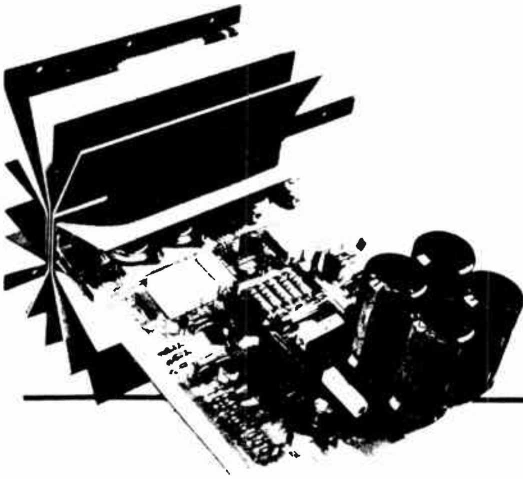

AUDIO PROJECTS

Projects

Articles

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'The Brute' — develops 300W into 4 ohms, 200W into 8 ohms!

Barry Wilkinson

For many audio applications there's no substitute for sheer power — low efficiency speakers, outdoor sound systems, or maybe you like the full flavour of the dynamic range afforded by a high power amp. Whatever your requirement — this 'super power' module should fill the bill.

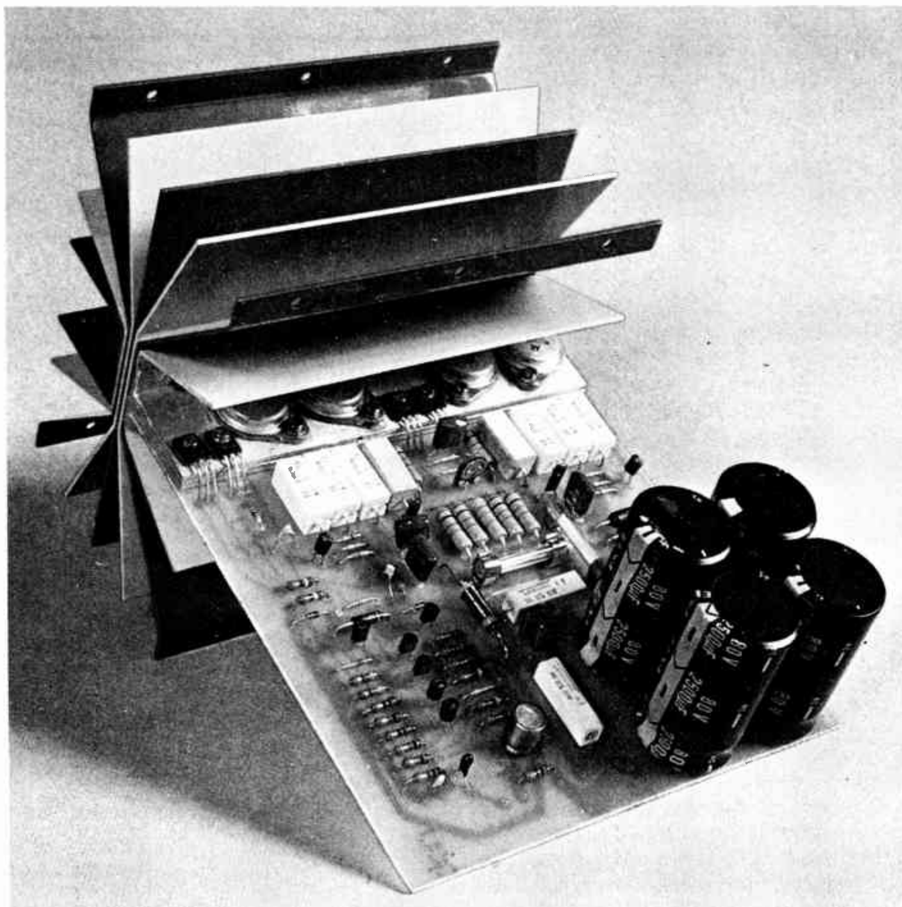
START HERE

Do not pass 'go', do not spend $200

THIS IS a relatively expensive project, compared to our previous amplifier modules, the ETI-480 and the more recent ETI-470. It is not recommended for beginners or inexperienced constructors. Although we have included protection for the output devices in the design it is obviously impossible to protect against circumstances which we cannot forsee. Follow the assembly details and advice given in this article — especially regarding heatsinks and power supplies etc, and you'll be well assured of success. We must *stress* that any deviation from this design, other than the variations suggested, you do at your own risk.

If this is your first experience with such high power don't be embarrassed to follow the instructions slavishly until you are familiar with the unit and get the 'feel' of the technology. Check *everything* as failures can be disastrous, not to mention spectacular, if something goes wrong.

If we haven't put you off by this stage — read on !



SPECIFICATIONS — ETI 466

Power output	
8 ohm load	200 watts RMS
4 ohm load	310 watts RMS
Frequency response	
20 Hz to 20 kHz	+/- 0.5 dB
Hum and noise	
re 200 W into 8 ohm	- 105 dB
Input sensitivity	
8 ohm load	1 V for 200 W output
4 ohm load	1 V for 300 W output
Total harmonic distortion	see graph
Damping factor	
20 Hz - 3 kHz	65
5 kHz	55
10 kHz	45
20 kHz	35

HI-FI AMPLIFIERS are becoming more and more powerful, and with good reason. Modern recordings, especially direct-cut discs, have a useful dynamic range approaching 40 dB between the quieter musical passages and the peaks of the crescendos. If the quieter passages are played at a power output of 100 mW, which is not untypical in a domestic environment, to faithfully reproduce the full recorded dynamic range of a good record without clipping the peaks would require an amplifier capable of delivering 1000 watts! This, coupled with the current trend amongst some manufacturers to build speakers having quite low efficiency, plus the number of people who like their music loud (*and* undistorted) makes the case for high power amplifiers very strong indeed.

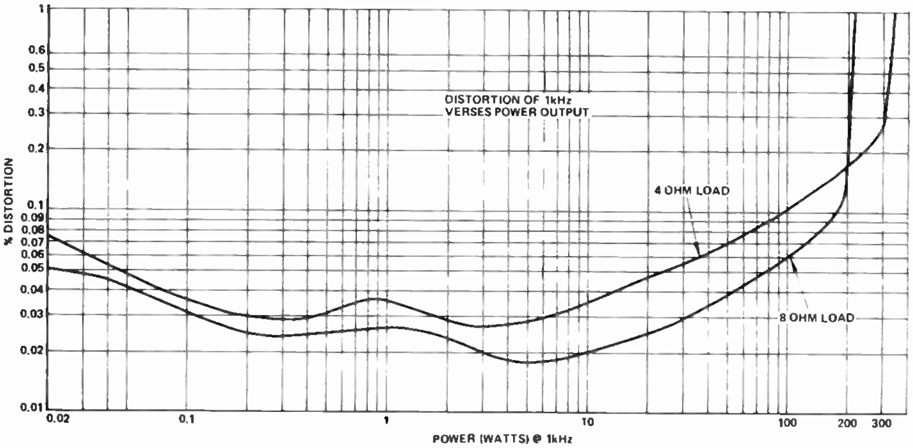
Past amplifier projects have generally been limited to output powers of 50 watts or so. Designed around cheap, readily available transistors, they have proved very popular. We have done the occasional 100 watt amplifier and once described a 'bridge' amplifier capable of delivering 200 watts into an eight ohm load, rather than design an amplifier using expensive, hard to get transistors for that power level.

To gain a worthwhile improvement in subjective performance over an amplifier of 50 watts output, we must go for a four times increase at least, to 200 watts, as the ear has a logarithmic response, and anything less is barely noticeable. That might be stating the case a little simply, but it conveys the general idea.

Over the past six or seven years we've had many requests for a *high* power amplifier, but for the reasons stated previously, we have decided against it. It would have been possible to design a unit using a large number of readily available power transistors in the output — in fact, one design we have seen used a total of 24 devices in the output stage! Difficulties for the home constructor in this approach are obvious, regardless of expense.

For various reasons, a bridge amplifier was ruled out when the design of this amplifier was considered. Hence, a plentiful source of suitable output transistors was first sought.

There are really not too many transistors available that meet the requirements. Firstly, adequate safe operating area (SOAR) is of prime importance. Next, and probably of equal importance, is availability. Let's have a look at the SOAR problem first. Some high power transistors don't compare too well with the ubiquitous 2N3055 (and its complement, the MJ2955) when operated as an amplifier. Take a look at the set of curves plotted on the accompanying diagram. This compares the safe operating area curves of a number of power transistors. Operation of any power device must be confined to the area inside the device's curve at worst case. If the current/voltage operating point is allowed to fall outside the area of the SOAR curve during any part of the operating cycle for the device, it will be destroyed — with amazing rapidity. Now, the 2N3773 and MJ802 transistors have been around for some time and at first glance would seem good choices for a high power amp, but note that their SOAR characteristics are not much better than the 2N3055. In fact, at 40 V (Vcc) the MJ802 is actually worse. In contrast, the MJ15003 is quite a long way outside the curve for the 2N3055 and therefore has a much higher power rating when used in an amplifier. Hence, the MJ15003 and its complement — the MJ15004, were chosen as the output devices for this design. Secondly, these transistors are widely used in industrial applications and are available from a number of sources, thus they meet the availability requirement. See Shoparound on page 147 for more information.

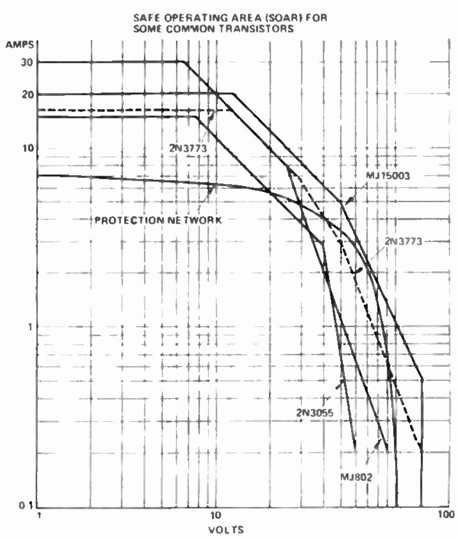


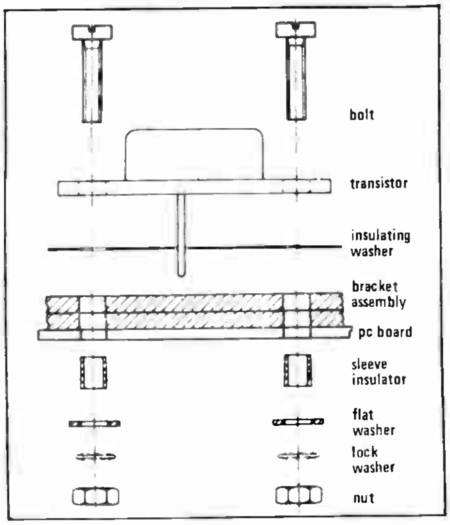
Total harmonic distortion versus power output at 1 kHz. The 'bump' at around 1 W is due to the output stage changing from Class A operation to Class AB operation.

Another problem that arises with a design such as this is protection for the output devices. Amplifiers using transistors such as the 2N3055/MJ2955 can easily be protected with a fuse. In high power amplifiers where supply rails of 60 — 70 volts are necessary, the energy available (from the filter capacitors) will easily destroy the transistor *and* the fuse — in that order. The answer is to use electronic current limiting in the output. This adds complexity, but is cheap insurance against accidental (or deliberate!) abuse. The curve showing the limiting effect on the SOAR characteristics of the MJ15003 for the protection network used in this amplifier is shown on the diagram with the other SOAR curves.

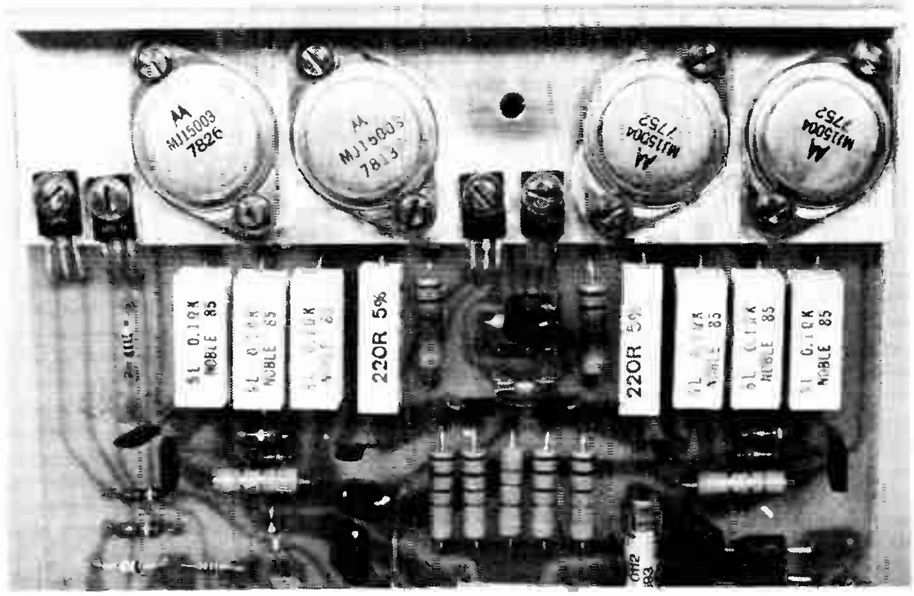
The main cost of the amplifier is in the output stage, transformer and heatsink. We therefore decided to go to a slightly more complex input stage to improve the performance. This type of amplifier usually uses a Class A driver which introduces second harmonic distortion. By using a complementary-differential input circuit we have been able to eliminate the Class A driver and therefore kept the second harmonic distortion very low indeed. The distortion curve shows the distortion is well under 0.1% until almost full power output. The 'bump' in the curve around one watt is the point where the output stage changes from Class A (peak output being less than the bias current) to Class AB operation. ▶

Comparison between the Safe Operating Area characteristics of a variety of transistors, including the MJ15003 used in the output stage of this amplifier.

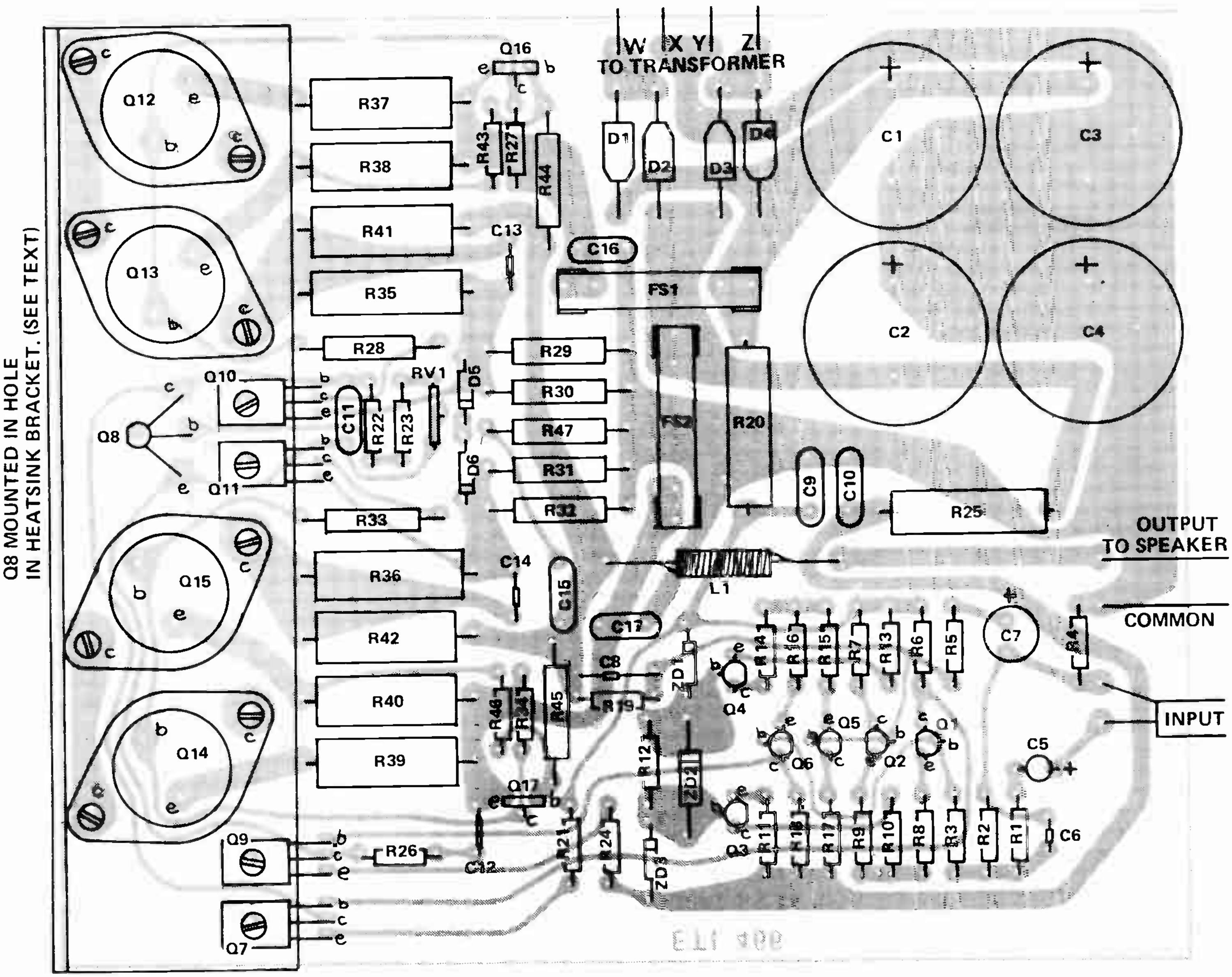




Exploded view of how the TO3 output transistors are assembled to the angle brackets and pc board.



Photograph of the completed output stage, prior to mounting to the heatsink assembly.



The complete amplifier, including the power supply components and output transistors, is assembled on a single pc board. An aluminium bracket holds the output transistors conducting heat from the output stage to the heatsink. Only three sets of external connections are made to the pc board; input, output and power supply ac input from the transformer.

Start the construction by making the aluminium bracket shown on page 11 We used two length of 3 mm angle which may be purchased from Alcan Handyman stores. This bracket is 3 mm thick and two must be placed back to back to make the required 6 mm thickness for adequate thermal conduction to the heatsink assembly. If you elect to use a Philips 65D6CB heatsink (see the box on 'Heatsinks'), a single 6 mm thick angle extrusion can be used, fixed to the flat side of this heatsink.

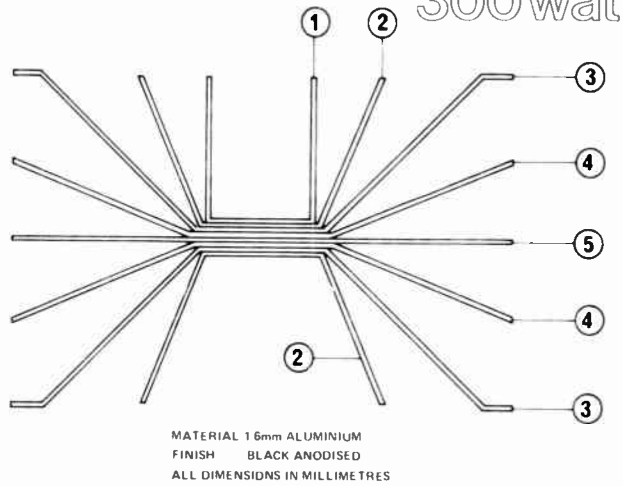
The easiest way to make the bracket assembly and ensure correct alignment of all the holes is to cut the two lengths of angle somewhat longer than necessary. The extra length will be cut off later. Clamp the two pieces back to back and drill a small hole at each end so that they can be clamped together with nuts and bolts through this excess. This allows you to shift the bracket assembly in a vice or what have you without getting them out of alignment. Next, mark out the position of the transistor holes (use the pc board as a guide if you have it to hand already) on the broad side of one bracket and then the holes in the narrow side – the latter secure the bracket assembly to the heatsink. Use a scriber or other sharp-pointed instrument. Then drill the holes.

The hole for the thermal feedback transistor (Q8) *must* be a neat fit. The best way to accomplish this is to drill a slightly smaller hole and carefully enlarge it with the correct size drill. A reamer gives a conical hole and is not really suitable. Those holes marked 'C' on the bracket drawings can be tapped to take a 4 BA bolt if you plan on using the sheet metal heatsink described later.

Once you have drilled all the holes in the bracket assembly, cut off the excess at each end and file the edges smooth. Also, ensure that no 'burrs' are left on the lips of each hole. Chamfer then with a large drill held in your hand.

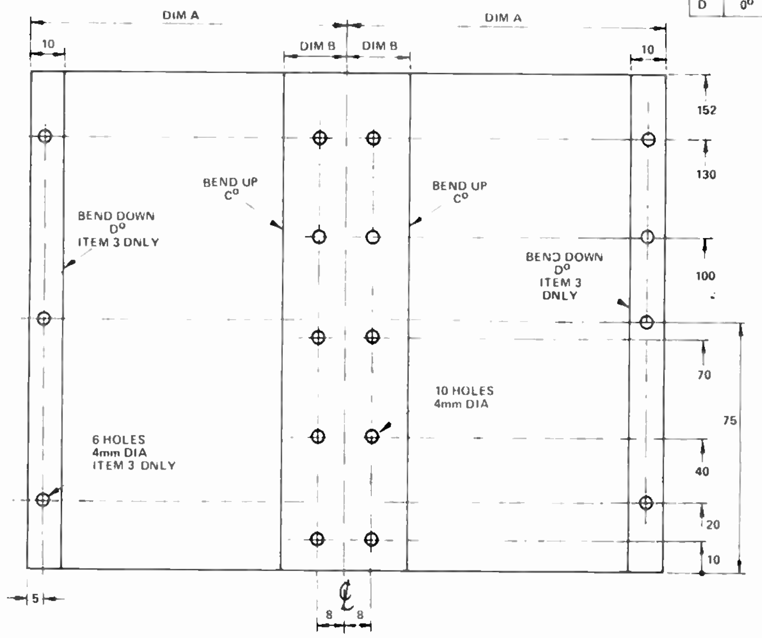
The next step is to make the heatsink assembly – that is, if you're not using one of the commercially-made alternatives suggested.

If you have access to a sheet metal ▶



Dimensions and bending details for the sheet metal heatsink assembly we used.

	ITEMS				
	1	2	3	4	5
A	58	65	95	80	75
B	15	17	19	21	–
C	90°	67.5°	45°	22.5°	0°
D	0°	0°	45°	0°	o



HEATSINKS

There are several alternatives you can choose from for heatsinking the amplifier output stage. The heatsink described, and shown in the front cover photograph, was made from sheet aluminium and has a thermal rating of 0.55°C/watt. This is the rating we recommend for any heatsink if the amplifier is to drive a four ohm load, particularly for pop group use. If it is driving an eight ohm load in typical domestic use, half the fins may be left out (every second one – the yellow ones!) resulting in a thermal rating for this heatsink arrangement of 0.75°C/watt.

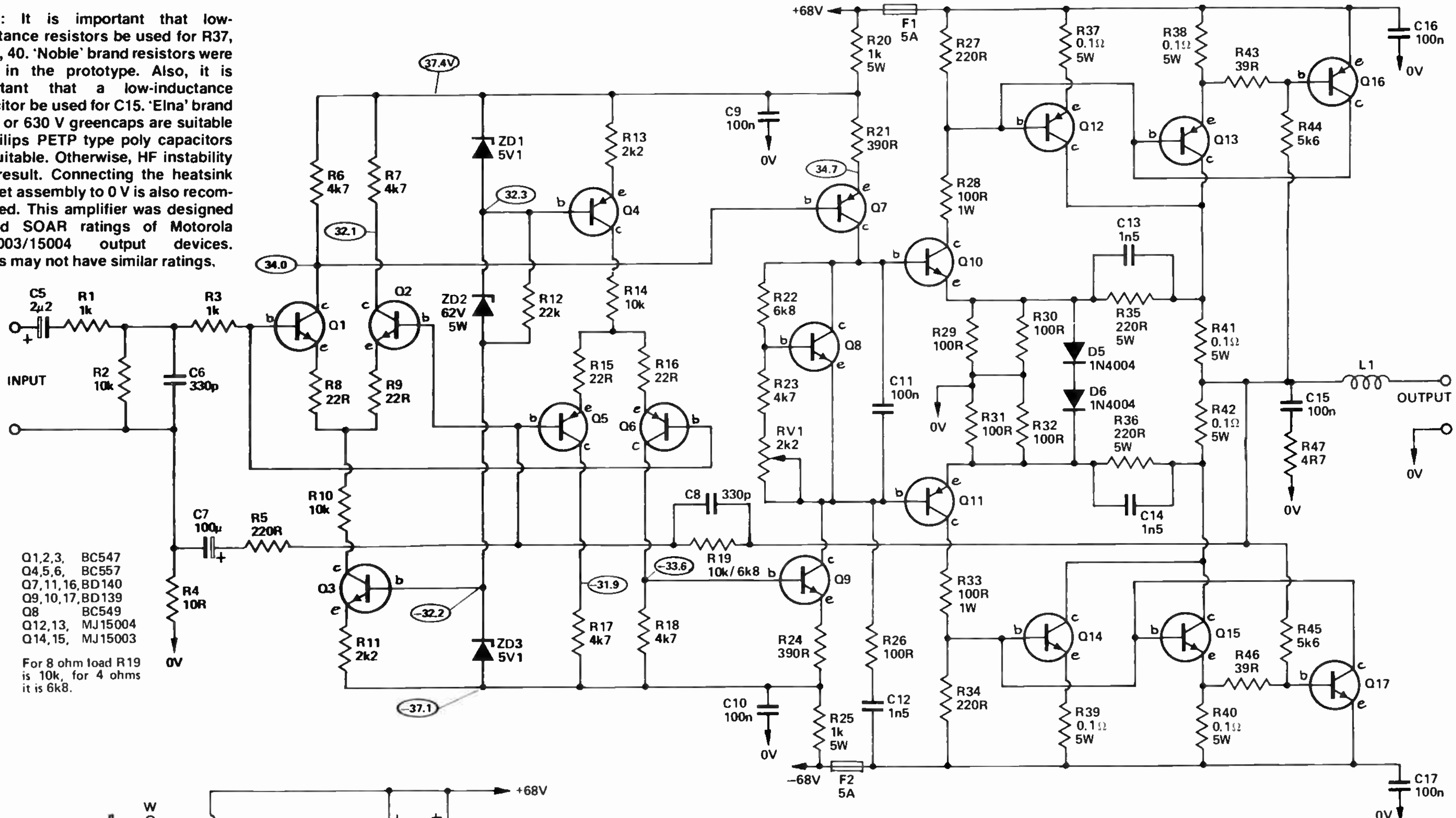
The nearest equivalent in a commercially-made heatsink is a 140 mm length of Redpoint R type – which nobody (to our knowledge) has had the foresight to stock in this country. Tch, tch.

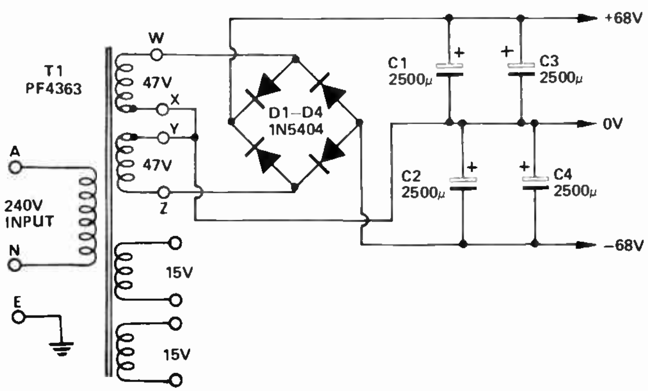
The Philips 65D6CB heatsink has a rating of 0.65°C/watt and would be suitable for this amplifier in most applications, except for a pop group with four ohm loudspeakers, unless fan cooling is added.

A heatsink with about 1°C/watt rating and substantial fan cooling is another alternative.

Remember that dissipation in the heatsink will be about 200 watts at full power output. That means a temperature rise of 110°C above ambient if the amplifier is run continuously. Poached eggs anyone? Temperature rise with music or intermittent use is considerably less, of course, as average power dissipated is much lower.

NOTE: It is important that low-inductance resistors be used for R37, 38, 39, 40. 'Noble' brand resistors were used in the prototype. Also, it is important that a low-inductance capacitor be used for C15. 'Elna' brand 250 V or 630 V greencaps are suitable or Philips PETP type poly capacitors are suitable. Otherwise, HF instability may result. Connecting the heatsink bracket assembly to 0 V is also recommended. This amplifier was designed around SOAR ratings of Motorola MJ15003/15004 output devices. Others may not have similar ratings.





Complete circuit diagram of the amplifier. Note that L1, not listed in the parts list below, is wound on a 1 W resistor — see text. Voltage readings are included as a guide.

The power transformer shown will power a pair of amplifiers (stereo) driving 8 ohm loads in typical domestic situations, but only a single module under other circumstances, particularly if driving a 4 ohm load.

When supplying two modules from a single transformer simply parallel W, X, Y and Z on each pc board and connect these to the transformer.

For stereo applications use separate earth returns for each speaker to the common on the pc board and separately join the two commons.

If the module is to be used in applications other than a domestic hi-fi set up and driving a 4 ohm load, we recommend you add another MJ15003/MJ15004 pair and associated components. The angle bracket and heatsink assembly will need to be extended.

bender, making your own heatsink is certainly the cheapest way out. The complete drawings are given back on page 7. Referring to these, note that dimension 'A' and dimension 'B' varies for each fin, the appropriate measurements being given in the table accompanying the drawings together with the angle of bend for each fin. Don't forget to allow a small angle for the 'spring' in the metal. Angles can be within a few degrees as they aren't that critical to heatsink efficiency. Don't be too sloppy though.

We used 1.6 mm thick aluminium sheet to construct the heatsink — *do not substitute a thinner gauge.* The bolts which secure the heatsink assembly to the bracket assembly also hold the whole heatsink assembly together.

It is easiest to drill the heatsink fins *before* bending them up, but you *must* mark out and drill the holes *accurately.* Mark one outer fin very carefully. Assemble the fins in order, making sure they are carefully aligned, then clamp the whole assembly and drill right through. Carefully de-burr all the holes.

At this stage you can do a trial assembly of the heatsink and bracket assemblies to see how it all mates — or not. If you have taken care with the drilling, then all should be well. Having confidence in your ability, we shall press on.

If you decide to paint the heatsink rather than having it anodised black, the mating surfaces should all be masked before spraying.

If you intend to use a Philips 65D6CB heatsink, the bracket holes may be marked on the heatsink using the already-drilled 6 mm thick bracket as a template. The holes can be drilled to the root diameter of a 4 BA bolt and suitably tapped.

The whole heatsink 'business' is not assembled at this stage, final assembly comes later. Be patient my little chickens !

The next part is the easy part (! . . . . Ed.). Having got the mechanicals off your chest, the electronics needs attention.

The components may be assembled to the pc board starting with the smaller resistors and capacitors. Carefully follow the overlay drawing. When you come to the 0.1 ohm, 5 W resistors note that they should be mounted about 2 - 3 mm off the board to allow a free air flow around them. Next mount the power supply electrolytics. Note that the recommended types have three pins projecting from the base. This is to provide mechanical rigidity. All three pins are soldered to the board and the capacitors can only be inserted one way round. The inductor L1 is made by winding a layer of 26 swg enamelled wire (or the nearest equivalent gauge) along the body of a 1 W resistor. The number of turns is not critical, just wind enough wire on the resistor to cover the body with one layer. The value of this resistor may be anything over 100 ohms. Two 5 A fuses are mounted on the pc board, held in place with fuse clips.

Next comes the semiconductors.

▼

HOW IT WORKS – ETI 466

The amplifier can be divided into three separate parts. These are: the input stage — which consists of Q1 - Q9, a high gain, low power driver; the output or power stage — which only has a voltage gain of four but enormous power gain; and the power supply.

The input stage is a complementary-differential network, each with its own current source. Each transistor in this stage is run at a collector current of about 0.7 mA. Emitter resistors are employed to stabilize the gain and improve linearity. The output of Q1 - Q6 drives Q7 and Q9. The latter are virtually two constant-current sources run at about 7 mA collector current. With an input signal these 'current' sources are modulated out of phase — the collector current of one decreases while the other increases. This configuration provides quite an amount of gain.

In between the bases of these two transistors is Q8, the thermal sensing - bias transistor. The voltage across Q8 may be adjusted by RV1, thus setting the quiescent bias current for the output stage.

The output stage, Q10 - Q15, has a gain of about five, set by R39 and R29 plus R30. Diodes D5 and D6 prevent reverse biasing of Q10 and Q11 (otherwise the output would be limited.

Protection of the output transistors is provided by Q16 and Q17 which monitor both current and voltage in the output transistors and bypass the base current if the limit is exceeded.

The power supply is a full-wave rectifier, with a centre-tap on the transformer giving the 0 V rail, providing +/- 68 volts. A total of 5000 uF is used across each supply rail for filtering. The amplifier input stage works on a reduced supply rail, derived from ZD1-ZD3 via R20 and R25.

Frequency stabilisition is provided by capacitors C8, 13, 14 and the RC networks R26/C12 plus R47/C15. Frequency response of the amplifier is set by C5 and C7 (lower limit), C8 sets the upper frequency limit.

The transformer has two additional windings of 15 Vac each. These are not used here but are suitable for powering a preamplifier.

PARTS LIST - ETI 466

Resistors all ½W, 5% unless noted

R1	1k
R2	10k
R3	1k
R4	10R
R5	220R
R6, R7	4k7
R8, R9	22R
R10	10k
R11	2k2
R12	22k
R13	2k2
R14	10k
R15, R16	22R
R17, R18	4k7
R19	10k(6k8 for 4 ohm loads)
R20	1k 5W
R21	390R
R22	6k8
R23	4k7
R24	390R
R25	1k 5W
R26	100R
R27	220R
R28–R33	100R 1W
R34	220R
R35, R36	220R 5W
R37–R42	0.1 ohm, 5W
R43	39R
R44, R45	5k6 1W
R46	39R
R47	4R7 1W

Potentiometers

RV1	2k2 trim

Capacitors

C1–C4	2500µ 80V RTP electro
C5	2µ2 35V tantalum
C6	330p ceramic
C7	100µ 25V RB electro
C8	330p ceramic
C9–C11	100n polyester
C12–C14	1n5 polyester
C15–C17	100n polyester

Semiconductors

Q1–Q3	BC547
Q4–Q6	BC557
Q7	BD140
Q8	BC549
Q9, Q10	BD139
Q11	BD140
Q12, Q13	MJ15004
Q14, Q15	MJ15003
Q16	BD140 or BC640
Q17	BD139 or BC639
D1–D4	IN5404
D5, D6	IN4004
ZD1	5V1 300 mW (IN751A)
ZD2	62V 5W (IN5372B)
ZD3	5V1 300 mW (IN751A)

Miscellaneous

ETI 466 . . . . pc board
Heatsink - see text
Transformer PF4363 (47 + 47V - 300 W)
4 fuse clips, 2 x 5A fuses

Leave Q7, 8, 9, 10 and Q11 plus the output stage devices Q12, 13, 14 and Q15 until last. Be *careful* with the orientation of the diodes.

Now you can assemble the heatsink bracket to the pc board, plus Q7 to Q15 inclusive.

First smear heatsink compound on the two mating surfaces of the bracket assembly. Note that insulating washers are used on all the transistors, Q7 to Q15, mounted on the bracket assembly (except Q8 of course). Smear both sides of each washer with heatsink compound. Place the bracket pieces on the board – component side – and secure Q7, Q9, Q10 and Q11 with nuts and bolts. Only tighten the nuts finger tight at this stage. Now, take the whole board and place the bracket ends against a flat surface – such as the flat heatsink fin – and juggle the brackets until the end faces are flush. Check that all holes line up and then tighten the nuts and bolts.

The TO3 power transistors Q12, 13, 14 and Q15 may now be assembled to the bracket and pc board using the accompanying assembly diagram as a guide. We used spaghetti insulation to sleeve the bolts but pieces of heatshrink tubing would be better.

Don't solder any leads yet.

Allow time for the heatsink compound to spread under compression and finally tighten all nuts. Last of all insert Q8. Smear the inside of the hole it sits in with heatsink compound to ensure good thermal contact.

Now you can solder all the transistor leads.

Check the component placement against the overlay now, just to ensure all is in order. If you wish, you can test the amplifier up to the driver stages for correct operation before assembling the unit to the heatsink. Remove the fuses before applying ac input from the transformer. Refer to the 'powering up' procedure. If there are any problems, look for errors in component placement or orientation – particularly with diodes. If all is well, assemble the module to the heatsink and you're ready for the big test.

Powering up

The set of output transistors is expensive to replace, therefore we recommend you follow this test procedure in the interest of conserving supplies of same.

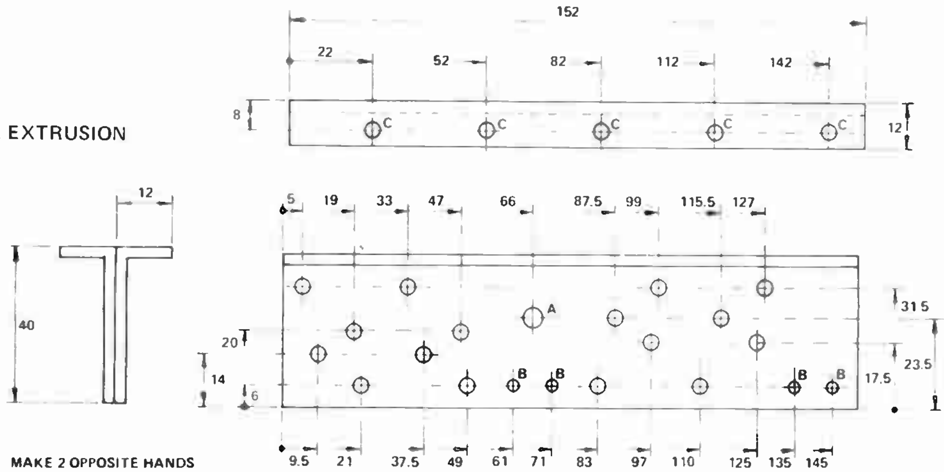
The power supply ac input should be connected to the transformer (see the overlay) but no power applied. You'll need a multimeter of at least 20k ohms/V sensitivity.

1) Remove the two fuses.
2) Solder a small link across C11.
3) Solder a wire between this link and the output pad.
4) With no load connected and no input signal, switch the power on.
5) Check the supply rail voltages. These should be about 68 volts each (plus and minus).
6) Check the voltages on the cathode of ZD1 (should be about +37 V) and the anode of ZD3 (about -37 V) with respect to 0 V.
7) If these two voltages differ with respect to each other by a volt or so, check other voltages around the input stage to determine the reason.
8) Check the dc voltage on the output (with respect to 0 V). It should be within 20 mV of zero.
9) Inject a sinewave signal into the input at a level of about 20 mV (RMS). Don't use a higher input level. Output should be 1 V RMS.
10) Switch off the main power and allow the filter capacitors to discharge. Remove the input signal.
11) Solder a 10 ohm ½W resistor across each fuse holder. Rotate the trimpot RV1 such that it is set at maximum resistance. Remove the short across C11 and the link from there to the output pad.
12) Switch on. . . . . if the 10 ohm resistors immediately vaporise you either have a short or some fault in the output stage!
13) If all is well, check the dc output voltage. It should be near zero.
14) Measure the voltage drop across one of the 10 ohm resistors placed across the fuse holders and adjust RV1 to give a reading of 1.0 V.
15) Switch off, allow the filter capacitors to discharge and remove the two 10 ohm resistors. Replace the fuses.
16) Connect suitably rated loudspeakers, warn the neighbours, connect a signal source to the input (turn down the volume), switch on the power and put the amp through its paces.

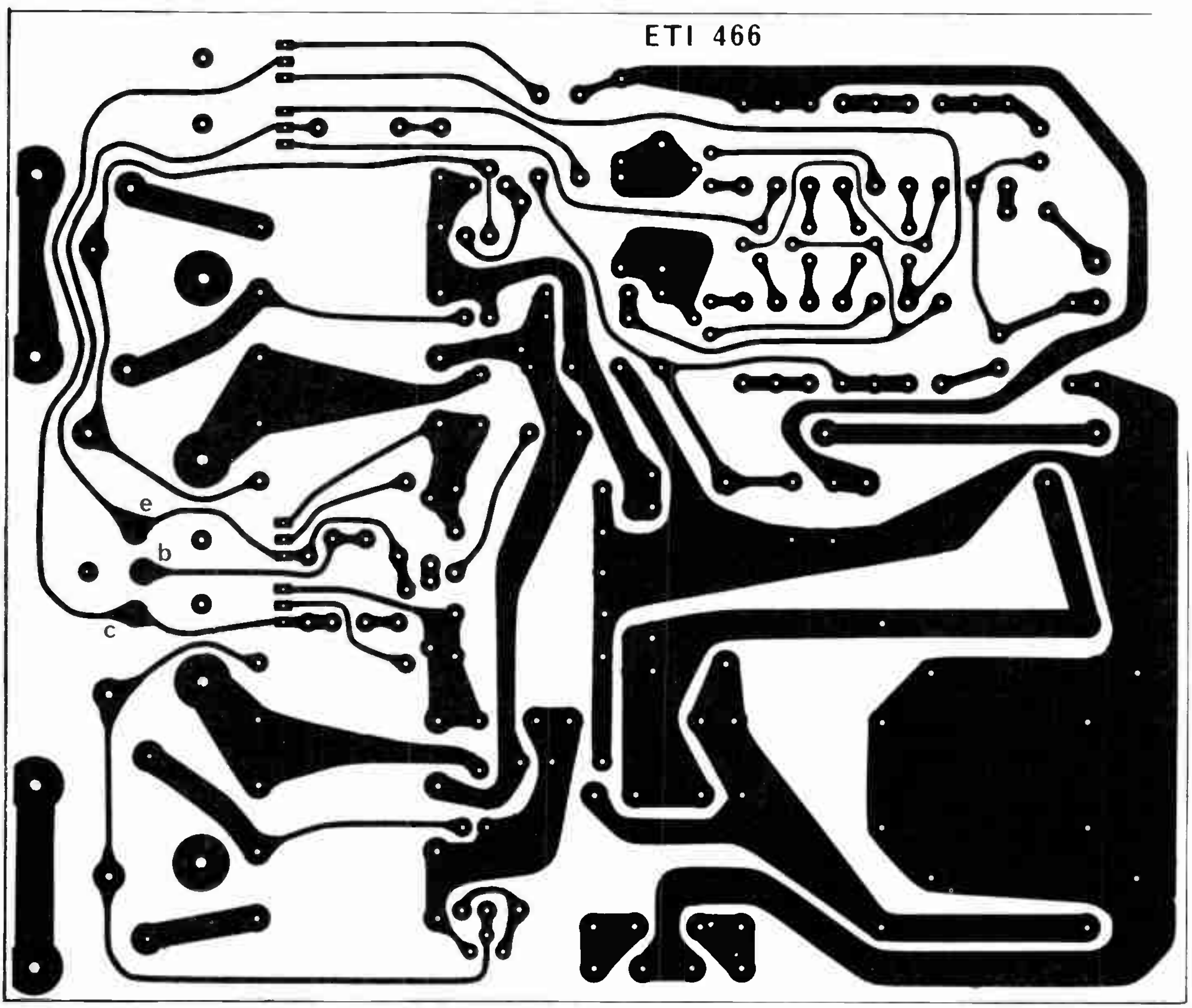
At this stage we'll leave the applications of this module up to you. No doubt you have plenty in mind already.

HOLES MARKED A 4.5mm DIA
HOLES MARKED B 3mm DIA
HOLES MARKED C TAP 4BA OR 4mm DIA
ALL OTHERS 4mm DIA

MATERIAL 40 x 12 x3 ALUMINIUM ANGLE EXTRUSION



Drilling details for the heatsink bracket assembly. All dimensions are in millimetres. Suitable aluminium angle stock is available from Alcan Handyman stores.



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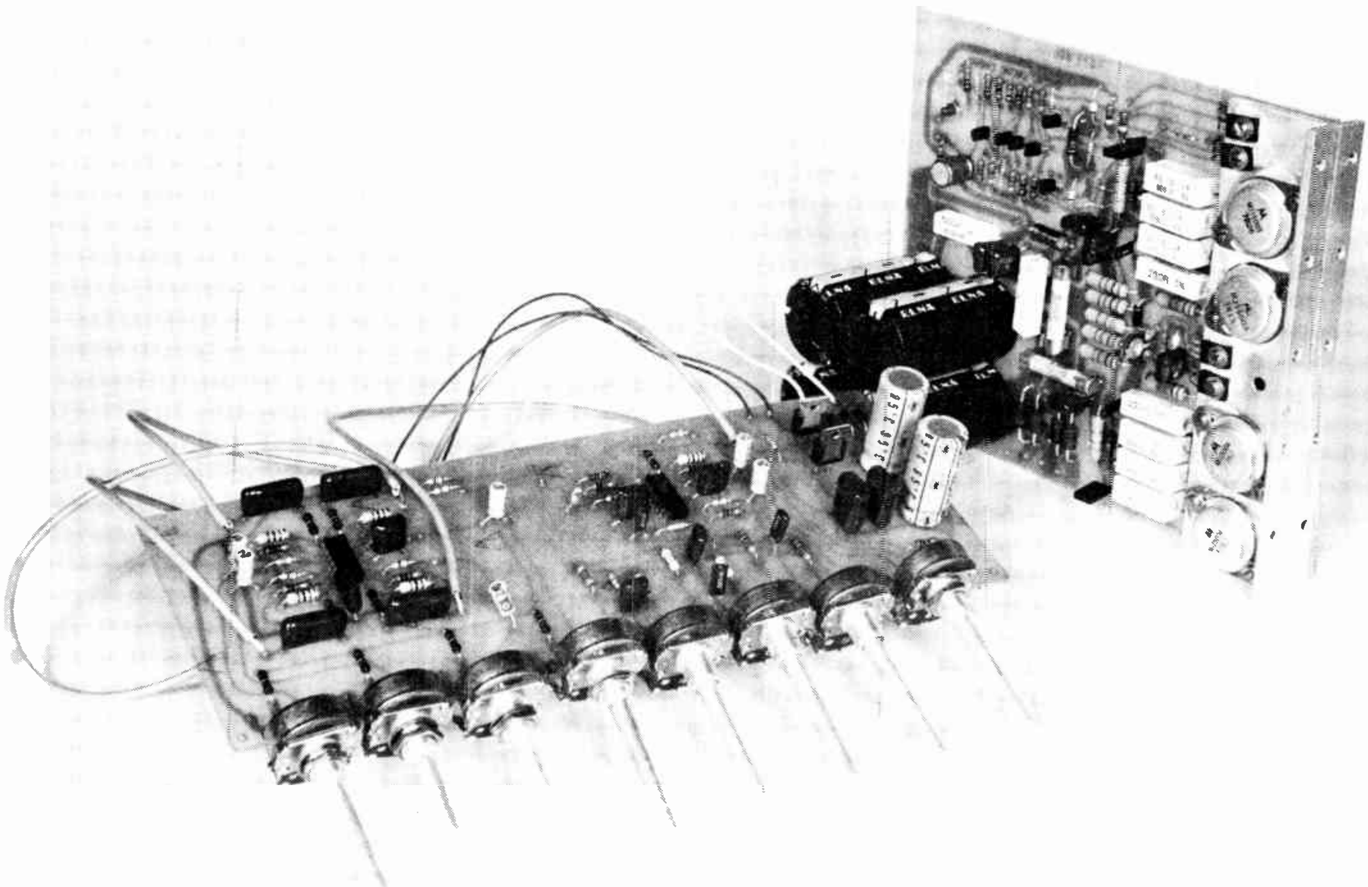
Four-input guitar/mic preamp to suit the ETI-466 module

Featuring simple construction and versatile operation, this preamp has been designed to mate with our 300 W 'Brute' power amp.

David Tilbrook



WE PUBLISHED the ETI-466 300 W power amplifier in February 1980. We then received many requests for a guitar preamp capable of driving the module to full output. The biggest demand was for a four-input preamp with bass, presence and treble controls followed by a master volume control. We are presenting the project as a printed circuit board rather than mounting the pc board in a chassis, as we feel most constructors will want to organise a chassis to meet their own requirements.

SPECIFICATIONS 4-INPUT PREAMP ETI-467

Hum and noise	**76 dB below 50 mV input signal (20 kHz bandwidth)**
Frequency response	**30 Hz to 30 kHz, +/− 1 dB**
Tone controls	**Bass: +/− 17 dB @ 50 Hz Presence: +/− 22 dB @ 1.5 kHz Treble: +/− 22 dB @ 10 kHz**
Max. output before clipping	**20 V peak to peak**

The power transformer recommended for use with the 466 power amp has an additional 15-0-15 volt winding that can be used to power this project if desired. For this reason we have included all the power supply components on the pc board, including voltage regulators. If the project is being constructed as a separate, stand-alone preamplifier, a small 12-0-12 volt power transformer can be used, such as the one specified in the parts list.

Design

The circuit consists of four input stages, followed by a mixing stage, tone control and output amplifier. The four input amplifiers drive the output of the mixing stage. This feeds the tone control circuit and then the final volume control and output amplifier. The 4136 quad operational amplifier IC was used throughout the project, mainly for the convenience of the quad package. One IC is used for the four input stages and the other for the mixing stage, tone control and output amplifer.

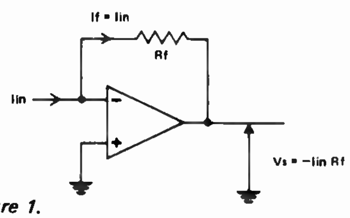


Figure 1.

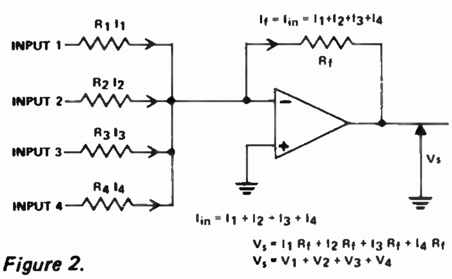


Figure 2.

Each input goes to one amplifier of the first 4136, where the input signal is amplified before going to the input level control. The input stages each have a gain of 20 dB, so a typical input level of around 50 mV will be amplified to 500 mV before being applied to the input level controls. With the input controls set at midway position the output from the potentiometers will be reduced to around 50 mV again. Without the input amplifiers, this signal voltage would be only 5 mV, causing a dramatic decrease in signal-to-noise ratio. The input amplifiers have an input impedance of 100k which should suit the vast majority of guitars. The outputs of the input level controls are fed to a 'virtual earth' point formed by the feedback loop around another op-amp. This is probably the most common technique used for mixer stages at audio frequencies and allows signal voltages or currents to be added independently. Figure 1 shows the circuit of an ideal op-amp with negative feedback applied through resistor R_f. If a positive-going current is passed into the non-inverting (−ve) input, the output of the op-amp will swing negative, pulling current through the resistor R_f, until the voltage at the non-inverting input returns to zero. The output of the op-amp will always attempt to maintain the voltage on the non-inverting input at earth potential and for this reason it is referred to as a 'virtual earth' point. The output signal voltage from the op-amp is given by the equation,

$$V_S = -I_{in}R_f$$

Where: V_S is the signal output voltage
I_{in} is the input signal current
and R_f is the value of the feedback resistor.

If input resistors are added to the circuit as shown in Figure 2, the signal current from each input will be determined by the input signal voltage and the particular input resistor. Since the non-inverting input is a virtual earth point, the current through each input resistor will not be affected by the other input currents. The total input current simply becomes the sum of all the individual input currents. So,

$$I_{in} = I_1 + I_2 + I_3 + I_4$$

and since the output signal voltage is given by:

$$V_S = -I_{in}R_f$$

then

$$V_S = I_1R_f + I_2R_f + I_3R_f + I_4R_f$$
$$= V_1 + V_2 + V_3 + V_4.$$

The output signal voltage is thus the sum of the individual input signal voltages — exactly what is required of a mixing stage.

The output of the mixer is fed to the input of the tone control circuit. The bass, presence and treble controls are formed by including a potentiometer and a suitable R-C combination in the feedback loop of an op-amp. The potentiometer varies the amount of feedback around the op-amp and this has the effect of altering the frequency response of the circuit.

