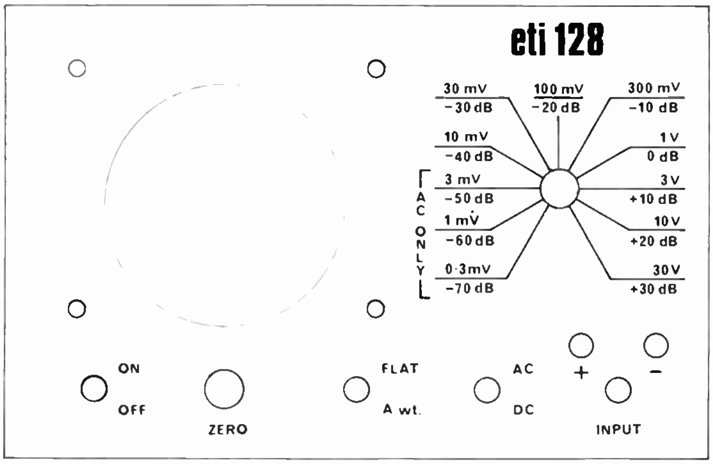


Fig. 6. Front panel artwork. Full size 189 x 121 mm.

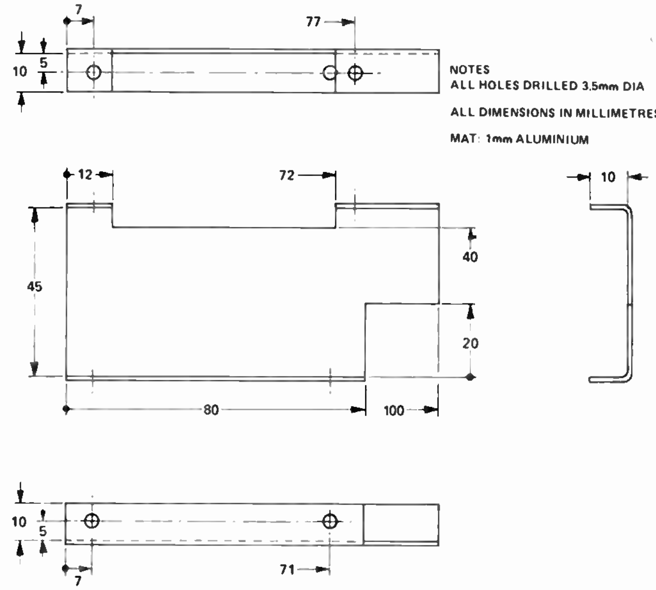


Fig. 7. Details of shield-support bracket.

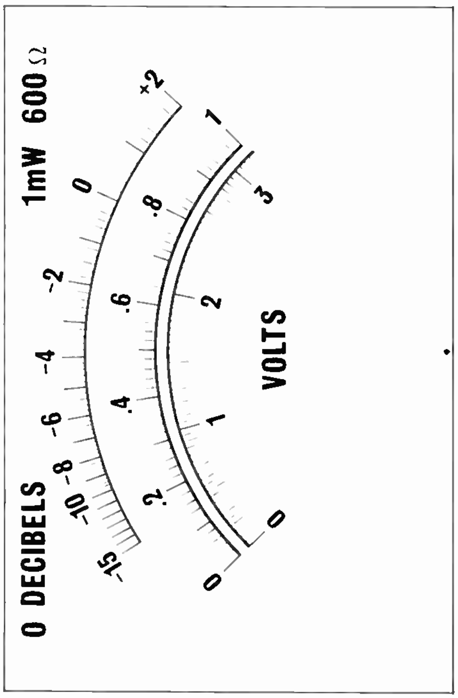
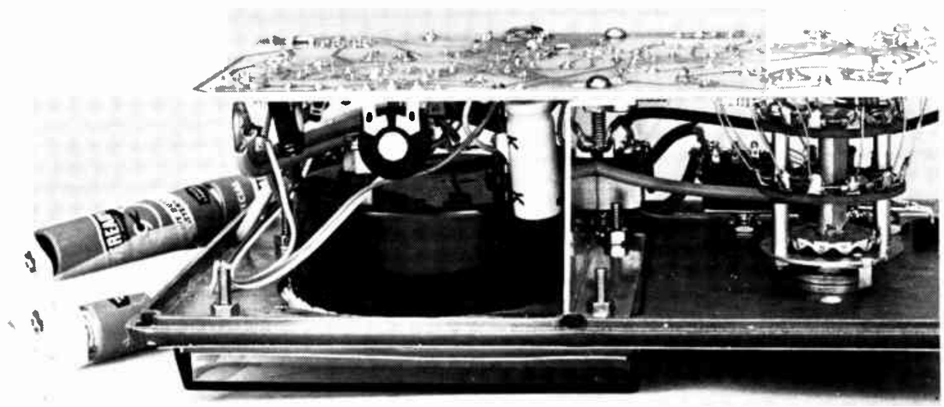
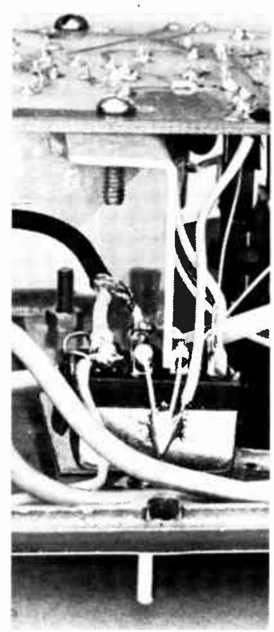


Fig. 8. Artwork for meter (shown full size).



This internal view of the meter shows on the right, how the range switch is wired to the printed-circuit board. Note also the shield.