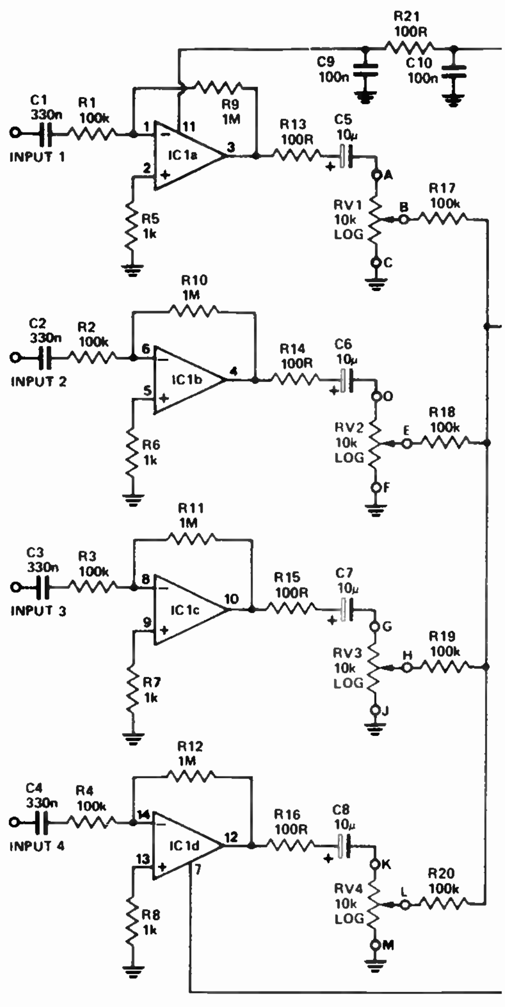
We judged that most guitarists would

HOW IT WORKS

The four input stages each use one section of a 4136 quad operational amplifier. These op-amps are internally compensated and require no external compensation capacitor. The input stages are configured as inverting amplifiers with a gain of 10 (20 dB) set by the ratio of the input resistor to the feedback resistor. The 100 ohm resistor in series with the output of each op-amp isolates any reactance on the output from the feedback loop, ensuring maximum freedom from instabilities that might otherwise cause oscillation.

Resistors R21 and R22, and capacitors C9 and C10 decouple the positive and negative supplies of the input amplifiers from all subsequent amplifier stages.

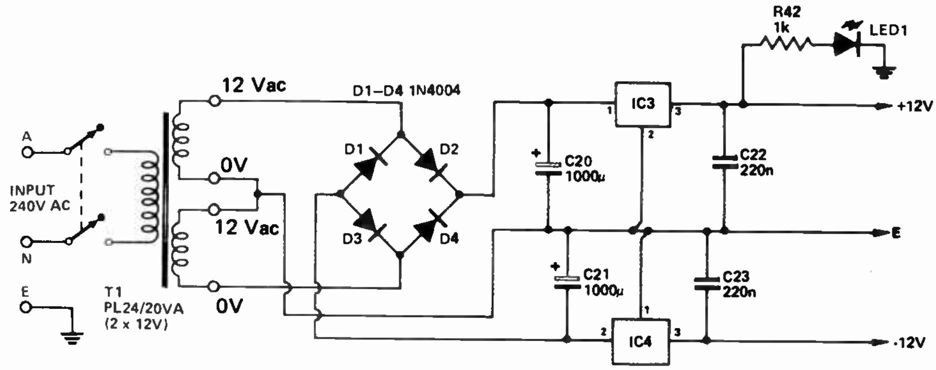
The outputs of the input stages are fed to the mixer input where they are summed as described in the text. The signal is then fed to the input of the tone control circuit, then through the master volume control to the last amplifier stage and finally to the output.

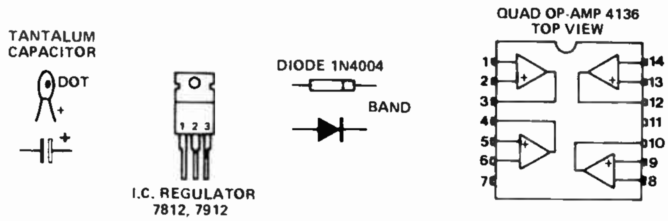


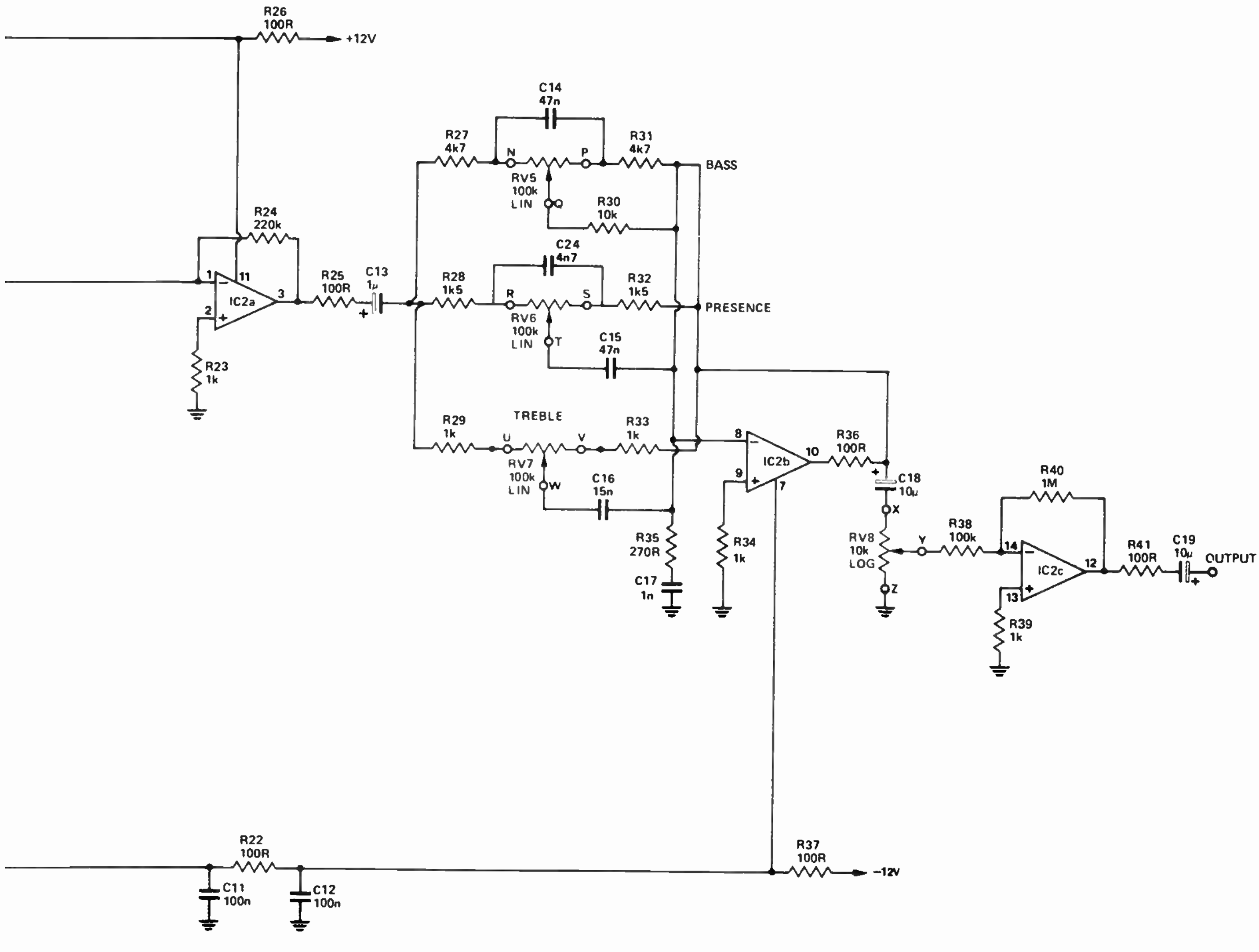
— ETI 467

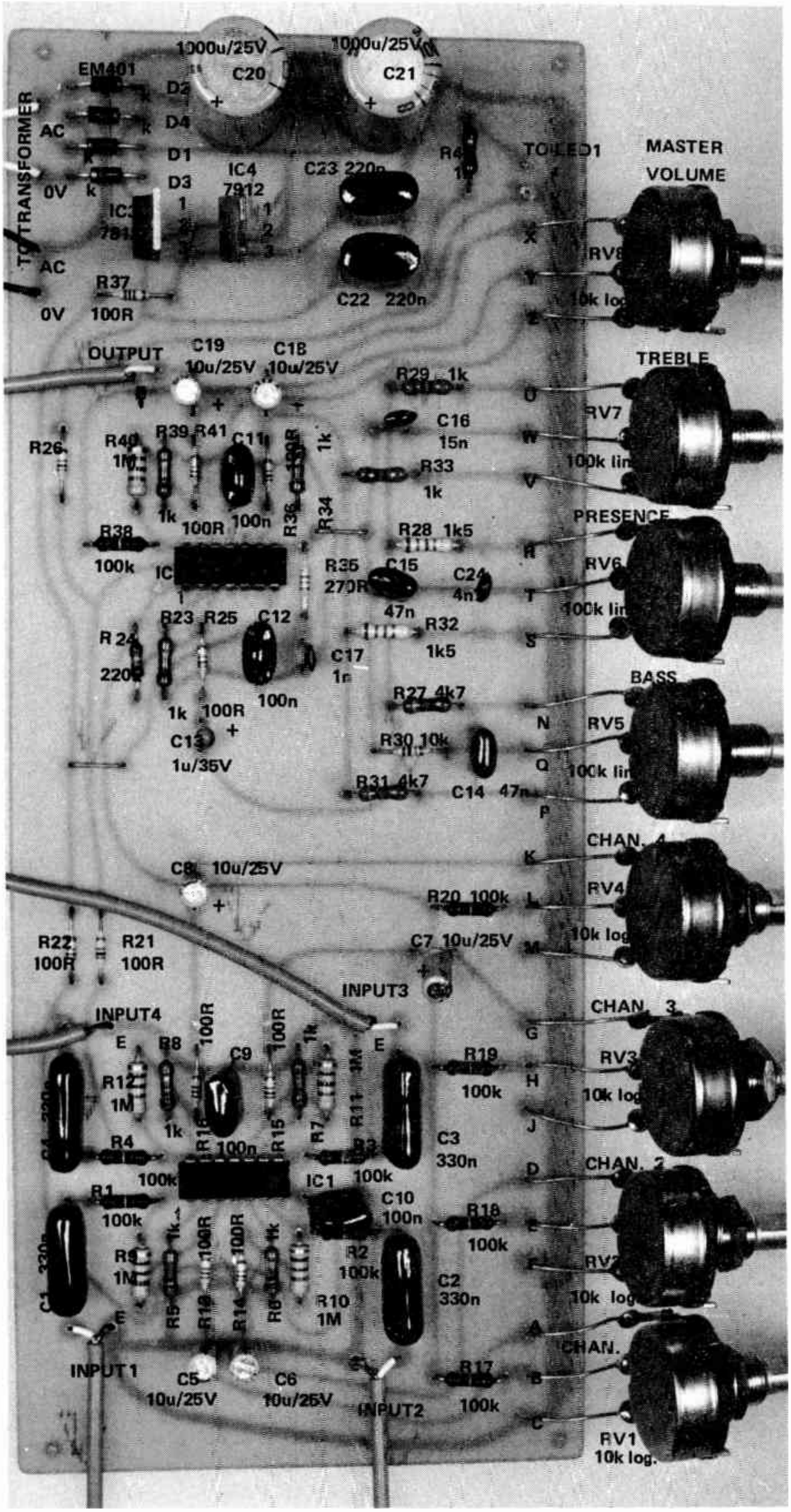
The tone control stage works in a very similar way to the mixer stage. The signal currents in the bass, presence and treble circuits are summed at the non-inverting (−ve) input of the op-amp without interacting with one another, so the tone control really consists of three circuits operating simultaneously and these can be analysed separately. In each control, the potentiometer and the two resistors connected to each end of it, form the effective input and feedback resistance. The position of the wiper will determine the overall gain of the circuit. If a capacitor is now added, either across the potentiometer or in series with its wiper, a low pass or high pass frequency response, respectively, is obtained. The presence control simply consists of both a low pass and a high pass in the one circuit, forming a variable bandpass filter.

The power supply circuit is simple and uses commonly available IC voltage regulators to ensure stable voltage rails and minimise hum levels.









want a tone control that gives a relatively large amount of boost and cut, considerably larger than most hi-fi tone controls. The bass control has over 16 dB of boost and cut, while the presence and treble controls give over 20 dB.

The output of the tone control circuit goes to the master volume control and then to the final amplifier stage. The output impedance of the circuit is around 100 ohms and the maximum output voltage is 20 volts peak to peak, more than adequate to drive a power amplifier.

Construction

All the construction is restricted to the printed circuit board and so is relatively straightforward. We recommend you use our pc board, otherwise stability and ground-loop problems may occur.

Mount the resistors and non-polarised capacitors first. Next mount the tantalum and electrolytic capacitors, ensuring that they are inserted with the correct orientation. Most electrolytic capacitors have their negative

PARTS LIST

Resistors all ¼W, 5%

R1,2,3,4,17 18,19,20,38	100k
R5,6,7,8,23,29, 33,34,39,42	1k
R9,10,11,12,40	1M
R13,14,15,16,21 22,25,26,36,37,41	100R
R24	220k
R27,31	4k7
R28,32	1k5
R30	10k
R35	270R

Potentiometers

RV1,2,3,4,8	10k log.
RV5,6,7	100k lin.

Capacitors

C1,2,3,4	330n greencap
C5,6,7,8,18,19	10u, 25 V electro
C9,10,11,12	100n greencap
C13	1u, 35 V tantalum
C14,15	47n greencap
C16	15n greencap
C17	1n greencap
C20,21	1000u/25 V electro
C22,C23	220n greencap
C24	4n7greencap

Semiconductors

IC1,IC2	4136 quad op-amp
IC3	7812 +ve 12 V reg.
IC4	7912 –ve 12 V reg.
D1,D2,D3,D4	EM401, 1N4001 or sim.
LED1	TIL220 or similar LED

Miscellaneous

Transformer (if required) — 2 x 12 V, 0.8 A Ferguson PL24/20 VA (plus mains cord, cable clamp, plug, etc); pc board — ETI 467; five phone jack sockets; eight knobs; DPST 240 Vac switch (if required) lead identified by a black arrow on the body of the capacitor. The polarity of tantalum capacitors is indicated by the position of the dot, as shown in the drawing accompanying the circuit diagram. Mount the power supply diodes and ICs next. Again, be certain these components are inserted with the correct orientation.

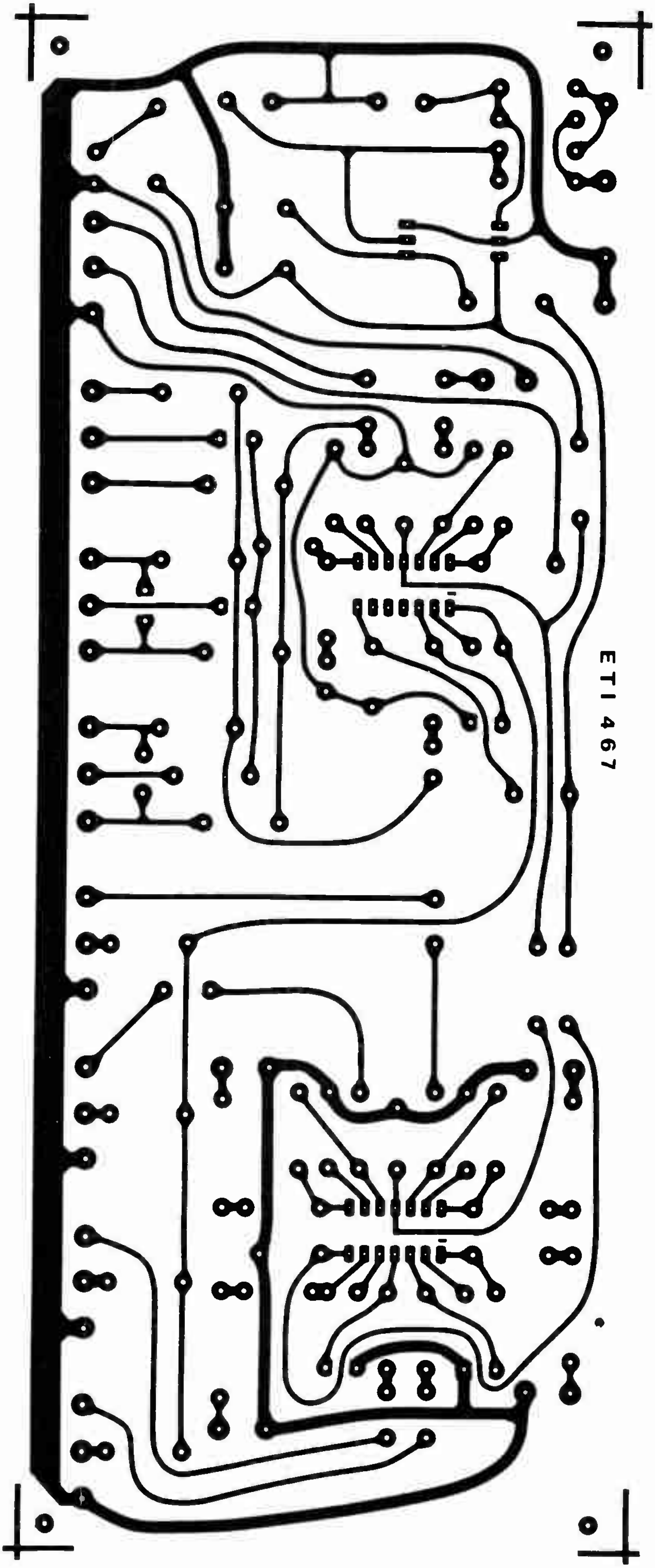
If the preamplifier is being constructed in a metal chassis, such as a 19-inch (483 mm) rack mounting cabinet for example, a separate power transformer is needed. This should be mounted on the chassis as far away from the pc board as possible. The input jacks will probably be mounted on the front panel and this will automatically connect the chassis to the signal earth at this point. If you experience any problems with hum in the finished unit this will probably be the cause. The solution is to try to obtain jack sockets that are insulated from the chassis, then experiment with the earthing arrangement. In most applications there should be no problems with hum, we have experimented by varying the position of the board with respect to its power transformer and it does not seem to be particularly sensitive.

Shielded cable should be used between the board and the input and output sockets. In the prototype unit the potentiometers are wired with short lengths of tinned copper wire. If the distance between the pc board and the pots is increased by more than a couple of centimetres, the connections to the pots should be made using shielded cable.

There is no special set-up procedure needed, but check all the components on the printed circuit board before applying power.

Using it

Turn all the input controls and the output volume control *fully counter-clockwise* and set the three tone controls to their mid positions. Connect the output of the preamplifier to the input of the ETI-466 (or what-have-you) power amplifier before turning the power amp on. This will prevent any momentary injection of hum into the power amp which could damage your loudspeakers. Plug in a guitar and turn up the input control for the channel you are using. Now, gradually increase the output volume control until the required volume is achieved. The tone controls may be adjusted to your liking, a bit of experimentation will show up what suits you and your system. ●



Series 3000 compact stereo amplifier

Phil Wait

"Small is beautiful" when you have a premium on space, but there's no need to sacrifice performance. This amp delivers 20 W/channel, has low distortion and costs less than $90!

THE TREND to living in 'compact' accommodation such as units, town houses and inner city terraces, has led to a demand for appropriately sized household goods and furniture — including hi-fi equipment. The major hi-fi manufacturers all leapt into this market last year, releasing various combinations of separate components and integrated arrangements generally tailored to a size to stack conveniently on a shelf or bench top and not take up precious space.

This project is aimed at those readers who want something to fill that requirement but want to 'roll their own' to save money or just to gain the satisfaction of having done it themselves.

Apart from that, this project is ideal for the beginner with a little construction experience who wants a 'meaty' project to tackle. This unit is quite simple to build as the majority of components are mounted on a single printed circuit board and the interwiring has been simplified with the use of 'ribbon' cable.

Despite the low price, compared to other such projects around, this unit is not a 'cheap' amplifier. The performance is demonstrably better than similar amplifiers that cost a great deal more and it can be teamed with some of

SPECIFICATIONS

Power output
25 W RMS, one channel driven
20 W RMS, both channels driven

Distortion (refer to graph)
1kHz: 0.03% @ full power
0.013% @ 12W RMS
10 kHz: 0.08% @ full power

Frequency response
Phono: within 1 dB, RIAA
Other inputs: within ±0.5 dB from 10 Hz to 20 kHz; −3 dB @ 40 kHz

Hum
Phono: −60 dB w.r.t. 10 mV input
Other inputs: −70 dB w.r.t. 200 mV input

Noise
Phono: −80 dB w.r.t. 10 mV input
Other inputs: −86 dB w.r.t. 200 mV input

Tone controls (see text)
Bass: ±10 dB @ 50 Hz
Treble: ±10 dB @ 12 kHz

Slew rate
15 V/μs

Separation
Phono: 46 dB
Other inputs: 40 dB

Sensitivity
Phono: 2.5 mV for full output
Other inputs: 200 mV for full output

Tape output level
200 mV

our previously published hi-fi projects to obtain quite a respectable hi-fi set up.

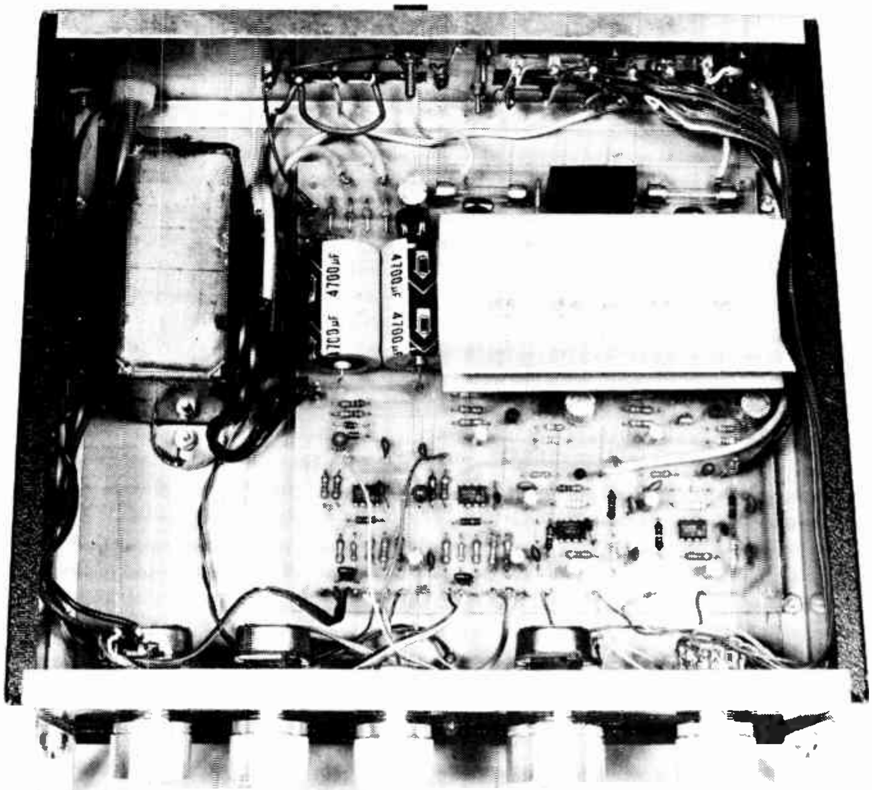
The design

Overall design is fairly conventional, except for the output stage, but we have paid attention to such details as transient intermodulation and slew limiting distortion as well as keeping total harmonic distortion within acceptably low limits.

The preamp stages employ common integrated circuits op-amps. The power amplifiers employ a differential input stage driving a complementary-symmetry class-B power stage. The output devices are TIP31C/TIP32C 'flatpack' transistors arranged such that a simple heatsink can be mounted on the pc board — which can be seen in the internal photograph. A conventional transformer/bridge rectifier power supply is used and speaker anti-thump circuitry is provided too.

The output stage of each channel is capable of delivering 25 watts (single channel) into an 8 ohm load. If you're building the amplifier as a Christmas present for your kids, in the interest of tranquillity, family relations and the sanity of the family cat(/dog/budgie/goanna) the power output might seem 15 - 20 watts too much. By the simple expedient of using a lower voltage transformer and changing a few resistors, the amp will deliver only five watts(*). This also reduces the heatsink requirement and the overall price (as the transformer's less expensive). The two different types of transformer are given in the parts list.

*Psst — kids. If the old fella has built you a five watt amp you can make it up to a 20 W/ch amp without much fuss. Read on.

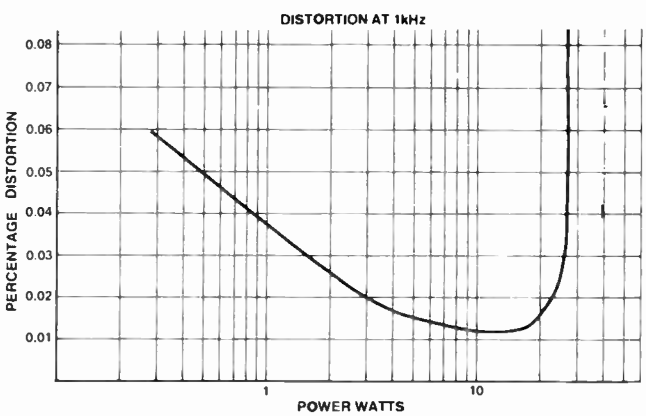


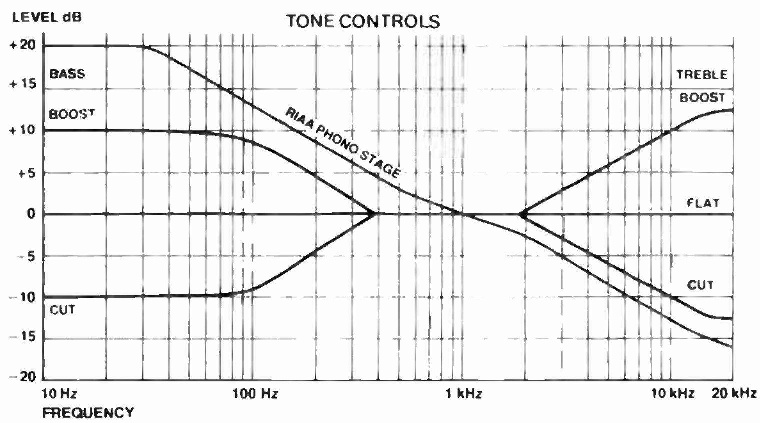
Internal view of the completed project shows the general layout and obvious simplicity — almost everything's on the pc board. We haven't anything much to say about the external appearance as it speaks for itself! Scotchcal transfers for the front and rear panels will be available from suppliers — see Shoparound on page 147.

When you turn an amplifier on or off, a loud thump may come from the speaker. This transient may have sufficient energy to destroy the speaker! It is caused by the supply line voltage rising (at switch-on) or falling (at switch-off) at different rates. This may cause the output to the speaker to swing wildly between the positive and negative supply rails under the worst-case conditions — spectacular, but costly if your speakers can't handle it. Consequently, we have provided an anti-thump circuit which only connects the speakers after a short delay whenever the amplifier is turned on. When the unit is turned off, the speakers are immediately disconnected, before the supply rails have a chance to fall appreciably.

Construction

We chose to assemble the amplifier in a Horwood instrument case measuring 255 mm by 255 mm by 78 mm overall. These are supplied with black, vinyl covered steel top and side panels and plain aluminium front and rear panels. Handles are attached either side of the ▶





HOW IT WORKS — ETI 476

This amplifier employs common IC op-amps in the phono and preamp stages driving a class-B power amplifier employing discrete circuitry. The power supply is conventional, employing a transformer and bridge rectifier with capacitor-input filters to derive positive and negative supply rails and incorporating a speaker anti-thump circuit.

The following description applies to one channel only as both channels are identical, except for component numbering. Components are numbers IC1, R1, C1 etc in one channel, IC101, R101, C101 etc in the other. Ganged controls are labelled SW1a, b or RV1a, b etc.

PHONO STAGE

The phono input, from a moving magnet turntable cartridge, is applied to the input of IC1, an LM301, via an R-C network (C1, R1) which provides a roll-off at subsonic frequencies. The feedback network for IC1 (R3, R4, R5, R6 and C4, C5) provides the correct RIAA compensation. Output from this stage goes to the Source switch, SW1, via C7/R7.

Resistor R7 maintains the negative side of C7 at 0 V to prevent speaker thump when operating the Source switch. The purpose of C6 is to reduce the gain of IC1 to unity at dc so that the dc offset at its output is maintained at a low value.

Capacitor C3 provides stability compensation for IC1. Resistor R2 and capacitor C2 provides RF interference immunity for the input of IC1.

Overall gain of the phono amp is about 300 and is designed to provide full power output with 2.5 mV RMS input at 1 kHz.

PREAMP-TONE CONTROL

The Source Switch selects the various inputs, applying them to the Volume Control, RV1, via the Tape Monitor Switch, SW2.

Output from the Volume Control passes to the input of the preamp-tone control stage via C8. Conventional tone control circuitry around the feedback of IC1 provides boost and cut at 12 kHz for Treble and 50 Hz for Bass. However, unlike conventional controls we have only provided a range of +/− 10 dB for reasons explained in the text.

General gain for this stage is around 10, and input sensitivity is 200 mV RMS for full power output. The slew rate for this stage has been set at 15 V/us so that it is slower than the power amplifier. This has been done by selecting the value of the compensation capacitor, C11 to limit the speed to that required. The R-C network formed by R15 and C12 provides additional slew rate limiting at the output of the tone control stage. This technique avoids transient intermodulation distortion developing in the power amplifier.

POWER AMPLIFIER

Keen-eyed readers will recognise similarities between this circuit and the ETI-452 Guitar Practice Amplifier (Jan 1980) and the ETI-453 General Purpose Amplifier Module (page 71). There's no need to re-invent the wheel!

Ten transistors are employed (Q1 to Q10) in a discrete component design. Transistors Q1 and Q2 form a differential input stage. Q4 provides a constant current source for the input stage emitters, biased by R24 and LED1.

Output from the preamp-tone control stage is applied to the Balance control, and thence to the input of the power amplifier stage via C13 to the base of Q1. The collector of Q1 is directly coupled to the base of Q3, an emitter follower which is directly coupled to the base of the pre-driver, Q5. Diodes D1, D2 and D3 maintain about 1.8 V between the bases of Q7 and Q8. Each of these transistors will drop about 0.6 V across their base-emitter junctions. This leaves a total of 0.6 V to be dropped across the two 100 ohm resistors R27 and R28. Since these are of equal value, each drops 0.3 V and holds this across the base-emitter junctions of Q9 and Q10, the output transistors. As these two transistors require 0.6 V to be biased on, they will remain off until the applied signal raises the voltage on the bases above 0.6 V (with respect to 0 V). Only a little more than 10 mA through R27 and R28 will supply the extra 0.3 V to turn on the output transistors.

Transistor Q6 provides a constant current sink (or source, depending on your point of view) for the collector current of Q5, increasing the gain of the drive stage, Q5, and decreasing distortion. There is approximately a one volt drop across R26 (and incidentally, R19).

Emitter ballast resistors are included in the output stage, these being resistors R29, 30 and R31, 32. Their main purpose is to help prevent thermal runaway in this application and stabilise the gain of each output transistor. They play a secondary role as fuses in the event of a fault condition causing heavy conduction in the output devices. Hence, the text advises these resistors be mounted up off the pc board on their leads.

Negative feedback is supplied by the potential divider formed by resistors R23 and R20. The capacitor C3 represents a short circuit to the common rail (0 V) for ac signals in the audio range. Gain of the stage across the audio range is thus the ratio of R23 to R20, about 12 in this case. At very low frequencies the impedance of C3 increases, decreasing the gain of the power amplifier by increasing the amount of negative feedback. Capacitor C16 increases the speed of the ac feedback at high frequencies.

The base of Q1 is tied to 0 V and as the whole amplifier is dc-coupled, the quiescent output voltage will be held to a value less than about 50 mV.

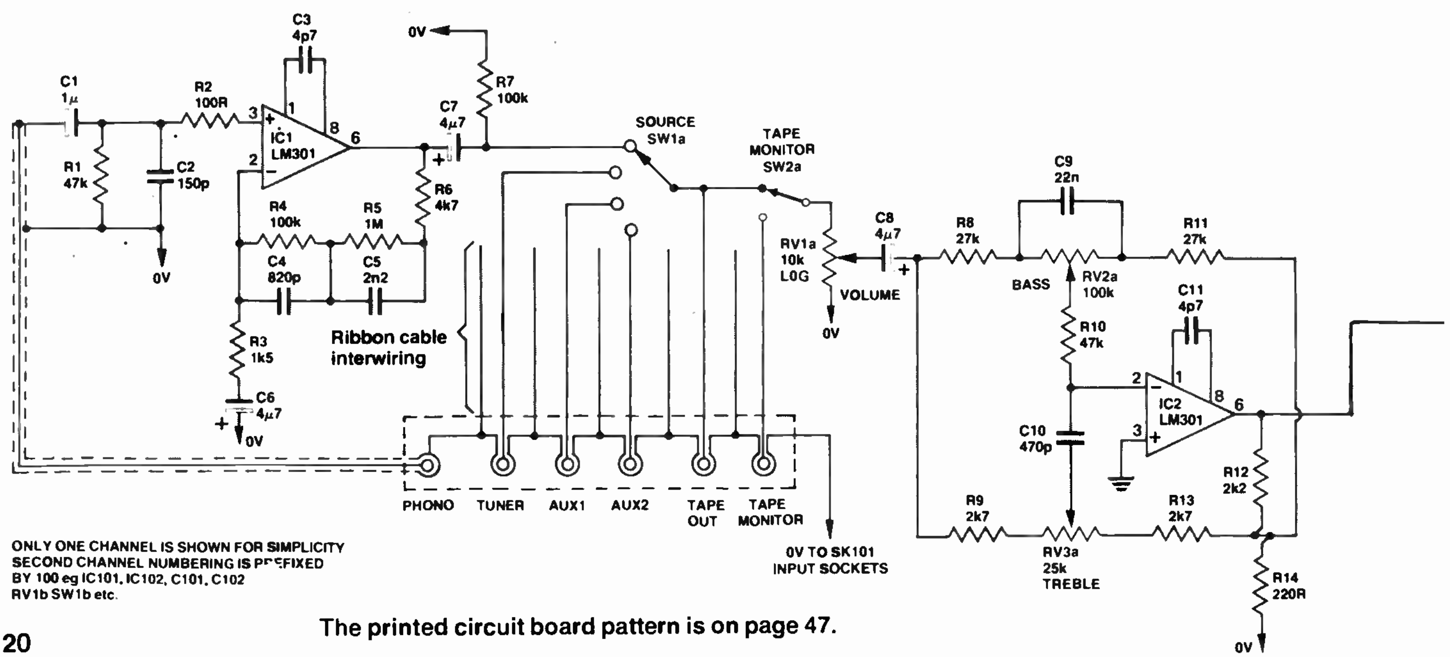
Output from the power stage is taken via a set of contacts on the anti-thump relay and a 2 A fuse, for speaker protection. The R-C network R33 and C17 provides output phase lage compensation.

The output stage devices — a TIP31C and complementary TIP32C — operate in pure class-B, the effects of crossover distortion being reduced by the feedback arrangements. These devices will deliver 25 W into an 8 ohm load. Only a modest heatsink is required as quiescent dissipation is low. The output devices are operated close to their SOAR limit under certain conditions, but no problems should arise. Lower power output can be arranged by reducing the supply rail voltage (see text).

POWER SUPPLY

The 240 Vac mains input is applied to the primary of the power transformer via the power switch, SW3, which isolates both active and neutral leads. A 'spike' suppression capacitor (C26) is connected on the mains input side of the power switch.

The secondary of the mains transformer consists of two windings connected in series, the 'centre tap' providing the 0 V return line. A bridge rectifier comprising diodes D4 to D7 provides positive and negative supply rails. The main supply rail voltages will depend on the transformer chosen for the desired power output as per the text.



The printed circuit board pattern is on page 47.

Smoothing for each supply rail is provided by C20 and C21. Capacitors C22 and C23 reduce the supply rail impedances at high frequencies. The phono and preamp-tone control stages require +/− 12 Vdc supply rails and these are derived by conventional shunt zener regulators involving R36, R37, ZD1 and ZD2. Capacitors C24 and C25 provide bypassing for these supply rails. LED2 is a 'power on' indicator.

ANTI-THUMP

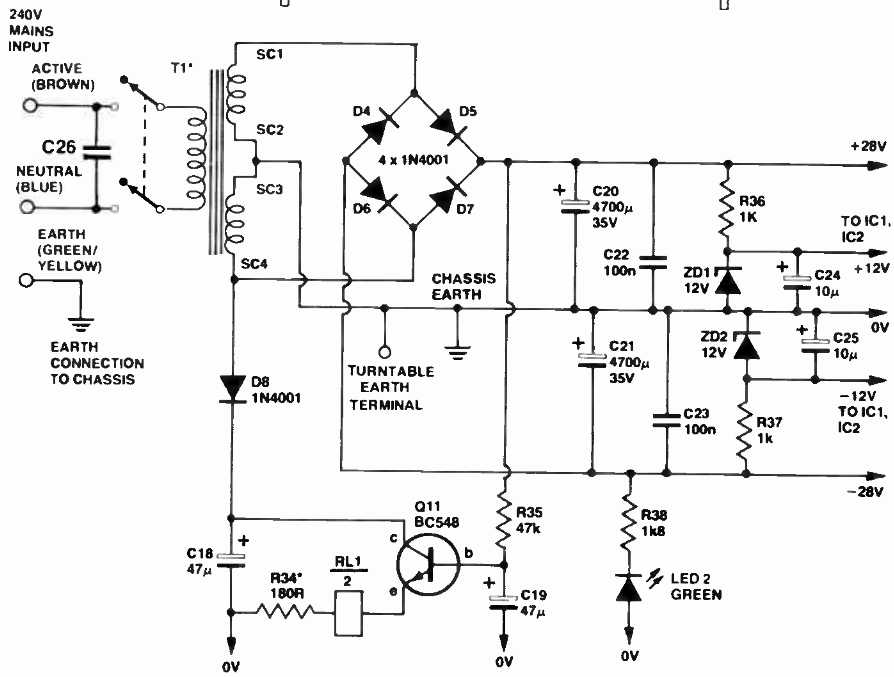
The anti-thump circuit involves D8, Q11, C18 and C19, R34 and R35 plus RL1. The object is to isolate the speakers from the power amplifier until the supply rails have stabilised following switch on and to isolate the speakers once again when the amplifier is turned off, before the supply rails have decayed.

The relay RL1 has two sets of contacts which are connected normally open. These contacts are in series with each channel's speaker output line, between the power output stage and the speaker protection fuse.

At switch on, D8 rectifies one side of the transformer secondary, rapidly charging C18, establishing collector supply for Q11 before the bridge rectifier smoothing capacitors have time to charge to a significant voltage. Until such time as C20 charges to 12 volts, Q11 has no base bias and is turned off. Thus, relay RL1 remains unoperated. As the voltage on C20 rises, capacitor C19 will charge via R35. When the voltage on C19 rises to about 12 V, Q11 will turn on and RL1 will operate, connecting the speakers. The R-C network formed by R35/C19 provides a time delay such that the voltage on C19 will only reach about 12 V after the voltage across C20 has risen to the full supply voltage. The delay is several seconds.

When the unit is switched off, C18 will rapidly discharge and the current through RL1 will drop below that required to hold it operated well before the supply smoothing capacitors (C20 and C21) discharge.

Resistor R34 limits the current through the relay to a safe maximum value when Q11 is on. Note that it is not required if the lower voltage transformer is used.

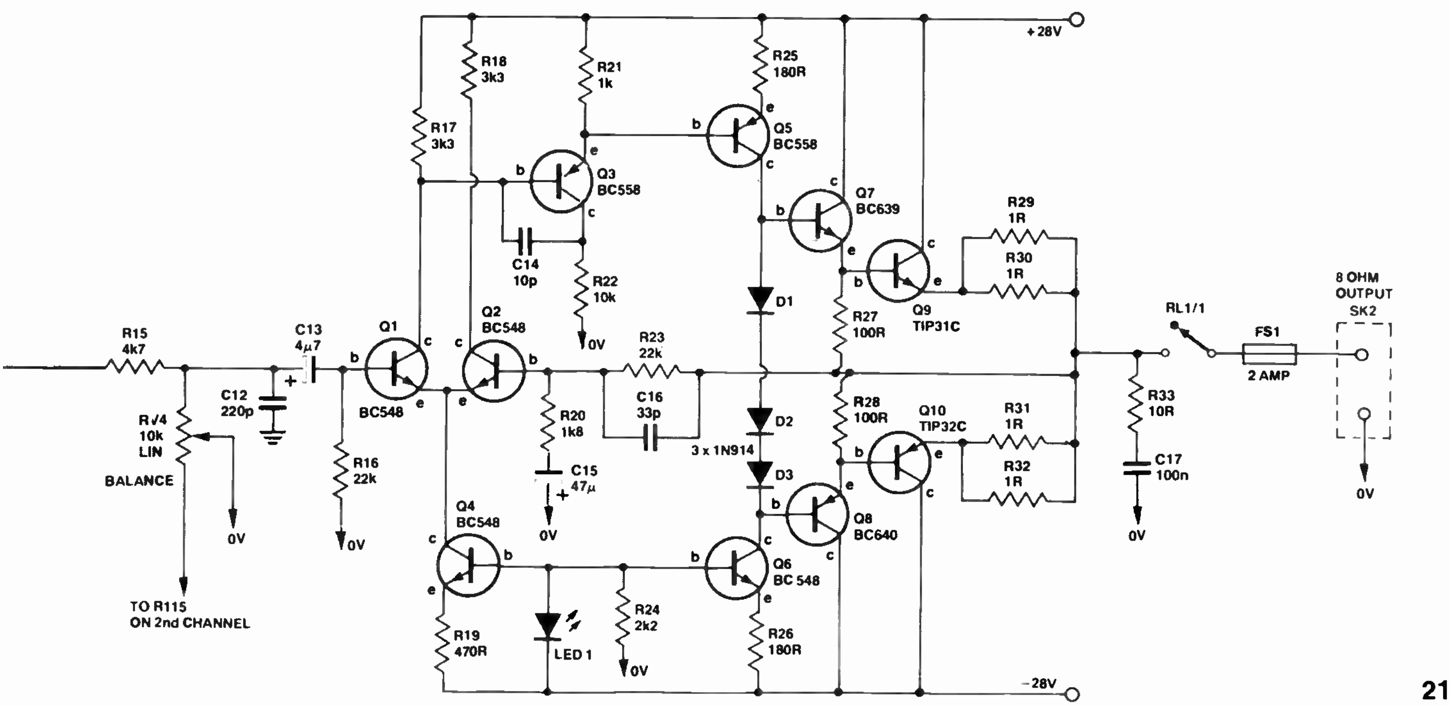


Power supply and speaker anti-thump circuit. A transformer with a secondary rated at 20 - 0 - 20 V at 1.5 A will provide 20 W per channel. For 5 W per channel operation, a 12 - o - 12 V at 0.8 A transformer is required. In the latter case, reduce the values of R36 and R37 to 390 ohms each and short out R34.

front panel and these can be obtained in plain aluminium or black anodised. We made up a black Scotchcal label with white lettering (i.e: 'reverse') for the front panel and used brushed satin aluminium knobs.The whole effect is quite attractive.

The layout of the controls on the front panel is kept quite simple. There are only two toggle switches — power and tape monitor. A stereo/mono switch was thought unnecessary as it would add to the cost and clutter up the simple front panel layout. They are rarely used these days in any case. The rear panel holds the input and output connectors, power cord and an earth terminal for other equipment such as a turntable, headamp or whatever.

We used an internally-mounted heatsink for the four output stage devices, made from a sheet of 16 gauge aluminium. This heatsink is the minimum recommended for 20W/channel opera-▶



tion. If you wish, the output transistors can be mounted on the rear panel, in the space above the speaker output terminals. Their leads may be connected to the pc board with hookup wire or ribbon cable in this case.

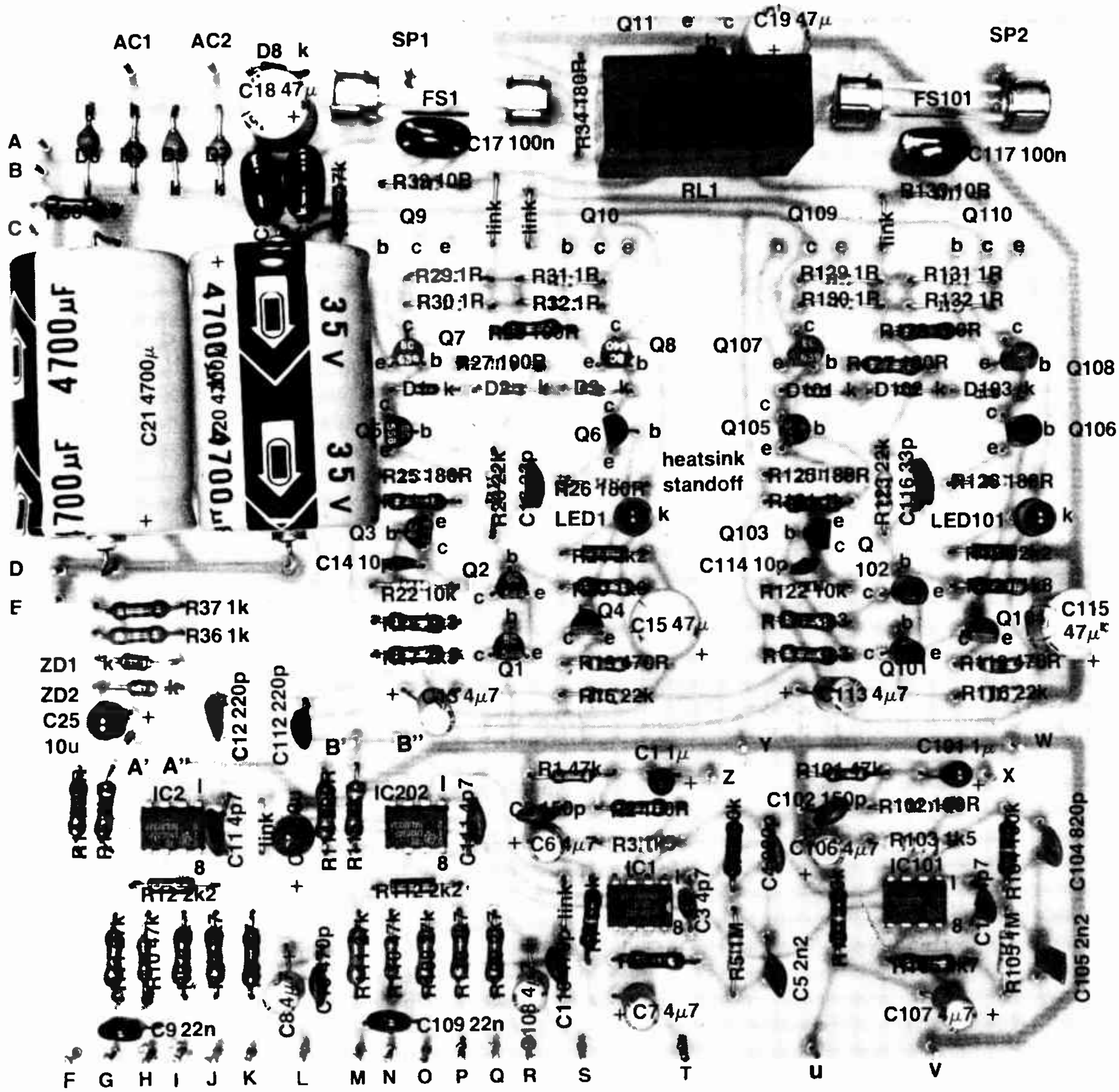
You will notice from the internal photographs that the phono input connection to the pc board is made with shielded cable, but the other inputs are made via a length of 20-wire ribbon or rainbow cable. This cable is wired in a signal-earth-signal-earth fashion so that each signal wire has an earth either side to provide some shielding. This we tried as an experiment and found it very successful. It simplifies the wiring enormously compared to using individual shielded cables. There was only a slight degradation in the cross-talk between channels and no increase in hum levels.

Construction is best commenced by loading the components on the pc board. The overlay photograph shows all the details. Insert the four link wires first. Note that a link should replace R34 if you intend using the lower voltage power transformer. This ensures that 12 V appears across the anti-thump relay coil at the lower voltage, ensuring correct operation.

Next mount all the resistors. Note that, if you're using the lower voltage power transformer, resistors R36 and R37 should be reduced to a value of 390 ohms each. Resistors R27 to R32 and R127 to R132 should be stood about 4 - 5 mm above the board to protect the pc board in the unfortunate event of a fault in the driver or output stages causing overheating of these resistors.

The capacitors can be mounted next. As usual, take care with the polarity of the electrolytic and tantalum capacitors. The lead length on the low value ceramic capacitors C3, C11, C14 and C103, C111, C114 should be kept as short as possible. Mount them so that the body of the component is right down on the pc board. The mains transient suppression capacitor is mounted off the pc board, but this is discussed later.

Now you can mount all the semiconductors, except the output transistors. Here too, take care with the orientation of the devices. The pinouts of the BD639s and BD640s do not have the base lead between the emitter and collector leads like most small signal transistors, so take care with these devices. Pay particular attention to the orientation of the diodes D1 to D3 and D101 to D103 as these set the voltages on the bases of the output transistors. If they are inserted the wrong way round



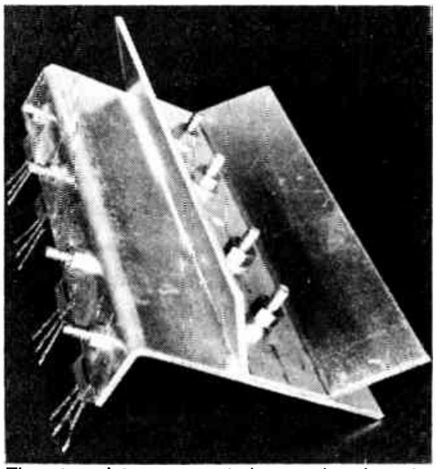
(or are open circuit for some reason) the output transistors won't last long, as we found out to our detriment! Apart from the disappointment, the smell is dreadful! If in doubt, use a multimeter to check the diode. Remember that the *positive* lead of an ohmmeter has the internal battery *negative* connected to it. Thus, this lead will be connected to the *cathode* of the diode when the ohmmeter indicates a low resistance (i.e: diode conducting).

Mount the relay, fuse clips and fuses next. Some fuse clips are hard to solder so to prevent overheating the pc board, first file the plating off the edge of the pins on each clip before you attempt to solder them in.

For all the external connections to the board we used pc board pins. They aren't essential, but they do make it considerably easier to wire the board to the other components. These pins are mounted at this stage.

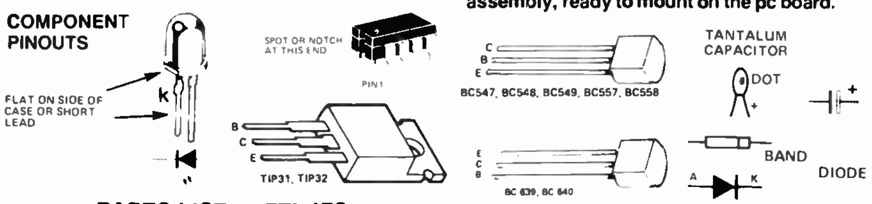
If you haven't already noticed, there are two pads on the board, just above IC2 and IC102, that appear to have no purpose. Circuit-wise, these pads are located at the input to the power stage of each channel and are marked A' and B' on the overlay photograph. By breaking the track between the points marked A' and A", and B' and B", the preamp output and power amp input can be separated to provide connections for "preamp out — main in" sockets so that equipment such as a graphic equaliser may be used in conjunction with this unit. For those readers including this provision, the output impedance of any equipment used to drive the power stages of this unit should be between 4k and 10k.

The output stage transistors are mounted on the board last of all. If you mount them as we have done, drill and bend up the aluminium heatsink first. The accompanying drawing gives all the details. The 'fin' mounts on top of the L piece and the two are held tightly together with four nuts and bolts. Smear thermal compound between the two mating surfaces and insert the bolts with their heads on the side that will face the pc board. Make sure that the surface on which the transistors mount is both smooth and flat. Smooth the surface with emery paper if necessary. By the way, don't paint the heatsink. We painted ours to experiment with colour photographs inside the completed unit and found it adversely affected the thermal capacity of the heatsink. ▶



The transistors mounted on the heatsink assembly, ready to mount on the pc board.

COMPONENT PINOUTS



PARTS LIST — ETI 476

Resistors all ½W, 5%

- R1, 10, 35, 101 .... 47k
- R110 .... 47k
- R2, 27, 28, 102 .... 100R
- R127, 128 .... 100R
- R3, 103 .... 1k5
- R4, 7, 104, 107 .... 100k
- R5, 105 .... 1M
- R6, 15, 106, 115 .... 4k7
- R8, 11, 108, 111 .... 27k
- R9, 13, 109, 113 .... 2k7
- R12, 24 .... 2k2
- R112, 124 .... 2k2
- R14, 114 .... 220R
- R16, 23, ... 22k
- R116, 123 .... 22k
- R17, 18 .... 3k3
- R117, 118 .... 3k3
- R19, 119 .... 470R
- R20, 120, 38 .... 1k8
- R21, 36, 37, 121 .... 1k
- R22, 122 .... 10k
- R25, 26, 125, 126 .. 180R
- R34 .... 180R see text
- R29 - 32 .... 1R
- R129 - 132 .... 1R
- R33, 133 .... 10R

Capacitors

- C1, 101 .... 1u electro or tantalum
- C2, 102 .... 150p ceramic
- C3, 103, 11, 111 .... 4p7 ceramic
- C4, 104 .... 820p ceramic
- C5, 105 .... 2n2 greencap
- C6, 7, 8, 13, 106 .... 4u7, 16V electrolytic
- C107, 108, 113 .... 4u7, 16V electrolytic
- C9, 109 .... 22n greencap
- C10, 110 .... 470p ceramic
- C12, 112 .... 220p ceramic
- C14, 114 .... 10p ceramic
- C15, 115 .... 47u, 16V electrolytic
- C16, 116 .... 33p ceramic
- C17, 22, 23, 117 .... 100n greencap
- C122, 123 .... 100n greencap
- C18, 19 .... 47u, 35V electrolytic
- C20, C21 .... 4700u, 35V electrolytic
- C24, 25 .... 10u, 16V tantalum
- C26 .... 10n to 100n, 240VAC Rated capacitor (value not critical)

Semiconductors

- IC1, 101 .... LM301
- IC2, 102 .... LM301
- Q1, 2, 4, 6, 11 .... BC548, BC108
- Q101, 102, 104, 106 BC548, BC108
- Q3, 5, 103, 105 .... BC558, BC178
- Q7, Q107 .... BC639
- Q8, 108 .... BC640
- Q9, 109 .... TIP31C
- Q10, 110 .... TIP32C
- D1 - D3, D8 .... 1N914, 1N4148
- D101 - 103 .... 1N914, 1N4148
- D4, 5, 6, 7 .... 1N4001, A14A or sim.
- ZD1, 2 .... 12V, 400mW zener diode
- LED1, 101 .... red LED, TIL220R or sim.
- LED2 .... green LED TIL220G or sim.

Potentiometers

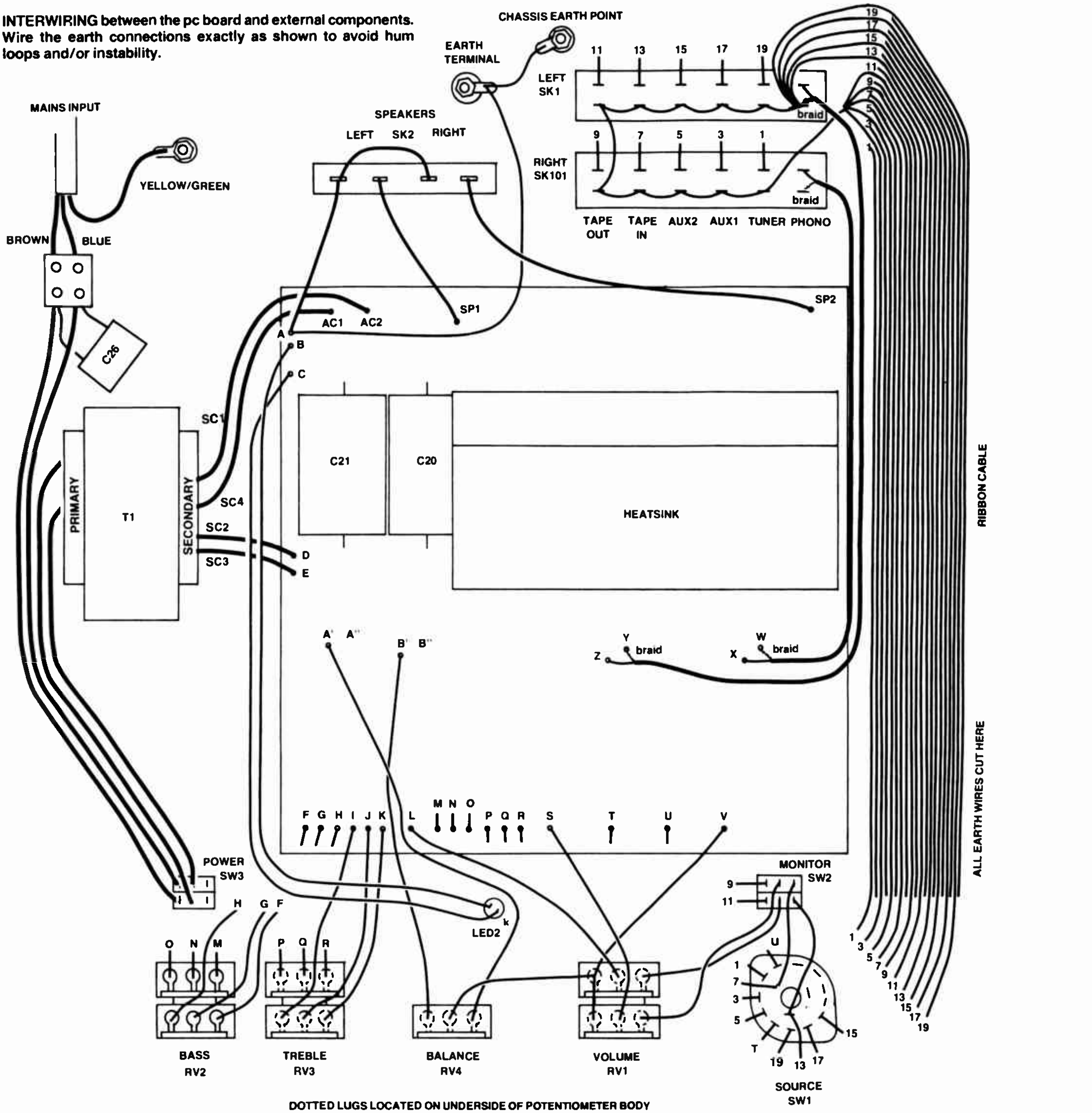
- RV1 .... 10k dual gang log.
- RV2 .... 100k dual gang lin.
- RV3 .... 25k dual gang lin.
- RV4 .... 10k single gang lin.

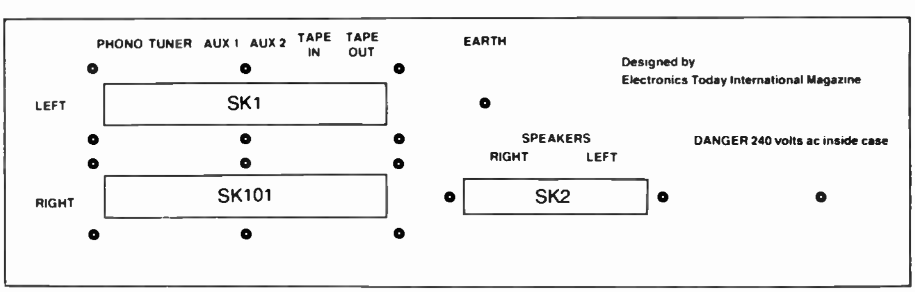
Miscellaneous

- SW1 .... two pole, four position wafer switch
- SW2 .... DPDT miniature toggle switch
- SW3 .... DPST 240V rated miniature toggle switch
- SK1, 101 .... six RCA sockets on insulated panel
- SK2 .... set of spring contact speaker terminals on insulated panel
- T1 .... For 25 Watts output 20 - 0 - 20V secondary at 1.5 amps Ferguson type PF3993 or similar For 5 Watts output 12 - 0 - 12V secondary at 800mA Ferguson type PL24/20VA or sim.
- FS1, 101 .... 2 Amp, 3AG fuses with pc board mounting clips
- RL1 .... DPDT pc board mounting relay with 12 Volt coil, Takamisawa type VB 12 STAN or Pye 265/12/G2V

ETI 476 pc board, heatsink (see text), black screw terminal for turntable earth connection, Horwood case type 93/10/V (255 mm wide x 255 mm deep x 76 mm high), power cord and 3 pin plug, four 12 mm pc board standoffs, clamping grommet for power cord, nuts, bolts, length 20 wire ribbon cable, length shielded cable.

INTERWIRING between the pc board and external components. Wire the earth connections exactly as shown to avoid hum loops and/or instability.





◀ REAR PANEL

SW1	POSITION FUNCTION	LEFT	RIGHT
1	PHONO	U	T
2	TUNER	1	19
3	AUX 1	3	17
4	AUX 2	5	15
—	TAPE OUT	9	11
—	TAPE IN	7	13

Source and monitor tape switches wiring legend.

Mount the output transistors as shown in the accompanying diagram. Note that a mica or plastic washer and insulating bush must be used with each one. Smear a small amount of thermal compound on both sides of the mica or plastic washer before you mount and screw the bolts down firmly to get good thermal contact between the transistor body and the heatsink.

Check that you have the transistors in the correct order and check also that there are no shorts between the collector pins and the heatsink.

Before the heatsink/transistors assembly can be mounted, bolt a 25 mm standoff pillar on the pc board as indicated on the overlay photograph. This is to secure the heatsink and relieve stress on the transistor leads.

Once you have the heatsink and output transistors all together, bring the assembly to the pc board and insert the leads of the transistors in the appropriate holes. A small pair of needle-nose pliers helps here. Push the transistor leads through the board such that they protrude about 2mm on the copper side of the board. Bolt the heatsink to the standoff and then solder the transistor leads.

That completes the pc board assembly.

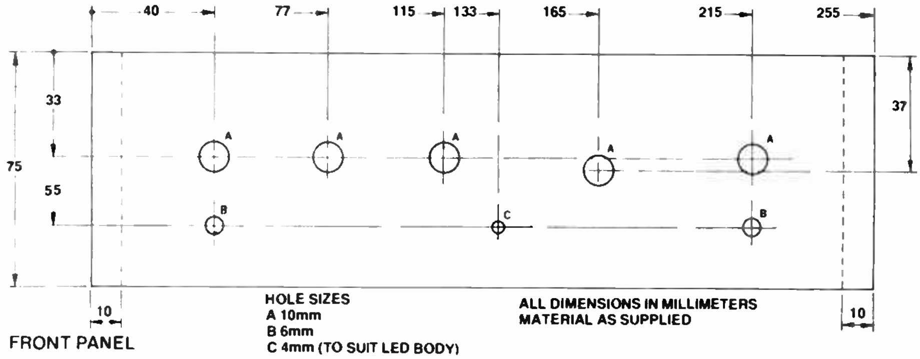
Next step is the metalwork. If you have purchased a pre-drilled case, just check that all fits and there are no burrs. If you've purchased an undrilled case, first thing to do is to disassemble it and scribe the hole positions on the front and rear panels as per the metalwork drawings. If you're going to use a Scotchcal front panel, this may be used as a template for drilling with the added assurance that all will fit when the holes are drilled. Don't take the backing off the Scotchcal at this stage as once it is stuck down to anything, you can't remove it without damaging thc panel.

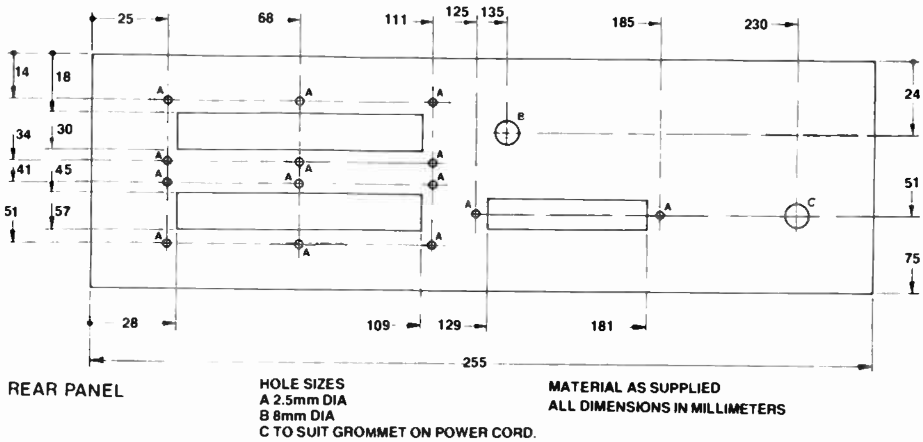
Carefully drill and de-burr all the holes. Slots are cut in the rear panel to accommodate the strips of RCA input/output connectors and the speaker connector strip. These can be cut using a 'nibbling' tool, which requires a single hole to be drilled, or by drilling a succession of holes and filing the edges of the slot straight.

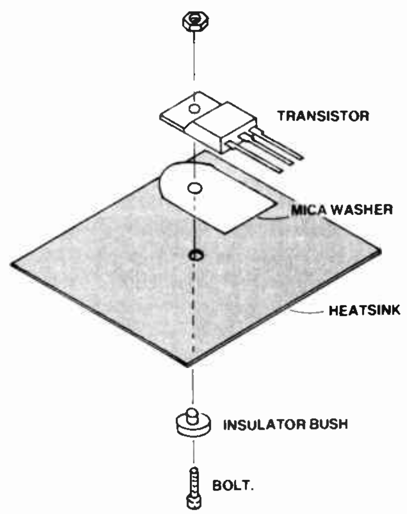
With the front panel drilling completed, the Scotchcal can be positioned and stuck down. Take care, making sure that it is properly aligned. With the Scotchcal in position, mount the front panel components, noting the potentiometers are slightly angled to ensure the centre of their range corresponds to the centre of the knob pointer. The bass control faces upside down to the other controls so that its terminals face away from the power switch. Note that the power and tape monitor toggle switches are sideways operating.

Mount the rear panel components next, making sure there are no shorts between the terminals and the chassis.

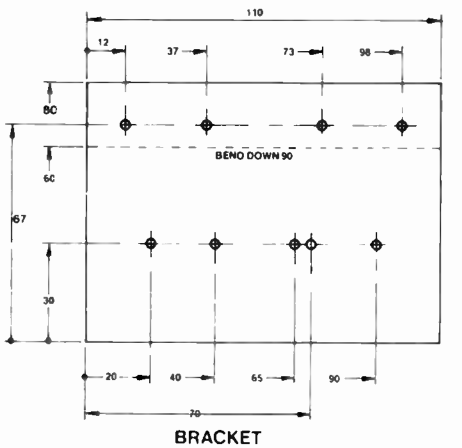
The bottom panel of the case needs to be drilled to suit the transformer mounting holes, mains terminal block ▶

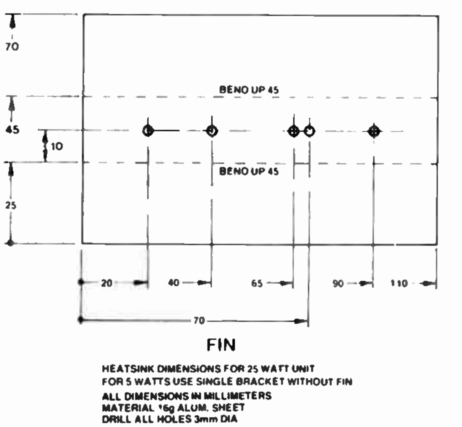






Assembling a flatpack transistor to a heatsink.





and the pc board standoff mounts. With this done, assemble the case leaving the top panel off for the moment. Note that the handles mount over the Scotchcal on the front panel, and that the bottom and top plates of the case overlap the front panel, protecting its edge and improving the overall appearance.

The bottom panel is normally held in place by four self-tapping screws but, owing to the weight of the transformer, we used four small nuts and bolts.

To protect the surface of whatever the equipment may stand on, we attached four 'stick-on' rubber feet to the bottom panel of the case.

Mount the transformer and pc board as shown in the internal photograph. The transformer is mounted in the rear left hand corner, furthest away from the phono stages which are sensitive to hum pickup from its field. The pc board is held in place by four 12 mm plastic push-in standoff mounts.

Now all the interconnecting wiring can be completed. The wiring diagram shows how this is done. Best place to start is with the ribbon cable between the function switch and the input/output RCA sockets. We also used ribbon cable to wire the potentiometers as it makes quite a neat job.

The high level inputs are connected to the source switch via the ribbon cable in which each second wire is connected to earth at the input sockets. The earth wires are then cut off neatly near the source switch. A bit of care in installing the ribbon cable will make a big difference in appearance if not performance.

The two phono inputs are connected to their respective input sockets with shielded cable. Both shields are connected to the pc board and their input socket. One of the shields is then extended to the earths on the high level inputs, but not to the other phono input.

Wire the earth wiring *exactly* as shown in the wiring diagram. The earth for the speakers is returned to the *power supply* end of the pc board and the earth for the inputs returns down one of the phono lead shields. The only connection to the chassis is made near the earth terminal on the rear panel, and this is returned to the earth connection adjacent to the power supply section on the pc board.

Take care with the mains wiring. We used a clamp grommet to secure the cable where it enters the case and the wires are terminated at a two-way terminal block. The earth lead (green/yellow) should be longer than the other two and secured to a solder lug mounted under a nut and bolt. The ac noise suppression capacitor mounts on this terminal block, on the side where the wiring leads to the power switch on the front panel. If your transformer comes with terminals, the suppression capacitor can be mounted there. All the 240 Vac wiring is passed along the left hand side of the chassis. Use spaghetti sleeving or heatshrink tubing on the power switch wiring to protect the exposed connections.

That just about finishes it. The only thing left to do is to carefully check your wiring and then give the unit a test run.

With no inputs and no speakers connected, turn the unit on. The LED on the front panel should light as should the two LEDs on the pc board. With a sensitive multimeter, check the voltage on the output terminals of each channel. You should read no more than 100 mV. Because the output stage operates in class-B there is no bias adjustment.

If all is well, turn the unit off, connect loudspeakers and a turntable or tape deck and you're ready to rock!

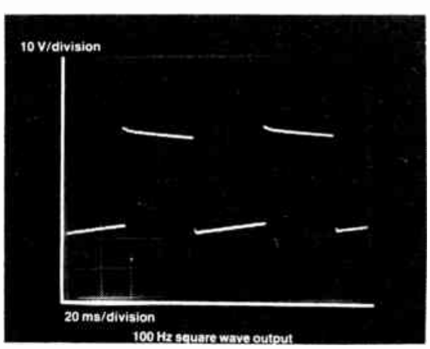
Trials

We gave the unit a thorough trial, running it into 'odd' loads etc, running it heavily into clipping and conducting extensive listening tests. The accompanying photographs taken from the oscilloscope in our lab show how the amplifier performs with square wave drive at different frequencies throughout the audio range plus one shot showing the amplifier's output when driven into clipping with a sine wave input. Performance of the unit is clearly very good.

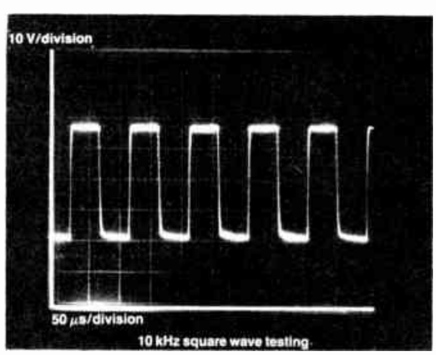
Just to convince ourselves, and you skeptics amongst the readers, we have also included a photograph taken from the screen of a Hewlett-Packard model 3580A spectrum analyser on loan from Tech-Rentals for some development work. As you can see, this amplifier has quite a creditable performance.

For listening tests we used a pair of our Series 4000/2 three-way loudspeakers in an average sort of domestic environment with a Sansui turntable and Shure M91 cartridge. Overall sound is very clean, with well defined bass and crisp top end and it was obvious that any sonic 'faults' were not contributed by the amplifier. The unit drove the Series 4000/2 speakers effortlessly to levels liable to raise neighbour complaints!

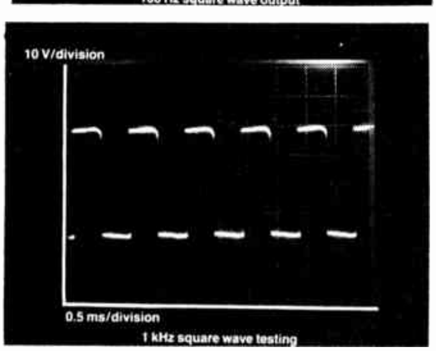
We think you'll be as pleased with the Series 3000 Compact Stereo Amplifier as we are. ●



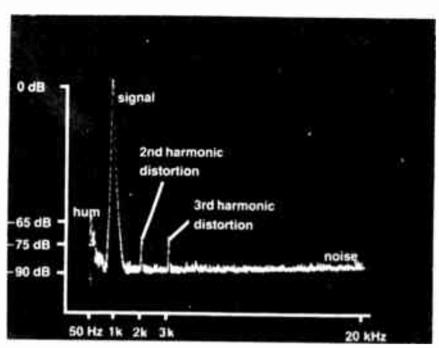
100 Hz square wave output

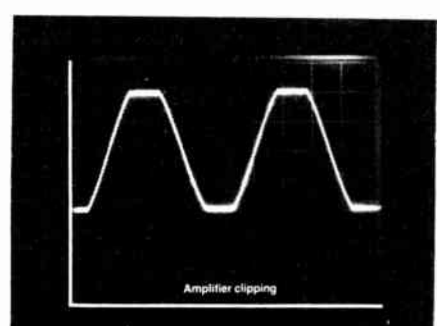


10 kHz square wave testing



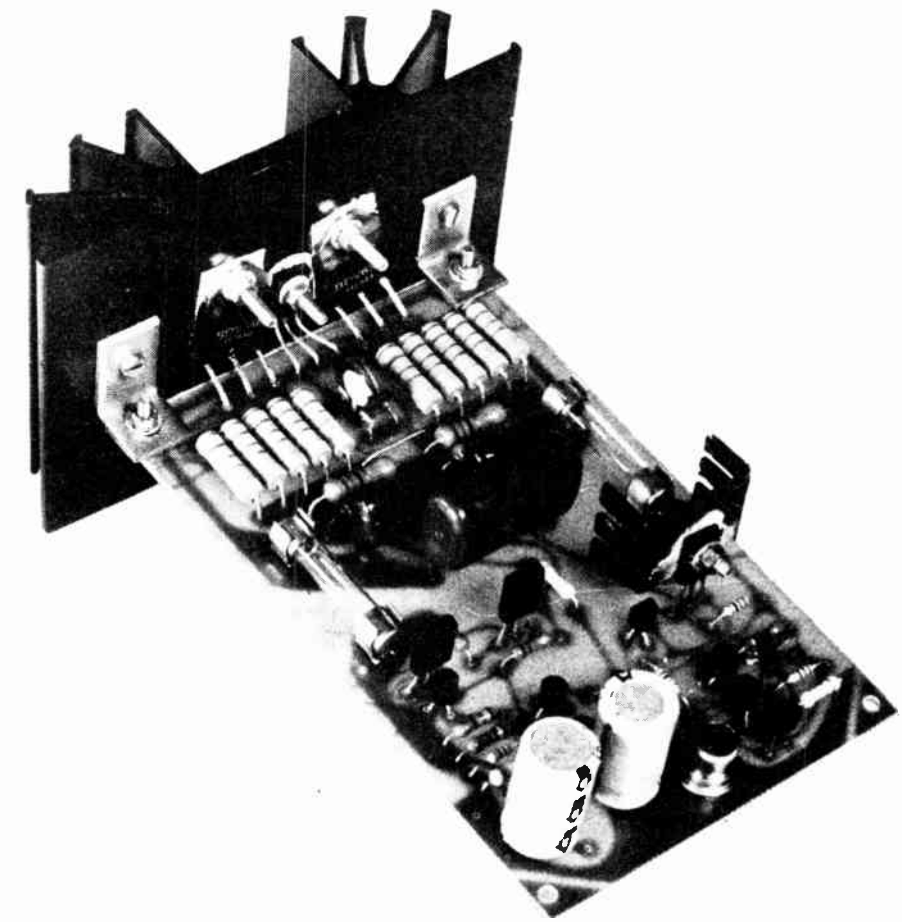
1 kHz square wave testing




Amplifier clipping

SIMPLE 60W LOW DISTORTION AMPLIFIER MODULE

Here is a simple, low distortion amplifier module that has been used as the basis of a quality do-it-yourself hi-fi amplifier system. This project employs simple mechanical construction and readily available components, yet gives high quality sound.



MANY DIFFERENT amplifier circuits have appeared in popular electronics magazines over the years. The most popular audio projects we have ever published were the 100 watt guitar amp. (ETI 413, published in December 72 and still going strong!), the 422 amp. and the 480 series of power amp. modules.

While these seemed to have satisfied a large demand, our attention has been drawn to the need for something a 'step up' from there — something that approaches the current 'state of the art' for hi-fi equipment. Lower distortion than previously obtained, better bass performance and flexibility was the message we received from reader's letters and kit and component suppliers ("Why don't you . . .", "What I'd like to see . . .", "I need a . . .", etc.).

Satisfying as many requirements as possible took considerable thought and planning. Our major design hurdles were cleared with room to spare with the assistance of talented West Australian designer, Trevor Marshall.

A great many factors place sometimes quite severe constraints on project design — particularly component availability and ease of construction; not forgetting that this design had to perform significantly better than those that came before it.

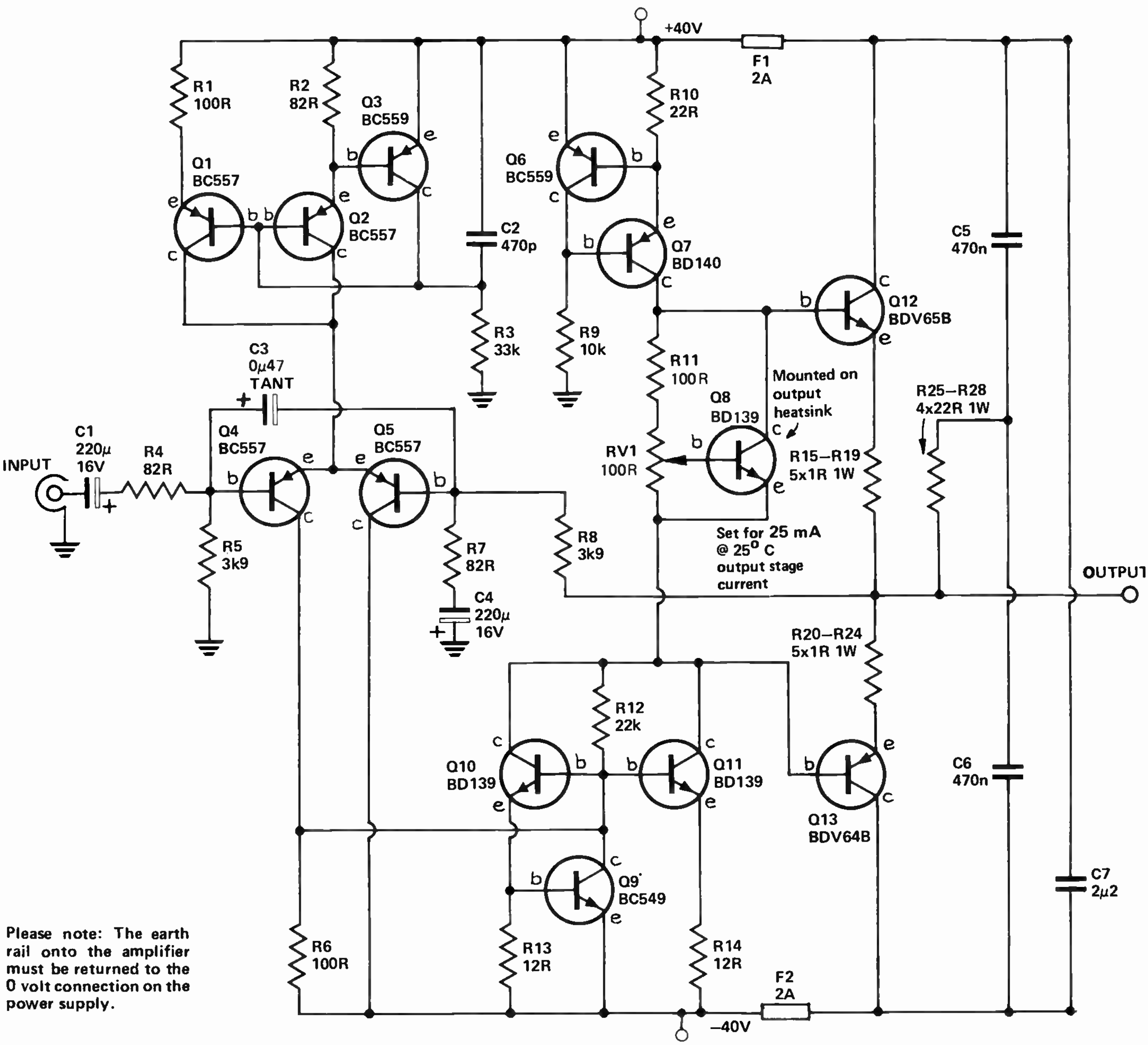
There is clearly little point in describing a project that includes components that are impossible to get or one that is difficult to construct.

A strong point that came across to us from reader feedback and from the popularity of our 480 series of amplifiers was that constructors favoured a modular concept. It seems that the days of the single-board stereo amplifier project have come and gone.

This power amplifier offers a significant improvement in specifications and ease of construction over most kit amplifiers offered to date. It has been designed particularly with low transient intermodulation distortion in mind.

Although a difficult parameter to measure, transient intermodulation distortion is an inherent characteristic of many amplifier designs — especially those which incorporate large amounts of feedback to even out frequency response and reduce harmonic distortion. The heavy feedback 'school' of design produces an impressive list of specifications — but the difference *to the ear* between such an amplifier and one designed for low TID has to be heard to be believed.

60W AMPLIFIER MODULE



Please note: The earth rail onto the amplifier must be returned to the 0 volt connection on the power supply.

HOW IT WORKS – ETI 470

The input stage of the amplifier consists of an emitter coupled differential pair (Q4, Q5) with a constant current source (Q1, Q2 and Q3). The use of a constant current source reduces distortion, as well as the possibility of high frequency oscillation and prevents any ripple on the positive supply from unduly affecting the input stage. Unequal emitter resistors (R1, R2) allow the currents in Q4 and Q5 to be optimised. Input lag compensation is provided by C3, limiting the slew rate of the amplifier to reduce high frequency intermodulation. The gain of the differential pair, driving Q10 and Q11, is very low.

Almost all the gain of the amplifier is obtained from the parallel pair Q10 and Q11. They are operated with series (R13, R14) and shunt (R12) feedback, and a constant current source (Q6, Q7). This results in a highly linear stage.

Q9 protects Q10 and Q11 from high peak currents or damage should a fault occur. When the current through R13 exceeds the safe limit, Q9 conducts and shorts out the drive to Q10 and Q11.

Bias from the output stage is set by RV1 and a shunt regulator (Q8). Q8 is mounted on the same heatsink as the output stages and stabilises the output bias current against heatsink temperature rise. Resistors R15-R24 in the emitters of the output Darlingtons, Q12 and Q13, maintain operation in their safe region as well as reducing the chance of thermal run away.

Protection against ultrasonic oscillation is provided by C7 and the network consisting of R25-R28 and C5, C6.

Both DC and AC feedback is taken from the output, via R8, to the negative input of the differential pair, the amount of feedback being set by the ratio of R8 to R7. C4 increases the feedback, and therefore decreases the overall gain, at very low frequencies. The feedback also automatically holds the DC output voltage at close to zero volts.

Choice of Power Supply

The design of the power supply can mean the success or failure of an otherwise well-designed amplifier. The supply voltage should be well-regulated, varying less than 10% from no load to full load, and be able to supply high peak currents.

However, if a voltage regulator is employed it too must be capable of delivering the very high peak currents occasionally demanded. This necessitates an expensive regulator device and large, expensive filter capacitors.

The alternative is to use a fairly large transformer and large value filter capacitors on a capacitor-input bridge rectifier. This is what we chose.

The circuit given here shows a power supply suitable for supplying a stereo amplifier using two of these modules. The filter capacitors C8 and C9 consist of two 2500 uF, 50 volt electrolytic capacitors connected in parallel. This is the minimum we would recommend.

In general, the largest value filter capacitor one can afford is a good rule of thumb! *It has been suggested to us that values as high as 20 000 to 50 000 uF makes an audible difference in performance.* (Watch the rectifier specifications though!).

Improved performance can be obtained for a modest increase in cost by having a separate supply for each channel module. This improves the regulation, reduces crosstalk and increases the amount of power available before output clipping commences.

The choice of transformer will determine power output. A 28-0-28 volt, 2 A transformer (Ferguson PF3577 or similar) will power a module to 60 watts (RMS) power output, while a 26-0-26 volt, 2 A type (e.g. Dick Smith M-0148 C-core) will permit 40 watts.

The power supply output should be limited to a peak DC voltage of about 40 volts (for 60 W output). A C-core transformer will generally improve the hum and noise output figures apart from having a reduced field, thereby reducing possible hum pickup problems.

If the amplifier module is to be used with a 4-ohm speaker system the supply voltage must be limited to about 30 volts maximum, otherwise the output devices will attempt to deliver over 100 watts followed by rapid self destruction!

Adventurous constructors may wish to try adding a second set of Darlington output devices, with their own emitter resistors as per the circuit, connected in parallel with the original pair. This combination may supply 100 watts or more into a four ohm speaker load. This technique is also recommended if you are contemplating driving highly

ETI 470 SPECIFICATIONS

Power Output . . . . 60 watts into 8 ohms (∓40V supply)

Frequency Response . . . . 10 Hz to 100 kHz ∓0.5 dB

Input Sensitivity . . . . 500 mV rms for 60 W output

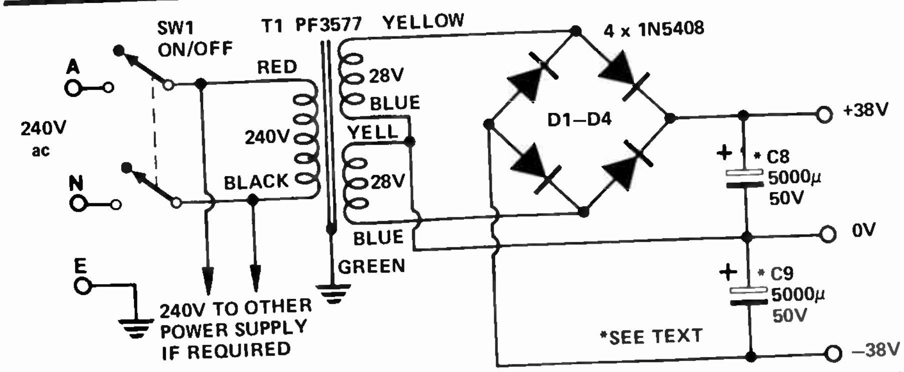
Hum and Noise . . . . better than –110 dB on full output (dependent on power supply)

Feedback Ratio . . . . 35 dB

Distortion . . . . at 1 kHz, 30 V p-p output into 8 ohms,
Closed Loop . . . . 0.04 %
(open loop 1 %)

Stability: The amplifier was found to be completely stable when operated into reactive loads consisting of R + C, L + C and pure L

Intermodulation (calculated values) . . at 1kHz, 30 V p-p output into 8 ohms,
3rd order . . . . less than 0.015 %
5th order . . . . less than 0.0023 %
(Intermodulation reduces with reduced power)



WHY LOW TID?

Looking at the circuit and a quick glance at the specifications, there's little in the circuit that looks outstandingly different from others. So what makes this amplifier special?

The difference in concept that makes this amplifier unique is the use of a very linear, high gain driver stage (Q10, Q11), with a constant current source (Q6, Q7), so that the gain of *this* stage is dependent upon the input impedance of the output transistors. However, *their* input impedance is dependent upon their gain, and therefore *the gain of the amplifier stage is dependent solely upon the characteristics of the output devices.*

Series and shunt feedback is used with Q10 and Q11 which results in a highly linear stage with a very low input impedance (about 28 ohms). The gain of the differential pair when fed into this low impedance is close to unity, so almost all the gain of the amplifier is concentrated in Q10 and Q11.

Provided the phase shifts in the differential pair and the gain stage are negligible the feedback loop is unconditionally stable.

There are two other design features which result in low TID.

The total open loop (feedback disconnected) distortion is only 1% at 30 V p-p output. So, very little feedback is necessary to reduce this to an acceptable level.

Protection of the output transistors is done by fuses, rather than electronically, and very high transient currents can be fed to the speaker without being affected by the (inevitably) non-linear impedance of an electronic protection circuit.

reactive loads such as electrostatic loudspeakers.

Construction

All components are mounted on a pc board — including the output devices. This method of construction is recommended. The module has been designed so that it is mechanically simple to assemble, much simpler than our ETI 480 module. Wiring errors are also avoided when a pc board is used.

Firstly, assemble and solder all the components on to the printed circuit board with the exception of Q12, Q13 (the output Darlingtons) and Q8. Carefully observe the polarity of all the electrolytic capacitors and orientation of the transistors.

The board is then mounted hard against the heatsink using small right-angle brackets. Be careful to avoid shorting the ends of the one ohm emitter resistors, R15-19 and R20-24, to the brackets.

If the module is to be mounted in a chassis the bottom (copper) side of the pc board should be 25 mm above the bottom of the heatsink. This will allow the use of 25 mm spacers to support the 'input' end of the board (furthest from the heatsink). It is expected that kits will include pre-drilled heatsinks and suitable brackets.

Once the board is attached to the heatsink the output Darlingtons, Q12 and 13, and Q8 may be mounted. Insert them in the pc board and then press them back against the heatsink to form their leads to the right shape. Do not solder their leads yet.

Smear heat conducting compound on either side of the mica insulators (don't use too much though) and insert these between the devices and the heatsink.

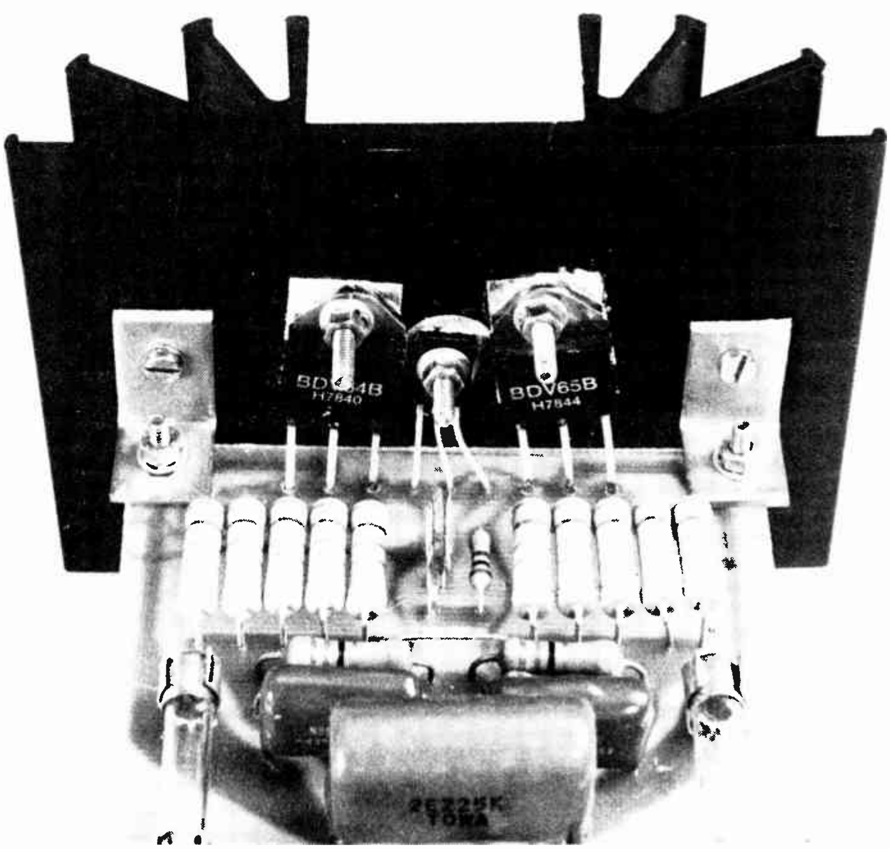
Assemble the washers and mounting bolts for these, finally checking with an ohm meter that there is not a short circuit between the metal tags (collectors) of the devices and the heatsink.

The input connection to the module is via a single-hole mounting RCA socket. This is mounted directly on the pc board. The centre pin connects to C1 via a short length of tinned copper wire.

If this facility is not required the RCA socket may be omitted and a length of shielded cable soldered directly between C1 and the pc board common.

The power supply and speaker connections are soldered directly to the appropriate copper lands on the underside of the pc board.

The 'earthy' side of the speaker must be returned directly to the zero volt connection of the power supply, as close to the filter capacitors as possible (preferably direct to the negative terminal). Do not connect this side of the speaker to the amplifier board.



Left: closeup view of the output stage showing how the Darlington transistors are mounted and how the pc board attaches to the heatsink

PARTS LIST - ETI 470

Resistors all ¼W, 5%, except R15-R28

R1	100R
R2	82R
R3	33k
R4	82R
R5	3k9
R6	100R
R7	82R
R8	3k9
R9	10k
R10	22R
R11	100R
R12	22k
R13, 14	12R
R15–R24	1R 1 watt
R25–R28	22R 1 watt

Potentiometer

RV1	100R mini trimpot (vertical)

Capacitors

C1	220µ 16V electro
C2	470p ceramic
C3	0µ47 35V tant
C4	220µ 16V electro
C5, 6	470n greencap
C7	2µ2 greencap

Semiconductors

Q1, 2	BC557, DS557
Q3	BC559, DS559
Q4, 5	BC557, DS557
Q6	BC559, DS559
Q7	BD140
Q8	BD139
Q9	BC549, DS549
Q10, 11	BD139
Q12	BDV65B
Q13	BDV64B

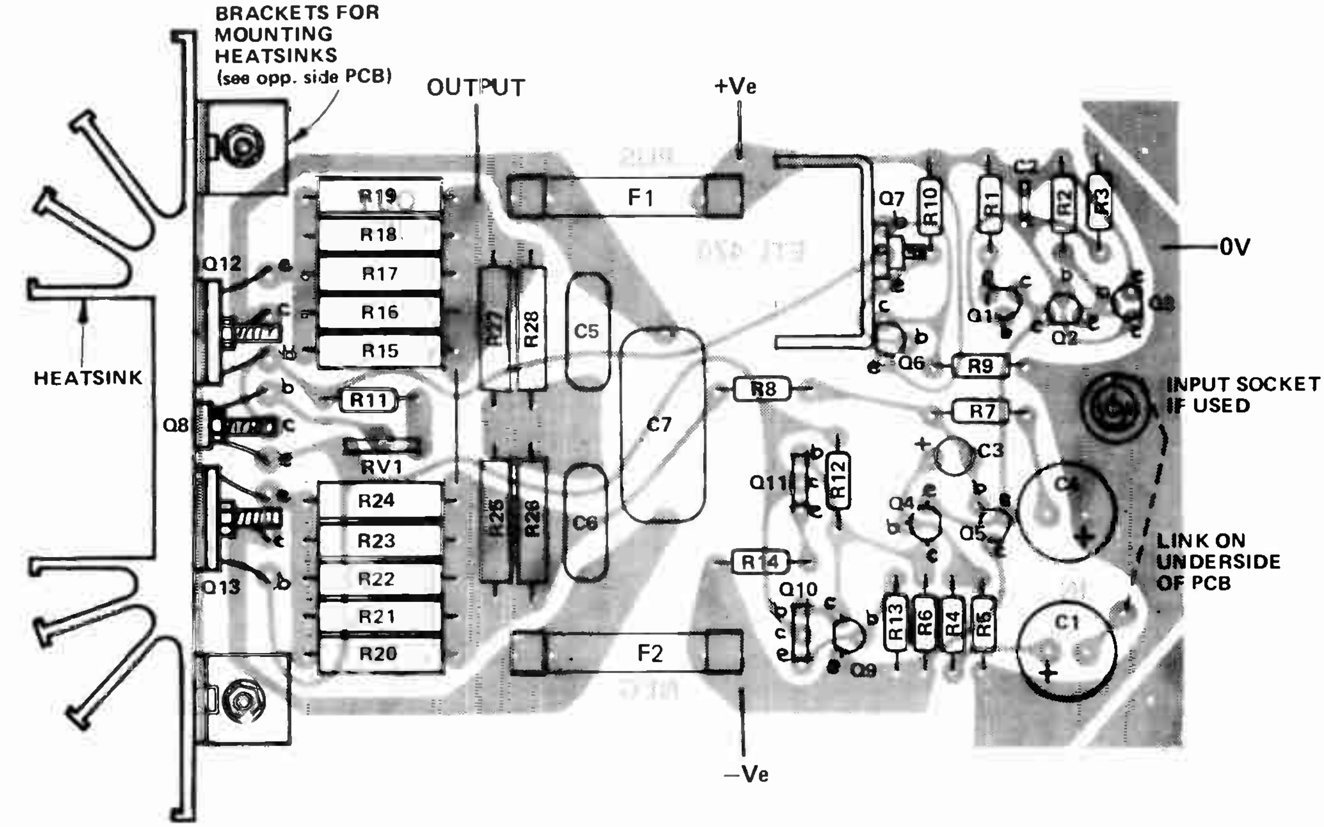
Miscellaneous

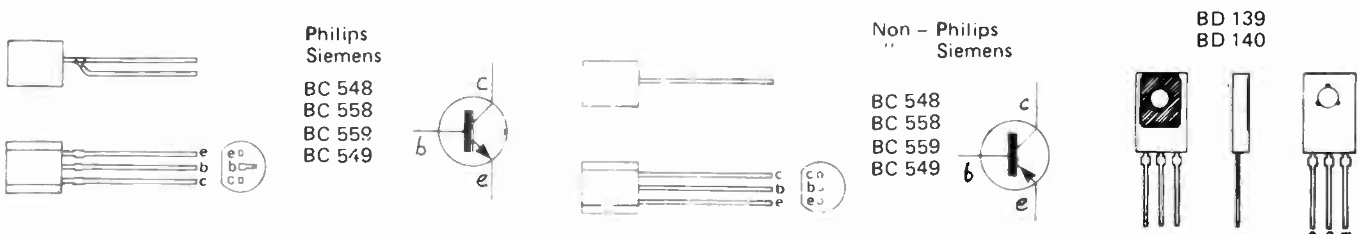
SK1	single hole, panel mounting RCA socket.
F1, F2	2 Amp 3AG Fuses.

Fuse holders, heatsink for Q7, mica insulating kits (for Q8, Q12 and Q13), flat sided heatsink (75mm x 110mm), angle brackets, ETI 470 pcb.

Parts List for Power Supply

D1–D4	IN5404 or sim
C8, 9	5000µ 50V electro (see text)
SW1	240V DPDT switch
T1	28V-0V-28V, 2 amp transformer Ferguson type PF3577 or similar (see text)


BRACKETS FOR MOUNTING HEATSINKS (see opp. side PCB)
OUTPUT
+Ve
HEATSINK
Q12
Q8
Q13
R19
R18
R17
R16
R15
R11
RV1
R24
R23
R22
R21
R20
F1
F2
R27
R28
R25
R26
C5
C6
C7
R8
R14
Q7
R10
R1
R2
R3
0V
Q1
Q2
Q3
Q6
R9
R7
INPUT SOCKET IF USED
C3
C4
Q11
R12
Q4
Q5
LINK ON UNDERSIDE OF PCB
Q10
Q9
R13
R6
R4
R5
C1
−Ve


Philips
Siemens
BC 548
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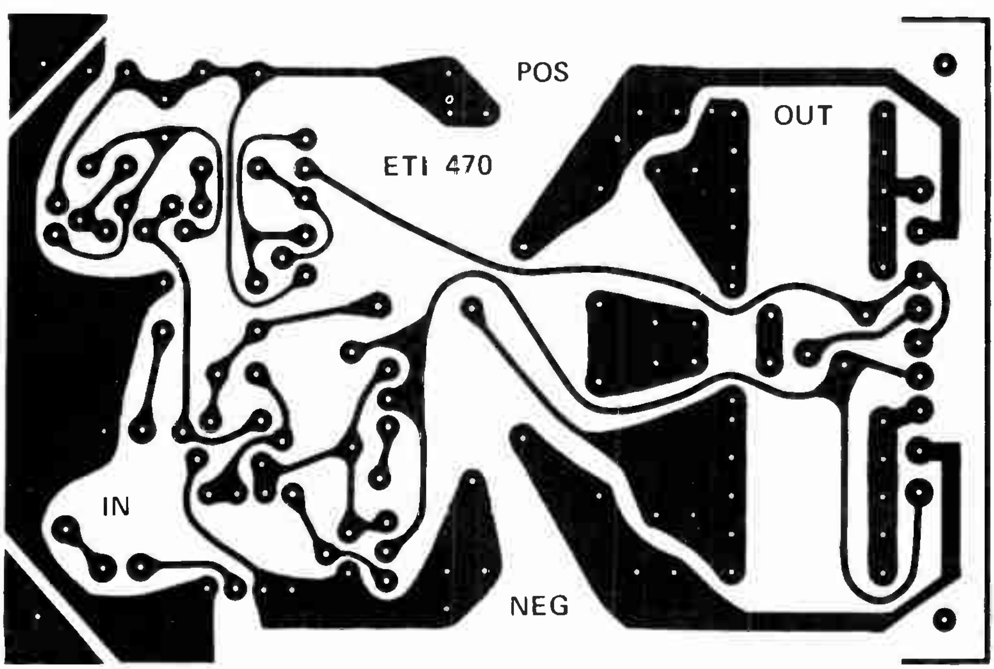
BD 139
BD 140


POS
ETI 470
OUT
IN
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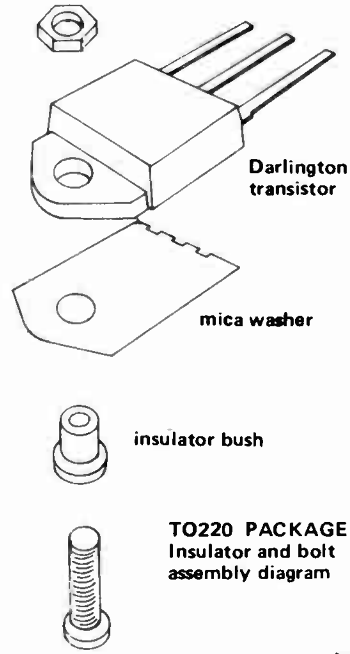

Darlington transistor
mica washer
insulator bush
TO220 PACKAGE
Insulator and bolt assembly diagram

60W AMPLIFIER MODULE

Components

The Darlington output transistors are the only 'special' components, all others are generally available from kit and component suppliers.

The Darlington transistors should be available through the following stores:-

Jaycar
125 York St
Sydney NSW 2000

Electronic Agencies
115-117 Parramatta Rd
Concord NSW 2137
and
117 York St
Sydney NSW 2000

Dick Smith Electronics
P.O. Box 321
North Ryde NSW 2113
(stores everywhere)

Rod Irving Electronics
425 High St
Northcote Vic 3070

All Electronic Components
118 Lonsdale St
Melbourne Vic 3000

Radio Despatch Service
869 George St
Sydney NSW 2000

Heatsinks

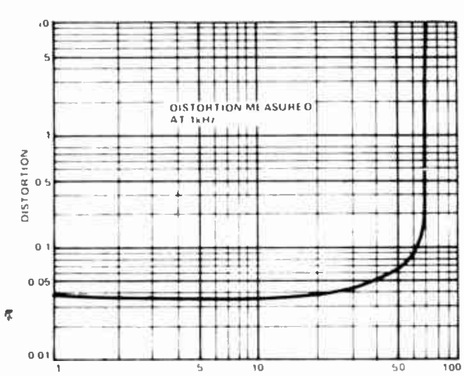
Heatsinks on any amplifier are a compromise between cost and temperature rise.