Note how the shield passes between the earthed, centre contacts of the 'A' weight switch.

on ac should be about 10% higher than on dc. If it is 10% lower the leads to switch SW1b should be reversed.

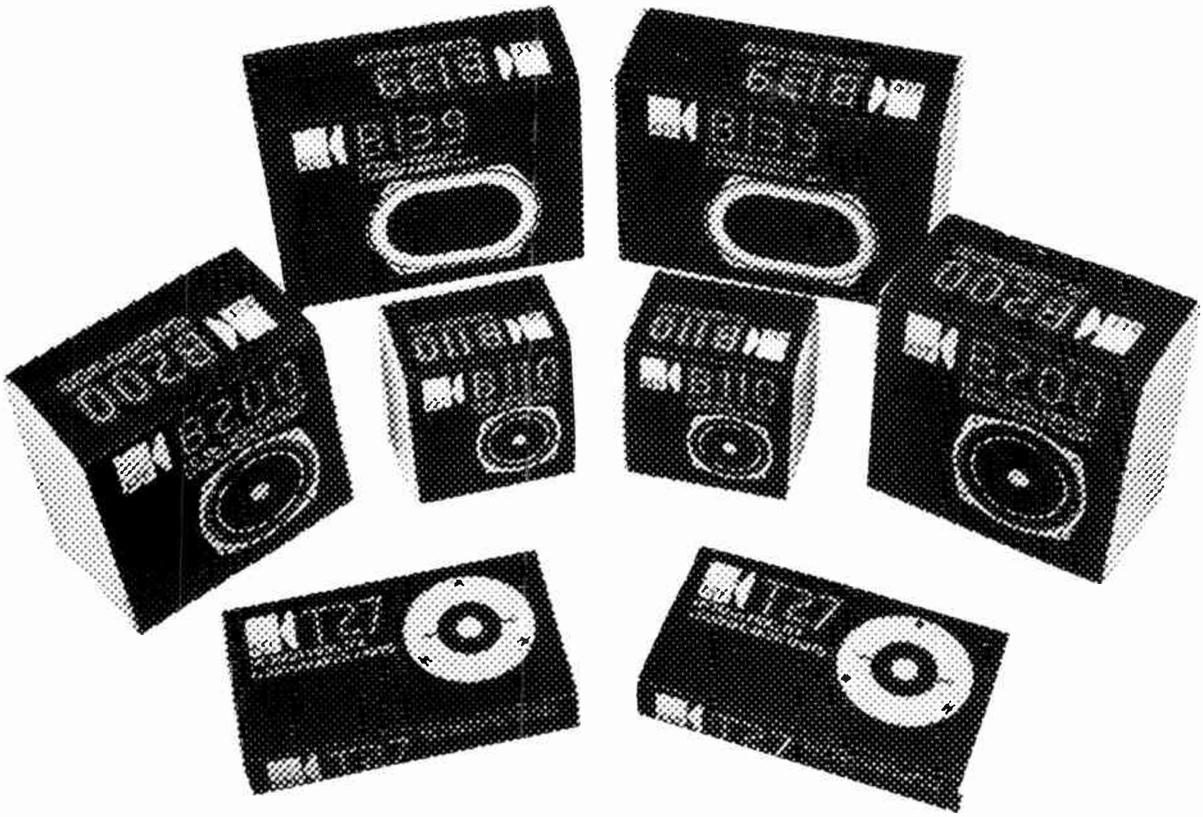
6. In the ac mode select 'A'-weight and apply a 1 kHz signal of sufficient level to obtain a 0 dB indication on the 1 volt range. Vary the frequency over the whole audio range and check that the response as shown in Fig. 4 is obtained.

7. Go back to 1 kHz and check that zero dB is indicated in the 'A'-weight mode. Now select 'flat' and adjust RV2 to obtain the same reading.

8. Apply an accurately known voltage with the instrument set to the flat and ac modes and adjust RV4 to give the correct reading.

9. Apply a dc input of known polarity and check that the correct LED illuminates. If not, reverse the leads to the LEDs.

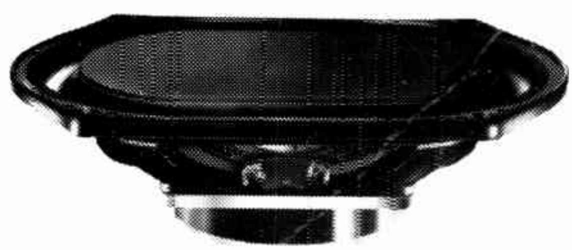
This completes the calibration and the instrument should now give accurate readings on all ranges and at all frequencies within the specified range. ●



KEF drive units

If you're making your own speakers, you cannot do better than build them round the world-famous KEF drive units: the B139 flat-fronted bass driver approaching perfect-piston performance right down to 20Hz; the B200 bass/mid range and B110 mid range units with rubber-modified polystyrene diaphragms and advanced suspension; and the T27 ultra high-frequency tweeter with special mylar dome to give wide dispersion.

Each of these superb drive units comes with full installation instructions and mechanical template to make fitting easy.



KEF B200

A 200mm (8") low frequency drive unit which uses a rubber modified polymer as the cone material. Laminated to it is a visco-elastic damping layer which absorbs excess energy at high volume to overcome 'cone-flapping.'

Quality control of these fine drive units is assured by KEF's sophisticated, computerised digital analysis techniques.

KEF B139

The world's most exclusive low frequency drive unit. It approaches perfect-piston performance right down to 20Hz.

The flat, light-weight expanded polystyrene diaphragm is acoustically dead, non resonant and suspended on a special neoprene roll-surround.

Quality control of these fine drive units is assured by KEF's sophisticated, computerised digital analysis techniques.

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