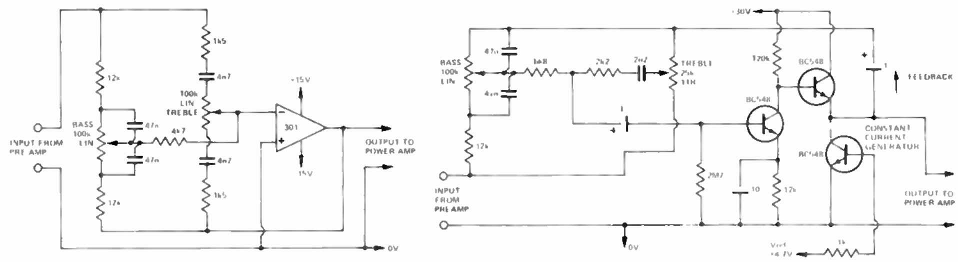
Unless you are going to play long passages of organ music, or run a disco, you will probably find that relatively small heatsinks run quite cool.

However, Darlington transistors are hard to temperature stabilise and should be run as cool as possible. This is why we have opted for a fairly large heatsink compared to other designs. The transistors should be bolted directly to the heatsink, not through a steel chassis. A slit could be cut in a chassis large enough to slide the assembled amplifier through the rear. Heatsink fins should always be vertical to provide the most efficient convection cooling.

The heatsink recommended for the output devices in this project is a flat-sided type with radial fins, 75 mm in length. Other flat-sided types are available with straight fins, and these too would be suitable. A similar length should be used. In general the heatsink should have a thermal resistance, mounting surface to ambient, of around 1° C per watt.

A small 'flag' heatsink is attached to Q7, a BD140 flatpack transistor. A commercial heatsink may be employed (they're only about 60 cents) or a small strip of aluminium may be bent up, drilled, and bolted to the transistor. See that the metal area of the BD140 and a face of this heatsink are in contact. Heatsink compound should be used.





Two suggested tone control circuits for a preamp to suit this module. Low output impedance is an important consideration. Choice of discrete or IC circuitry is given.

Setting Up

Once the amplifier has been assembled and carefully checked, the bias current for the output devices must be set. Remove the fuses, F1 and F2 and connect a 100 ohm resistor across each fuse holder. Remove any input signal. Connect the power supplies and measure the voltage drop across each of these resistors. Adjust the trim pot RV1 for a reading of 2.5 volts across each resistor. This corresponds to a bias current of 25 mA. The reading should be nearly the same across each resistor. Next check that there is no DC voltage across the output terminals.

If the reading across each of the resistors cannot be adjusted, or if there is a DC voltage across the output greater than one volt then there is a fault and the fuses should not be inserted.

If all is well, remove the two resistors and insert the fuses. Connect the speaker and away you go.

Preamp Considerations

The input impedance of this amplifier is relatively low, falling at very high frequencies. Consequently, it must be fed from a low impedance source.

When driving the amplifier with a preamp-tone control unit, the output is best taken from an emitter follower circuit (to provide the required low source impedance) or directly from the output of an operational amplifier. In either case, it *must* be taken from the point where the output is fed back to the tone control circuitry.

Two suggested tone control circuits suitable for the application are illustrated in Figure 5. Both use a 'Baxandall' type tone control network with feedback derived from the output point.

The circuit at right uses discrete components which may suit some constructors better. The left circuit, using a commonly available op-amp, has higher distortion than the discrete circuit.

A preamp-control unit project to suit the amplifier module is described in the next article.

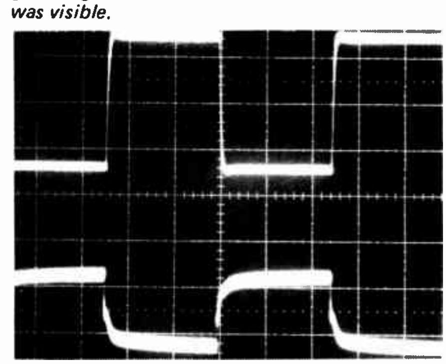
PULSE TESTING

Operation into severely reactive loads was examined by looking at the ac component of the Vbe of Q10 as a measure of the 'overshoot' of the loop and to see if transient overload occured.

f = 1 kHz. CRO is 0.2 mS/div. Output is 30 V into 8 ohms.

Upper trace 10 V/div. Output into 8 ohms.

Lower trace 10 mV/div. Vbe of BD139 gain stage. *No evidence of transient overload was visible.*



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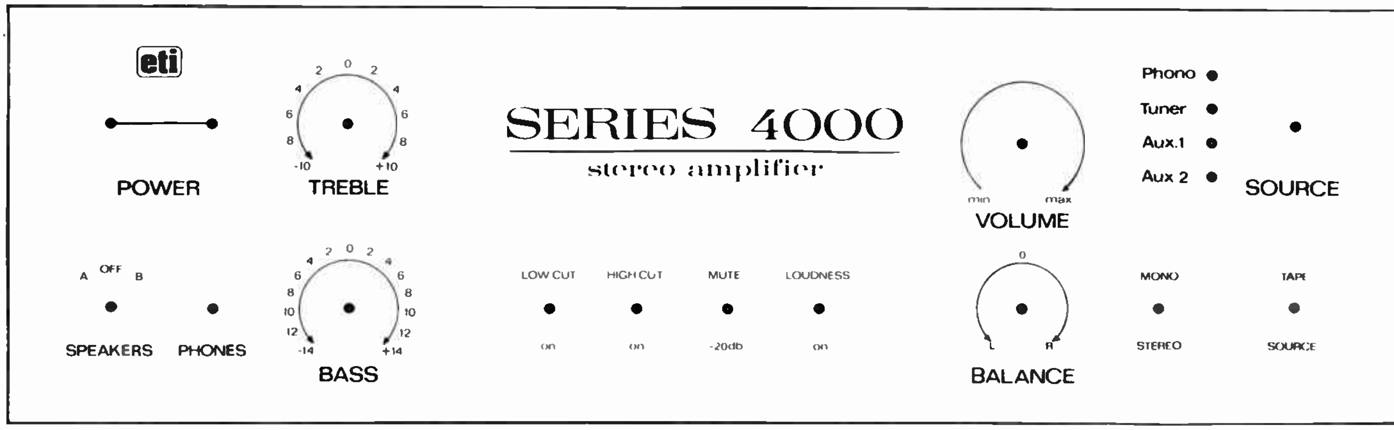
Project 471

High performance stereo preamp control unit

Phil Wait

This project is designed to complement our 60 watt low distortion amplifier module and forms part of a complete stereo system, our "Series 4000" project.



THIS stereo preamplifier is designed to drive two 60 watt, low distortion amplifier modules (ETI 470), described in the previous article.

The requirements for this preamplifier/control unit were set down after many hours of office discussion. In fact it would be fair to say that the final design was evolved, rather than conceived.

Amongst the first requirements were low hum and noise and low distortion – much lower distortion than the amplifier modules it would be required to drive. Low distortion in a preamplifier is relatively easy to achieve and makes the subsequent addition of a high quality class A headphone amplifier worthwhile.

In the final design, we feel we have achieved performance figures well up front amongst commercial equipment.

Features considered essential included loudness, high cut and low cut filters. These are common in commercial preamp/control units but lacking on most kit designs. The low cut filter incorporated in our design will effectively reduce bass rumble while the high cut filter is useful for reducing tape hiss or 'monkey chatter' and heterodynes from an AM tuner.

The disc amplifier stage of a preamp must be capable of handling very high input signals before clipping to preserve dynamic range, especially as moving coil cartridges with voltage boosting transformers and/or amplifiers are finding increasing popularity. The disc input of this design can handle 400 mV peak-to-peak before clipping, giving it a dynamic range in excess of 100 dB!

Finally, and by far the most difficult of our requirements to implement, was the idea that all switches and potentiometers be mounted directly onto the pc board, with as few links and external leads as possible. All this, while preserving an attractive and stylish front panel layout! The advantage of this is that assembly is easy, and straightforward and there is less room for wiring errors to creep in and, should it be necessary, the board can be removed for servicing in its complete, functional form. All interconnections to and from the board are via RCA sockets using standard audio 'jumper' leads.

The 60 watt power amplifier module and this preamp/control unit project form the basis of our "Series 4000" high performance stereo amplifier project (page 42).

Construction

All the components, including the pots, switches and LEDs, are mounted onto the pc board. The board is then fixed, component side forward, behind the mounting panel of the case using standard 25 mm spacers and countersunk screws. A dummy facia – with the control markings etc on it, is subsequently held in place by the switch nuts.

If all directions are followed, then construction is quite straightforward – it's easier to do than describe!

Firstly, the mounting panel and facia must be cut and drilled to the dimensions shown on the drawing (unless you have bought a kit, in which case this may already be done). The drilled pc board may be used as a template. Dimensions shown in brackets refer to the facia panel which must be cut slightly smaller if you wish to use the same case for your stereo as we have.

The holes for the pot shafts are only 7 mm in diameter on the facia panel to ensure correct knob alignment. Countersunk holes are drilled in the mounting panel, but not in the facia, for the bolts securing the pc board through the spacers.

Once the mounting panel and facia are drilled, carefully check the alignment of all holes with the corresponding holes in the pc board. The drilling must be reasonably accurate.

ETI 471 – STEREO PREAMPLIFIER SPECIFICATIONS (Measured on prototype)

Distortion	0.015% at 1 kHz 0.015% at 10 kHz (For all inputs, with 500 mV RMS output – distortion is mainly 2nd harmonic).
Hum and Noise	83 dB unweighted (With respect to 10 mV phono input).
Frequency Response	Phono: Within 0.5 dB of RIAA from 20 Hz to 20 kHz (Follows new IEC curve). Other inputs: 20 Hz to 20 kHz ± 0.5 dB Subsonic rolloff: 6 dB/octave below 20 Hz
Output	7 V p-p before clipping
Tape output	150 mV RMS
Sensitivity	For 500 mV RMS output phono: 3 mV RMS other: 150 mV RMS (Phono overload level is 400 mV p-p).
Tone controls	Bass: ± 13 dB at 50 Hz Treble: ± 11 dB at 10 kHz
Filters	High: 6 dB/octave, –3 dB at 5 kHz Low: 6 dB/octave, –3 dB at 100 Hz
Loudness	8 dB boost at 150 Hz and 10 kHz.
Mute switch	20 dB attenuation

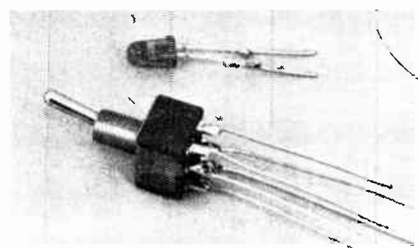
Once this mechanical work is completed the components may be mounted on the pc board. Start with the RCA sockets. Take care not to use too much force on the nuts and check that electrical contact has been made to the earth plane of the pcb using an ohm-meter. Join the centre pin of the RCA sockets to the pc board pads using lengths of tinned copper wire – refer to the overlay.

Mount the potentiometers next so that their terminals are directly above the pads on the pc board. The lower pot terminals can be cut, bent down and soldered directly onto the pads. Connect the upper pot terminals to the pc board, as shown in the overlay, using tinned copper wire.

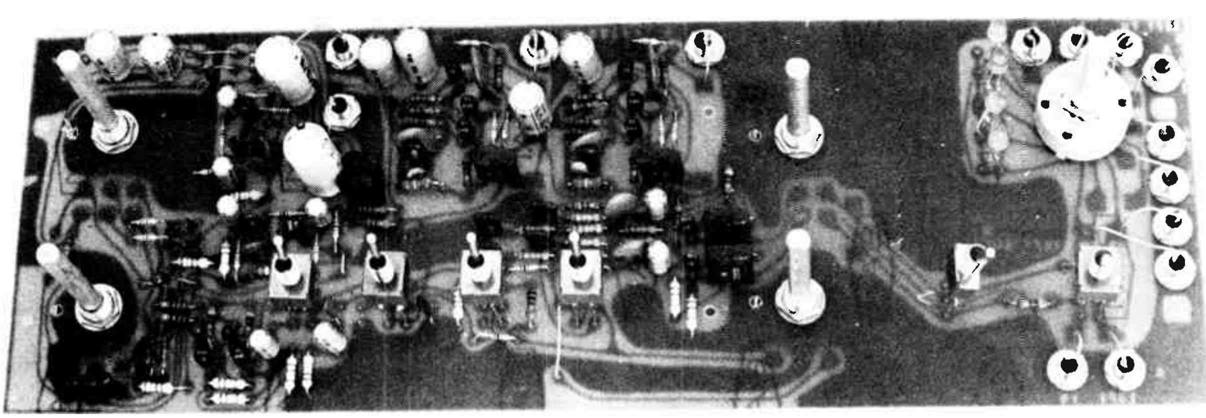
Either of two types of rotary switch may be used for the source selector. We have specified a C & K pc-mounting type but a standard rotary wafer switch may be used instead. The C & K switch mounts directly onto the pc board. If a standard rotary switch is used it will bolt to the front panel of the case and is wired in as detailed shortly.

Once the major parts are assembled onto the pc board, all the minor components may be loaded and soldered in place. Make sure that any large components (electrolytics particularly) are less than 25 mm high, otherwise they will foul the front panel. Check that all transistors, tantalums and electrolytics are correctly oriented. Refer to the overlay as you proceed.

The switches and LEDs must be mounted and spaced correctly off the pc board. Solder 50 mm lengths of tinned copper wire onto each of the switch terminals and LED leads (see illustration). Pass the wires through the ▶



Above: The switches and LEDs have lengths of wire soldered on to them so that they can be inserted into the pcb before being attached to the front panel. They can then be soldered in place. This procedure ensures that there is no strain on the joints. Below: the completed unit. Full details of metalwork will be given in a later article, in which we will describe how to use this preamp with two of the ETI 470 60W units to build a high-performance, low cost stereo amplifier.



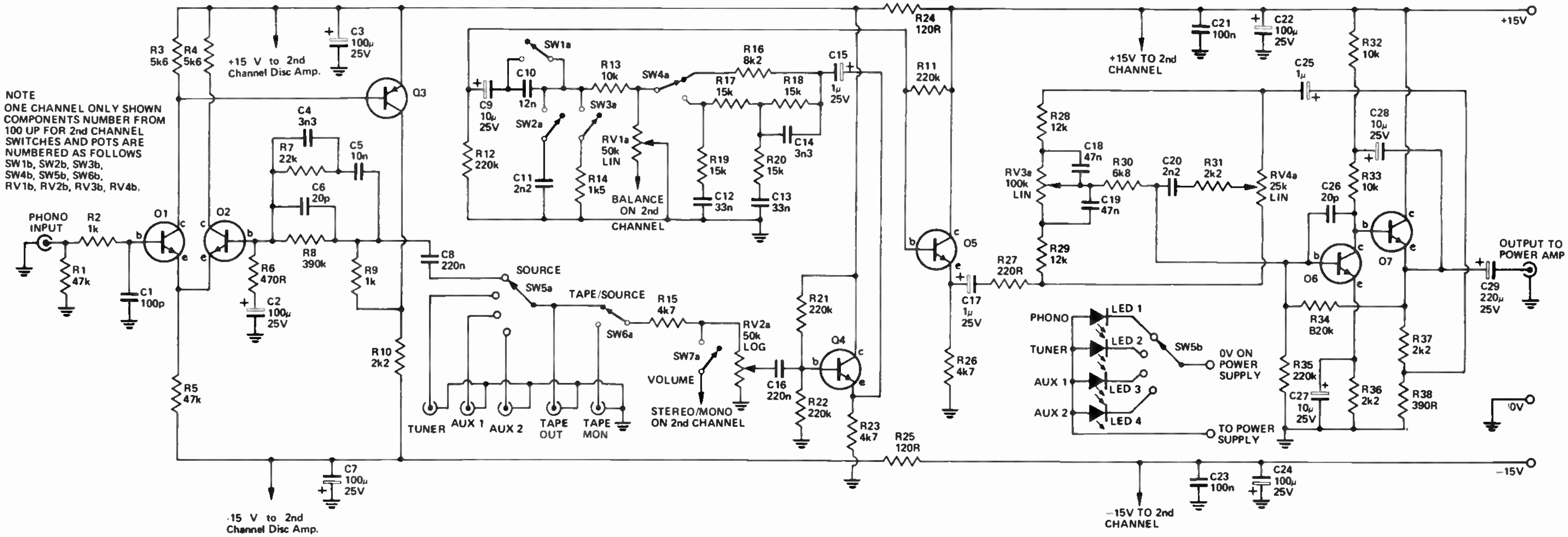


Fig. 1. Preamplifier circuit diagram. Only one channel has been shown for clarity. The component numbering of the other channel begins at 101.

HOW IT WORKS - ETI 471

The signal from a magnetic cartridge is fed to the base of Q1 via a low pass filter, (R2 and C1) for attenuation of radio frequencies. Q1 and Q2 form a differential pair, each half operating at low collector current to minimise noise. The output of the differential pair is taken from the collector of Q1 and further amplified by Q3. Feedback is taken to the base of Q2, the negative input of the differential pair, through the RIAA equalisation network. Overall gain of the phono stage is set by the ratio of the feedback network impedance to the value of R6.

Subsonic bass roll-off of 6 dB/octave, to conform to the new IEC 65 specification, is achieved by a high pass filter consisting of C8 and RV2.

Output from the disc preamplifier is then fed via the Source Switch (SW5), Tape-Source switch (SW6), R15 and the volume control (RV2), to an emitter follower, Q4. This emitter follower presents a high impedance for the aux inputs and a constant impedance for driving the filters.

When switched in, the loudness network boosts the high and low frequencies with respect to the midrange. In actual fact, all frequencies are attenuated but the midrange is attenuated more. When the loudness is switched out, R16 approximates the impedance of the network.

Muting is achieved by switching R14 to earth. The ratio of R14 to R13 sets the attenuation to 20 dB. C11 shunts high frequencies to earth for high cut, while C10 reduces low frequency content when switched in, providing low cut.

A second emitter follower, Q5, presents a constant impedance to the filters and acts as a low impedance source to the tone control stage.

A Baxandall tone stage is used here, a common circuit in many designs. Q6 is a gain stage with a bootstrapped collector load, via C28, to the output. Bootstrapping increases the gain by increasing the effective collection load impedance. Q7 is an emitter follower connected directly to the collector of Q6. This provides a very low output impedance. DC bias for Q6 is taken from the output.

Some of the output signal is fed back to the tone controls and split into high and low frequencies by RV3 and RV4. By adjusting the controls the percentage of the input to the negative feedback signal appearing at the base of Q6 can be varied, thereby varying the overall gain of the amplifier at either high or low frequencies. The gain of the tone stage is set by the ratio of R37 to R38. As R38 is reduced in value the negative feedback is reduced and therefore the overall gain is increased.

To preserve the very low output impedance of the pre-amplifier the balance control is placed ahead of, rather than after, the tone stage.

Power supply filtering and decoupling is provided by 100 μF capacitors and resistors in each rail.

Source indication is by LEDs from the spare section of the source switch. No current limiting resistor is on the pc board for the LEDs as one will be included in the power supply.

corresponding pc board holes for these components but do not solder them in place yet. Check that the LED leads are the right way round.

Assemble the pc board onto the case mounting panel (using the 25 mm spacers and countersunk screws). Place the facia over the front panel, securing it in place with the switch nuts (three hands and a prehensile nose might help! . . . a little sticky tape and deft juggling is all that's really necessary). Once you've got it all together the protruding wires may be soldered to the pc board. Ensure that no short circuits have occurred.

That completes the assembly. For servicing purposes the pc board and all switches, LEDs and pots — all the operating controls — may be removed simply by undoing several nuts, removing the facia and the countersunk screws beneath.

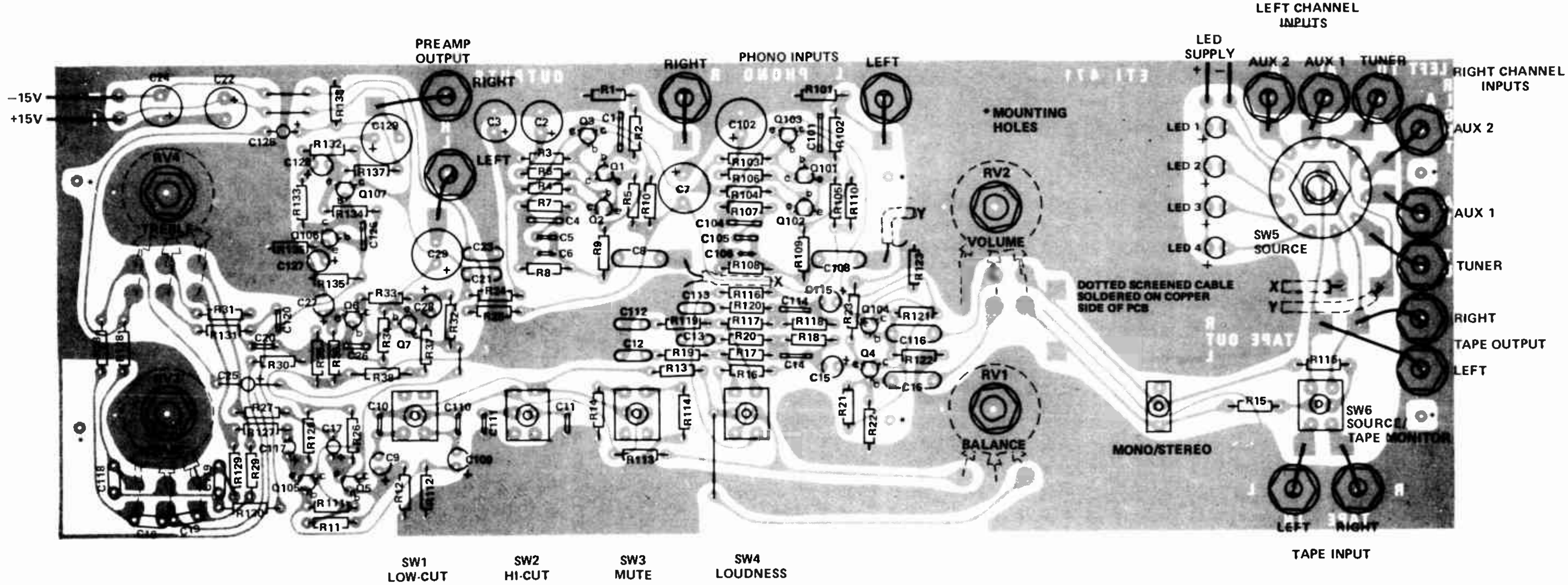


Fig. 2. Component overlay. Note that lengths of shielded cable are used to connect the outputs of the disc preamplifiers to the selector switch.

PARTS LIST - ETI 471

Resistors all ¼W 5%

R1, R101 . . . 47k
R2, R102 . . . 1k
R3, R4, R103,
R104 . . . . . . 5k6
R5, R105 . . . 47k
R6, R106 . . . 470R
R7, R107 . . . 22k
R8, R108 . . . 390k
R9, R109 . . . 1k
R10, R110. . . 2k2
R11, R12,
R111, R112 . . 220k
R13, R113. . . 10k
R14, R114. . . 1k5
R15, R115. . . 4k7
R16. R116. . . 8k2
R17 – R20,
R117, R120 . . 15k
R21, R22,
R121, R122 . . 220k
R23, R123. . . 4k7
R24, R25 . . . 120R
R26, R126. . . 4k7
R27, R127. . . 220R
R28, R29,
R128, R129 . . 12k
R30, R130. . . 6k8
R31, R131. . . 2k2
R32, R33,
R132, R133 . . 10k
R34, R134. . . 820k
R35, R135. . . 220k
R36, R37,
R136, R137 . . 2k2
R38, R138. . . 390R
R118,R119 . . 15k

Potentiometers

RV1 . . . . 50k single linear
RV2 . . . . 50k dual log
RV3 . . . . 100k dual linear
RV4 . . . . 25k dual linear

Capacitors

C1, C101 . . . . 100p ceramic
C2, C3, C102 . 100µ 25V electro
C4, C104. . . . 3n3 greencap
C5, C105. . . . 10n greencap
C6, C106. . . . 20p ceramic
C7. . . . . 100µ 25V electro
C8, C108. . . . 220n greencap
C9, C109. . . . 10µ 25V electro
C10, C110 . . . 12n greencap
C11, C111 . . . 2n2 greencap
C12, C13,
C112, C113 . . 33n greencap
C14, C114 . . . 3n3 greencap
C15, C115 . . . 1µ 25V tantalum
C16, C116 . . . 220n greencap
C17, C117 . . . 1µ 25V tantalum
C18, C19,
C118, C119 . . 47n greencap
C20, C120 . . . 2n2 greencap
C21 . . . . 100n greencap
C22. . . . . 100µ 25V electro
C23. . . . . 100n greencap
C24. . . . . 100µ 25V electro
C25, C125 . . . 1µ 25V tantalum
C26, C126 . . . 20p ceramic
C27, C28,
C127, C128 . . 10µ 25V electro
C29, C129 . . . 220µ 25V electro

Semiconductors

Q1, Q2,
Q101, Q102 . . BC549, DS549, BC109
Q3, Q103 . . . BC559, DS559, BC179
Q4–Q7, Q104
–Q107 . . . . . BC549, DS549, BC109

LED1–LED4 . TIL220R or sim LED's

Switches . . . . (see text)

SW1–SW4 . . . DpDT min toggle switch
SW5. . . . . . . 3 pole 4 pos rotary switch
SW6. . . . . . . DpDT min toggle switch
SW7. . . . . . . spdt min toggle switch

Miscellaneous

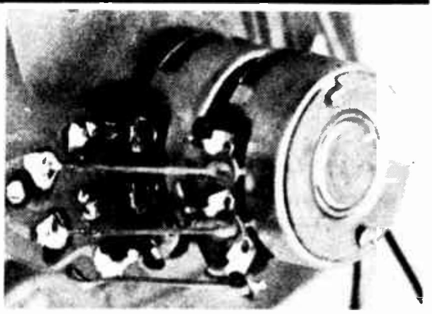
14 RCA panel mounting single hole sockets, ETI 471 pcb, tinned copper wire, length shielded cable, 25 mm spacers, 30 mm screws, nuts, mounting panel and facia plate.

Project 471

Power Supply

The preamplifier/control unit is capable of giving extremely good performance – but only with a good power supply. The supply should be well regulated and filtered for noise, especially if zener regulation is used.

Our next article describes a suitable dual 12 V power supply design which will provide dc for both the 4000 preamplifier and two of the 60 W modules described in the last article.



Potentiometer connections to the pcb are made via lengths of tinned copper wire.

Further Suggestions

We have designed this preamplifier to use commonly available components. However, some constructors or kit suppliers may wish to improve the appearance and the ease of construction of the project. One way of doing this is to use different switches.

The SOURCE switch which we suggest is a C & K Lorlin three-pole, four position pc mounting rotary switch. If only wire terminal models are available, the eyelets can be cut off and the switch mounted as if it were a pc mounting model.

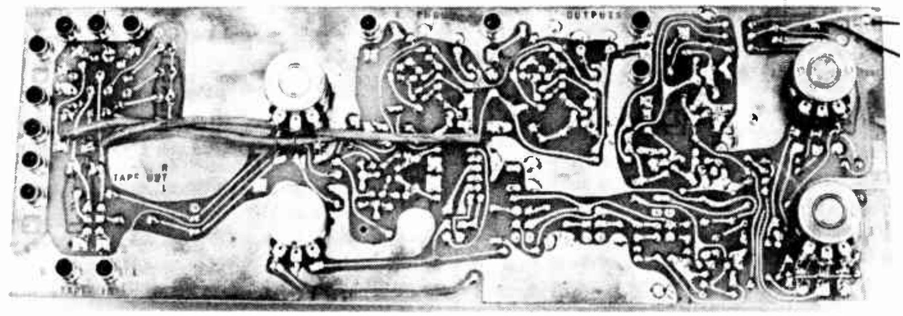
C & K toggle switches may be used for the other switch functions. These are available on order with 'paddle' levers and 25 mm wire wrap terminals. The switches make the preamp look very professional and can also be directly mounted onto the pcb.

The appearance of the LEDs can be improved by using C & K Cliplite covers. These are available in a variety of colours.

C & K switches are available from:

Radio Despatch Service, Sydney
Jaycar, Sydney
Altronics, Perth
Electronic Agencies, Sydney
Rod Irving Electronics, Melbourne
All Electronic Components, Melbourne

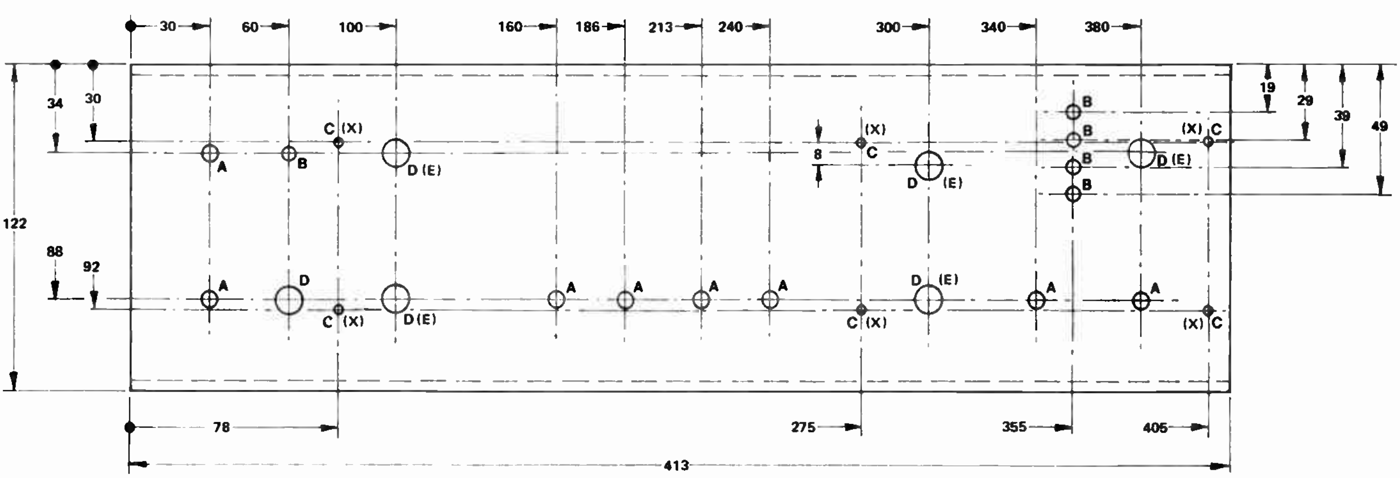
C & K Electronics,
2/6 McFarlane Street,
Merrylands, NSW



Rear view of the assembled preamp, showing how the potentiometers and shielded cables are mounted. Note the use of pc mounting phono sockets for ease of assembly.

Printed circuit board patterns for this project can be obtained from Electronics Today, 4th Floor, 15 Boundary St, Rushcutters Bay 2011. Send large stamped, self-addressed envelope.

This photo shows how the input sockets are wired into the pcb.



HOLES MARKED
A – 6mm
B – 4.5mm
C – 3mm COUNTERSUNK
D – 10mm
E – 7mm

HOLE SIZES IN BRACKETS ARE FOR FACIA PANEL ONLY

FOR HEIGHT OF FACIA PANEL TRIM 4mm FROM TOP AND BOTTOM OF FRONT PANEL DIMENSION. (AS INDICATED BY DOTTED LINE.)

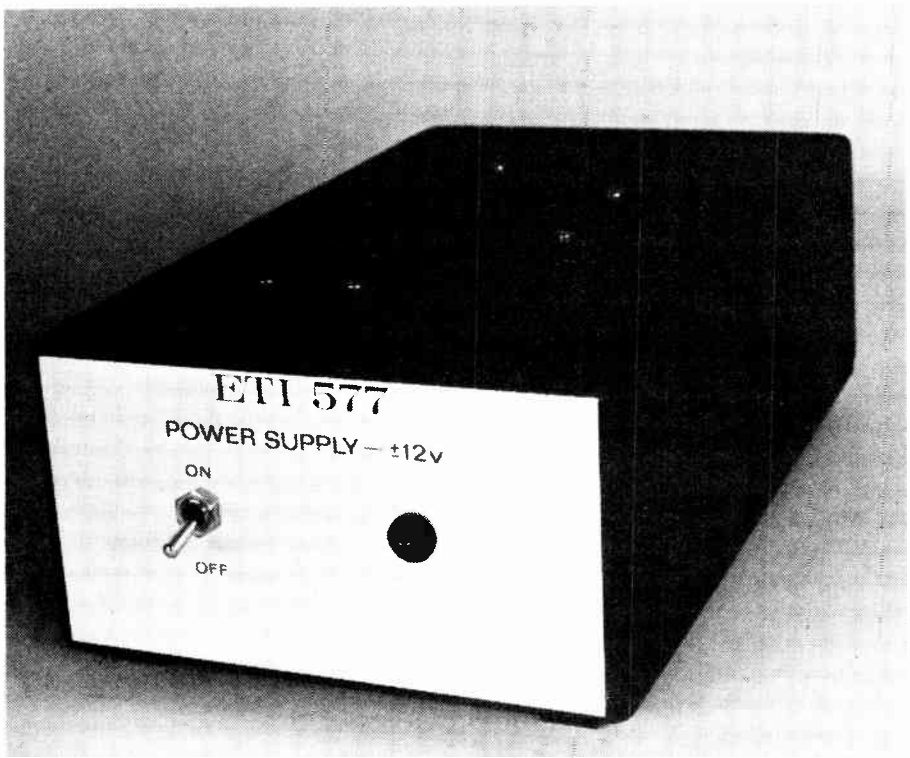
ALL DIMENSIONS ARE IN MILLIMETRES

DO NOT SCALE DRAWING

A general purpose, dual 12 V supply

David Tilbrook

Whilst this supply was designed specifically to power the Series 4000 moving-coil cartridge preamp (see page 48) it should find application in many electronic projects.



Our prototype was housed in a diecast box to match that used for our Series 4000 moving-coil cartridge preamp, although any suitable box may be used if the power supply is intended for another application. Scotchcal front panels should be available from kit suppliers or separately from Radio Despatch Service in Sydney.

THIS POWER SUPPLY provides the +/-12 volts needed by the Series 4000 moving coil cartridge preamplifier. We intend designing a range of hi-fi system 'add-ons' like the M.C. preamp and rather than have a power supply in each unit they will be powered from this supply. This decreases the cost of of building the units and just as importantly removes the major source of hum from within the chassis.

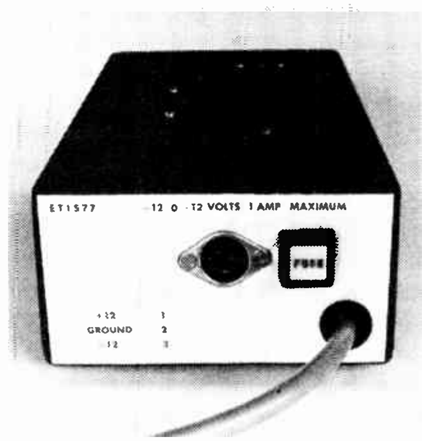
The supply delivers positive and negative 12V dc at 1A while the IC series regulators provide short circuit and temperature protection. These regulators have a tendency to oscillate at around 3 MHz and for this reason must have their output pins bypassed to ground through an appropriate capacitor. If they are allowed to oscillate the device quickly overheats and its thermal protection cuts in.

The regulators are mounted onto the chassis which acts as a heat sink. If the recommended power transformer is used, the voltage after rectification is approximately 17 volts. The regulators must drop 5 volts at a worst-case current of one amp, so they are dissipating a maximum of five watts which is well within their ratings.

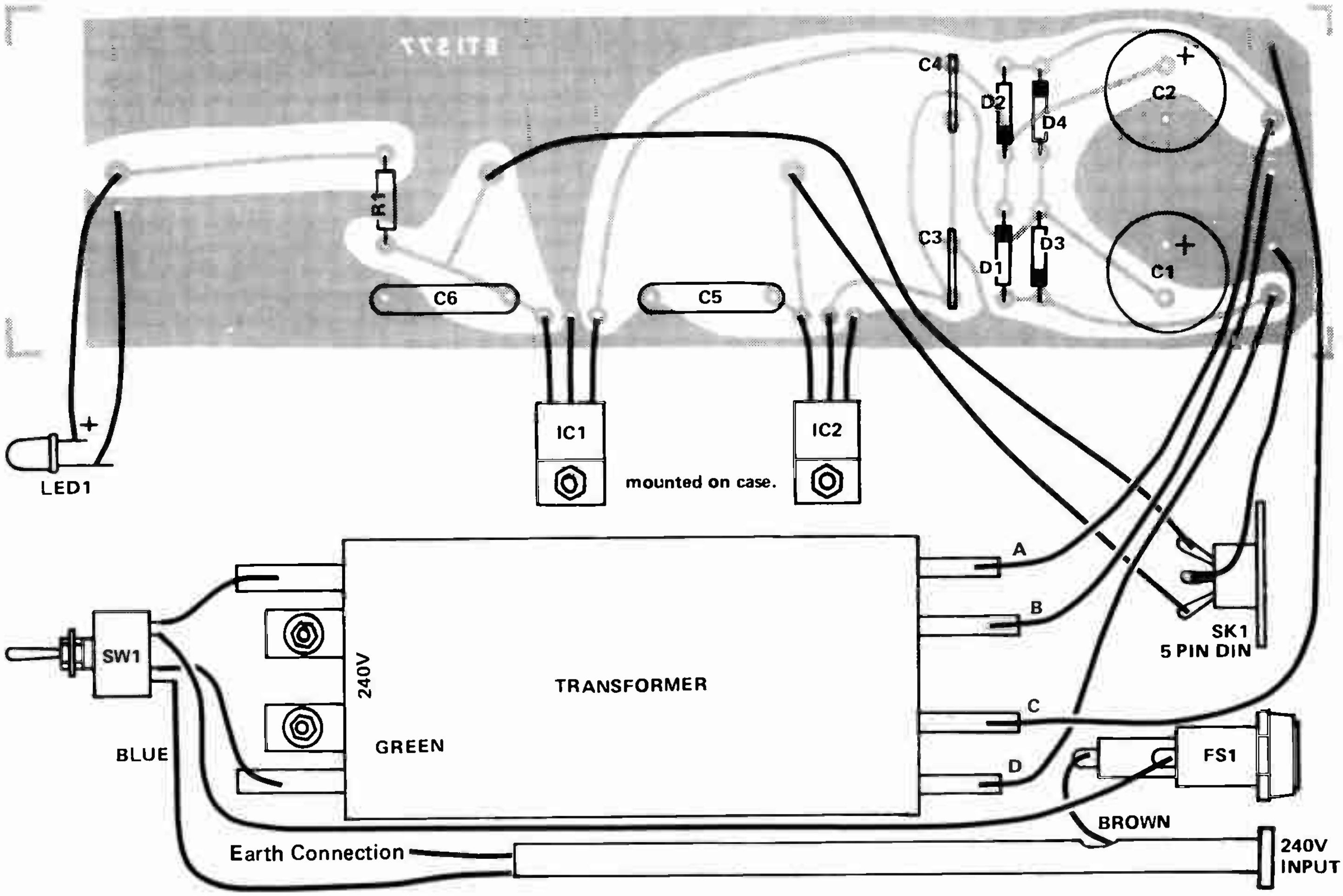
Construction

The power transformer used in the prototype was a Ferguson type PL30/40 VA. This is one of their low profile transformers and fits easily into the die-cast aluminium box. The printed circuit board has been designed to slot into the grooves in this box.

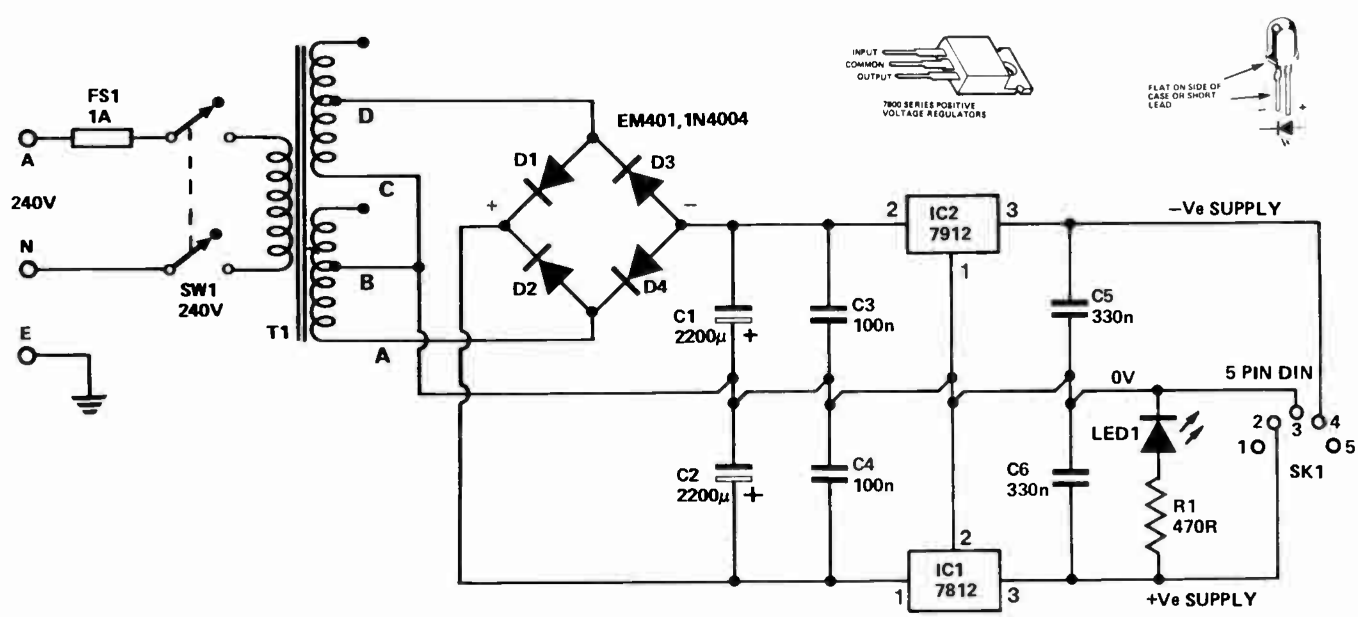
Assembly of the pc board is not difficult as it has relatively few components. If you are using the same box we did it is easier to solder pc board pins onto the board, slot the board into place, bolting the regulators down, and then make the necessary wiring interconnections. Both regulators must be insulated from the case using the appropriate mounting hardware. The case of these regulators is connected to pin 2. For the 7812 this is the ground connection, and accidental connection to case will cause a hum loop when the unit is connected to the moving coil cartridge pre-amp. In the 7912, pin 2 is the input to the regulator and as such has 17 volts directly from the bridge rectifier connected to it. Accidental connection of this to ground will probably damage the rectifier diodes, so check with a multimeter that the case of this regulator is well insul- ▶



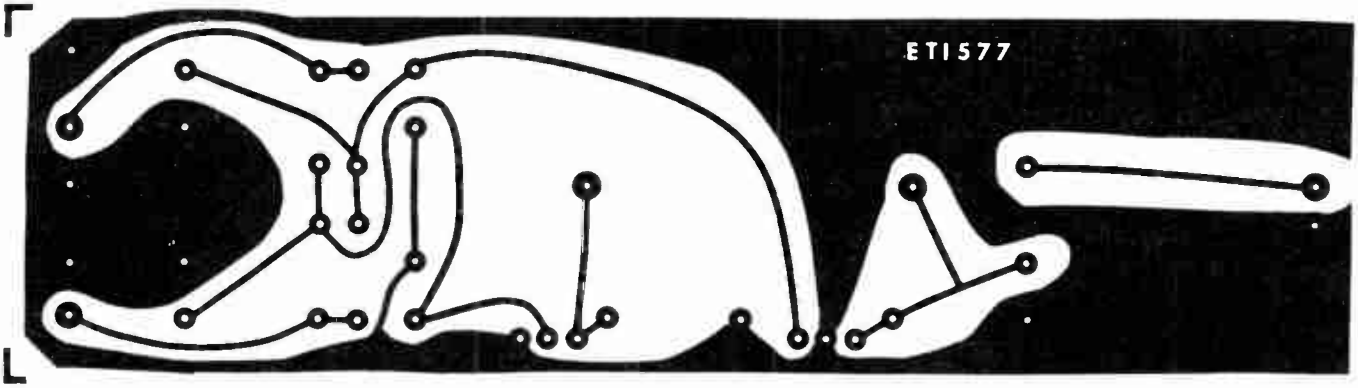
Rear view of the power supply showing placement of the major components.

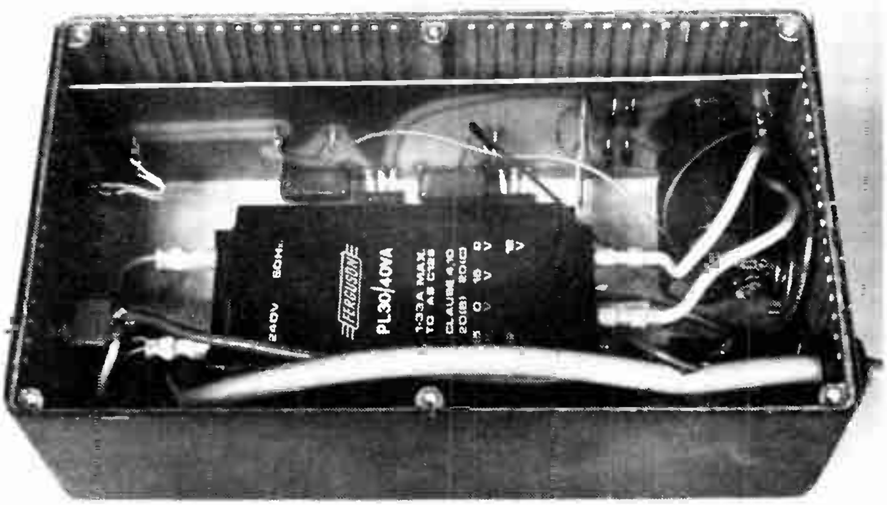


ETI 577
C4
D2
D4
C2
R1
C3
D1
D3
C1
C6
C5
IC1
IC2
mounted on case.
LED1
A
B
SK1
5 PIN DIN
SW1
240V
TRANSFORMER
C
GREEN
BLUE
FS1
D
BROWN
Earth Connection
240V
INPUT



FS1
1A
A
240V
N
E
SW1
240V
T1
D
C
B
A
EM401,1N4004
D1
D3
D2
D4
C1
2200µ
C2
2200µ
C3
100n
C4
100n
IC2
7912
−Ve SUPPLY
C5
330n
C6
330n
0V
5 PIN DIN
LED1
R1
470R
SK1
IC1
7812
+Ve SUPPLY
INPUT
COMMON
OUTPUT
7800 SERIES POSITIVE VOLTAGE REGULATORS
FLAT ON SIDE OF CASE OR SHORT LEAD





Internal view of the power supply showing how the pc board was mounted, the position of the power transformer and the general wiring arrangement. General parts placement is not at all critical and a variety of layouts is possible. Be sure to insulate the regulator ICs.

ated from the chassis before powering up.

The LED is mounted onto the front panel with a standard LED mounting grommet and connected to the board by two short lengths of hook-up cable.

Make absolutely certain that all 240 volt connections are secure and that the mains cable ground lead is connected to chassis as shown in the wiring diagram. The mains flex must be secured to the chassis, either with a clamp-type grommet where it enters the box or with a cable clamp on the inside.

Before applying power to the unit make a final check of the board and all connections to the power transformer. Check the 240 volt connections and ensure that the regulators are satisfactorily insulated from the chassis. If all is correct, turn the power supply on. The LED on the front panel should come on immediately. Check the voltage present on the output DIN socket which should be very close to 12 volts (certainly within 0.25 V). Make sure the positive and negative supply connections terminate on the correct DIN socket pins. ●

HOW IT WORKS – ETI 577

Mains 240 Vac is applied to the primary of the transformer via a 1A fuse. The transformer secondary consists of two 15 V windings with tappings at 12 V. The 12 V tapping of one is joined to the 0 V of the other – this junction (effectively a centre-tap) forming the volt rail.

A bridge rectifier D1-D4 rectifies the ac voltage from the transformer and supplies around 17 volts to the inputs of the regulator ICs. Capacitors C1-C4 filter the input to the regulators while C5 and C6 ensure high frequency stability of the regulators.

The IC regulators provide a stable, regulated output very close to the specified 12 Vdc and can supply up to one amp of dc current. Overload and thermal protection is provided internally on the IC chip. These regulators are convenient, inexpensive and require the minimum number of components.

PARTS LIST - ETI 577

Resistors	all ½W, 5%
R1	470R
Capacitors	
C1, C2	2200µF 25V electro
C3, C4	100n greencap
C5, C6	330n greencap
Semiconductors	
D1–D4	IN4004, EM401 or sim
LED1	Red led, TIL220R or sim
IC1	7812 or LM340-12 voltage regulator (positive)
IC2	7912 or LM320-12 voltage regulator (negative)
Miscellaneous	
T1	transformer, 15V-0-15V, 1.3 amps (Ferguson PL30/40VA)
SW1	DPDT 240V switch
F1	1A, 3AG type fuse
SK1	Chassis mounting 5 pin DIN socket

Chassis mounting 3AG fuse holder, 5 pin DIN plug, 240V mains plug, 240V/3 core cable, rubber mains cable grommet, Diecast aluminium box 190 x 60 x 110 mm.

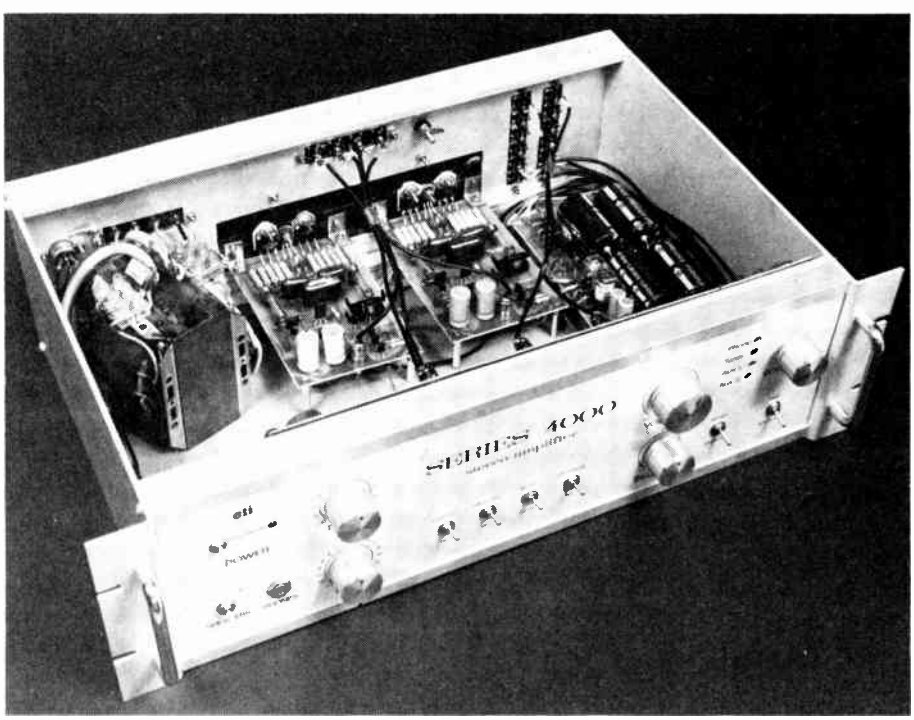
The "Series 4000" stereo amplifier

Here's how to assemble a high-performance 60 watts per channel stereo amplifier using our ETI-470 modules and the ETI-471 preamp control unit.

Circuit design
Trevor Marshall / Phil Wait

Mechanical design/layouts
Phil Wait

Front panel art
Bill Crump



The completed stereo amplifier is shown here mounted in a handsome rack-mounting case. This particular style of case is also available with wooden end cheeks if that is what you prefer.

THIS ARTICLE presents the complete Series 4000 amplifier made from the ETI 470 60 watt module and the ETI 471 preamplifier.

We chose to build the amplifier into a single box, being the most economical method as only one box and power supply is used for the preamp and both power amplifiers. However, this method has several drawbacks. Firstly, since the preamp and power amp share the same power supply, the regulation for the preamp must be very good, otherwise low frequency instability can occur, caused by the drop in supply line voltage when the outputs draw high current getting back into the preamplifier.

Hence we have chosen IC regulators for the preamplifier supply lines.

Secondly, the magnetic field from the large transformer and associated AC wiring required to supply the power amplifier modules is quite large and almost impossible to keep out of the sensitive preamp stages. Therefore you will notice that the specification for hum in the completed amplifier is lower than that of the individual units. We took this measurement using a standard EI lamination transformer (Ferguson PF 3577) after rotating it for minimum hum to the position shown in the wiring diagram.

The hum induced by the transformer can be further reduced by using a C-core

ETI 4000 SERIES STEREO AMPLIFIER — Specifications of prototype

Power output	60 watts @ 0.1% THD one channel driven 55 watts @ 0.1% THD both channels driven
Distortion	0.05% THD @ 30 V p-p output across 8 ohm load, both channels driven.
Hum	–70 dB on full output using standard transformer
Noise	–80 dB on full output
Damping factor	57 (measured at 100 Hz, 1 kHz and 10 kHz).
Frequency Response	Phono: Within 0.5 dB of RIAA from 20 Hz to 20 kHz (Follows new IEC curve).
Other inputs	20 Hz to 20 kHz ± 0.5 dB Subsonic rolloff: 6 dB/octave below 20 Hz
Tape output	150 mV RMS
Sensitivity	For 500 mV RMS output phono: 3 mV RMS other: 150 mV RMS (Phono overload level is 400 mV p-p).
Tone controls	Bass: ± 13 dB at 50 Hz Treble: ± 11 dB at 10 kHz
Filters	High: 6 dB/octave, –3 dB at 5 kHz Low: 6 dB/octave, –3 dB at 100 Hz
Loudness	8 dB boost at 150 Hz and 10 kHz.
Mute switch	20 dB attenuation

type, or better still a toroidal transformer, which have a contained field, but these are often hard to get and expensive to the hobbyist.

We feel that the specifications of the amplifier are very good, however the purist (with plenty of money) may like to do it this way:

The two power amplifier modules, together with individual power supplies using say, 30 000 uF capacitors, could be mounted in a separate box to the preamplifier, which could then be powered from the ETI 581 (June 77) regulated supply.

This would no doubt improve the power output and transient performance of the amplifier but the cost would be much greater.

Construction

Construction details for the preamplifier and power amplifiers have been described previously, all that remains is to house them together, with the power supply, in a suitable box. As we said before, many variations are possible – here is how we did it.

Assemble the power supply board first, taking care to correctly orientate the semiconductors, IC regulators and capacitors. To simplify construction we used pc pins for all terminations to the boards.

The photo of the rear panel shows the position of the input and output connections. Slots are cut in the panel for the connector blocks and a large cut running across the back panel is used to inset the power amplifier modules from the rear. Holes then must be drilled for the earth terminal, external power socket, power cord, mounting screws for the terminal blocks and holding screws for each power amplifier – which pass through the top of each heatsink fastening it to the panel.

The case measures 420 mm x 135 mm x 285 mm and is made from aluminium extrusion with easily removable panels. Available with either metal rack mounting or wooden sides, it can be purchased from suppliers listed at the end of this article.

One thing to watch though is that anodised aluminium does not conduct electricity and, after assembling the box, the various metal parts will probably not be connected to each other, causing a multitude of problems. To overcome this, strap the rear and side panels to the common earth point at the headphone jack on the front panel. (Yes, we found this out the hard way).

After the preamplifier/front panel, power amplifiers and power supply have been mounted in the box and the input/output sockets mounted onto the rear panel the unit can be wired as shown in the wiring diagram.

Common to all amplifier designs, the earth wiring is very critical. Most instability and hum problems can be traced to earth "loops" or incorrect wiring.

The common lead from each channel speaker is returned directly to the OV point on the power supply. A wire is then taken from this point and fed to one power module, to the other, and then to the preamplifier. To avoid an earth loop the braid of the shielded cables from the preamplifier to the power amplifier is not carried through the connector block on the rear panel. OV leads for the LEDs and external power are also returned to the power supply common. The common is then earthed to the chassis at the headphone socket together with the transformer shield and mains earth. This is the **ONLY** earth point onto the chassis.

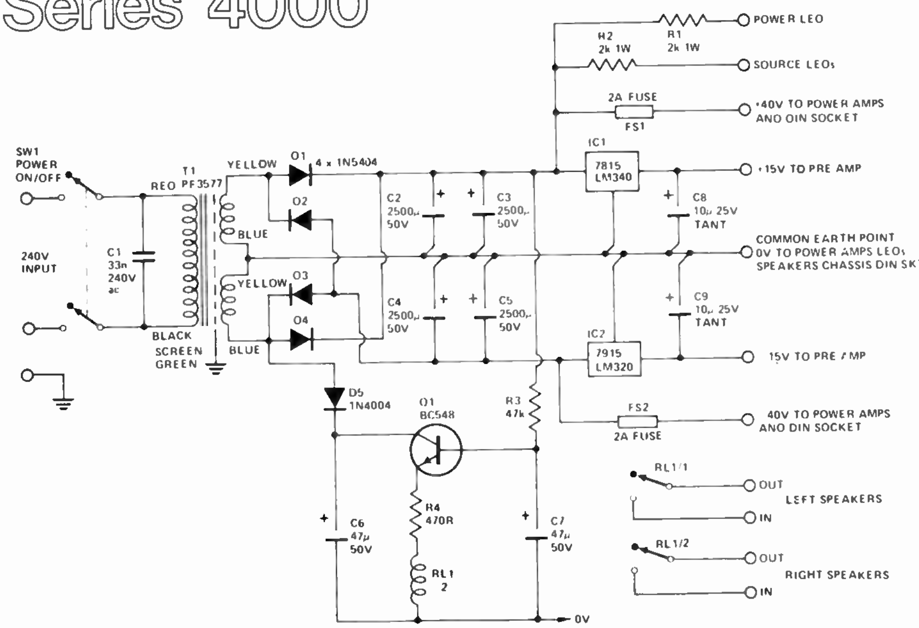
All the ac and speaker wiring is fed along the back and down the left side of the case as shown, well away from the sensitive parts of the amplifier. The dc wiring from the power supply to the preamplifier is carried along the front.

Lengths of shielded cable with RCA plugs on one end are used to connect the input sockets to the preamplifier. These can be made by cutting RCA patch cords to the appropriate length, one cord making two leads. The shields of these cables should not be connected together or to the case at the input sockets.

All that remains is to solder the 330 ohm resistors from the speaker switch to the plugs on the headphone socket.

Check that all wiring is correct and there are no frayed ends. The procedure for setting the bias current for the output transistors is given on page 32. As soon as this is done, insert the 2 A fuses and the amplifier can be switched on.

If you have the older 50 watt ETI 480 modules these could probably be used in place of the ETI 470 module, though we haven't tried it.



Power supply

The power supply for this amplifier uses a 28V-0-28V transformer rated at 2 A to provide +/- 40 Vdc rails for the power amplifiers. Two regulators, IC1 and IC2, supply very stable +/- 15 V rails for the preamplifier.

Current limit resistors are mounted on the pc board to power the front panel LEDs. This permits some flexibility to allow us to think up other things to do with the LEDs later.

Fuses are also provided on the board to protect the power supply from a short circuit in the dc output lines. If the dc output facility on the rear panel is not used the fuses can be short circuited, as each power module is protected by its own fuses.

When an amplifier is first switched on,the two supply lines rarely come up to full voltage simultaneously. This causes a loud 'thump' in the speakers which may damage them.

PARTS LIST - ETI 472

Resistors
R1, R2 . . . . . 2k 1W 5%
R3 . . . . 47k ¼W 5%
R4 . . . . 470R 1W 5%

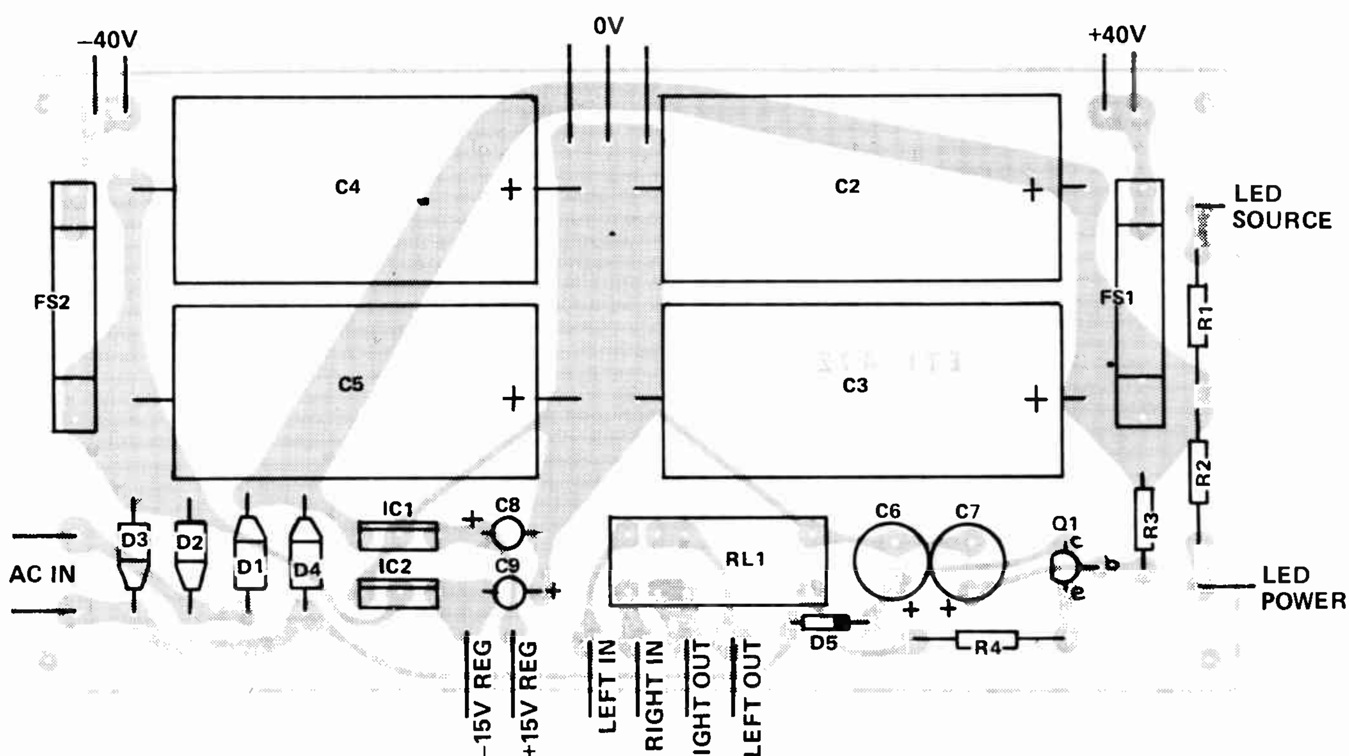
Capacitors
C1. . . . . 33n 240V ac metalized paper
C2—C5 . . . . . 2500μ 50V electro
C6, C7 . . . . . 47μ 50V electro
C8, C9 . . . . . 10μ 25V tantalum

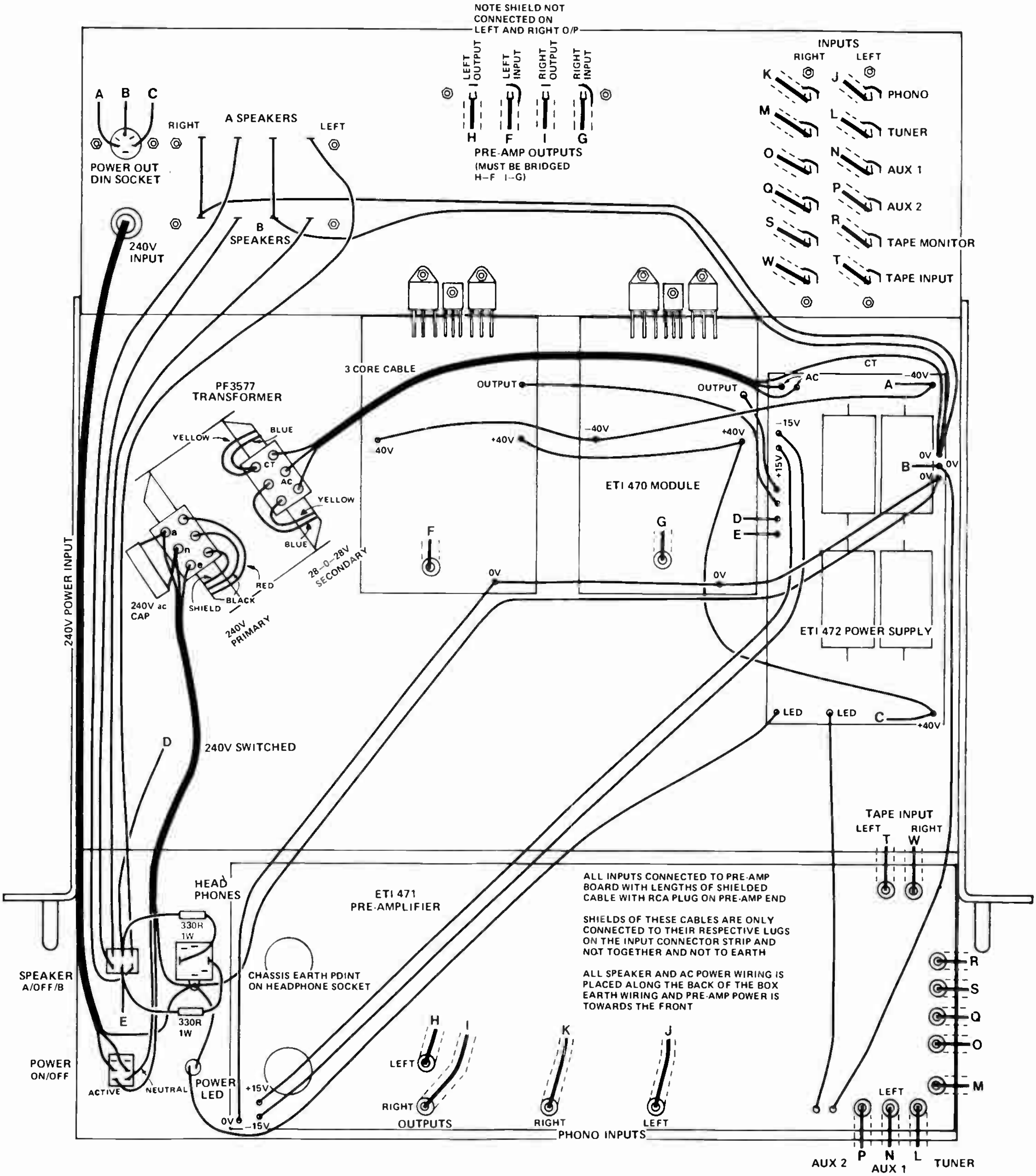
Semiconductors
D1—D4. . . . . IN5404 or sim
D5 . . . . IN4004, A14A or sim
Q1 . . . . BC548, BC108, DS548
IC1 . . . . 7815, LM340-15, 15V regulator
IC2 . . . . 7915, LM320-15, -15V regulator

Miscellaneous
T1. . . . . PF3577 or similar (Ferguson)
FS1, FS2. . . . 2 amp fuses (if used)
RL1. . . . . . . pcb mounting, 2 pole changeover relay, 12V coil, Pye 265/12/G2V, DS cat S7130 or sim
SW1. . . . . . . 2 pole 240 VAC miniature toggle switch.

CHASSIS PARTS LIST

Headphone socket . . . 6.5 mm jack skt.
Speaker switch . . two pole, two position, centre off min. toggle switch
16 RCA plugs or eight patch leads cut in half, two short RCA patch leads, power lead and clamp.
Two, 330R, 1W resistors
Two, 3-way plastic mains terminal strips
Two, 4-way speaker terminals
Two, 6-way RCA panel sockets
One, 4-way RCA panel socket
One, 5-pin DIN socket





To avoid this an "anti-thump" circuit connects the speakers several seconds after the amplifier is turned on.

It works this way; as the power rails come up to voltage a capacitor, C7, charges via R3. Transistor Q1 conducts pulling in the relay, RL1, and connecting the speakers after the power rails have had enough time to stabilise.

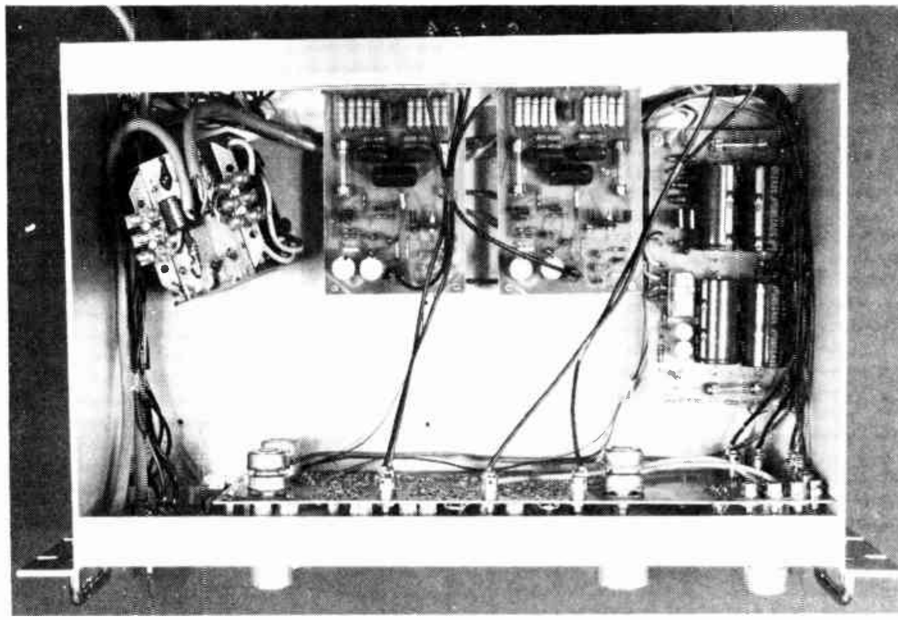
The printed circuit board pattern for this project is on page 47.

At first we tried mounting the power supply board in front of the transformer near the preamplifier, but found the proximity of the speaker wiring to the tone control stage caused high frequency instability if the treble control was advanced. The power supply board is now mounted at the opposite side of the case to the transformer and the ac secondary wiring run across the back.

Two three-terminal connector strips are mounted on top of the transformer, using the holes in the mounting plates, to take primary and secondary connections. The shield (green wire) makes up the third wire on the primary side and is run together with the 240 V wiring to the front panel. We used three-core mains flex for connections from the transformer to the power switch and the power supply pc board. A suppression capacitor (C1) is mounted across the transformer primary on the connector block.

Make sure that the power switch you have is rated for 240 Vac, as some being sold are only 125 Vac rated and sometimes fail catastrophically.

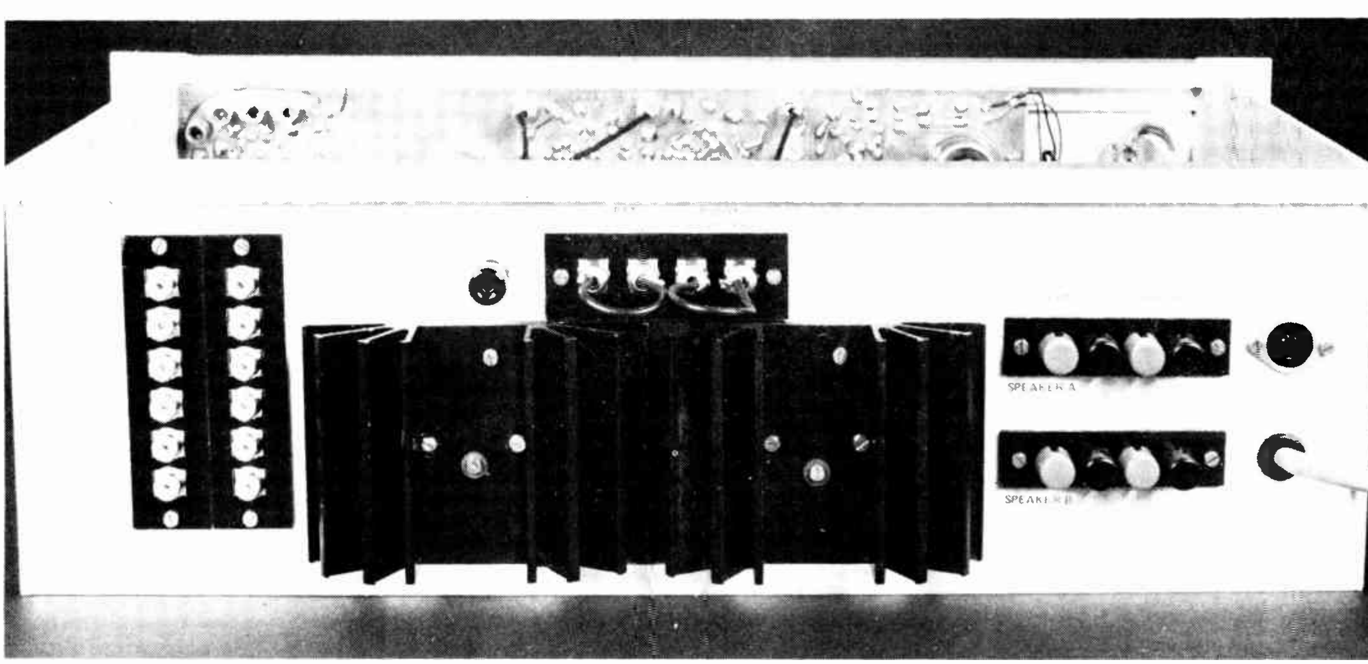
Short patch leads will have to be made up to connect each of the preamplifier outputs to their respective power amplifier inputs.

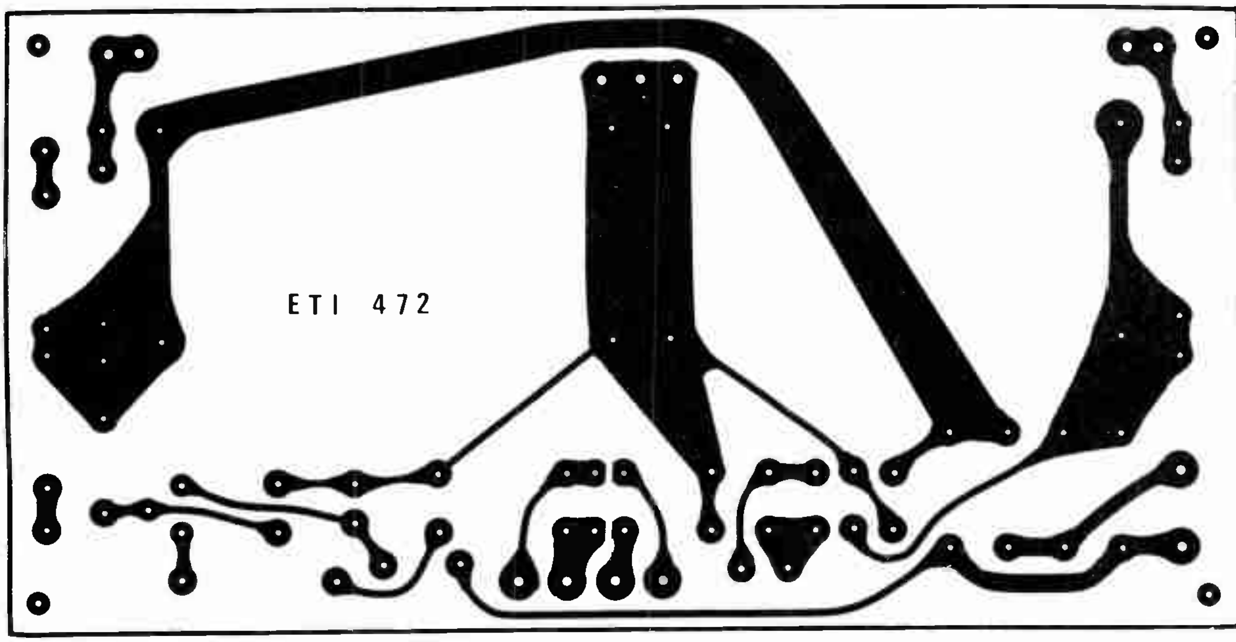


This internal view shows the placement of the main modules and the orientation of the power transformer. The latter will have to be oriented individually to reduce hum levels to the minimum obtainable.

Power Darlington Transistor Equivalents

If difficulties arise in finding sources of the Philips BDV64/65 power Darlington transistors, Texas Instruments have equivalents. These are known as TIP142 and TIP147, respectively. They have been tested in the ETI-470 module which performed without any variation.

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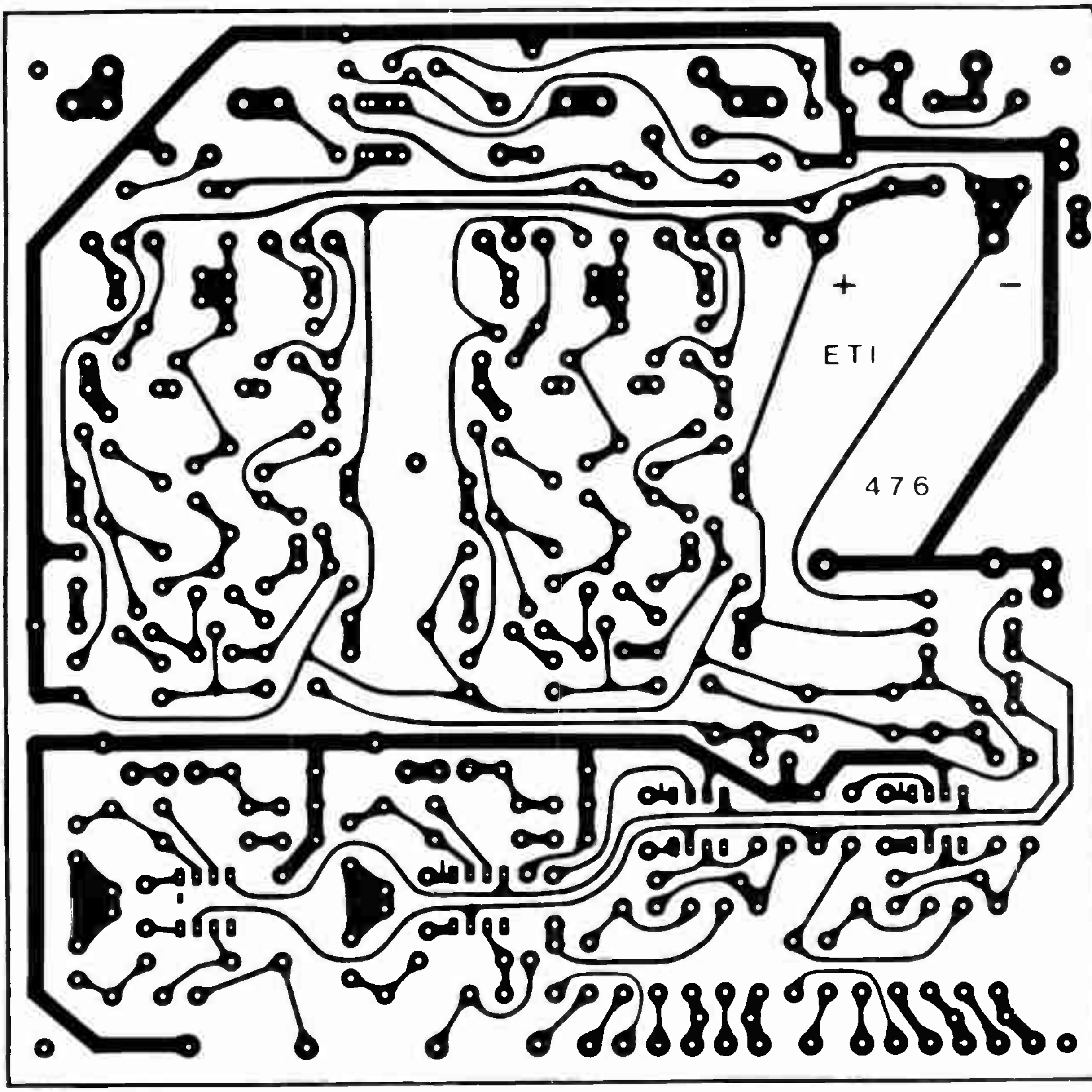
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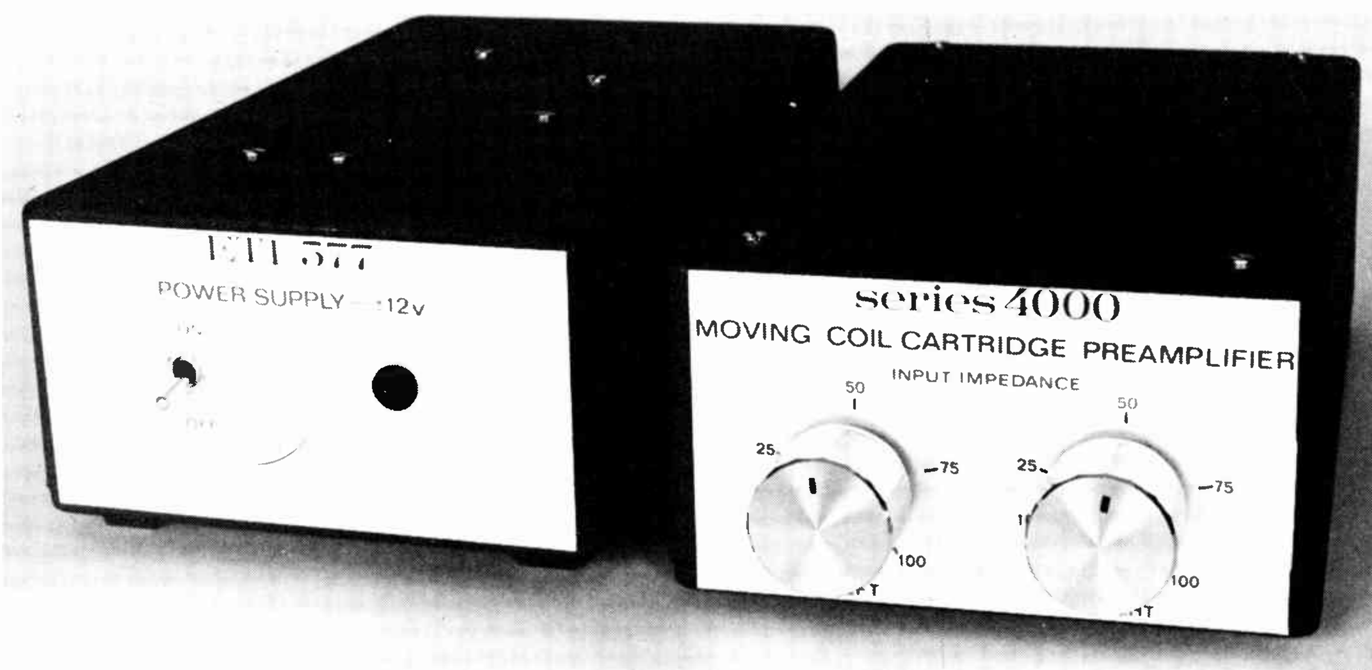
Series 4000 moving-coil cartridge preamplifier

David Tilbrook

Designed to complement our popular Series 4000 stereo amplifier, this project features performance equal to, or better than, top quality commercial preamps currently available.



OVER THE LAST several years there has been a dramatic increase in the number of moving coil cartridges released. The design of this type of cartridge results in a number of advantages over the more usual phono cartridge which works on a moving magnet principle.

Modulations on the wall of the record are tracked with a diamond stylus attached to a long arm called a cantilever. In the moving-magnet cartridge a small magnet is attached to the cantilever so that stylus movement causes movement of the magnet. Two pick-up coils are mounted close to the magnet so that the windings of the coils intersect the lines of magnetic flux from the magnet. As the stylus moves the magnetic flux seen by the pick-up coils varies in direct proportion to the stylus movement, and small electrical signals are generated in the coils.

The moving-coil cartridge works in a similar way but inverts the roles of the pick-up coils and magnet. The magnet assembly is held stationary while the pick-up coils are mounted on the cantilever assembly and move with the stylus modulations (hence the name 'moving coil').

The pick-up coils are reduced drastically in size and weight compared to the coils used in moving magnet cartridges. This results in a total cantilever weight that is much smaller than in the typical moving magnet cartridge. Since the weight is greatly reduced the ability of the stylus to react to transients is increased and an overall improvement in signal accuracy results. Moving coil cartridges generally have superior frequency response characteristics and improved phase response at high frequencies. But they also have disadvantages.

The small pick-up coils have a very low impedance resulting in much lower signal levels than available from normal phono cartridges. In fact, the voltages present on the typical moving-coil cartridge at a recording velocity of 10 cm/sec can be in the order of 150 μV! This is generally insufficient to drive an amplifier to anything like full power. Furthermore, since the output level is some 30 dB below that expected by the amplifier then a great reduction in the signal-to-noise ratio will result. An amplifier with a short circuit signal to noise ratio of 80 dB for example, which is quite a good figure, will end up with a signal noise ratio of about 50 dB — which is distinctly *bad*.

SPECIFICATIONS – ETI 473 moving coil cartridge preamp.

Gain	28 dB (x 25 approx).
Frequency response	29 Hz to 48 kHz ±1 dB.
Input impedance	Adjustable 3.3 to 100 ohms.
Noise	Total equivalent input noise 0.3 nV√Hz. Over a 20 kHz noise bandwidth—42nV. Signal-to-noise ratio, with respect to an input level of 150 μV: -71dB.
Total Harmonic distortion	With respect to an input level of 0.2mV, unmeasurable (below noise). Calculated to be 0.0015% (see text). Rising to 0.015% for a 30 mV input signal at 1 kHz.
Channel separation	Better than 61 dB.
Input overload margin	better than 80 dB.

The internal impedance of moving-coil cartridges is around 5 ohms and to achieve the low recommended load impedance required it is clearly not satisfactory to simply load down the input of the average phono input with a resistor since this does nothing to overcome the signal-to-noise ratio problems.

The solution to these problems is to insert some voltage gain between the output of the cartridge and the phono input. This can be done in two ways. Firstly, it is possible to use a transformer to boost the voltages up to the desired level and they are capable of very good results. But, transformers are still limited in transient performance and noise. To obtain the necessary voltage gain the turns ratio must be relatively high. Since the impedance ratio is related to the square of the turns ratio, the output impedance must, of necessity, be high also — usually around 30 k for a 50 Ω input impedance. This is substantially higher than the output impedance of normal phono cartridges and degrades the noise figure of the phono input stage. A solution to this is to use a pre-preamplifier instead of a transformer to achieve the necessary voltage gain.

Preamp requirements

Preamplifiers have their disadvantages also. The biggest problem by far is the design of an extremely low noise input stage with the correct input impedance to load the cartridge according to the manufacturers' recommendations. The distortion must be kept to a minimum and the frequency response should be as flat as possible. These design goals are not unique to a moving coil cartridge preamplifier but they are difficult to achieve owing to the very low output voltage of the moving coil cartridge.

The required low input impedance can be achieved in several ways. Firstly, we can make the input stage a common base configuration. In this type of circuit the input is connected to the emitter of the transistor so that the input impedance is determined by the emitter resistor in parallel with the base-emitter junction of the input transistor, which can be quite low. However, this does not solve the problem of input stage noise.

The other possibility, and the one I elected to use in this design, is common emitter configuration. The impedance of the base-emitter junction of a bipolar transistor is a function of the amount of current flowing in the emitter of the transistor. This will be largely determined by the collector current and not by the base current, which will contribute only a small amount of the total emitter current. A study of base-emitter turn-on characteristics shows that the impedance of the base-emitter junction is approximately equal to:

$$\frac{26\beta}{I_e\ (\mathrm{mA})}$$

where 'β' is the small signal current gain of the transistor.
and 'I_e' is the current in the emitter of the transistor in mA.

So, to reduce the input impedance of the first stage it is simply necessary to increase the emitter current. But this increases the current density in the input transistors, increasing the noise generated by the input stage.

To understand why this happens it is necessary to look more closely at the causes of noise.

Noise

There are two main sources of noise in transistors: shot noise and l/f noise. Shot noise is the main cause of noise at middle and high frequencies and is generated when an electron attempts to cross a potential barrier. It is therefore directly related to the amount of charge flowing in the device. More specifically, it is given by the equation:

$$\overline{I_s^2} = 2qi_{dc}B\ (\mathrm{amps})^2$$

(mean shot noise current)

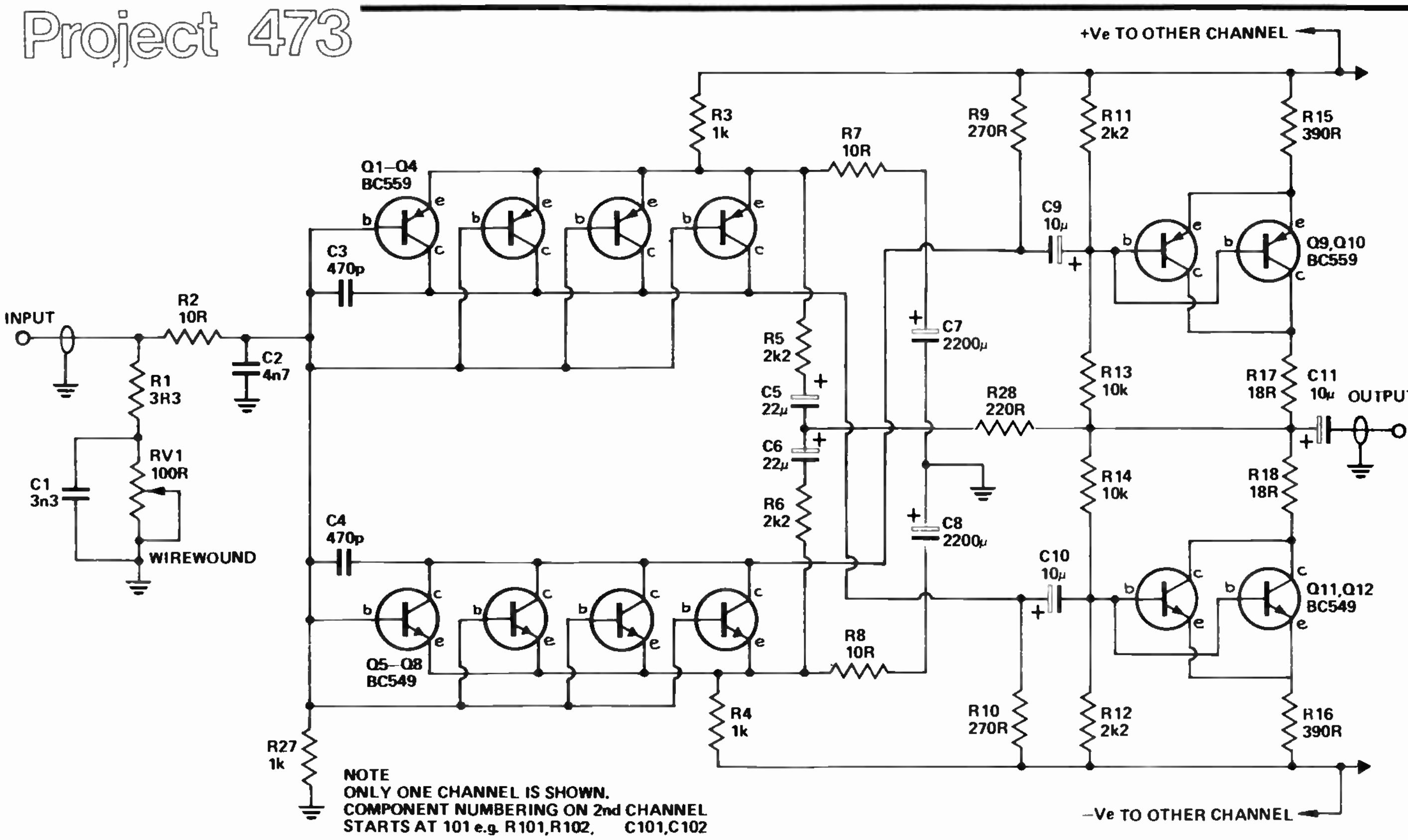
where 'q' is the charge of an electron, in coulombs
'i_{dc}' is the dc current in amps
and 'B' is the noise bandwidth in Hz.

1/f noise has a random amplitude like shot noise but its spectral density has a 1/f characteristic. This means that the noise amplitude increases as frequency decreases and becomes the dominant source of noise at low frequencies. As with shot noise, its equation reveals that it is directly related to the current flowing in the transistor.

$$\overline{I_f^2} = K\frac{(I_{dc})^a}{f}B$$

where 'I_{dc}' is the dc current in amps
'K' and a are constants that are a function of the particular device
'f' is the frequency in Hz
and 'B' is the noise bandwidth.
Notice that as I_{dc} is increased, so too is the 1/f noise (I_f^2)

It is clear from this that, in order to keep noise generated by shot and l/f noise to a minimum, it is necessary to keep the current density in the input stage low. But, as we saw earlier, to obtain the necessary low input impedance we have to increase the emitter current. The solution to this is to use several transistors in parallel to form the input device. This decreases the current density in each of the transistors since the necessary emitter current can be shared by all of the input devices. It also places the impedances of the base-emitter junctions in parallel, further decreasing the input impedance of the first stage. Furthermore, since each transistor is a completely independent noise generator their noise voltage will tend to reduce each other (a process too complex to examine in detail here).



HOW IT WORKS – ETI473

The input stage consists of Q1 to Q8 plus associated circuitry. Q1 to Q4 and Q5 to Q8 are in parallel to reduce the current density providing a low input impedance stage having very low noise. A detailed account of how this works is given in the text.

Capacitor C1 and C2 fix the upper frequency roll-off characteristics as well as shunting the input with the desired load capacitance for the moving-coil cartridge. The configuration of R1 and R2, C1 and C2 was found to give the best loading for a variety of moving-coil cartridges.

The potentiometer RV1 allows the input impedance to be varied over the range most commonly recommended by cartridge manufacturers.

Negative feedback is applied via the network consisting of R28, capacitors C5 and C6 and resistors R5 and R6. Some degenerative feedback for the input stage is applied to the first stage by the emitter resistors R7 and R8. Capacitors C9 and C10 are coupling capacitors to the second stage while bias for this stage is determined by R11, R12, R13 and R14.

The power supply consists of a series regulator Q13 and Q14. The potential dividers R21/R23 and R22/R24 divide the voltage present at the output of the regulator and drive the transistors Q15 and Q16, and the LEDs. The transistor base-emitter junction in series with the LED will drop 0.6 + 1.65 volts. Therefore, whenever the voltage present at the centre of the potential divider tries to increase above 2.3 volts the transistor increasingly, conducts decreasing drive to the pass transistors Q13 and Q14.

This is a relatively low noise regulator since the voltage reference is LED and not a zener diode which is a noisy device. Resistors R19 and R20, together with capacitors C12 and C13 form 6 dB per octave low-pass filters on the supply rails to further reduce noise that may be generated by the regulated supply. . .

This configuration works very well and the noise levels of this preamplifier rival any of the commercially available units.

To see just how difficult it is to obtain a satisfactory signal to noise ratio at these signal levels it is necessary to look at another form of noise called 'thermal noise'. This is caused by the agitation of charged particles in any conductor due to their temperature. Every passive component will generate thermal noise and short of dunking the whole thing in liquid helium to cool it off, there is simply no way of getting rid of it. Thermal noise is given by the equation:

$$\overline{e_R^{\,2}} = 4kTRB \text{ volts}^2$$

where 'T' is the temperature in degrees Kelvin (K).
'R' is the value of the resistance.
'B' is the noise bandwidth
'k' is Boltzmans constant, equal to 1.38×10^{-23} W-sec/K.

From this equation we can calculate the theoretical noise that will be generated by the moving coil cartridge itself. This clearly is the absolute lowest noise figure that is possible with the input stage generating no noise of its own (which is very unlikely!).

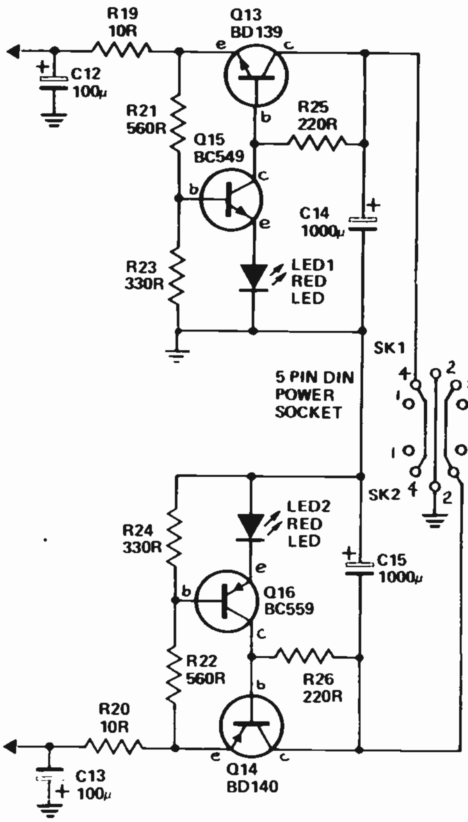
If we let the temperature of the transistor be 300 Kelvin (i.e.: mean atmospheric temperature) and the noise bandwidth be 20 kHz (the hi-fi audio band), then since the dc resistance of the cartridge is about 5 ohms the equation becomes:

$$\overline{e_R^{\,2}} = 4 \times (1.38 \times 10^{-23}) \times 300 \times 5 \times (20 \times 10^3)$$

Therefore $\overline{e_R} = 4.07 \times 10^{-8}$ volts or 41 nV.

So, the thermal noise of the cartridge itself is 41 nV.

Actually, this calculation is not quite right since the noise bandwidth is defined as having a 'brick wall' response. An amplifier with 3 dB point of 20 kHz that is falling at a rate of 6 dB per octave will actually have a noise bandwidth much greater than 20 kHz. Furthermore, if we want to be able to quote noise figures to enable comparison between different input stages, it is valuable to quote noise voltages independently of noise bandwidth. This can be done quite easily by dividing the noise voltage by the square root of the bandwidth. The dimensions of this new figure will be "volts per root Hz",



and our result for the thermal noise of a moving coil cartridge becomes:

$$\frac{41}{\sqrt{20\,000}}\ \mathrm{nV}/\sqrt{\mathrm{Hz}}$$

or 0.29 nV/√Hz

Now, if we are aiming at a signal to noise ratio of 70 dB with respect to a signal voltage of 150 nV (0.15 mV), which is the expected signal level at a recording velocity of 10 cm/sec., then the equivalent input noise of the amplifier will be given by the equation:

$$-70 = 20\log\left(\frac{N}{0.15\times10^{-3}}\right)$$

and is equal to 0.33 nV/√Hz.

The necessary equivalent input noise is in the same order of magnitude as the noise being generated by the cartridge itself!

Designing an input stage with this sort of noise isn't easy, especially when it is considered that the noise generated by even the quietest transistor is in the order of several nV/√Hz for usable emitter current. This is substantially worse than the requirement.

Performance features

The total equivalent input noise of this unit was measured at 0.3 nV/√Hz. With respect to a noise bandwidth of 20 kHz, this corresponds to an input noise of 42 nV, giving a signal to noise ratio with respect to an input signal of 150 nV (0.15 mV) of 71 dB. At this level, the noise generated by the cartridge itself will be one of the dominant noise sources.

The circuit uses a symmetrical configuration with NPN and PNP transistors set up in such a way that asymmetrical distortions tend to cancel. Normally distortion products are generated differently for positive and negative signal excursions and this tends to produce second harmonic distortion products. The configuration used in this circuit results in very low second and third harmonic distortion. This has enabled a total harmonic distortion figure of around 0.0015% to be obtained.

The problem with quoting distortion figures of this order is that they are too low to be measured directly, being well hidden under the noise level. The only way a figure can be obtained is to remove the overall negative feedback, measure the distortion and then divide by the gain difference when the feedback is reapplied. Unfortunately, feedback does not affect all the distortion products equally, but the figure is still meaningful.

Another advantage of the symmetrical design of the input stage is that it does away with the need for an input capacitor. This is a definite advantage when dealing with low input impedances since the value of the capacitor would have had to be very large to obtain a flat frequency response at low frequencies.

The signal voltages present in the preamplifier are naturally extremely low and for this reason the power supply has been kept as a separate unit to reduce the possibility of 50 Hz induction from the power transformer.

A voltage regulator supplies the necessary ±6 volts. As it is critical to achieve low noise it is important that the regulator does not put noise onto the supply rails which would degrade the noise performance of the unit. Normally the voltage reference used for regulators of this type is a zener diode but, as the zener is reverse biased, it generates a comparatively large amount of noise. In this design an LED was used as the voltage reference. A red LED operated in the forward-biased mode drops a constant 1.65 volts and generates very little noise.

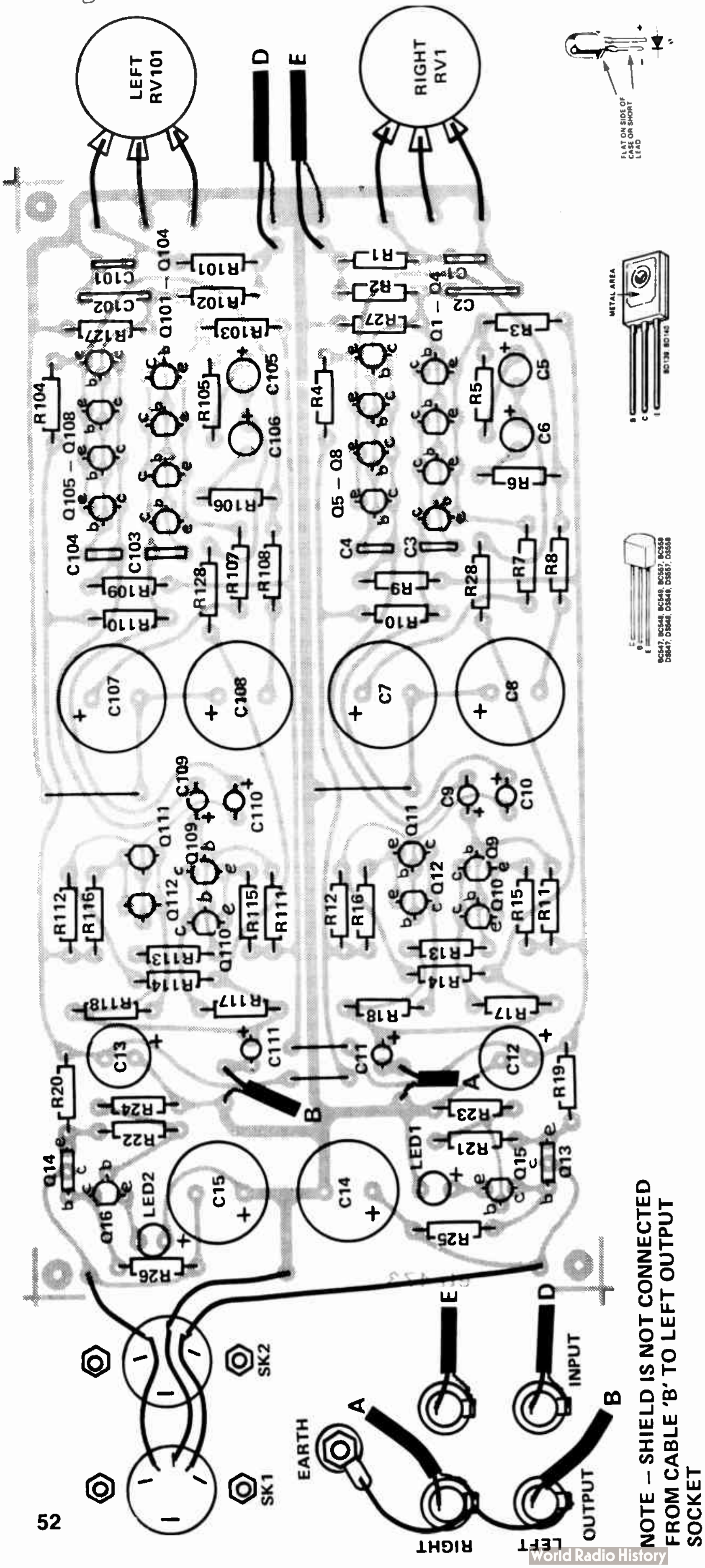
Construction

Construction is relatively straightforward since most components are on the mounted pc board. Other construction methods are possible but performance may not match that of our prototype.

Mount the resistors and capacitors first, followed by the transistors. Since there are quite a few transistors on the board placed close to each other, don't make the mistake I did and get them mixed up! Cut the necessary lengths of shielded cables and solder them onto the board keeping the ends as short as possible. Solder the necessary lengths of hookup cable to the board and after checking all components mount the board in the chassis.

I used a diecast aluminium box and quite frankly wish I hadn't. The shielding to external magnetic fields really isn't good enough. I found I had to be careful where the preamp was placed or it would pick up hum from the magnetic field produced by the power amp's transformer. Use a steel box if you can, if not, just be careful where it is placed.

Once the board is mounted in the chassis, the pots and rear panel hardware can be mounted and the wiring completed according to the wiring layout diagram shown. Here again I came unstuck. The first system I used to ground the shielded cables caused a monumental hum loop (and I still don't really understand why!). The final method tried is shown in the wiring diagram and this works very well. The shielded cables coming from the outputs on the board have only one of their shields connected to the output RCA sockets which are wired together and connected to the chassis at the ground terminal. This type of terminal is supplied with the necessary hardware to insulate them from the chassis. In this case however, we want the terminal to connect firmly to the case to provide the necessary ground connection. It is important that the RCA sockets be insulated from the case and that the ground connection made to them is according to the wiring diagram. If the unit is going to be used with the recommended power supply there should be no hum problems. This power supply, ETI-557, is described earlier in this book. It is wired so that the 0 volt line is not connected to the chassis of the power supply. This is important, otherwise a hum loop around the units' mains grounds will result. If you wish to use a power supply other than the 577 then it will be necessary to ensure that the 0 volt line from the supply does not connect to the power supply chassis. Do not 'cure' the problem by disconnecting the ground wire at the 240 volt plug as this will remove any ground connection from the power



supply chassis. This is not only dangerous, it's illegal.

Powering up

Before turning the unit on make a final check of the board. Check the orientation of the transistors, electrolytic and tantalum capacitors and the LEDs. If all is right, turn down the volume control completely and switch the power supply on. The LEDs in the

PARTS LIST - ETI 473

Resistors all ½W, 5%

R1, R101 . . . 3R3
R2, R102 . . . 10R
R3, R4, R103,
R104 . . . . . . 1k
R5, R6, R105,
R106 . . . . . . 2k2
R7, R8, R107,
R108 . . . . . . 10R
R9, R10, R109,
R110 . . . . . . 270R
R11, R12,
R111, R112 . . 2k2
R13, R14,
R113, R114 . . 10k
R15, R16,
R115, R116 . . 390R
R17, R18,
R117, R118 . . 18R
R19, R20 . . . 10R
R21, R22 . . . 560R
R23, R24 . . . 330R
R25, R26 . . . 220R
R27, R127. . . 1k
R28, R128. . . 220R

Capacitors

C1, C101. . . . 3n3 ceramic
C2, C102. . . . 4n7 ceramic
C3, C4, C103,
C104 . . . . . . 470p ceramic
C5, C6, C105,
C106 . . . . . . 22µF 16V tantalum
C7, C8, C107,
C108 . . . . . . 2200µF 25V electro
C9–C11,
C109–C111 . . 10µF 16V tantalum
C12, C13. . . . 100µF 25V electro
C14, C15. . . . 1000µF 25V electro

Transistors

Use only types specified - substitutes may result in inferior performance.

Q1–Q4,
Q101–Q104. . BC559
Q5–Q8,
Q105–Q108. . BC549
Q9, Q10,
Q109, Q110 . . BC559
Q11, Q12,
Q111, Q112 . . BC549
Q13. . . . . . . BD139
Q14. . . . . . . BD140
Q15. . . . . . . BC549
Q16. . . . . . . BC559

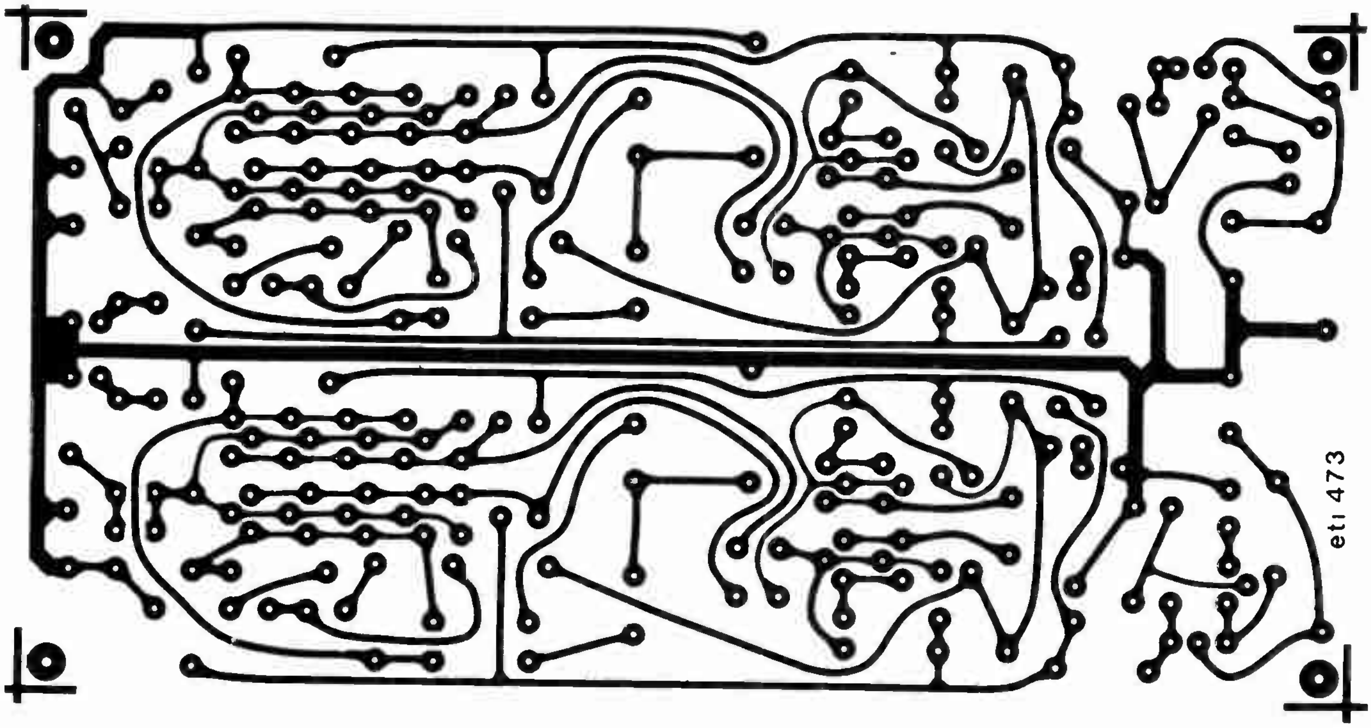
LED1, LED2 . standard red LED

Potentiometers

RV1, RV101 . 100R wirewound linear

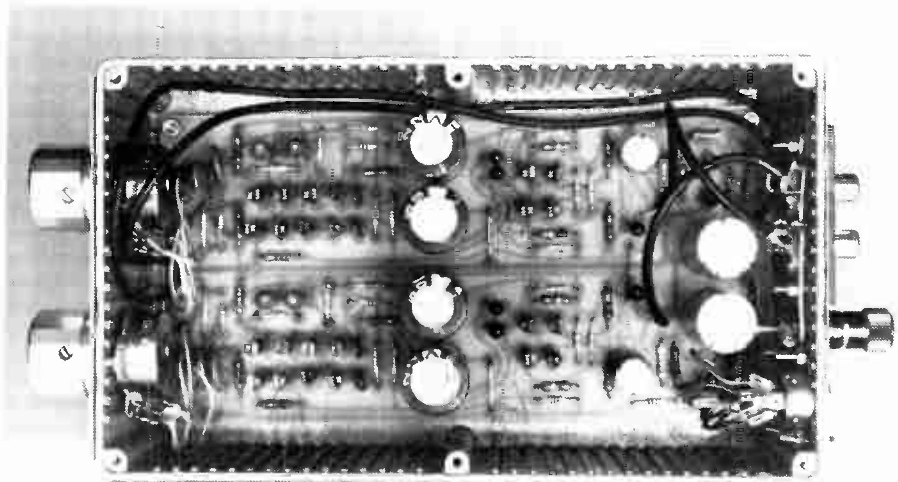
Miscellaneous

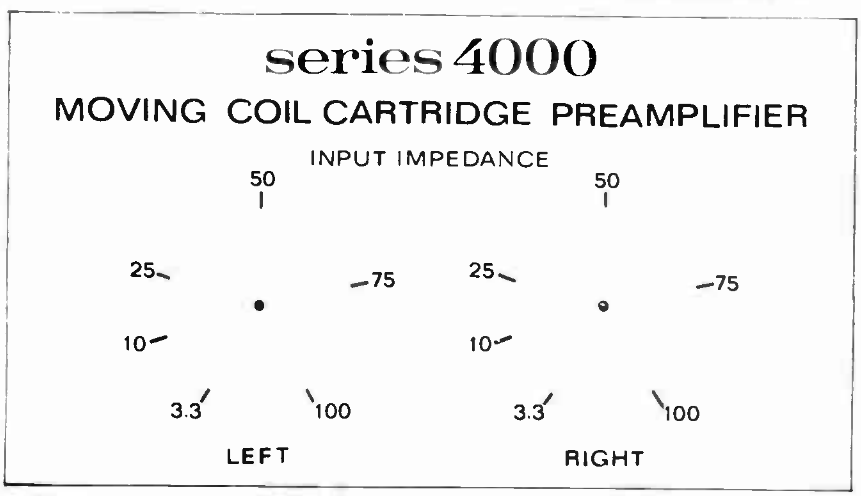
SK1, SK2 . . . 5 Pin DIN socket
Four RCA sockets (insulated from case), One black terminal, mains lead, plug and mains cord securing grommet, two knobs, box to suit, 190 x 60 x 110 mm.

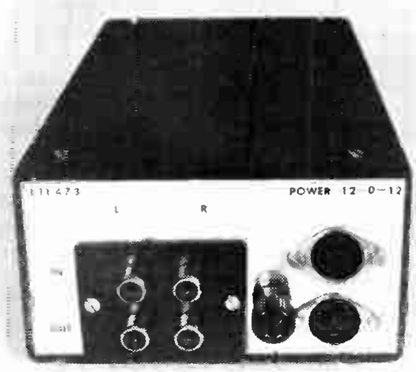


preamp's regulator should come on immediately. I used standard RCA to RCA cables from the output of the preamp to the phono input and had some trouble with hum induction into the leads. Fortunately, we had been sent a set of Audio-Technica type AT620 cables for evaluation several days before and these cured the problem completely.

Perhaps I am biased, but the sound quality of this preamp is extremely good! Using a Nakamichi MC1000 cartridge, this preamp showed distinct improvement over the transformer I was using previously. There is an openness that never existed before and the bass end showed a great improvement being firmer and much more defined. I trust you'll be as satisfied with your project as I have been. ●







High-to-low impedance 'interface' to suit the ETI-470 60 watt amp

The popularity of our 60 watt low distortion amplifier module (May '79) has exceeded all expectations. To achieve the amplification 'accuracy' these power amps are capable of, the drive impedance must be very low — in the order of five to ten ohms. Our previous preamps, the 422 and 482, and many preamps available, generally have a medium to high output impedance and will not properly mate with the 470. This interface provides the necessary impedance conversion, allowing these amps to be used with many existing preamp designs.

Phil Wait

DESIGNED primarily for use with our Series 4000 stereo amplifier, the 470 low TID 60 watt amplifier module has found its way into the most surprising applications — from a dc motor drive to discos in central Africa. Thousands of the modules have been built, occasional output transistor shortages not withstanding, in Australia, New Zealand, Europe, Africa, Canada and the UK.

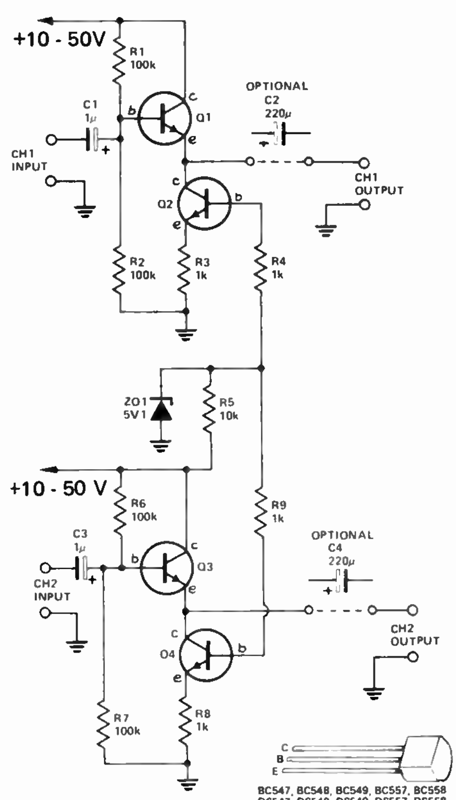
Although the 470 module was designed to be driven from a low impedance source, it is obvious from readers' letters that there are many who want to use it with existing equipment having a preamp with a high output impedance.

This project describes a two-channel (stereo) interface for driving up to two 470 modules (per channel) from a high impedance source, and can in fact be used in any application requiring a very low impedance drive at audio frequencies.

The input stage of the 470 consists of an emitter — coupled darlington pair with the input signal fed to the non-inverting input and the feedback connected to the inverting input. To reduce high frequency intermodulation the slew rate of this stage is limited by placing a 470n (0.47μ) capacitor between the two bases.

The input impedance varies with frequency, from a few thousand ohms at quite low frequencies to hundreds of ohms at the high frequencies, where the effect of the slew limiting capacitor becomes apparent.

If the stage is driven from a high impedance source, the output of the driving current will be loaded down at high frequencies by the reduced input impedance of the amplifier, causing high frequency distortion. This is why we specified a low impedance driving source for the 470, and designed our preamplifier accordingly.



Interface design

The circuit for our interface uses two emitter followers (one per channel) with constant-current generators in the emitters referenced from a zener-regulated supply voltage.

The easiest way to convert from a high impedance to a low impedance with little attenuation is with an emitter follower. The input signal is fed into the base of a transistor and the output taken from the emitter, the collector being tied to the supply. Emitter followers have a high input impedance and very low output impedance. The output impedance is roughly the value of the emitter resistor divided by the beta of the transistor.

To allow the circuit to be used with the power amplifier or with the driving source the circuit must be able to operate over a very wide range of dc supply voltages as found in graphic equalisers, organs, preamplifiers and such.

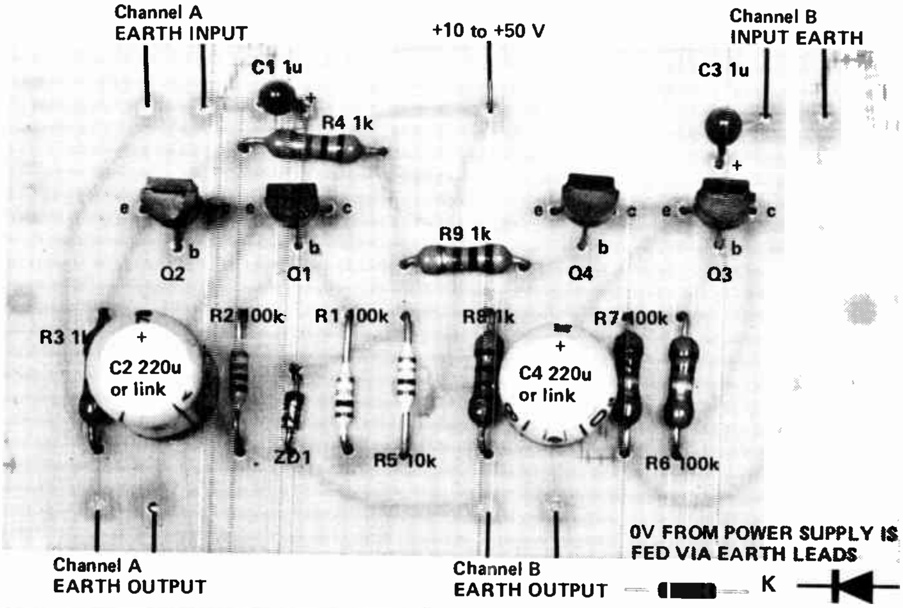
To limit the supply current and dissipation of the emitter follower when

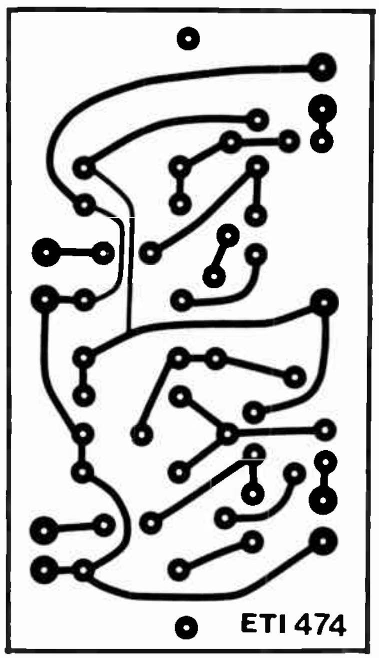
HOW IT WORKS – ETI 474

The circuit consists of two emitter followers, Q1 and Q3, with constant current generators in their emitters. The constant current generators share the same voltage reference, ZD1.

The reference voltage, 5.1V is derived from ZD1, and fed to the bases of Q2 and Q4. The voltage on their emitters is then set at 4.4V. The transistors will always pass the exact amount of current required to maintain this voltage on the emitters, regardless of supply voltage.

The input signal is fed to the bases through dc blocking capacitors C1 and C3, and the output is taken from the emitters directly or via the optional blocking capacitors C2 and C4. The gain of the circuit is a little less than unity.





PARTS LIST - ETI 474

Resistors all ½W, 5%
R1, R2 . . . . . 100k
R3, R4 . . . . . 1k
R5 . . . . 10k
R6, R7 . . . . . 100k
R8, R9 . . . . . 1k

Capacitors
C1. . . . . 1µ 35V tantalum
C2. . . . . 220µ 35V electro (optional)
C3. . . . . 1µ 35V tantalum
C4. . . . . 220µ 35V electro (optional)

Semiconductors
ZD1. . . . . . . 5V1 400mW zener diode

Q1—Q4. . . . . BC107, BC547, DS547 or similar

Miscellaneous
ETI 474 pc board.

used with a high supply voltage we used a constant current generator (Q2, Q4) in each of the emitters in place of the normal emitter resistor. The use of a constant current generator also increases the input resistance and decreases the output resistance. A current of about four milliamps flows through the transistors for all supply voltages above five volts.

The output capacitors (C2, C4) provide dc isolation for the output, but since the 470 modules already have an isolation capacitor (C1), they can be left out and the pc board bridged with a length of tinned copper wire. If any other connection is made from the output, for auxilliary equipment, the capacitors should be left in.

If the capacitors are removed it will be necessary to replace the input capacitor on the 470 power amplifier (C1) with a 220µ, 35 volt electrolytic oriented with its positive lead towards the input terminal.

Construction

Construction is straightforward, the only thing to watch is the orientation of the transistors and the zener diode. The unit can be mounted with the power modules and run from their supply or mounted with the driving circuit. Input and output connections should be via shielded cables which also carry the power supply earth on the braid to avoid earth loops.

If only one power module is to be driven, as with an electronic organ, the pc board can be cut in half and only one channel assembled. ●

Hints and tips for the ETI-470 60W Module

MOST PEOPLE haven't had problems with their 470 module, but inevitably there are some who do. From calls and letters to our reader enquiry service we have identified five areas of trouble.

1) The earth rail on the amplifier must be returned to the 0V rail on the power supply. If this is not done the input transistors and their current source (Q1-Q5) will be destroyed. This is probably our failing as, although it is obvious to most people, it was not indicated on the circuit given on page 28 but was indicated in the wiring diagram of the Series 4000 amplifier.

2) It can be seen from the overlay that the base lead of Q5 must be slightly bent to fit the pc board. The transistor can easily be inserted the wrong way round. Watch this.

3) The darlington output transistors **must** have a good heatsink. Always make sure the thermal contact between the transistor and the heatsink is good. Use a thermal compound (such as Bevaloid GS13), but not too much — just a smear on either side of the mica washer. Use a metal, rather than a nylon screw with an insulating bush, to fasten the transistor — a nylon one will stretch under tension. Make sure the heatsink is smooth and flat, curved or sandblasted heatsinks will not make good thermal contact with the transistor body.

4) Make sure that the transistor Q8 has a good thermal contact to the heatsink. It must be the same heatsink as the output transistors.

5) Never, never run the amp without a heatsink, even if only to set the bias.

Overheating of the output devices due to poor heatsinking will result in thermal run away which will blow the fuses but will probably not damage the output transistors provided the two amp fuses are in circuit. Faults where the amplifier operates correctly for a while then blows fuses, will probably be due to poor heatsinking.

Most transistors in the amplifier are designed to run quite warm in normal operation.

No problems have become apparent with the preamplifier (ETI-471).

Series 4000 four-way loudspeaker

David Tilbrook

This project is the first in a series of loudspeaker projects designed to complement our Series 4000 range of quality hi-fi projects.

LOUDSPEAKERS still remain the weakest link in the hi-fi chain and the total sound of any system will depend more on the loudspeakers than any other single hi-fi component. So it is important to get the best loudspeakers, even if this means accepting a slightly lower performance amplifier or turntable. In most systems the performance of the cartridge, turntable and amplifier greatly exceeds that of the loudspeakers so an improvement in the loudspeaker department will often yield a radically improved system.

Unfortunately, there are very few really good kit loudspeakers. This project is an attempt to rectify that situation by providing a loudspeaker suitable for home construction that rates amongst the best available. This is not an inexpensive project — the driver and crossover cost being around $400 — but the finished project will rival commercial units at three times the price.

Choosing the drivers

In order to build a good loudspeaker it is obviously important to use good drivers, but availability is just as important a criterion as performance. For this reason we had a close look at the drivers commonly available in Australia and finally decided to use drivers from the huge range of Philips loudspeakers, some of which were not available in this country at that time. Philips agreed to stock the drivers we decided on and these form the basis of the 4000 series of loudspeakers.

The 4000/1 is a four-way sealed enclosure loudspeaker using 12 dB/octave crossover slopes. The original design for our prototype used an 18 dB/octave M-derived crossover (see our article 'Principles and problems in loudspeaker design' which starts on page 130) but it was enormously expensive and complex and would have contributed little to the overall sound finally achieved with the 12 dB/octave crossover. The four-way approach allows closer control over the final frequency response than does a three-way. More importantly a major part of the mid-range normally handled by the woofer can be dedicated to a separate mid-range driver. The basic design idea was to use the woofer only up to 150 Hz. A separate mid-range driver would then take over up to 750 Hz where a second mid-range would come in. The lower mid-range driver, crossing in at 150 Hz needs a usable response down to around 60 Hz (i.e. one octave) so that the crossover region will have a reasonably flat response. Similarly, the woofer crossing out at 150 Hz needs to have a usable response to at least 300 Hz.

After a great deal of testing it was finally decided to use the Philips AD12250/W8 unit for the woofer. This is a 100 watt driver with a free air resonance of 26 Hz. When mounted in the enclosure the fundamental resonance rises to around 31 Hz, an excellent figure. This driver seems to have a bad hole in its response at 350 Hz but this is unimportant in this loudspeaker.

The AD70601/W8 unit was chosen as the lower mid-range as it has a free air resonant frequency at 45 Hz. This driver is actually a woofer and does not have the integral sealed enclosure common to many mid-range drivers. The enclosure must be provided by the cabinet construction and the volume chosen in the 4000/1 increases the 45 Hz fundamental resonance of this driver to around 55 Hz, which is ample.

The response between 750 Hz and 3 kHz, where the tweeter takes over, is handled by the latest Philips dome (AD02161/SQ8) mid-range. This driver has a 50 mm textile dome giving a good frequency response and wide dispersion at higher mid-range frequencies.

Above 3 kHz the AD01610/T8 tweeter is used. We tested a large range of Philips tweeters and this was the best, followed closely by the AD01605/T8, which suffered a little from roll-off of the frequencies above 10 kHz.

The 4000/1 loudspeaker, without the front grille, showing the drivers. It stands about one metre tall.

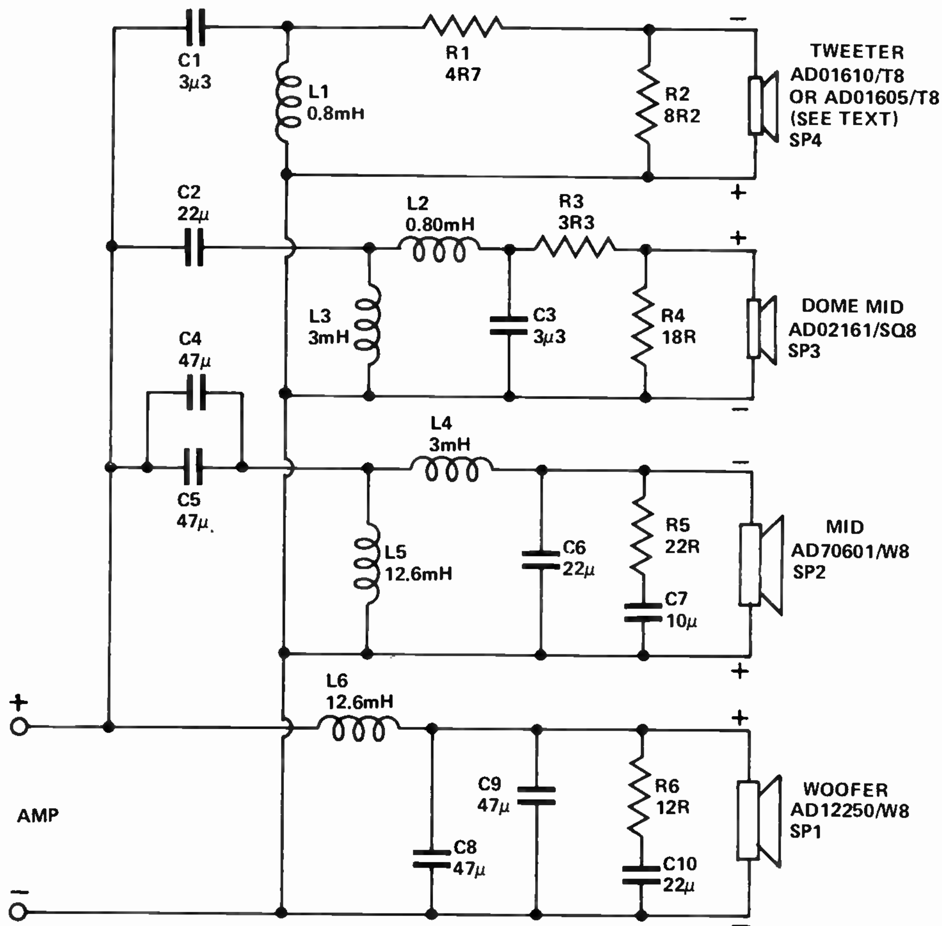
Construction

If you are constructing the boxes yourself start by assembling the sides, top, bottom and back of the cabinet. The bottom panel is placed 100 mm above the bottom of the box and the cavity formed under the box can be used to mount the crossover instead of putting it inside the box as is the usual practice. Now insert the two pieces of timber that form the mid-range enclosure. It is essential that there is a perfect seal between the bass and mid-range chambers, as well as between these two chambers and the outside air. Line every joint carefully with caulking compound

HOW IT WORKS.
The input signal from the output of the amplifier is fed to the 4 way crossover that divides the signal into the 4 different frequency bands covered by each of the drivers. The loudspeaker cabinet is divided into two sections, the larger one forming the bass chamber for the woofer and the smaller one forming the midrange chamber. These two chambers are sealed from each other so that interactions cannot occur between the back radiations of the woofer and lower midrange. The other two drivers have their own enclosures as an integral part of the driver For a detailed account of the design approach and the problems that occur in loudspeaker design, read 'Principles and problems in loudspeaker design' page 130).

or glue so that no possiblity of an air leak exists. This is probably the best stage of the construction to drill the holes for the wiring to the loudspeakers. I used two cores of 240 volt three-core mains cable for this purpose, mainly because a round hole could be drilled and the cable squeezed through it to make a reasonable seal. Three holes need to be drilled in the bottom of the midrange chamber to allow for cables for the two midrange drivers and the tweeter. Cut suitable lengths of 240 V mains cable and insert these through the holes. Seal between the cables and the holes with sealing compound or a glue like Silastic. If the crossover is to be mounted under the loudspeaker, drill four holes through the bottom of the box and run the cables exactly as with the mid-range enclosure. Drill the holes so that they are closer to the rear of the box to allow ample room for mounting of the crossover. The input terminals should be mounted on the back of the enclosure, below the bottom panel if the crossover is mounted under the loudspeaker.

It is not necessary to have the front baffle removable since the drivers are external mounting types. It is probably easier to cut the holes for the drivers before mounting the baffle onto the front of the cabinet. The base panel and midrange enclosure panel should have been cut so that 38 mm remains between these and the front edge of the side and top panels. When the front panel is fitted, 19 mm should remain between the front of the baffle and the front edge of the sides and top.



Circuit diagram for the four-way system. Driver polarity is important. Note that the "dome mid" driver, AD02161/SQ8, is available in two models, the other being AD02160/SQ8, which is different in appearance but electrically equivalent.

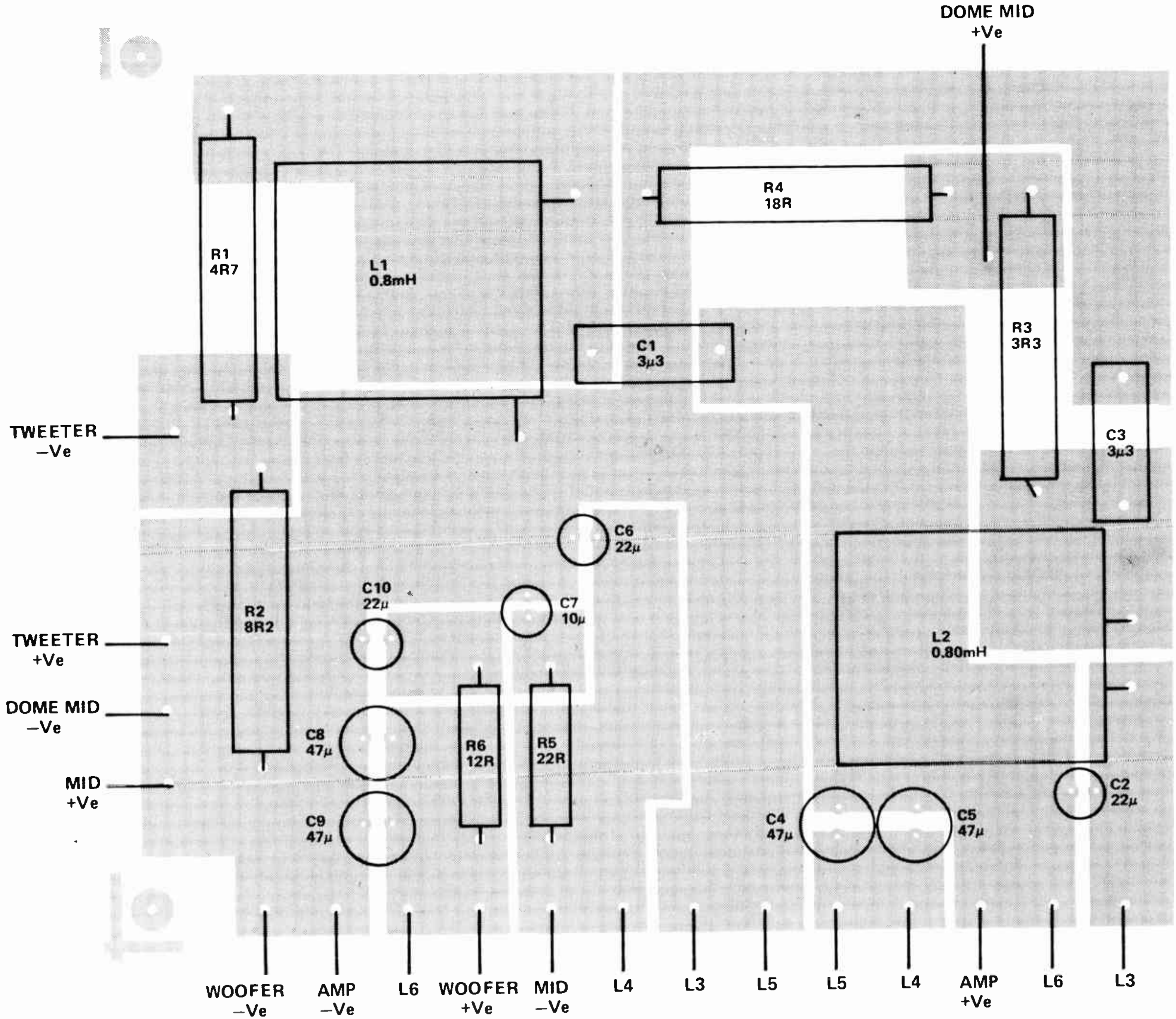
This space will be taken up by the grill cloth frame. Seal the remaining joints between the front baffle and the rest of the box. The only remaining part of the box construction is to attach the small 100 mm high wooden panel to the bottom of the box. The front grill is made by constructing a rectangular frame that fits into the remaining cavity on the front of the baffle. Stretch the grill cloth (use proper speaker grill material to avoid absorption of the treble) tightly over the frame.

If you have purchased a kit of ready made boxes it will still be necessary to drill the holes for the cables and to seal the box thoroughly with some sealing compound. If the slightest leak exists between the bass and mid-range chambers the large pressure increases created in the bass chamber will force the mid-range to vibrate, causing distortion.

The last stage before mounting the drivers is to line the box with 25 mm thick loudspeaker innerbond. Line the back, sides, top and bottom of both the bass and mid-range chambers. Attach the innerbond firmly to the sides of the box using tacks or thin nails and glue.

The tweeter and dome mid-range drivers are supplied with mounting washers so that good seals can be made between the drivers and the baffle. Use adhesive foam tape available from most hardware stores, to make a good seal around the lower mid-range unit and the woofer. Stick the tape to the front of the baffle around the edge of the holes cut for the woofer and mid-range so that when the drivers are mounted a good seal results.

Solder the wires to each of the drivers making sure you know which wire is connected to the positive terminal on the loudspeaker. This terminal is marked on the driver either by a red terminal or a red dot near one of the terminals. Mark the other ends of the cables so that it is clear which cables connect to which drivers. *This is important;* if the outputs of the cross-over are connected to the wrong drivers ▶

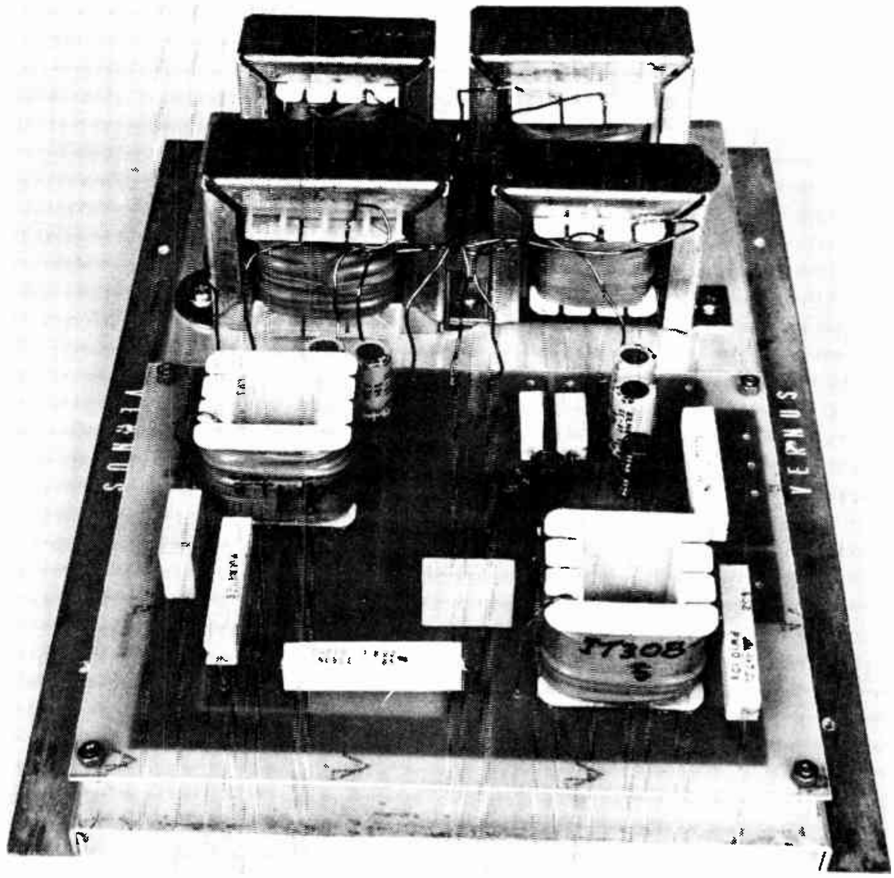

DOME MID +Ve
R1 4R7
L1 0.8mH
R4 18R
R3 3R3
C1 3µ3
C3 3µ3
TWEETER –Ve
R2 8R2
C6 22µ
C10 22µ
C7 10µ
L2 0.80mH
TWEETER +Ve
DOME MID –Ve
C8 47µ
R6 12R
R5 22R
MID +Ve
C9 47µ
C4 47µ
C5 47µ
C2 22µ
WOOFER –Ve
AMP –Ve
L6
WOOFER +Ve
MID –Ve
L4
L3
L5
L5
L4
AMP +Ve
L6
L3

this could result in damage to the drivers.

Once all of the drivers are mounted the final stage is the construction and mounting of the crossover. If the crossover is mounted inside, instead of under the box it will be necessary to leave mounting of the woofer until last. After all of the drivers have been mounted connect a 1.5 volt battery to the woofer wires and watch the lower mid-range cone. If it moves, the seal between the bass and mid-range chambers is not complete.

The inductors used in the crossover are too big to be mounted on the pc board. All the other crossover components are on the pc board. Start construction of the crossover by mounting and soldering the capacitors to the pc board. Next solder the resistors into place spacing them approximately 10 mm off the board. This is necessary to prevent charring the pc board should these resistors get hot when the speaker is used with high power amplifiers. The remaining two inductors should be glued onto the pc board and then the leads soldered.

The prototype crossover was mounted on a sheet of aluminium 200 mm by 330 mm, but this is optional. If you elect to use this method of construction screw the remaining four inductors onto the aluminium sheet and solder the leads from these onto the pc board. Solder the leads from the drivers and input terminals onto the pc board and mount the pc board onto the aluminium base using 6 mm spacers. Finally, the whole crossover can be screwed to the bottom of the loudspeaker box. If you are not using the aluminium base the pc board and inductors are mounted directly to the bottom of the loudspeaker box. The advantage of using the aluminium base is so that the crossover can be handled as one complete unit.



We mounted the crossover network assembly on an aluminium plate, bent as shown. The whole assembly was then screwed to the bottom of the loudspeaker and each driver connected as per the overlay.

PARTS LIST - ETI 496

The following is a parts list for one only loudspeaker so two of every component will be needed for a stereo pair.

Drivers
SP1 . . . . Philips AD12250/W8
SP2 . . . . Philips AD70601/W8
SP3 . . . . Philips AD02161/SQ8
Philips AD02161/SQ8 or AD02160/SQ8
SP4 . . . . Philips AD01610/T8 or AD01605/T8, see text.

Inductors
L1, L2 . . . . . 0.8 mH max dc resistance 0.5 R
L3, L4 . . . . . 3.0 mH max dc resistance 0.5 R
L5, L6 . . . . . 12.6 mH max dc resistance 0.7 R

Capacitors
C1. . . . . 3μ3 polycarbonate
C2. . . . . 22μ bipolar electrolytic 50 V
C3. . . . . 3μ3 polycarbonate
C4, C5 . . . . . 47μ bipolar electrolytic 50 V
C6. . . . . 22μ bipolar electrolytic 50 V
C7. . . . . 10μ bipolar electrolytic 50 V
C8, C9 . . . . . 47μ bipolar electrolytic 50 V
C10. . . . . . . 22μ bipolar electrolytic 50 V

Resistors
R1 . . . . 4R7 10 W 5%
R2 . . . . 8R2 10 W 5%
R3 . . . . 3R3 10 W 5%
R4 . . . . 18R 10 W 5%
R5 . . . . 22R 5 W 5%
R6 . . . . 12R 5 W 5%

Miscellaneous
pc board . . . . ETI 496
Wire, one pair of spring terminals, particle board, screws, glue, etc.
Speaker grill cloth, innerbond.

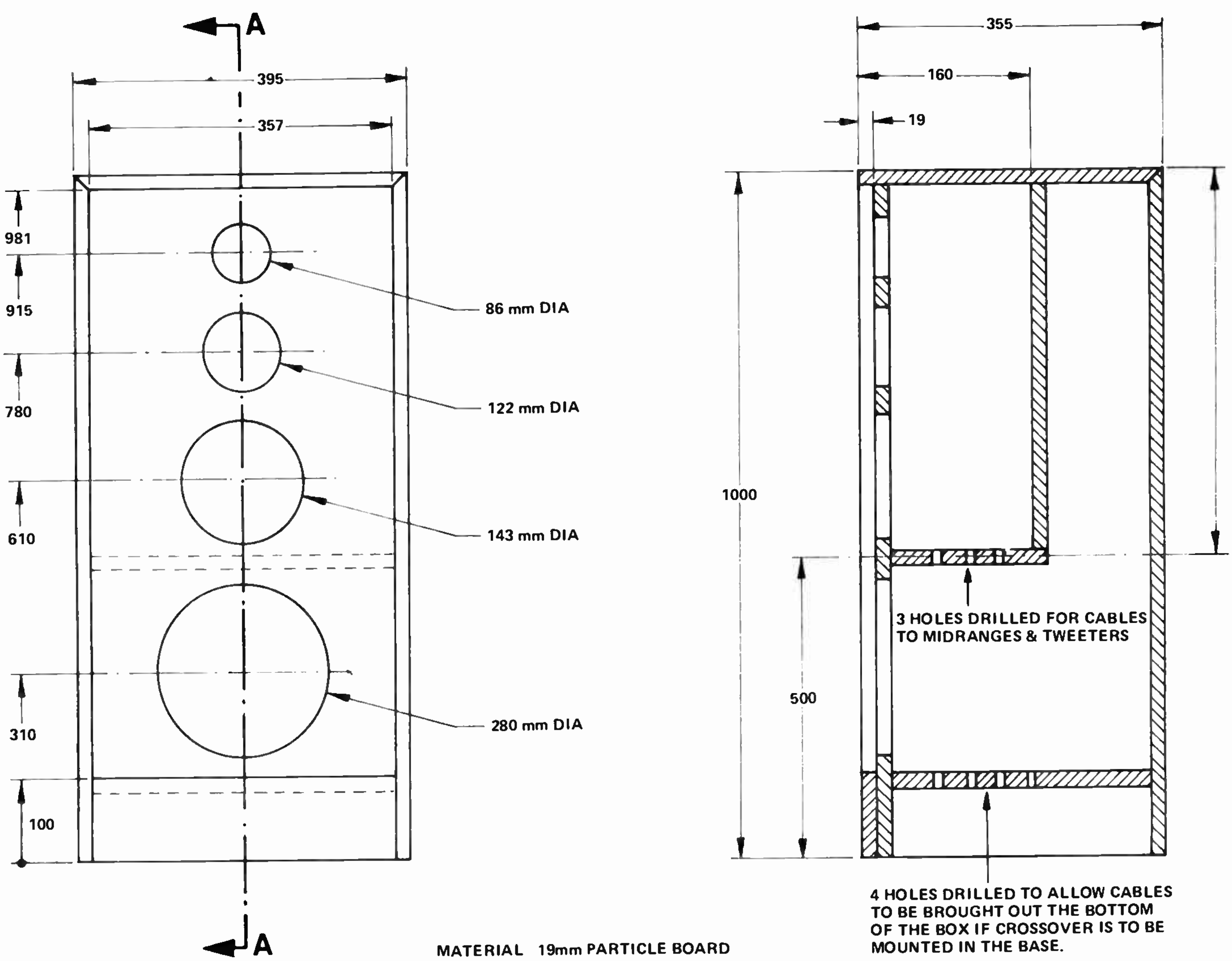
Powering up

Before connecting the loudspeaker to an amplifier touch the input of the loudspeaker to a *single 1½ volt* penlight battery. With the positive of the battery connected to the positive input (red terminal) of the loudspeaker the woofer cone should move forward and the loudspeaker should make a loud thump. Listen to all the drivers separately while connecting and disconnecting the battery to check that all of the drivers are operating. Don't use a battery any bigger than 1½ volts for this test or you could damage the woofer.

If all is well, connect the speakers to an amplifier and turn the volume up slowly.

Performance

Power handling figures for loudspeakers ▶



Complete cutting and assembly details for the four-way loudspeaker box. It is important that all joints be well sealed.

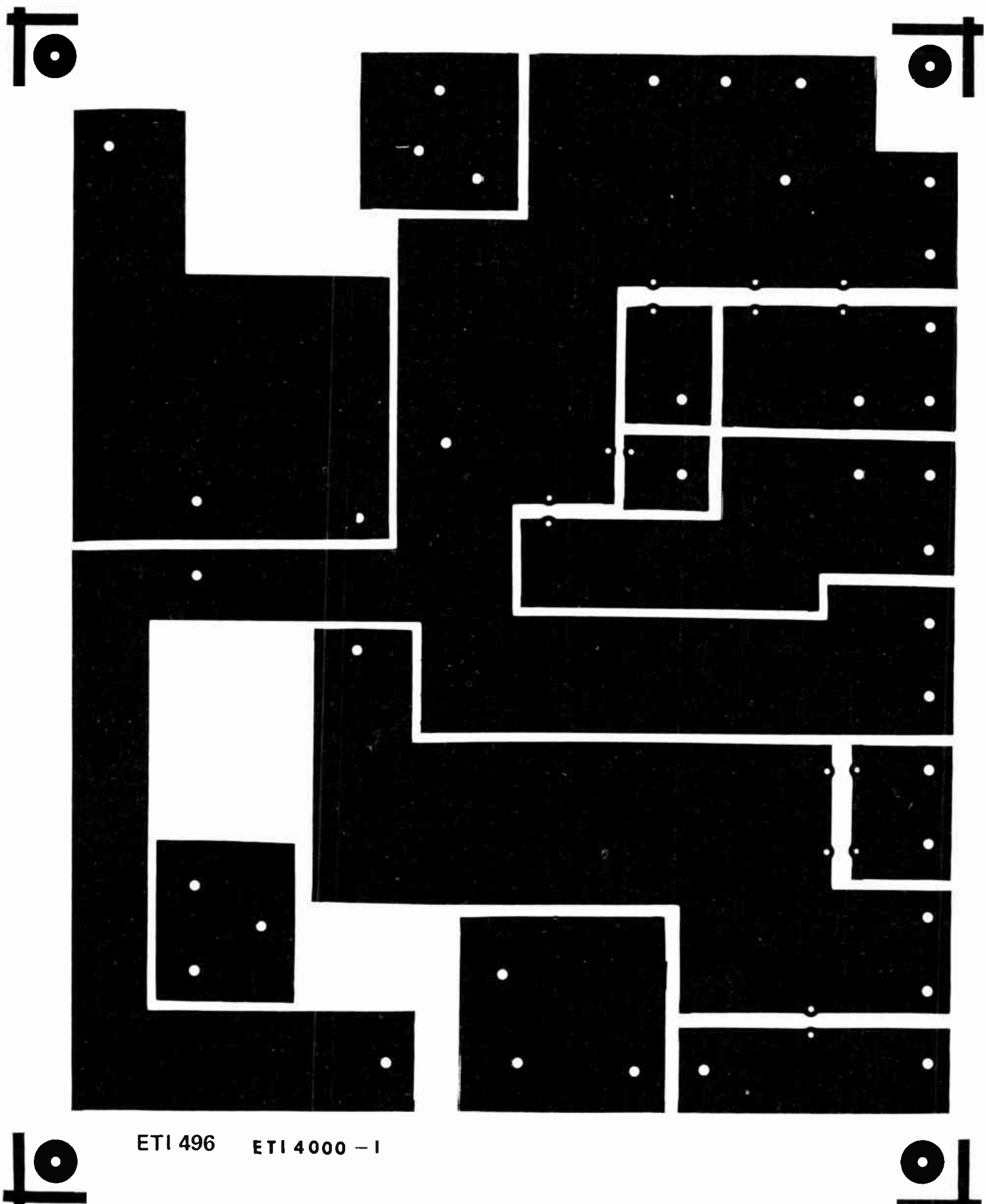
are a very dubious quantity. Some manufacturers (not many) quote continuous sine wave power handling at a particular frequency, but it is doubtful that this is a really meaningful figure. Probably the best way of measuring power handling is with pink noise. This is a type of noise which contains equal energy per octave over the entire audio range. Using this technique, these loudspeakers are rated at 100 watt power handling. The bipolar electrolytic capacitors used in the crossover are rated at 50 volts. This corresponds to 156 watts into an 8 ohm load so this should be considered the *absolute maximum* power for the loudspeaker. It is sometimes mistakenly thought that the power handling figure represents the power below which the loudspeaker cannot be damaged. The most dangerous condition for any loudspeaker is a heavily clipping amplifier. In this state the output of the amplifier approaches dc and even a 20 watt amplifier can do irreparable damage if operated incorrectly.

Your ears are the best indication that the loudspeakers are operating safely. If the sound becomes distorted or unpleasant at higher power levels, turn down your amplifier. Nine times out of ten it will be the amplifier and not the loudspeaker that is running out of power.

The 4000/1 loudspeaker has been designed in accordance with extensive tests that reveal the "ideal" frequency response characteristics for most listening environments. This response is not flat but has a tapered top end, so that the extreme treble is attenuated slightly with respect to the mid-range and bass.

The subjective test revealed just how good the loudspeakers are. The frequency response is smooth and extended and the bass and treble are present only when they should be!

Above all, the sound is clean and easy to listen to for extended periods, even at very high listening levels. I hope you get as much enjoyment from your 4000/1 speakers as I have. ●


ETI 496
ETI 4000 — I

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Three-way loudspeaker system — the 4000/2

Second in our line of quality hi-fi speakers for the do-it-yourself enthusiast.

David Tilbrook —

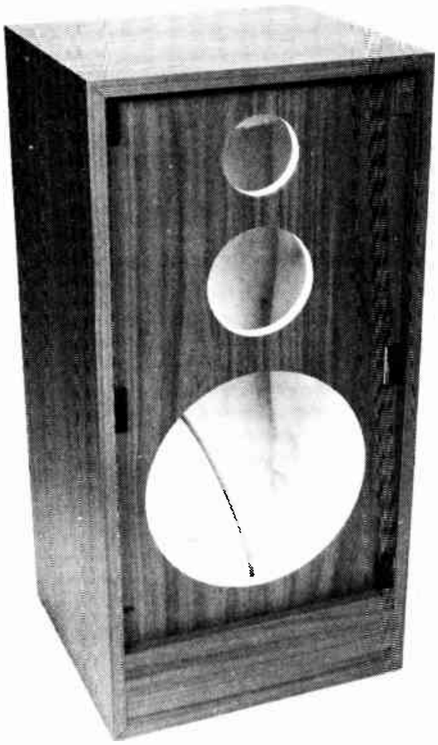
IN THE PREVIOUS ARTICLE, we described the first in a range of loudspeaker projects, the ETI 4000/1. This was a four-way loudspeaker that used a separate driver and mid-range enclosure to cover the frequency range of 150 Hz to 700 Hz. The objective of this design was to remove the lower mid-range portion of the audio spectrum from the woofer so that the large cone excursions typical of bass reproduction would not cause intermodulation distortion by mixing with the mid-range. This technique works very well and the 4000/1 is capable of some of the cleanest sound possible with present technology.

There is, however, a demand for a lower-priced system and we are introducing the 4000/2 loudspeaker to meet this requirement. Much of the cost of the bigger loudspeaker was involved in the lower mid-range unit and its enclosure, and in the complexity of the crossover. The 150 Hz crossover point between the woofer and the lower mid-range necessitated the use of very large inductors and capacitors, and these are expensive.

By eliminating the lower midrange unit and its associated hardware and using a more conventional design approach, we have been able to keep the cost of the 4000/2 to a minimum, while still achieving a loudspeaker capable of excellent results.

Like the four-way loudspeaker, the 4000/2 is housed in a sealed enclosure. Crossover points are at 700 Hz and 3 kHz, and the crossover is a 12 dB per octave constant-K design. (See "Principles and problems in loudspeaker design", which you'll find on page 130 of this book).

▶ View of the completed speakers, with and without the front grille. We secured the latter to the front baffle using Velcro strips stapled in appropriate positions.



The completed box, prior to mounting the drivers and crossover network. The box lining is placed in position after the crossover unit, tweeter and mid-range drivers are placed in position. The whole box is then stuffed with waste wool before mounting the woofer to the front baffle.

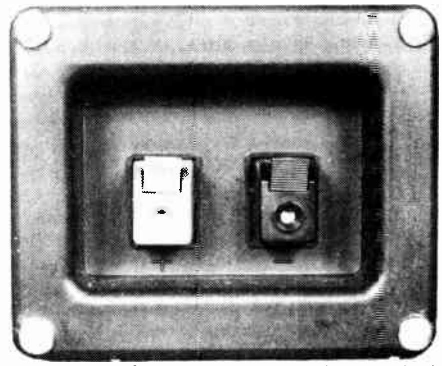
The drivers used here are part of the new range of Philips loudspeakers. We chose these drivers only after a thorough look at the price, availability and quality of drivers presently available in Australia. There is some relationship between power handling and price of a driver but there is *little relationship between sound quality and price*. Some of the best tweeters I have heard, for example, have been amongst the cheapest devices available. A complete set of drivers for this project will cost around $240, and at this price are a bargain.

The woofer used is the AD12250/W8. This has a free air resonance at 26 Hz and a power handling figure of 100 watts. When in its enclosure the resonant frequency rises to around 35 Hz, which is quite good!

The mid-range unit is the latest Philips dome mid-range, the AD02161/SQ8. This is a 50 mm diameter textile dome unit with its own mid-range enclosure as an integral part of the driver. The tweeter is a 25 mm diameter textile dome unit, the AD01610/T8. These three drivers were used in the earlier 4000/1 loudspeaker and the mid-range and tweeter units were shown to work well together with a crossover point at 3 kHz.

The big problem encountered in the design of this loudspeaker was the lower crossover point between the woofer and mid-range. The resonant frequency of the mid-range unit is approximately 350 Hz, so it should not be used much below 700 Hz. This enables the crossover to give 12 dB of attenuation at the resonant frequency. Unfortunately, this means that the woofer must handle everything up to 700 Hz and, ideally, should have a usable frequency response to 1.4 kHz. The AD12250/W8 doesn't have a response this good, but it can be made to operate satisfactorily up to 1 kHz, which is a reasonable compromise. Since the woofer is handling a good part of the mid-range spectrum, a substantial amount of cabinet damping is necessary, otherwise cabinet resonances will severely impair the mid-range performance of the loudspeaker.

The prototype loudspeakers were first lined with a double layer of speaker innerbond material, then completely filled with waste wool. As always, the exact amount of filling needs to be established by experiment. During the development of the 4000 range of loudspeakers I have had a good look at the types of innerbond material available and most of them are definitely not dense enough. Unless the quality of innerbond improves rapidly, you can safely consider carpet underfelt and fibreglass bats as suitable alternatives!

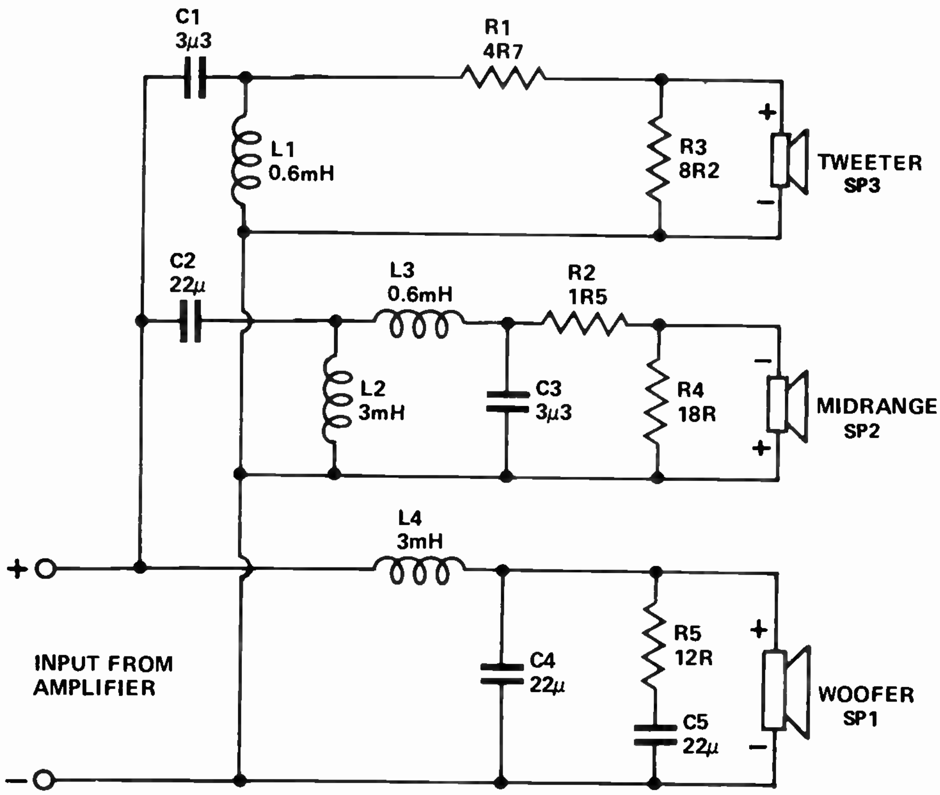


A close-up of the speaker connection terminals we used. This consists of two colour-coded spring-grip terminals mounted on a recessed plastic moulding. The assembly is screwed into a cutout in the rear baffle of the box.

Construction

If you are building the boxes yourself, rather than buying a complete kit, start by cutting the back, top, bottom and sides. The material used in the prototype boxes is 19 mm particle board and the finished units do not suffer excessively from panel vibration in use,

A group photo of the three drivers used in this system. Clockwise, from the left: AD02161/SQ8 dome mid-range, AD12250/W8 woofer and the AD01605/T8 tweeter — note that this last one is the alternative driver to that shown in the picture of the assembled speaker on page 63.



PARTS LIST - ETI 497

Drivers
SP1 .... Philips AD12250/W8
SP2 .... Philips AD02161/SQ8 or AD02160/W8
SP3 .... Philips AD01610/T8 or AD01605/T8

Inductors
L1 .... 0.6mH,max.dc resist: 1 ohm
L2 .... 3.0mH,max.dc resist: ½ ohm
L3 .... 0.6mH,max.dc resist: 1 ohm
L4 .... 3.0mH,max.dc resist: ½ ohm

Capacitors
C1 .... 3u3 polycarbonate
C2 .... 22u bipolar electrolytic, 50V
C3 .... 3u3 polycarbonate
C4 .... 22u bipolar electrolytic, 50V
C5 .... 22u bipolar electrolytic, 50V

Resistors all wirewound, 5%
R1 .... 4R7, 5W
R2 .... 1R5, 5W
R3 .... 8R2, 5W
R4 .... 18R, 5W
R5 .... 12R, 5W

Miscellaneous
wire; one loudspeaker terminal block; particle board; screws; glue, etc; speaker grill cloth; innerbond; waste wool.

so particle board of this thickness should be sufficient for most purposes.

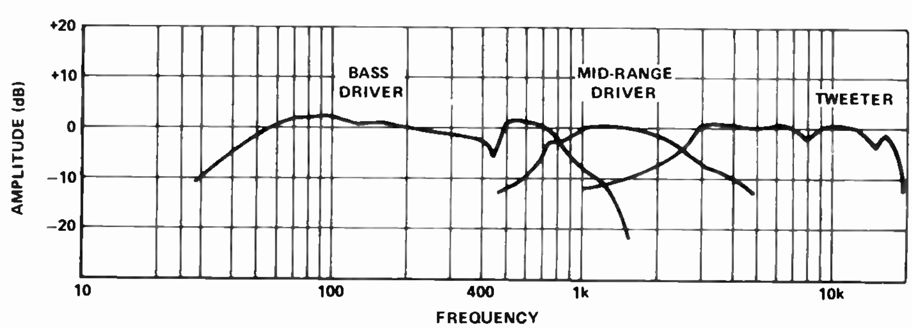
Apply a liberal quantity of Aquadhere, or a similar wood glue, to all the joints and screw the panels together. Let the glue dry and then line all the inside joints with a suitable sealent, such as caulking compound or Plastibond. Always use particle board screws or self tappers when working with particle board. Normal wood screws will not hold into the material properly.

For best bass performance it is important that the finished box is totally air tight, so eventually, the whole box will be sealed in this manner. This much of the sealing is best done now before the front baffle goes on.

Next, mount the spring terminals to the back panel. I used special loudspeaker terminals that are supplied fitted to a moulded plastic base. This simplifies the job of sealing the terminals considerably. The hole needed for the terminal base is rectangular but with rounded corners. Use a jig saw if you have one. If not, drill four holes to give the rounded corners necessary then join them up by cutting with a fret saw. Screw the terminal base into position then seal the inside of the cabinet between the timber and the base of the terminal block. Use Plastibond or Bostik for this as wood glue will not adhere to the plastic properly. An alternative way to seal the mounting terminal is to line the rim of the hole with plasticine and screw the terminals over the top. Be careful not to screw the terminal base down too hard as the plastic is easily cracked.

Cut out the front baffle so that it is a snug fit into the front of the loudspeaker. Cut four lengths of 25 mm square timber and screw these onto the inside of the sides, top and bottom, 38 mm from the front edge of the loudspeaker. The baffle is then glued and slid into the box so that it rests on the timber braces. Anchor the baffle board by screwing through it into the timber braces; 19 mm should remain between the front of the baffle and the front edge of the loudspeaker. This space will be taken up by the grill cloth former.. ▶

Hand-plotted graph showing measurements of the average response of the individual drivers in the completed system, measured in an 'average' living room. The 'dips' at the cross-over points "flatten out" in the overall response of the system since the contribution from each driver adds in these areas.



Glue the small 100 mm high wooden panel into position at the bottom of the box. In the 4000/1 loudspeakers there was a false bottom, leaving approximately 90 mm under the box in which to mount the crossover. In the interests of keeping the cost of this project to a minimum I have omitted the false bottom and the crossover is mounted on the inside of the box as is more commonly done.

The next stage of construction is to mount the tweeter and mid-range. First, solder lengths of speaker cable to each of the drivers so that there is enough cable to reach the bottom of the box. The tweeter and midrange units are supplied with their own mounting gaskets that ensure a good seal between the baffle and the drivers. Use Philips head or Posidriver type screws for mounting the drivers, this will minimise the possibility of slipping off the screw and possibly putting a screw driver through the loudspeaker cone.

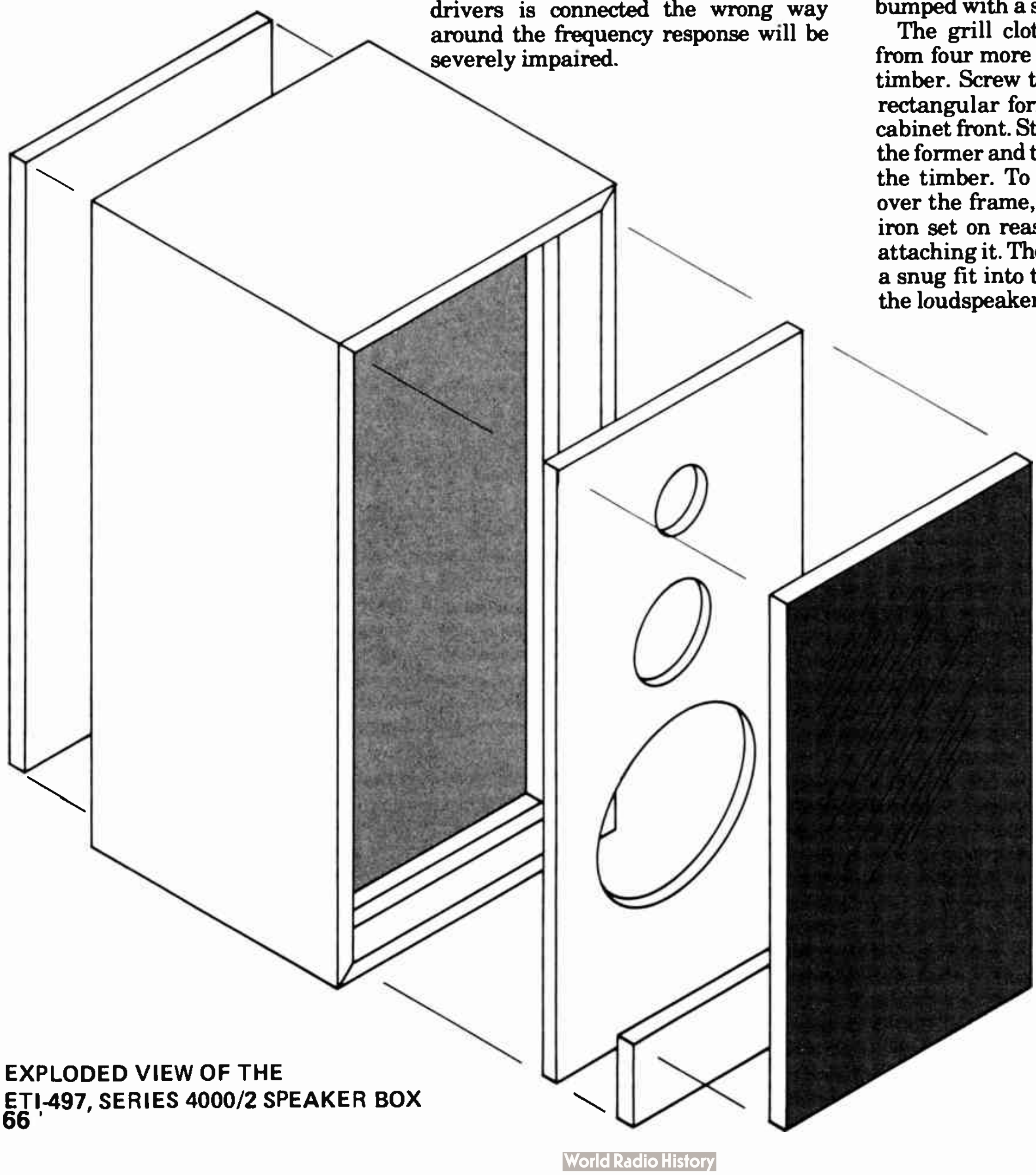
The crossover for the project will be manufactured by Philips and should be available through Philips retail outlets. The pc board on the front cover is the prototype unit and there may be some changes to the pc board layout to simplify manufacture, but the circuit will be the same. Should you decide to construct your own crossovers, the necessary components and the pc board overlay will also be available from Philips retail outlets. The inductors will need to be glued to the board with epoxy. Other than this, crossover construction should be obvious.

Solder the wires from the drivers to the crossover. *Be careful to ensure that the polarity of the wires to each of the drivers is correct.* If any one of the drivers is connected the wrong way around the frequency response will be severely impaired.

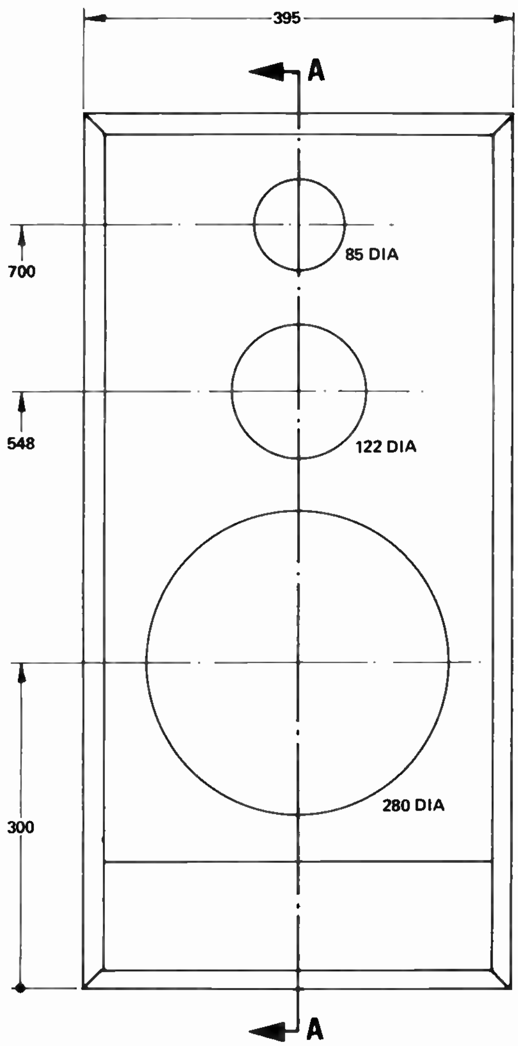
Solder wires from the terminals to the crossover, and mount the crossover to the bottom of the box with self tappers. Line the inside of the box with two layers of innerbond then stuff the whole box with waste wool. Some experimenting will be necessary to establish the optimum amount of stuffing. Finally, cut another piece of innerbond 450 mm square and place this in the box so that it is attached to the inside of the baffle around the rim of the woofer hole. This serves to keep all the stuffing away from the back of the woofer.

You will need to make a small hole in the innerbond for the cable to the woofer. Solder this to the woofer, then mount the woofer on the front panel. Once again, use Philips or Posigrip type self-tappers. The roll surround of the woofer is very easily damaged if bumped with a screwdriver blade.

The grill cloth former can be made from four more pieces of 25 mm square timber. Screw these together to form a rectangular former that fits inside the cabinet front. Stretch the grill cloth over the former and tack or staple the cloth to the timber. To make the cloth tighter over the frame, iron the cloth, with the iron set on reasonably low heat before attaching it. The finished grill should be a snug fit into the cavity in the front of the loudspeaker. ●



EXPLODED VIEW OF THE
ETI-497, SERIES 4000/2 SPEAKER BOX



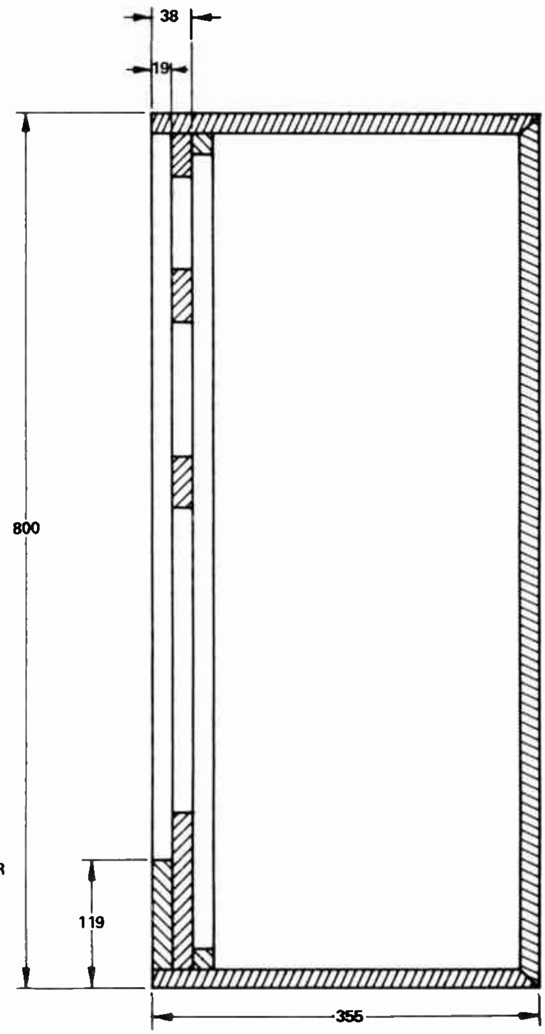
NOTES
ALL OUTSIDE WOODWORK IS 19mm PARTICLE BOARD COVERED WHERE VISIBLE WITH THE DESIRED VENEER

19mm x 19mm CLEATS AROUND FRONT EDGES

LINE ALL INSIDE SURFACE WITH ACOUSTIC FIBREGLASS

FRONT GRILL NOT SHOWN

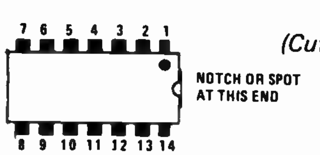
ALL DIMENSIONS IN MILLIMETRES

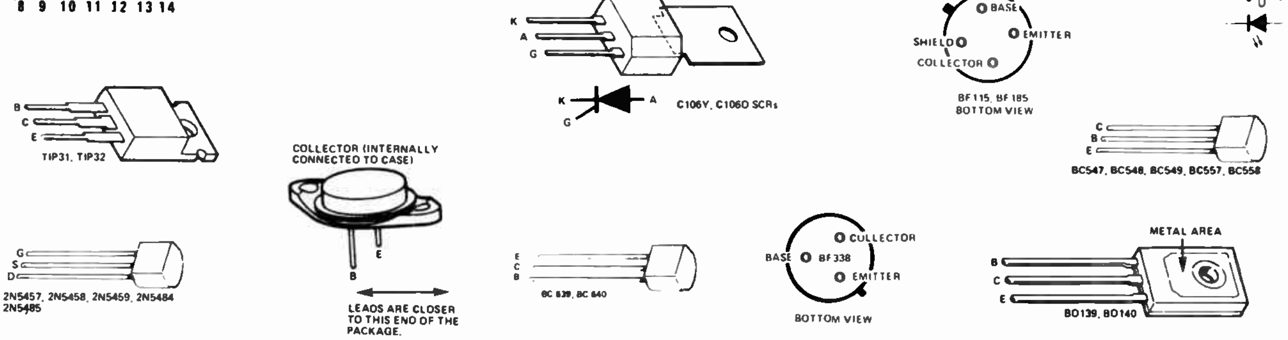


SECTION AA

COMMON SEMICONDUCTOR PIN-OUTS

(Cut this out, attach to strong card and stick it to your workshop wall)





A hum filter for hi-fi systems

There are few things more annoying in life than attempting to track down and remove all sources of hum from a hi-fi set up only to be partially successful, no matter how hard you try. This project should remove the last vestige of that 50 Hz pest from your system. Go 'notch' that nasty!

David Tilbrook

SO YOU'VE just spent most of your spare money, unpacked everything from the boxes, connected it up and turned it on. What on earth is that awful noise?

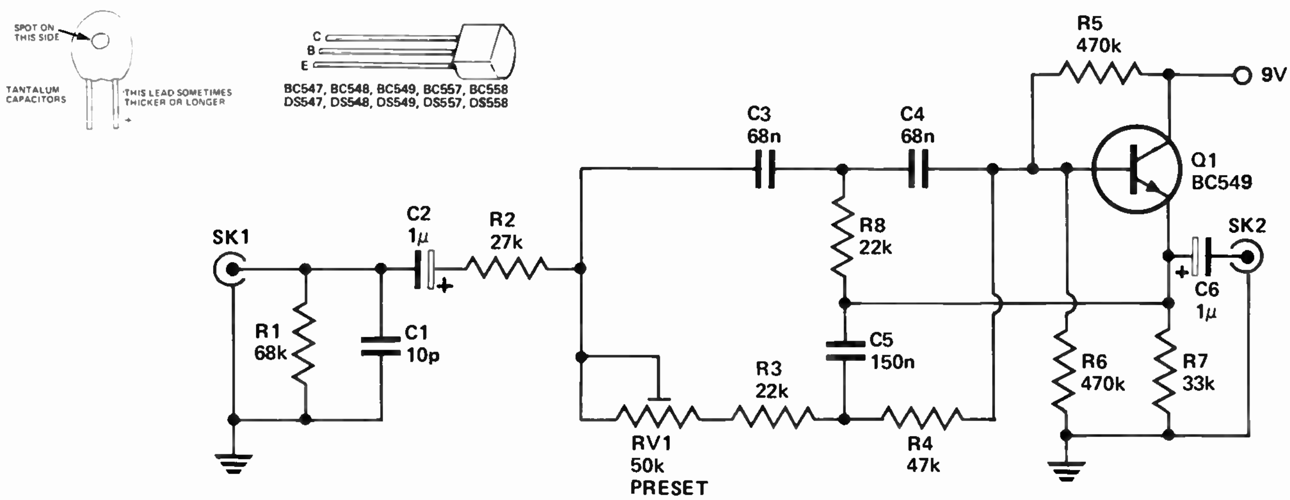
Maybe this is a bit of an exaggeration, but it does illustrate the problems some of us have with mains induced hum. Often it's necessary to position the various components of a hi-fi system close together and this can cause problems.

The magnetic field around the transformer in the power amplifier can couple to the preamp or tape deck. Also, the location of nearby 240V mains wiring can cause problems that can be very difficult to overcome. In theory, if the equipment and leads have been properly shielded and earthed this problem shouldn't exist. In practice it's a very different story.

This project aims at overcoming some of the problems of mains induced hum by using a notch filter at the hum frequency of 50 Hz. At this frequency any signal present will be attenuated. At frequencies either side of the notch the response should return to the unattenuated input level.

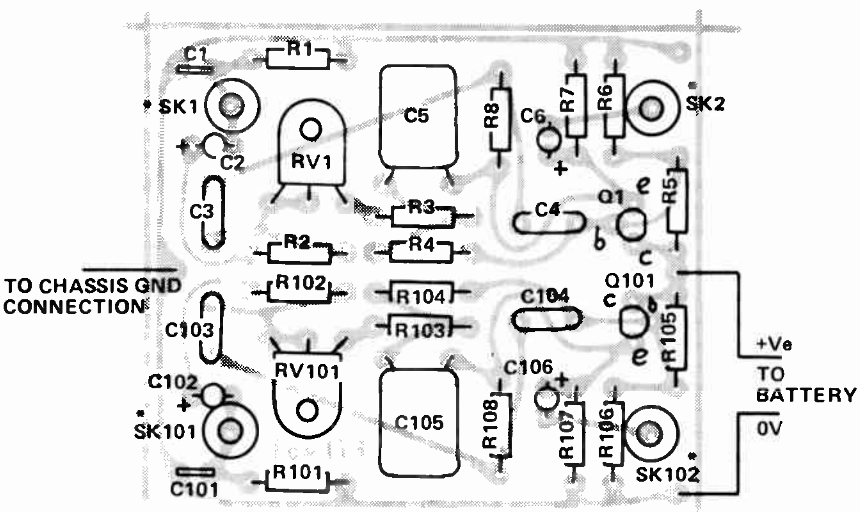
The 'Q', or Quality Factor, of a tuned circuit – which the RC network in this circuit forms, determines the bandwidth, or narrowness, of the amplitude response of the circuit (see the diagram). As this circuit forms a notch filter, the Q of the circuit determines the narrowness of the notch.

With a high-Q notch the frequency response of the circuit will dip suddenly around the notch frequency. Frequencies ▶

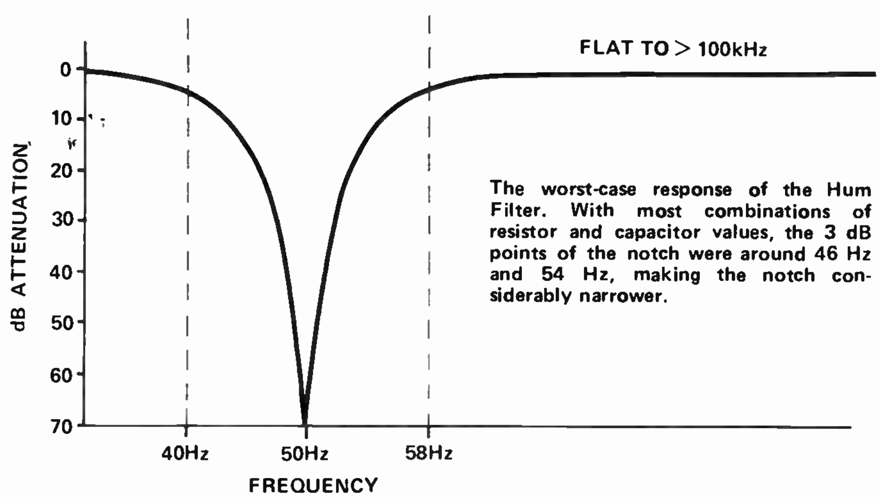


The printed circuit board pattern is on page 74.

NOTE
ONLY ONE CHANNEL HAS BEEN SHOWN FOR CLARITY. THE COMPONENT NUMBERING OF THE OTHER CHANNEL BEGINS at 101 i.e. R101 R102 etc.



*RCA SOCKETS POSITIONED HERE ON FRONT PANEL OF CHASSIS



The worst-case response of the Hum Filter. With most combinations of resistor and capacitor values, the 3 dB points of the notch were around 46 Hz and 54 Hz, making the notch considerably narrower.

PARTS LIST - ETI 451

Resistors all ¼ W, 5%

R1, R101 . . . 68k
R2, R102 . . . 27k
R3, R103 . . . 22k
R4, R104 . . . 47k
R5, R6, R105,
R106 . . . . . . 470k
R7, R107 . . . 33k
R8, R108 . . . 22k

Capacitors

C1, C101. . . . 10pf ceramic
C2, C102. . . . 1μ tant
C3, C4, C103,
C104 . . . . . . 68n greencap
C5, C105. . . . 150n greencap
C6, C106. . . . 1μ tant

Potentiometers

RV1, RV101 . 50k min preset

Semiconductors

Q1, Q101 . . . BC549, BC109, DS549, etc.

Miscellaneous

ETI 451 pcb, box to suit, 4 panel mounting, RCA sockets.

Components for 100 Hz operation

R4, R104 . . . 22k
R8, R108 . . . 10k

Replace R3 with wire link.

a little either side of the notch centre frequency will be little affected. If the Q is low, frequencies some way either side of the notch frequency will be attenuated. The actual attenuation at the notch frequency is greater with a high-Q circuit than with a low-Q circuit.

High-Q circuits have the disadvantage that slight changes in component values, due to temperature changes etc, will affect the centre frequency. Tuning of the circuit to frequency is also quite critical. Lower-Q circuits do not suffer so much from this disadvantage.

The design Q chosen for this project was a compromise between the constraints of critical tuning and drift effect and good attenuation at the notch with little affect on nearby frequencies. Peak attenuation at the notch centre frequency of 50 Hz is around 80 dB while attenuation of only 3 dB is obtained at 40 Hz and 58 Hz. There is some audible effect on the bass response of a system, but this is minimal.

Construction

Mount the resistors and capacitors on the board first. Be sure the orientation of the tantalum capacitors is correct. These are polarized and can only be installed one way round. Next, install the preset pot. If you elect to use the same case we did, the preset must lie flat on the board. This is best done by bending the pins 90° first and then soldering onto the printed circuit board. Finally, solder the transistor in place.

The input and output connections are best made by mounting the four RCA sockets directly above the input and output pads on the pc board. Strong wires can be soldered onto the RCA sockets and the entire board slid onto the four wires. This serves the purpose of holding the board in place as well as forming the input-output connections. A short insulated wire should be connected to the gound point provided on the pc board (see overlay diagram) and to the chassis. The RCA sockets are grounded by their mounting nuts, so be sure to use a metal case.

The circuit is run from a single No. 216 nine volt battery. The current consumption of the prototype was 200μA so the battery life should be good for several months. If it is found that battery life is not long enough a power switch could be fitted.

The filter can be used almost anywhere in the amplification chain since its overload margin is very high (typically 8 V p-p). It should obviously be placed after the point where the hum is being picked up. If the hum is in the turntable it can even be placed between the turntable and the magnetic phono input of the amplifier since the input impedance is 47 k shunted by 10 pF, which should suit most magnetic cartridges.

Once the filter is in place, the presets are adjusted so that the hum is brought to a minimum by adjusting each channel independently.

Installation

Before connecting the battery, check the pc board thoroughly. Check the orientations of the tantalum capacitors and the transistor. If all is right, plug in the battery and seal the base.

In the unit we built, holes were drilled in the chassis immediately above the preset pots. This allows the filter to be fine-tuned after it has been connected into the circuit. The presets themselves are connected to the base of the transistors via some resistance, so the transistor bias voltage is present on the preset. If the pot is to be adjusted through a hole in the chassis this voltage will probably be shorted out by the screwdriver touching the earthed chassis. Although this won't damage the circuit, it could damage the loudspeakers if the filter is being used in the magnetic phono line. It certainly makes the adjustment meaningless, so either use a non-metal adjustment tool or use LED mounting grommets to insulate the holes.

If the hum problem you are experiencing is 100 Hz instead of 50 Hz the filter is easily adapted. Simply replace resistor R3 (22 k) in each channel with a wire link. Remove R4 (47 k) and replace with a 22 k resistor. Remove R8 (22 k) and replace with a 10 k resistor. ●

HOW IT WORKS

The circuit consists of a "Twin-T" notch filter formed by capacitors C3, C4 and C5 and resistors R3, R4, R8 and preset PR1.

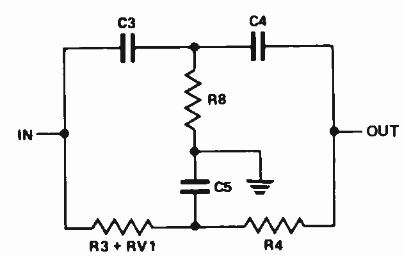
The operation of the Twin-T requires that $C_3 = C_4 = \frac{C_5}{2}$

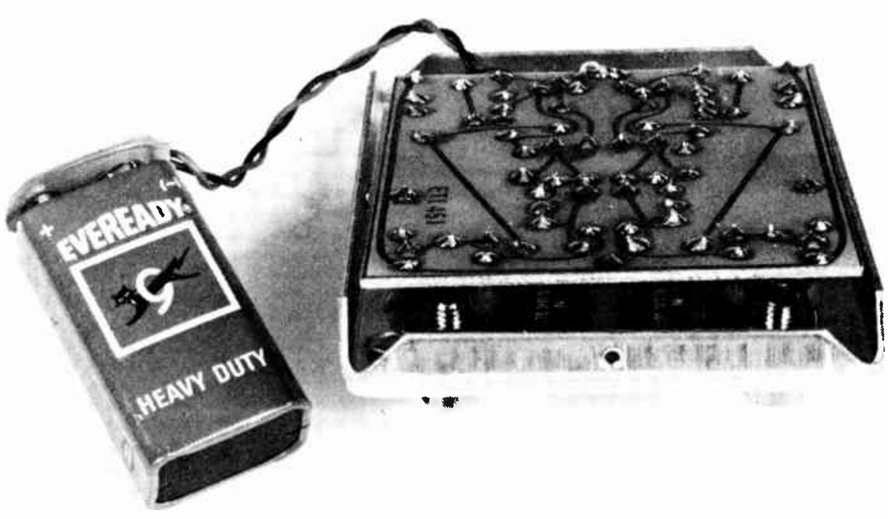
and $R_3 + PR1 = R_4 = 2R_8$

These conditions must be met with reasonable accuracy if a good, deep notch is to be obtained. The preset corrects to a certain extent for errors due to component mis-match and assumes that the notch can be adjusted to the exact frequency of the hum to be rejected.

The frequency of the notch is then given by $f = \frac{1}{2\pi R_4 C_4}$

The transistor is operating as an emitter follower, giving zero voltage gain, but providing feedback into the notch to increase the Q to acceptable limits.

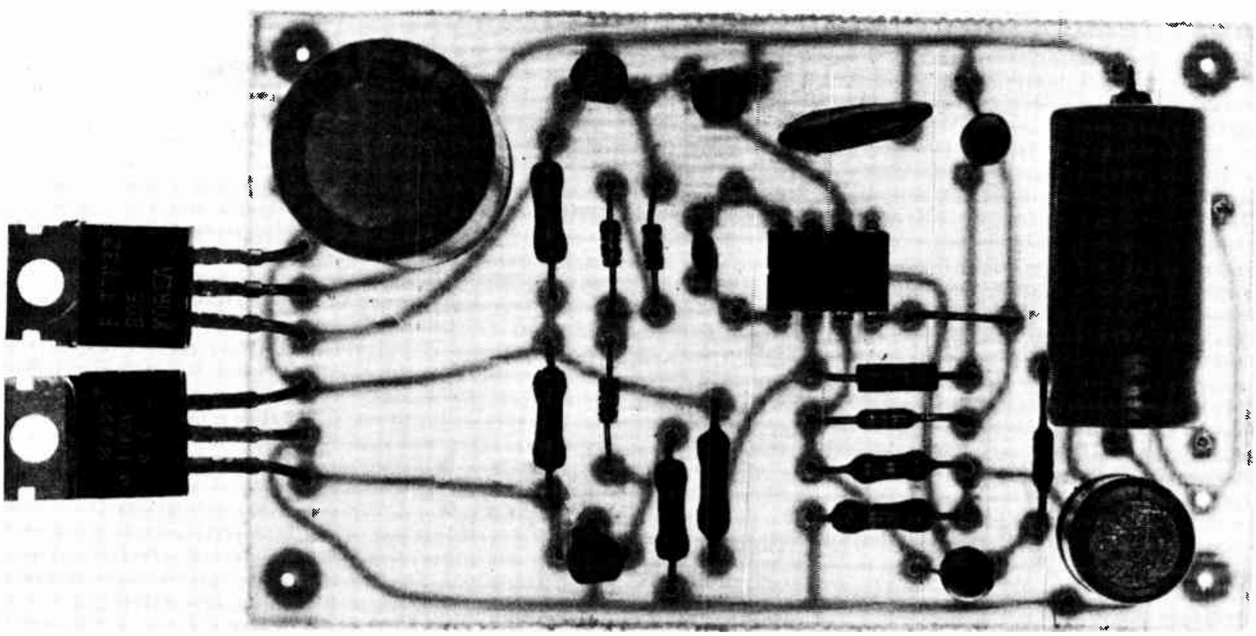




A general purpose audio amplifier module

One of the handiest 'tools' for the electronics experimenter is a genuine general purpose audio amp. This module will work from a wide range of supply voltages, has good sensitivity, is robust and reliable — easy to build too!

David Tilbrook



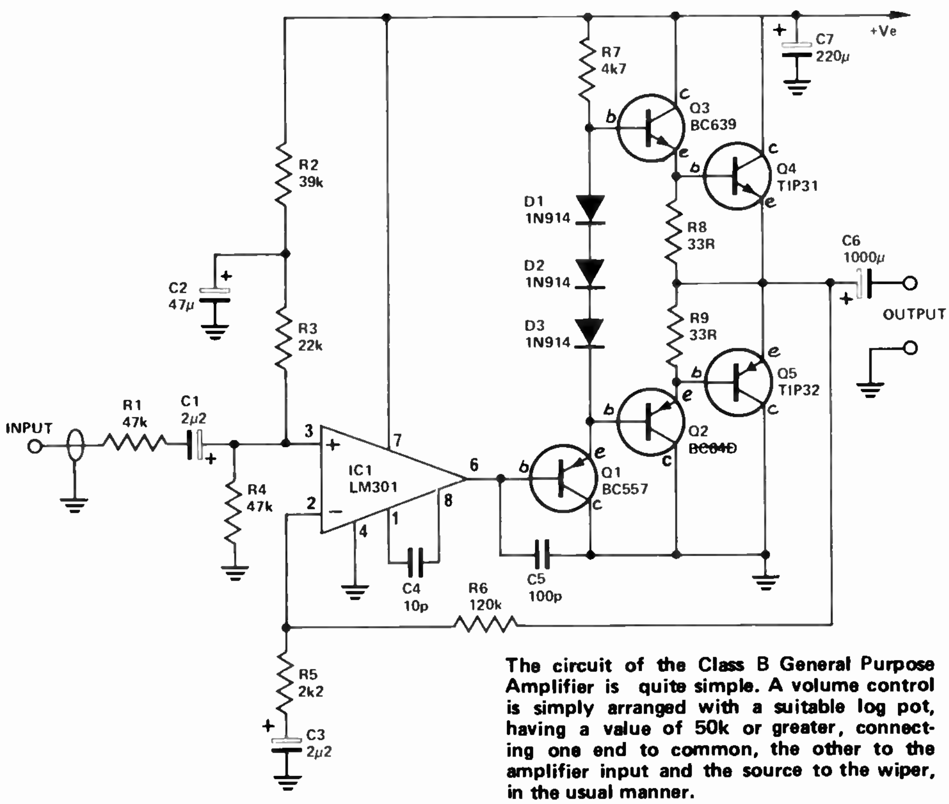
WHEN DESIGNING and building electronic projects, one of the most often needed circuits is a simple, low cost audio amplifier stage capable of driving a 4 or 8 ohm loudspeaker. This amp module is capable of driving a 4 ohm load with over 20 watts. This is more power than is necessary in many applications but it is a simple matter to decrease the maximum output power by simply decreasing the supply voltage. The table gives the relationship between output power and supply voltage. At lower power levels the amp module will not require any form of heatsinking. Some experimentation will be necessary to determine the amount of heatsinking that should be used at higher power levels.

The circuit gives maximum power out with an input level of about 300 mV which should be easily obtained by the stage used to drive the amp module. This figure can be changed slightly by varying the amount of negative feedback. Resistor R6 is the negative feedback resistor and, together with R5, determines the overall gain of the circuit.

Specifically, the gain of the circuit is given by R6/R5. In the circuit shown this becomes

$$\frac{120k}{2k2} = 54.55.$$

The output voltage will be this figure, multiplied by the input signal voltage. So, for an input voltage level of 100 mV RMS the output level will be 100 mV x 54.55 = 5.45 volts RMS, which is equivalent to 3.7 watts into an 8 ohm load. If you have only 50 mV available, for example, and wish this signal level to drive the amp to 3.7 watts, it would be necessary to halve the amount of negative feedback, thereby doubling the gain of the amplifier. The negative feedback resistor would have to be doubled in value to 240k. In practice, ▶



The circuit of the Class B General Purpose Amplifier is quite simple. A volume control is simply arranged with a suitable log pot, having a value of 50k or greater, connecting one end to common, the other to the amplifier input and the source to the wiper, in the usual manner.

PARTS LIST - ETI 453

Resistors all ½W, 5%

R1	47k
R2	39k
R3	22k
R4	47k
R5	2k2
R6	120k
R7	4k7
R8, R9	33R

Capacitors

C1.	2µ2 35V tantalum
C2.	47µ 25V electrolytic
C3.	2µ2 35V tantalum
C4.	10p disc ceramic
C5.	100p disc ceramic
C6.	1000µ 25V electrolytic
C7.	220µ 25V electrolytic

Semiconductors

D1—D3.	IN914 signal diode or similar
Q1	BC557
Q2	BC640
Q3	BC639
Q4	TIP 31
Q5	TIP 32
IC1	LM301

Miscellaneous

ETI 453 . . . . pc board

HOW IT WORKS

The amplifier can be considered in two sections, the voltage amplifier built around the LM301 op-amp and the current amplifying output stage.

The LM301 is an IC operational amplifier that is used to amplify the input signal up to the voltage levels needed by the output stages. Input to the op-amp is via the 47k resistor R1 and the 2u2 tantalum capacitor, C1. The input impedance at the non-inverting input of the op-amp is determined by resistors R4 and R3. R4 is connected directly to ground and the 47u electrolytic capacitor C2 represents a short circuit to ground for any ac signals flowing through resistor R3. As far as ac signals are concerned, R3 and R4 represent parallel resistance to ground. As a result, the impedance at this point is around 15k. R1 serves to increase the input impedance to approx 60k: R1 in combination with C1 also determines the lower frequency 3 dB bandwidth point of the circuit, setting it to around 10Hz.

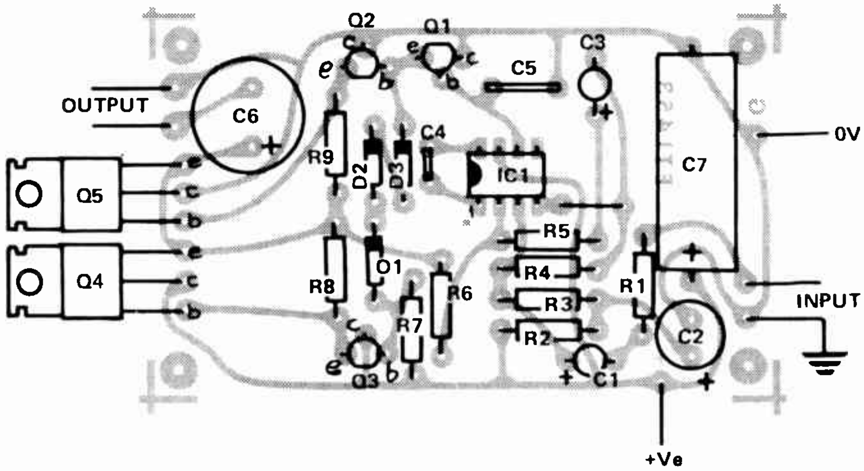
The LM301 is normally used with a split supply i.e: positive and negative supply rails. Since the objective of this design was to construct a general purpose amplifier module and its supply would, in many instances, be taken from an existing power supply, the circuit was designed to operate from a single supply. For this reason, the op-amp must be biased up to around half supply. This is accomplished by the resistors R2, R3 and R4. R2 and R3 form the upper half of a potential divider, R4 forming the lower half. The resistors chosen set the voltage on the positive input of the LM301 to about 0.44 of the supply voltage. Since these resistors are biasing the non-inverting input of the op-amp, any noise present on the positive supply rail would be communicated directly to the input of the amplifier through these biasing resistors. To prevent this, capacitor C2 was placed so as to represent a short circuit to noise voltages on the supply. This is much more effective than simply filtering the power supply (which must be done as well). In this configuration, the capacitor is fed from the 39k resistor R2, instead of directly from the positive supply as would be the case if C2 was used as a simple supply decoupling capacitor.

At audio frequencies, C2 will represent an impedance very much lower than 39k, effectively shorting out noise currents through R2. This would not be the case if R2 were not present, as the impedance of the supply would be a fraction of the impedance of the capacitor.

The amplified signal from pin 6 (output) of the LM301 is fed to Q1, which acts as the first current amplifier stage. This provides a reasonably low output impedance necessary to drive the remaining output stages. Diodes D1. D2 and D3 maintain about 1.8 volts between the bases of Q2 and Q3. These two transistors will drop 0.6 volts across the base — emitter junctions. This leaves 0.6 volts across the two 33 ohm resistors, R8 and R9. Each resistor will drop 0.3 volts, holding this voltage across the base emitter junctions of the output transistors. Since these devices, like the driver transistors, require 0.6 volts to turn on, they will remain off until the signal supplies the necessary additional 0.3 volts.

Capacitor C6 isolates the dc voltage on the emitters of the output transistors from the loudspeaker. C7 provides supply decoupling. If the amp module is used to deliver 10 watts or more, additional supply decoupling will probably be necessary.

Negative feedback is supplied by the potential divider formed from resistors R6 and R5. The capacitor C3 represents a short circuit to ground for ac signals in the audio range. Audio frequencies will therefore see R5 and R6 as a potential divider. The overall gain of the circuit will be determined by the value of R6/R5. (In this case, equal to 54.55). As the frequency decreases the impedance of C3 will increase, decreasing the gain of the circuit by increasing the amount of negative feedback. At very low frequencies, the gain of the circuit is reduced virtually to unity. This ensures that the voltage on the emitters of the output devices is stable.



Overlay showing placement of components.

The pc board pattern is on page 74.

a 220k resistor would be used. Note also that in order to obtain 3.75 watts it would be necessary to provide a supply rail of around 24 volts.

Since the project was to be a general purpose amp, it was essential that the design be robust and reliable. Class B was chosen for this reason as it has no bias current in the output stage and requires no set up procedure. The output devices are actually off, being switched on by the signal itself.

Construction

Start by soldering the resistors and capacitors into their positions on the printed circuit board. The electrolytic and tantalum capacitors must be placed on the board with the correct orientation. These are capacitors C1, C2, C3, C6 and C7. The printed circuit board overlay shows the correct orientation for these components on the board.

Solder the transistors and diodes in place, being careful that the transistors are inserted in the correct locations. Every transistor used in this project is different and as such, they are not interchangeable. Finally, solder the IC and wire link into position. Orient the IC so that the 'notch' points towards the output transistors.

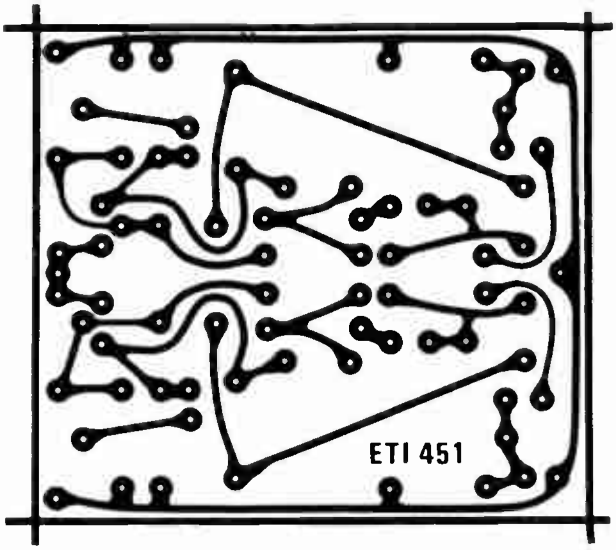
Connection to the input of the power amp is best made with shielded cable to decrease the possibility of hum being induced into the amplifier. ●

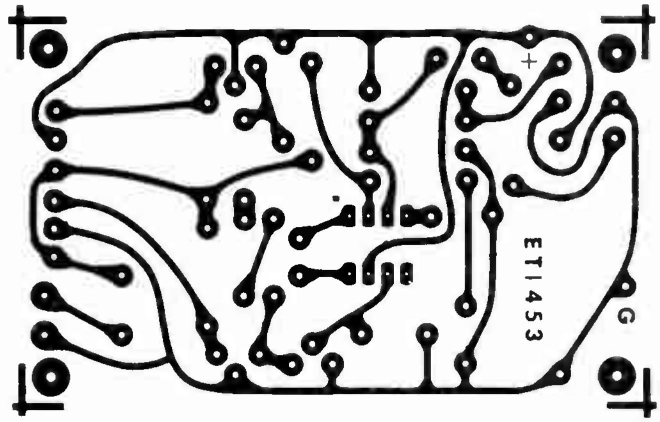
TABLE 1

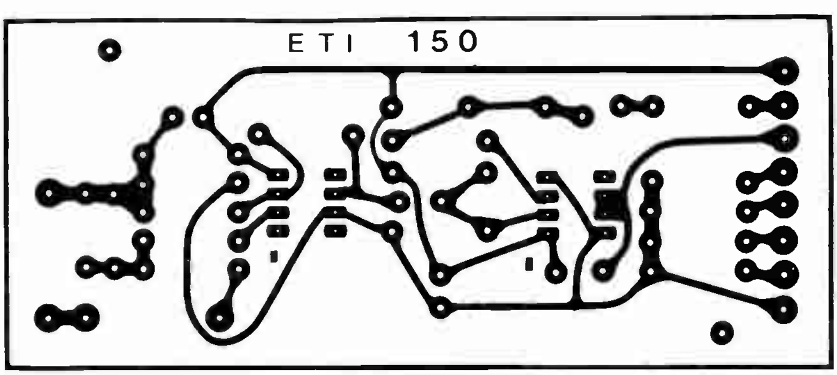
SUPPLY VOLTAGE	POWER INTO 8 OHMS	POWER INTO 4 OHMS
9	0.13 W	0.25 W
12	0.5 W	1 W
18	1.7 W	3.52 W
22	3.13 W	6.25 W
26	4.5 W	9 W
30	8 W	16 W
35	10.13 W	20.25 W

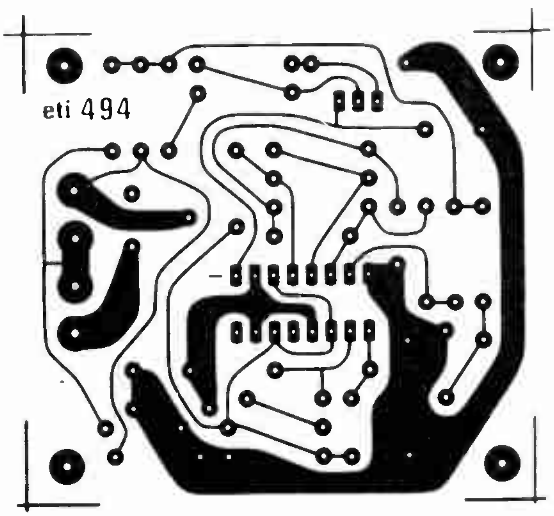
These are measured power output figures for different supply voltages. Powers quoted are at the onset of clipping.

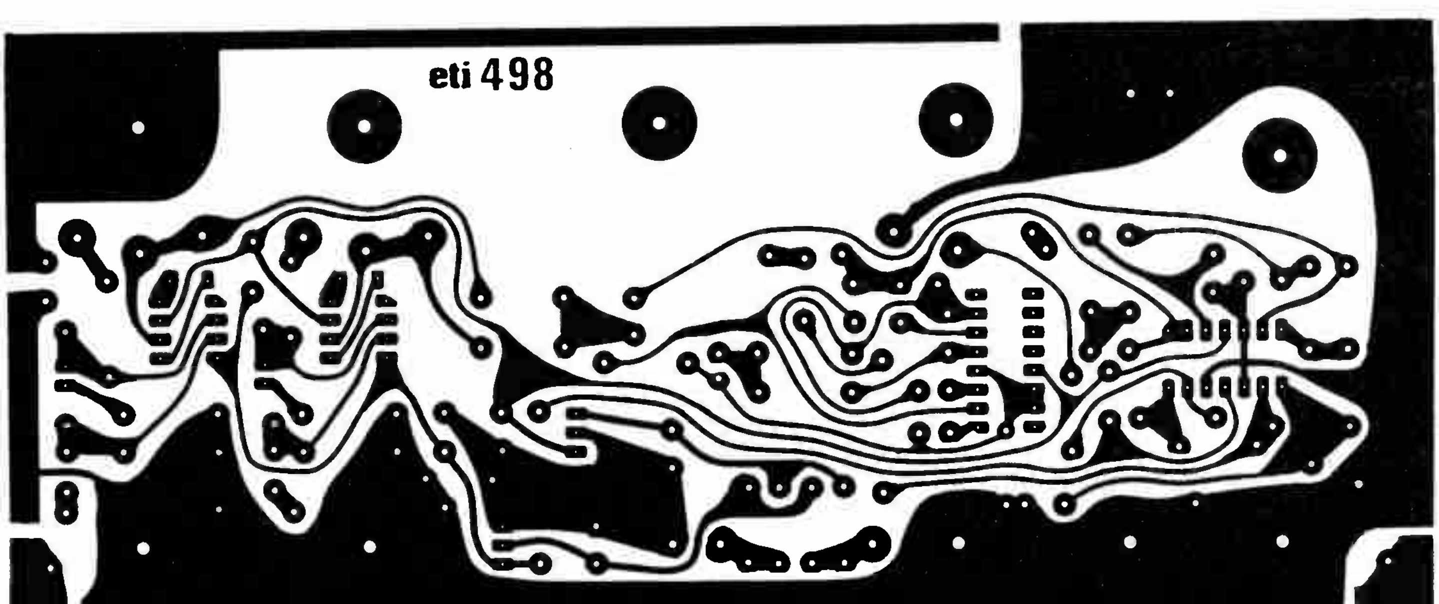
PCB's











mc o to
See Threepio,

Darth Vader to Daleks

— all from our Sound Bender!

Based on a remarkably versatile function generator IC, the XR2206, this project is capable of modifying an audio signal to produce tremolo effects on music or those peculiar, metallic robot voices so abundantly found in shows like 'Dr Who', 'Star Wars', 'Star Trek', etc.

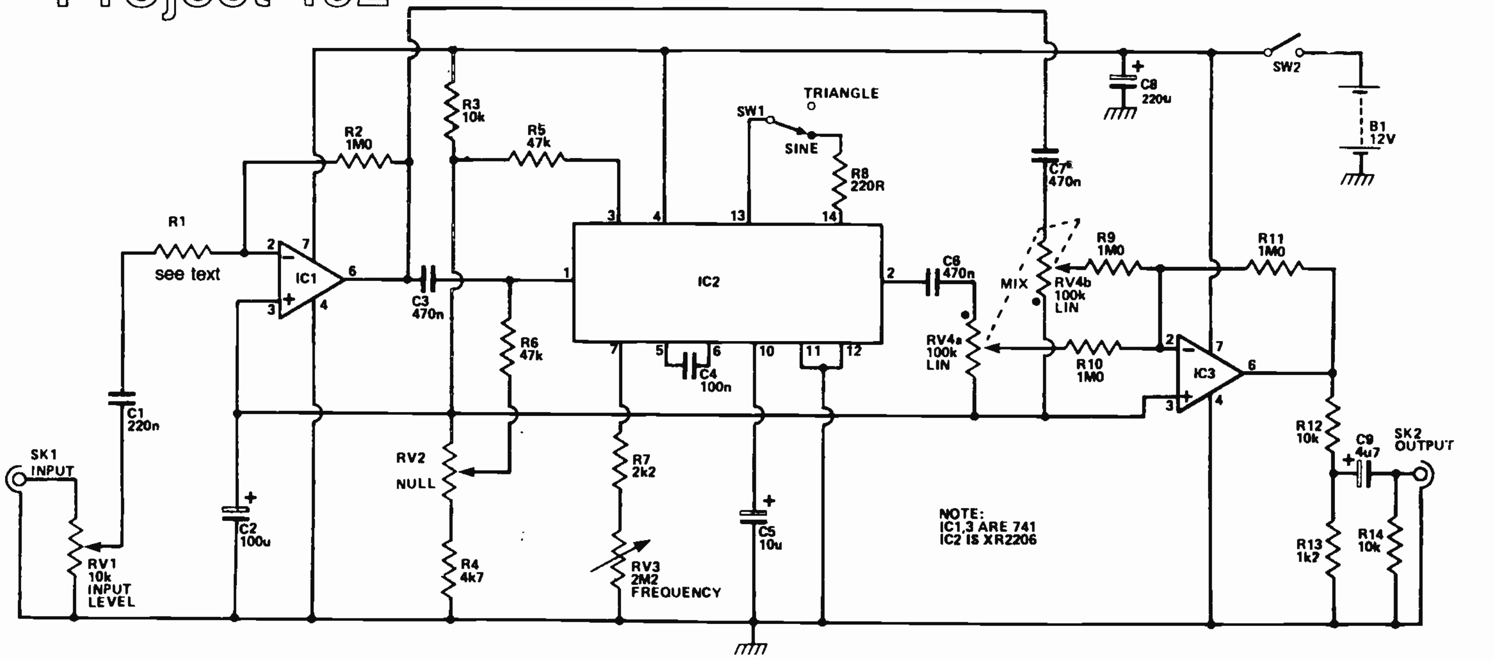
Design: **Ray Marston** Development: **Roger Harrison**

'VARIETY is the spice of life' goes a famous old saying, and when electronics entered the musical arena, engineers and musicians sought ways of extending the variety of available musical sounds, some by developing electronic 'instruments', others by developing circuits that modified the sound produced by the voice or an instrument. Deliberately introducing plain old distortion gave rise to the 'fuzz box', amplitude modulating the sound gave a 'tremolo' effect, etc.

Now, a device developed to permit more conversations per line on the telephone system was discovered to produce a range of 'intelligible', but highly modified, sounds from voice and music signals. Called variously a 'ring modulator' or 'four-quadrant multiplier', it is achieved by mixing an audio signal with an oscillator signal, and the output is the product of these two signals, containing both sum and difference frequencies. The oscillator or 'carrier' signal is reduced or suppressed. If, for example, the carrier frequency is 1 MHz and the audio signal is speech with a range of around 150 Hz to 3 kHz, the ring modulator's output would be two 'sidebands' — a 'lower' one (the difference) at 997 kHz to 999.85 kHz and an 'upper' one (the sum) at 1000.15 kHz to 1003 kHz. The 1 MHz carrier level could be 20 dB to over 40 dB lower in level than the sidebands, depending on how 'good' the ring modulator is. If the carrier is set at 1 kHz, though, the sum and difference frequencies at the output spread up and down the audio spectrum, and if speech is the input you get a jumble of voice sounds, some shifted up in frequency, some inverted and apparently shifted down in frequency. The best examples we can cite are the voices of Darth Vader from 'Star Wars', the Daleks from 'Dr Who' and the Cylons from 'Star Trek'. If the carrier is placed at a sub-audible frequency, then the result is a tremolo effect, where the audio signal is seemingly amplitude modulated at a slow rate.

The XR2206 function generator IC contains a voltage-controlled oscillator and a four-quadrant multiplier or ring modulator, so in one chip we have both the carrier oscillator and the modulator that can be combined in a circuit to produce the effects we seek. As the panel on page 78 shows, which explains the XR2206 and typical applications, the IC also includes internal control and signal shaping circuitry, making the circuit design job a whole lot simpler.

This project is designed to make full use of the functions incorporated in the XR2206 for this application, and the IC's VCO — used here as the carrier ▶



HOW IT WORKS — ETI 492

By mixing or 'multiplying' an audio signal with an oscillator or 'carrier' signal that may be varied from the sub-audible to the mid-range of the audio spectrum, the original signal may be altered in a variety of ways. Mixing an audio signal with a sub-audible carrier produces a tremolo effect — a form of amplitude modulation; mixing speech with a carrier around 1 kHz to 2 kHz produces 'robot' voices. That's just to name a few of the more familiar effects possible.

The heart of this unit is IC2, an XR2206 function generator chip that incorporates a multiplier — used to perform the modulating function — plus a voltage controlled oscillator (VCO), signal-shaping circuitry, and control circuitry that permits simple variable resistance control of the VCO. The signal-shaping circuitry permits generation of sine or triangle waveforms out of the VCO.

There are three sections to the circuit: the input amplifier (IC1), the mixer/carrier generator (IC2) and the output mixer/buffer (IC3).

The audio input signal enters via SK1 and RV1, the level control. The signal is coupled to the input op-amp IC1, which has a gain of 10 or 100 depending on the choice of value of R1. If R1 is 100k, the gain of this stage is 10, for 10k the gain is 100. The output of IC1 is coupled to the 'AM input' of IC2 and also to the input circuitry of the output buffer/mixer via C7 and RV4b.

In this application, the VCO in the XR2206 can produce either sine or triangle waveforms by means of switching a resistor in or out of circuit with SW1. A triangle waveform contains odd harmonics, which give a 'rough' or 'dirty' sound. A sinewave with little distortion has almost inaudible harmonics and thus sounds 'clean'. This is important, as we shall see shortly. The frequency of the VCO can be varied over the range from 3 Hz to about 5 kHz by means of RV3, the 'frequency' control. The frequency is determined by the values of C4, R7 and RV3. The AM input of IC2, pin 1, has a dc bias applied to it via R6, the bias voltage being determined by a divider network between the two supply rails consisting of R3, RV2 and R4. RV2 permits variation of the bias so that critical balancing of the XR2206's multiplier can be achieved to 'null out' the carrier signal (from the VCO). This 'null' control is normally adjusted to produce zero output with no audio signal input.

When an audio signal is applied, the multiplier in the XR2206 produces a *double sideband suppressed carrier* output signal. The output is taken from pin 2, via the internal buffer. Let's take a simple case to show what the multiplier does. Say the VCO is set to a frequency of 1 kHz. With the multiplier balanced there is zero output. Now, if a signal at 440 Hz ('A') is applied to pin 1 of the XR2206, the resultant output will be two frequencies: 1440 Hz and 560 Hz (the sum and the difference). Note, no trace of the carrier — this is a result of using a *balanced* mixer or multiplier. Now, say the audio input is 440 Hz (again), and the VCO is set to 5 Hz. The output will be 445 Hz and 435 Hz. Now, as every musician knows, two instruments tuned a few Hertz apart will produce a 'beat' when sounded together. The beat is perceived as an amplitude variation of the sound — if the effect is deliberately obtained, it is called 'tremolo'.

This applies for the case where the carrier is a 'pure' sinewave. If the carrier contains harmonics, then these too will produce sum and difference products when multiplied with the audio input signal and a complex output will result. Thus for a 'clean-sounding' output, switch SW1 to SINE, for a 'dirty-sounding' output, switch SW1 to TRIANGLE.

The output from the multiplier in the XR2206 is taken from pin 2 (from the internal buffer, as mentioned before). It is coupled to RV4a via C6. Now, RV4 is a dual-gang potentiometer with the 'bottom' end of RV4a connected to the 'top' end of RV4b. With RV4 at the fully anticlockwise position, no signal from pin 2 of IC2 is coupled to the input of IC3, while the full output of IC1 is coupled to the input of IC3. With RV4 at the fully clockwise position, the full output from pin 2 of IC2 is coupled to the input of IC3, while none of the output from IC1 is coupled to the input of IC3. Thus by varying RV4 from one extreme to the other you can obtain a varying proportion of 'direct' to 'modulated' signal.

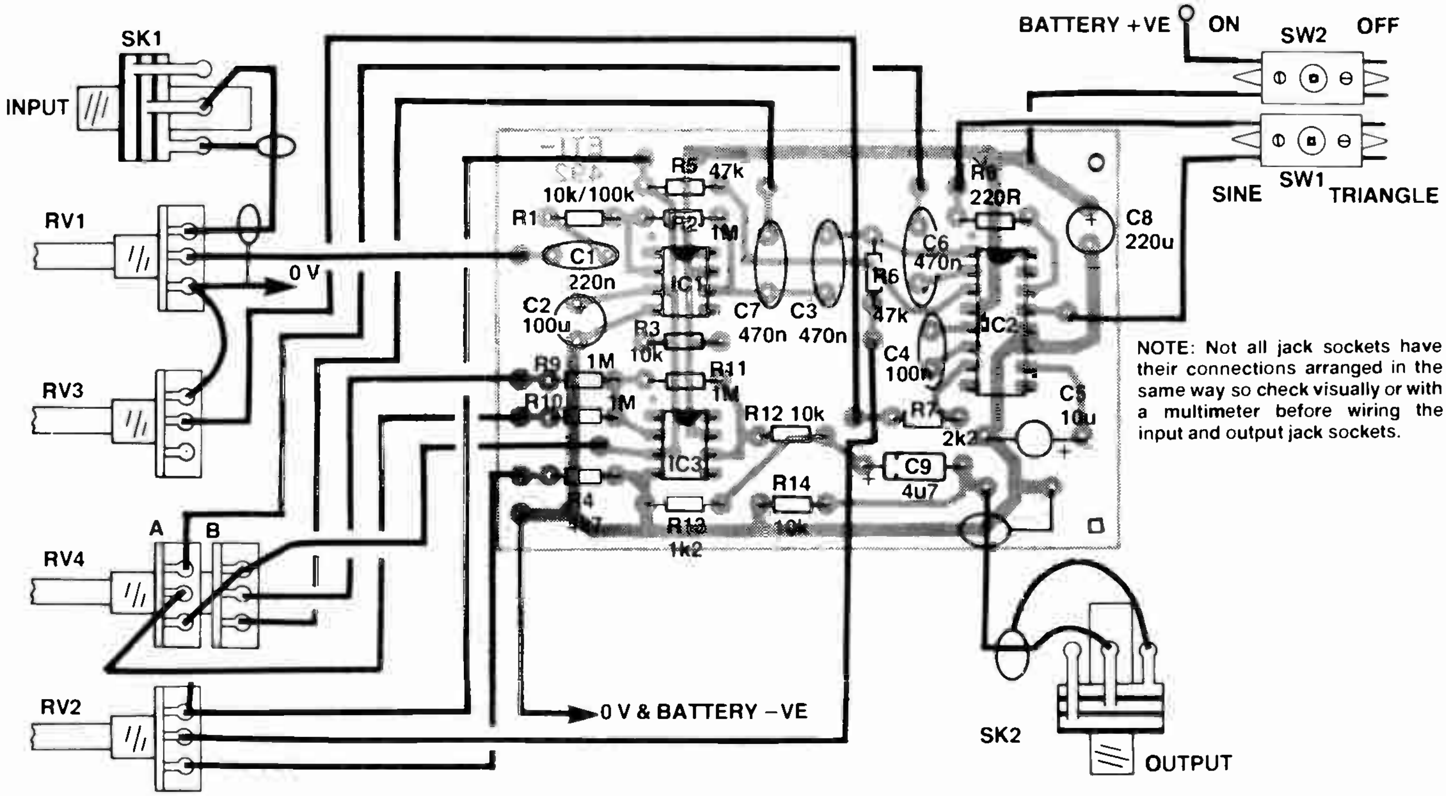
The output from IC3 is passed to SK2 first via an attenuator (R12, R13) that provides a division of 10 so that with the gain of IC1 set at 10 (R1 100k) the project has unity gain. From the attenuator the signal passes to SK2 via C9. R14 provides a dc return for the output circuit. If you wish, R13 may be omitted and R12 replaced by a link.

Capacitor C8 is a supply rail bypass, and capacitor C5 is a bypass for the internal reference of the XR2206. The non-inverting inputs of IC1 and IC3 are biased up to half the supply rail voltage by strapping them to the junction of R3 and RV2. This is done to provide a 'virtual earth' rail for these two ICs, which normally require a dual supply rail, whereas the XR2206 does not. Capacitor C2 serves as a bypass for this virtual earth rail. The multiplier direct output requires tying to the virtual earth rail also, as shown in the XR2206 application notes, and R5 does this. Note that the supply voltage can be anywhere between 9 V and 15 V. The circuit only draws a few milliamps (roughly, between 10 mA and 15 mA or so) and may be readily battery operated.

oscillator — spans a frequency range from 3 Hz to 5 kHz using a single control pot. To 'harden' or 'soften' the effect produced a 'triangle' or 'sine' oscillator waveform can be selected by a switch, and a two-channel mixer with a 'pan' control pot is incorporated on the output so that you can blend the 'direct' to 'modified' sounds to provide some control over the effect. In addition, a 'null' control has been provided as it is necessary to reduce the level of the carrier signal fed through to the output from the IC's modulator or multiplier.

The project can be operated from input levels as low as a few millivolts (e.g: microphone) or line levels of 100 mV or greater (e.g: preamp output, such as the 'effects send' on a mixer).

The Sound Bender may be powered from a supply ranging from 9 Vdc to 15 Vdc and draws typically between 10 mA and 15 mA current. A small dc plugpack would make an ideal power supply.



Alternatively, it may be battery operated.

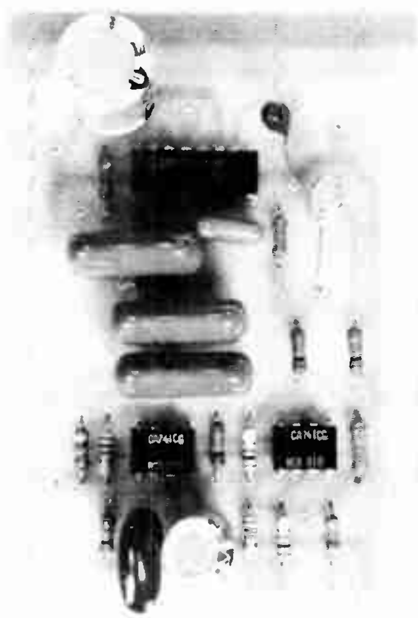
Construction

We have not described details of a case, front panel, etc, as this project will undoubtedly find a wide variety of uses and we leave it to individual contructors to arrange their own housing. Fortunately, housing is not critical, providing the controls are not mounted too far from the pc board. Leads from the board to the controls should be kept as short as possible, less than 300 mm preferably, as this avoids possible feedback and hum pick-up problems. If the unit is to be mounted in other equipment, keep it away from transformers and mains leads, or thoroughly shield it, again to avoid hum pick-up.

Construction should commence with the pc board. Solder IC1 and IC3 (the two 741s) in place first, taking care that you get them the right way round. They both face in the same direction. Next, insert all the resistors and solder them in place. You'll have to decide at this stage whether you use a 10k or a 100k resistor for R1, as noted with the circuit. The XR2206, IC2, may be inserted next. As it is a CMOS IC, remove it from its packing carefully, taking care only to handle the ends of the pack, not touching the pins. Carefully insert it in the board and solder pin 4 and then 11 and 12. Then solder all the other pins. Take care not to overheat any of the ICs when soldering them in place. Now all the capacitors may be inserted and soldered in place. Watch that you get the orientation of C2, C5, C8 and C9 correct.

Now you're ready to wire up all the external major components. These can be mounted in any order, to suit yourself, but keep the wiring to RV1 (input level) and RV4 (mix) separated to avoid possible feedback. Use shielded cable where indicated (input and output).

Our overlay and wiring diagram gives an overall guide as to assembly and wiring of the unit.



Using it

To try out the Sound Bender, connect a supply (battery, plugpack or bench supply — what-have-you) and connect the output to the input of an audio amplifier. We used the ETI-453 General Purpose Amp Module (see page 71). As we wanted to use a microphone, a 10k resistor was used for R1. Set the Sound Bender's input level to zero, set the mix control fully clockwise, and turn up the audio amp's input gain. SW1 may be set to sine or triangle, it doesn't matter. If you don't hear a whistle, rotate the frequency control until you do. Then vary the null control until you obtain minimum output. This null will be quite sharp so take it slowly. A big knob on the pot shaft or a small vernier would assist. A 10-turn pot here might seem extravagant, but some users may find it useful. ▶

PARTS LIST ETI-492

Resistors	all ½W, 5%
R1	100k
R2,9,10,11	1M
R3,12,14	10k
R4	4k7
R5,6	47k
R7	2k2
R8	220R
R13	1k2
RV1	10k lin.
RV2	5k lin.
RV3	2M2 lin.
RV4	100k dual lin.
Capacitors	
C1	220n greencap
C2	100u/16 V electro.
C3,6,7	470n greencap
C4	100n ceramic
C5	10u/25 V electro or tant.
C8	220u/16 V electro.
C9	4u7/16 V axial electro.
Semiconductors	
IC1,IC3	741
IC2	XR2206
Miscellaneous	
ETI-492 pc board; two SPDT miniature toggle switches, two phono sockets; case to suit; wire; knobs; nuts and bolts, etc.	
Price estimate	**$28 — $35**

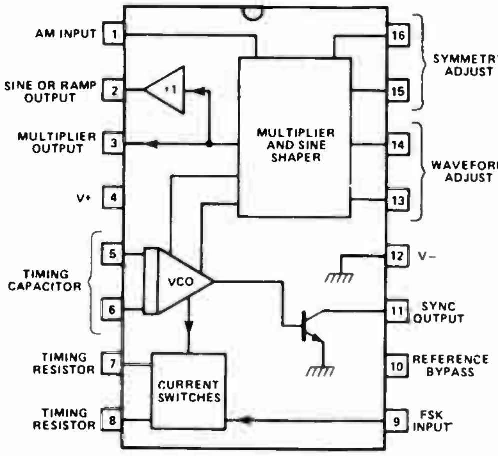


Figure 1. **Internal block diagram and pinout for the XR2206 function generator IC.**

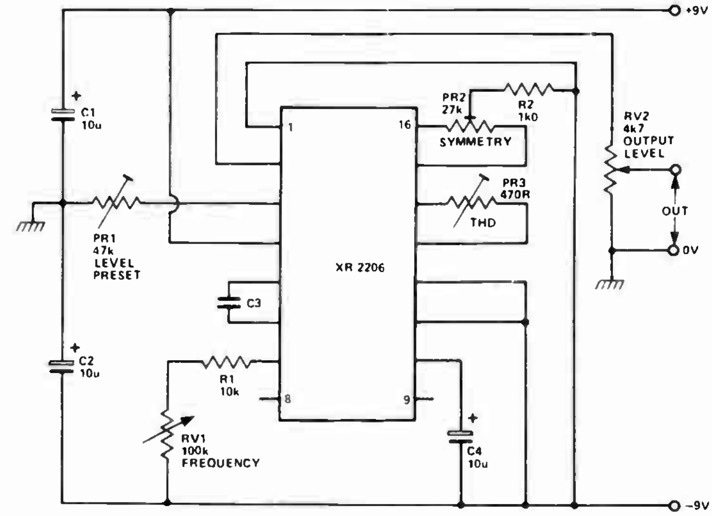


Figure 2. **High-performance sinewave generator. See Table 1 for values of C3.**

A BRIEF LOOK AT THE XR2206

The XR2206 integrated circuit is undoubtedly one of the most useful function generator or waveform generator chips available. It can generate sine, square, triangle, ramp and pulse waveforms at frequencies ranging from a fraction of a Hertz to several hundred kiloHertz, using a minimum of external circuitry. The frequency can be swept over a 2000:1 range using a single control voltage or resistance, and sinewave distortion can typically be as low as 0.5%. The chip incorporates special built-in modulation facilities that enable the generated waveforms to be subjected to AM or FM control, or to phase-shift or frequency-shift keying.

The XR2206 chip is housed in a standard 16-pin DIL package and can be powered from either single or split supplies in the range 10 to 26 V. The sinewave output of the device has maximum amplitude of about $2V_{RMS}$ and output impedance of 600R. The frequency stability of the IC is excellent, being about 20 ppm/°C for thermal changes and 0.01% V for supply voltage changes.

Figure 1 shows the pinout and internal block diagram.

WAVEFORM GENERATION

The XR2206 is a reasonably easy IC to use for basic waveform generation. A high-performance sinewave generator is shown in Figure 2. It requires a split supply rail, but total harmonic distortion at the output is typically less than 0.5%. Adjustment of trimpots PR2 and PR3 with a distortion meter connected to the output is necessary, but the THD holds over the frequency range. Trimpot PR1 requires setting for correct operation first, however. Disconnect PR3 (to obtain triangle output), then adjust PR1 until no clipping of the output waveform is visible on a 'scope hung on the output.

Note that the signal appearing on pin 3 of the IC is similar to that on pin 2, but has lower distortion and higher output impedance. Also, the signal on pin 3 is very nearly symmetrical about 0 V but that on pin 2 has an offset of several hundred millivolts. If desired, a slight dc offset may be applied to pin 3 to reduce the offset on the output signal from pin 2 — as shown in Figure 3.

The XR2206 will generate linear triangle waveforms by deleting PR3. A sine/triangle/square wave function generator is shown in Figure 4. Rise and fall times of the square wave output are typically 250 ns and 50 ns respectively, with pin 11 loaded by 10 pF.

C3	FREQUENCY RANGE
1u0	10 Hz TO 100 Hz
100n	100 Hz TO 1 kHz
10n	1 kHz TO 10 kHz
1n0	10 kHz TO 100 kHz

Table 1. Values of C3 for different frequency ranges

MODULATION

The amplitude of the pin 2 output signal of the XR2206 can be modulated by applying a dc bias and a modulating signal to pin 1 as shown in Figure 5. The amplitude of the pin 2 signal varies linearly with the applied voltage on pin 1 when this voltage is within 4 V of the half-supply value of the circuit; in split-supply circuits, of course, the half-supply value equals 0 V. When the pin 1 voltage is reduced below the half-supply value the pin 2 signal again rises in direct proportion, but the phase of the output signal is reversed. This last-mentioned phenomenon can be used for phase-shift keyed (PSK) and suppressed carrier AM generation.

The pin 1 terminal of the IC can also be used to facilitate gate-keying or pulsing of the pin 2 output signal. This can be achieved by biasing pin 1 to near pin 2, and then imposing the gate or pulse signal on pin 1 to raise the pin 2 signal to the desired turn-on amplitude. The total dynamic range of amplitude modulation is 55 dB.

The frequency of oscillation of the XR2206 is proportional to the total timing current (I_T) drawn from pin 7 or 8, and is given by:

$$f = \frac{320 \times I_T}{C} \text{ Hz}$$

where I_T is in milliamps and C is in microfarads.

The timing terminals (pins 7 and 8) are low-impedance points and are internally biased at 3 V with respect to pin 12. The frequency varies linearly with I_T over the current range 1 uA to 3 mA. Consequently, the frequency can be voltage-controlled by applying a voltage in the range 0 to +3 V between pin 12 and the timing terminal via a suitable resistor, so that the timing current is determined by the resistor value and the *difference* between the internal (+3 V) and external (0 to 3 V) voltages. This simple technique can be used to either frequency sweep the generated signals using an externally applied sawtooth waveform, or to frequency-modulate the waveforms with an external signal.

Figure 6 shows the basic method of applying FM to the standard XR2206 circuit. Here, the external modulation signal is applied to the junction of R1-RV1 via blocking capacitor C1.

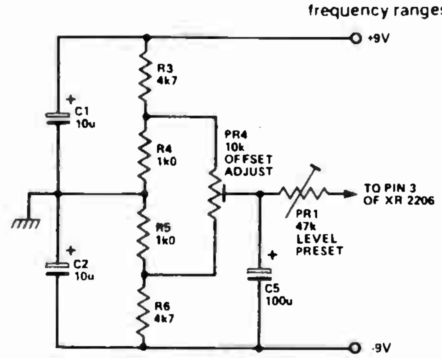


Figure 3. **Add-on modification for applying a limited dc offset for output signal dc nulling of the circuit in Figure 2.**

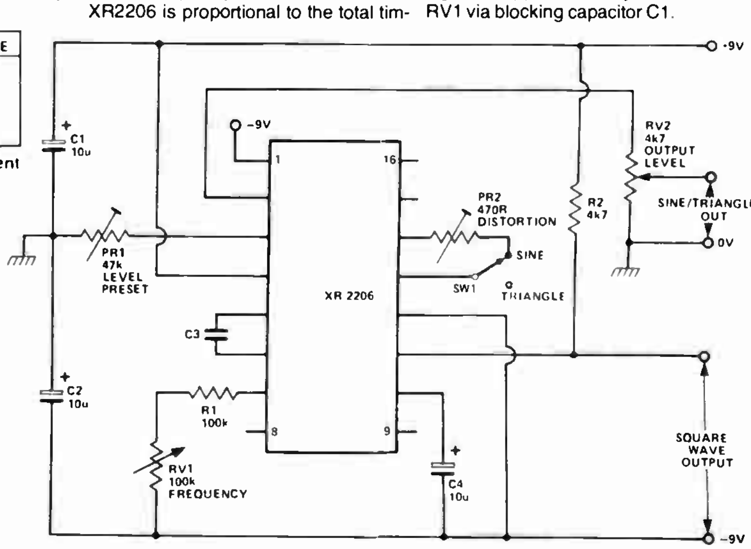


Figure 4. **Simple sine/triangle/square wave generator. See Table 1 for values of C3.**

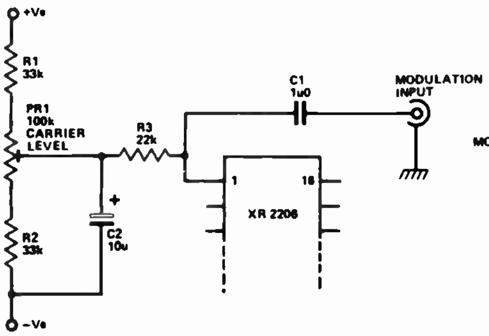


Figure 5. **How to add an amplitude modulation (AM) facility (split-supply circuit, as per Figure 2).**

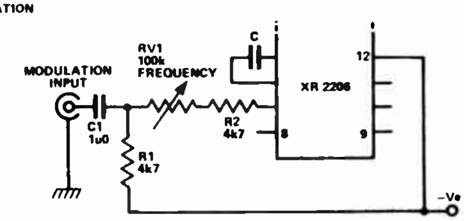
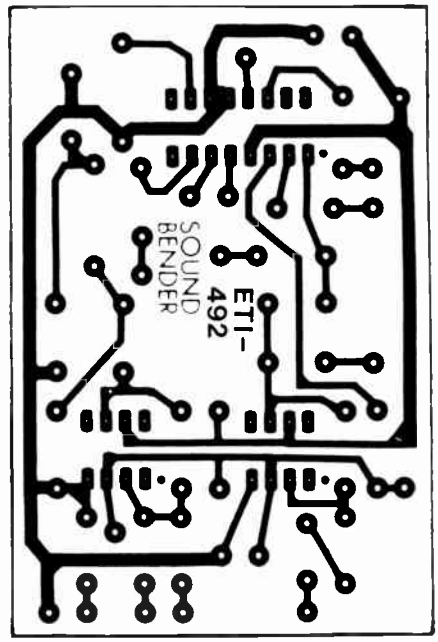


Figure 6. **How to add a frequency modulation (FM) facility (split-supply circuit, as per Figure 2).**



Note that the null is not perfect and there is some carrier feedthrough. However, this can be reduced and the effect-to-carrier leakage ratio improved by judicious adjustment of the mix and input level controls. Keeping the mix control somewhat back from the all-modulated end and the input level up does the trick.

Having nulled the multiplier, plug in a mike or signal source and advance the input level. Set SW1 to triangle for a 'dirty' sound. If the frequency is set to minimum (fully anti-clockwise), you will hear a tremolo effect. Setting the frequency control about two-thirds advanced you will be able to obtain 'Daleks', 'Darth Vaders', etc, with speech input. With the mix control you can 'fine tune' the effect quite well — we rarely used it fully clockwise (all modulated).

The unit performs best with a 'single signal' input — such as voice or one instrument (such as a guitar). Complex signals, such as from a band or orchestra, end up a confused jumble.

With SW1 set for a sinewave modulating signal, the effect produced is 'soft', while the effect produced when SW1 is set for a triangle wave modulating signal is 'hard'.

We noted that there seems to be some slight delay in the signal through the IC — or the modulator produces a similar effect — and the output sounds a bit 'echoey', especially when the frequency is very low, as on the tremolo effect.

Have fun with your Sound Bender! ●

electronic music/audio/video

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Project 475

AM Tuner features wide bandwidth and low distortion

Design: Ken Woods
Article: Staff

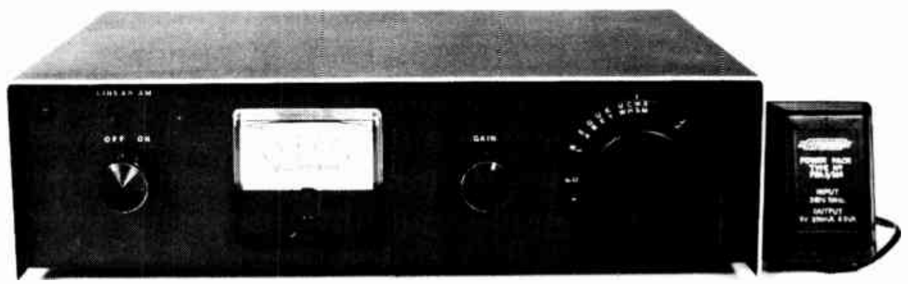
Now the 'FM boom' has arrived, the AM stations are fighting back with wide bandwidth, good quality sound. This tuner, though simple to build and get going, provides extraordinarily good performance.

WE WERE SURPRISED to learn recently that many of the AM broadcast stations have been transmitting 'full bandwidth' signals for quite some time, it seems they're 'fighting back' at the recent boom in FM with all the new stations coming on to the VHF band.

This tuner has been designed to take advantage of this situation. Broadcast stations are permitted to transmit an audio bandwidth that rolls off at 15 kHz. That means an AM broadcast station will have a nominal bandwidth of 30 kHz. At first glance this seems a little out of kilter as frequency spacing in the 530 - 1650 kHz AM broadcast band is 9 kHz. However, stations serving a particular area are generally allocated frequencies no closer than 54 kHz. Hence, a wideband tuner may be used to exploit the good quality reception possible from stations transmitting 'full bandwidth' programme material.

Design

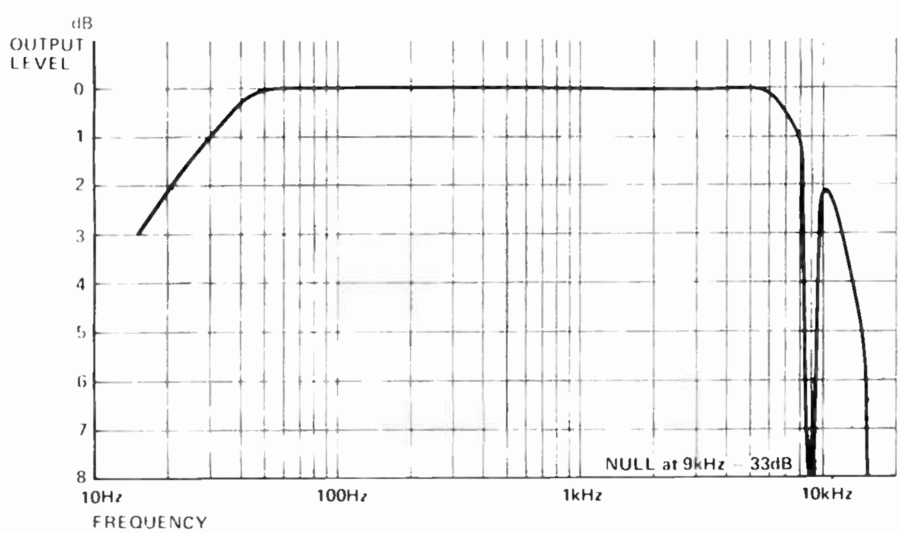
The designer, Ken Woods, has chosen to employ a 'tuned radio frequency' design to achieve low intermodulation distortion, low phase distortion and good transient response. There are only two tuned circuits. The overall selectivity is determined solely at the front end, at the frequency selected. The parameters of the input double-tuned circuit have been arranged to provide the required bandpass selectivity with good attenuation outside the passband, to reduce unwanted noise and interference. This circuit arrangement provides low phase distortion as it has a slowly varying phase change across the pass band and no phase reversals. Transient response of this particular arrangement is also good as there is minimum signal delay from input to output and the Q has been carefully 'tailored' to reduce the 'flywheel effect' of multiple tuned circuits (ringing).



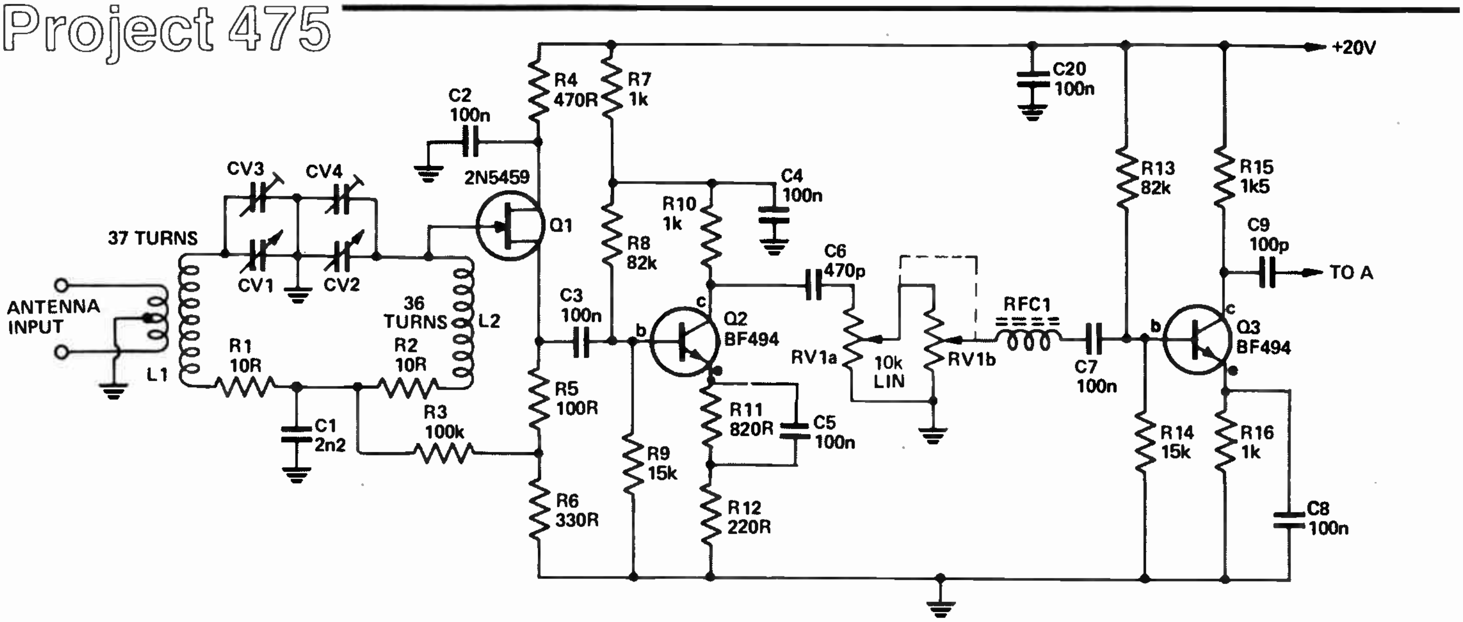
The tuner is housed in a simple, yet attractive, case. A plugpack is used to power the unit.

In addition, the circuit has a virtually constant bandwidth characteristic right across its tuning range. It's a little too complex to go into here, but readers looking for a good reference could hardly do better than consult "The Radiotron Designer's Handbook", by F. Langford-Smith, published by AWV-RCA, Chapter 8, section (iii) 'Complex Coupling' (Page 420 in ours, the Fourth Edition).

The tuner has a manual RF gain control and no AGC so that the evil of AGC distortion, prevalent with conventional superhet AM tuners, is eliminated. Actually, rather than varying the gain of one or more of the amplification stages, the RF gain control is an attenuator, ▶



Overall frequency response of the receiver. The response is 3 dB down at 15 Hz and 12 kHz. The whistle filter provides 33 dB of attenuation at 9 kHz and is 3 dB below midband response at just over 8 kHz and just below 10 kHz.



HOW IT WORKS — ETI 475

The tuner employs a TRF design where all the amplification and selectivity is achieved at the actual frequency of reception, prior to the detector.

The required selectivity characteristics are provided by the input tuned circuit which is arranged to tune the whole broadcast band, from 530 kHz to 1650 kHz. Several stages of untuned RF amplification follow the tuned circuit. A low distortion diode detector removes the audio programme information from the selected station's RF carrier and this is fed to a 9 kHz whistle filter — which provides a deep 'notch' to remove interstation heterodynes — followed by an audio output stage.

For best, low noise reception, a balanced antenna input is provided. Antennas are discussed in the text. The two tuned circuits, comprising L1, L2 and the dual-gang tuning capacitor CV1/2, are mutually coupled by C1. Individually, each tuned circuit has quite a high Q by virtue of the totally 'closed' magnetic field provided by the pot core. The reactance of C1, whilst small, is sufficient to overcouple the two tuned circuits, providing a 'double-humped' response (see the accompanying diagram). To remove the 'dip' in the middle of the response, the overall circuit Q is 'damped' by two, low value resistors — R1 and R2.

A FET source follower, Q1, isolates the input tuned circuits from the first RF amplifier stage, Q2. The input impedance to the gate of Q1 is quite high and this avoids loading the coupled tuned circuits which would reduce the Q. Gate bias dc return is via R3, R2 and L2. The source of Q1 is coupled to the base of Q2 via C3. This first stage of RF amplification has a gain of about five and is stabilised by having part of the emitter bias resistance unbypassed (R12).

An RF gain control is placed between the first and second RF amplifier stages. A dual-gang potentiometer, RV1a and RV1b, connected in a cascade configuration, provides very smooth control over the signal level.

The input stage FET has its drain decoupled from the supply rail by R4 and C2 while Q2 has its collector circuit and base bias decoupled by R7 and C4.

The third stage of amplification is provided by Q3 which operates at full gain. To prevent VHF parasitic oscillation in this amplifier, a wideband RF choke, RFC1, has been inserted in series with the input to the base.

A further two stages of amplification follow, before the detector. Transistors Q4 and Q5 are direct-coupled and the collector of Q5 drives the diode detector via C11.

The detector is a voltage-doubling type with degeneration to reduce distortion. In addition, there is negative feedback from the detector to the emitter of Q4 via C12, further reducing distortion. A signal strength meter has been provided as a tuning aid. It measures the dc output level from the detector. Capacitor C14 provides smoothing for the meter, removing any audio signal influence.

RF 'smoothing' from the output of the detector is provided by R24 and C15 forming a low pass filter that passes audio (3 dB down at 28 kHz) but bypasses the RF. The output of this is coupled to the input of the audio output stage via the 9 kHz whistle filter. This is a parallel tuned circuit made up by L3 and C16. This provides a 'notch' in the audio response, attenuating any 9 kHz interstation whistles by more than 30 dB. The coil is constructed in a pot core which ensures high Q and a narrow bandwidth notch.

The audio output stage consists of a Darlington pair emitter follower stage, Q6 and Q7. This has a high impedance input, so as not to load the whistle filter, and a low impedance output suitable to drive the 'tuner' input of an amplifier. The collectors of Q6 and Q7 are decoupled from the supply rail by R29 and C18.

Power is supplied from a plug back, thus removing a possible source of hum pickup, and a voltage-doubler rectifier involving D3, D4, C21 and C22. Extra supply filtering is provided by R31 and C23. A front panel LED power indicator, LED1, is supplied directly from the rectifier output. Switch SW1 is used to turn the tuner on and off.

located between the first and second stages of RF amplification. Its purpose is to allow adjustment of the signal level so that at no time is the detector overloaded.

The detector employed features some signal degeneration, via R23, to reduce distortion and negative feedback to one of the RF amplifier stages to further reduce distortion. The overall distortion

PARTS LIST — ETI 475

Resistors all ½W, 5%

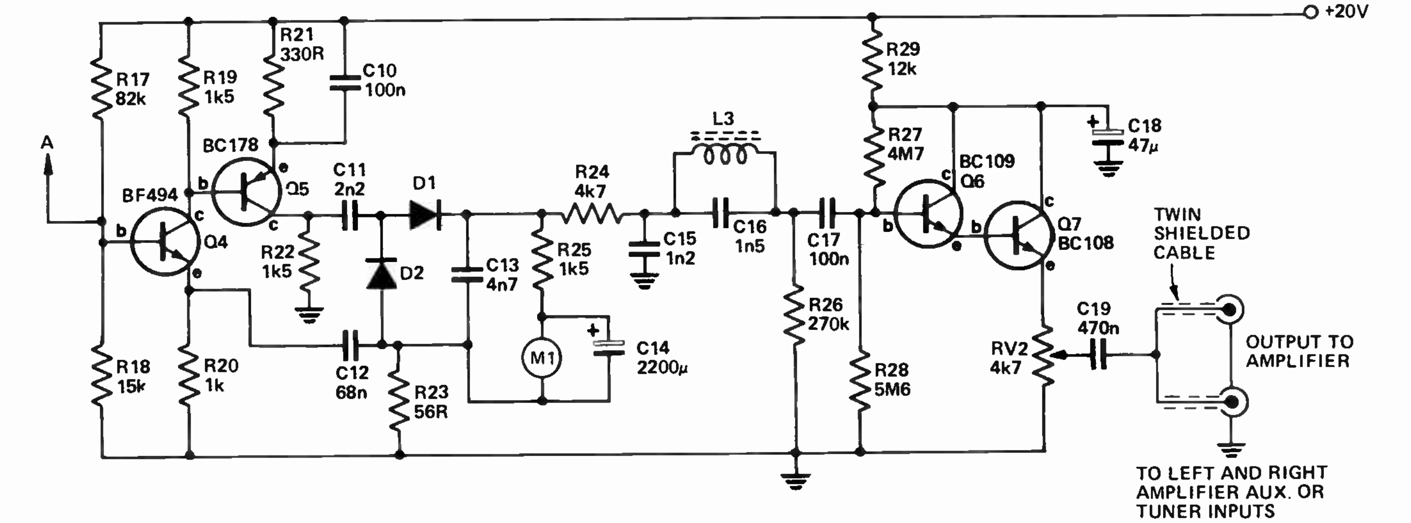
R1,R2	10R
R3	100k
R4	470R
R5	100R
R6,R21	330R
R7,10,16,20	1k
R8,13,17	82k
R9,14,18	15k
R11	820R
R12	220R
R15,19,22,25	1k5
R23	56R
R24	4k7
R26	270k
R27	4M7
R28	5M6
R29	12k
R30	2k2
R31	270R

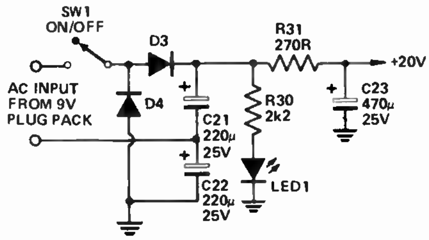
Potentiometers

RV1	10k linear dual pot.
RV2	4k7 flat mounting large trimpot.

Capacitors

C1,C11	2n2 greencap
C2,3,4,5,7, 8,10,17,20	100n greencap
C6	470p ceramic or styroseal
C9	100p ceramic or styroseal
C12	68n greencap
C13	4n7 greencap
C14	2200u/16V electro
C15	1n2 greencap
C16	1n5 see text
C18	47u/25V electro, axial lead type
C19	470n greencap
C21,C22	220u/25V electro





C23 .... 470u/25V electro

Variable Capacitors
CV1,CV2 .... 415p dual gang variable capacitor (Roblyn RMG — 2), see text
CV3,CV4 .... 40p film dielectric trimmer, Philips type 2222 808 01027 (grey case) or similar.
Inductors .... see coil winding details

Semiconductors
D1,D2 .... AA119, OA95, see text
D3,D4 .... 1N4004, A14A or sim.
Q1 .... 2N5459
Q2-Q4 .... BF494
Q5 .... BC558, BC178
Q6 .... BC549, BC109
Q7 .... BC548, BC108
LED1 .... red LED, TIL220R or sim.

Miscellaneous
SW1 .... one pole, two position rotary switch
M1 .... 1 mA meter, University TD48, Minipa MU45 or similar
Planetary dial drive with flange (Watkin Wynne type 4511 DAF or similar); large black knob 40 mm dia., aluminium disc 55 mm dia. for scale plate (see text); two small black knobs; two antenna terminals: one black, one red; two-pin plug and socket for power input; 9V plug pack, Ferguson type PPA9 - 500 or similar; length of twin shielded cable with two RCA plugs on one end; five pc board standoffs, box to suit (see text); ETI-475 pc board.

RFC1 — Philips VK200 wideband choke or six-hole ferrite bead (type 4312-020-31550) with a length of 22 swg tinned copper wire passed through it five times.

of this tuner proved to be significantly lower than that of our Wavetek laboratory signal generator, so we are unable to present a reliable distortion measurement. Listening tests confirm the low distortion characteristic of this tuner.

As some interference may be experienced in particular areas from distant stations propagating via the ionosphere (this generally occurs at night), a 9 kHz whistle filter has been incorporated. It provides an attenuation in excess of 30 dB at 9 kHz and the notch bandwidth is about 2 kHz maximum. If you do not experience any difficulties with this sort of interference, the whistle filter may be dispensed with.

A tuning or signal strength meter has been provided and it has several functions:

(a) To provide a positive tuning indication.
(b) To facilitate optimum signal strength control.
(c) To indicate when signal overload occurs.

It's a very handy aid when adjusting antennas or when setting the RF gain control.

Instruments are not absolutely necessary to align the tuner, although a signal generator makes it somewhat quicker.

An inexpensive plug back is used to provide power to the tuner. This has the advantage of removing the transformer from the tuner's chassis, eliminating a possible source of hum.

Construction

The tuner is housed in a chassis made from a 240 x 290 mm sheet of 16 gauge aluminium bent into a U-shape measuring 290 mm wide by 180 mm deep and 80 mm high. The lid was made from a 205 x 455 mm sheet of 16 gauge aluminium. It is bent and mounted so that it overhangs the front panel of the chassis by 10 mm and the rear by 15 mm. This results in quite a neat, professional-looking unit.

The power switch, power LED, signal strength meter, gain control potentiometer and planetary reduction drive are all mounted on the front panel. The antenna terminals and two-pin power input socket are mounted on the rear panel. The output lead passes through a rubber grommet.

Commence by marking out and drilling the holes in the chassis. This is probably best done before bending it up. Mark out accurately, as per the metalwork drawings. The size of hole required for the meter depends on the meter used. The one on our prototype is a Minipa MU-45 type. It fits neatly on the front panel and has a pleasing appearance. However, other types may be used, such as the University type TD48. This is a little smaller than our meter, but will do equally well.

Two lengths of aluminium angle are bolted to each side of the chassis and the lid is secured to these with either bolts (which mate with tapped holes in the angle pieces) or self-tapping screws. The aluminium brackets are cut from a single 320 mm length of 13 mm (½") angle. This is readily obtainable in hardware stores. Mark and drill the lid, then bend it carefully to shape. The aluminium angle pieces are best marked up and drilled using the chassis, and then the lid, as a template.

At this stage, the chassis and lid could be sprayed matt black inside and out, or anodised — if you're willing to go to that expense. ▶

COIL DETAILS ETI-475

L1

Primary: two turns wound bifilar with thin plastic insulated hookup wire (wound last).
Secondary: 37 turns pile wound, 34 SWG enamelled wire (wound first).

L2

36 turns, 34 SWG enamelled wire (0.23 mm).
L1 and L2 are wound on Philips P18/11 pot core assemblies, 3D3 material, ue = 68, with adjusters.

Philips part numbers

Pot core	4322/022/24454
Adjuster	4322/021/32170
Former	4322/021/30270
Washer	1811/HWI
Clip	1811/HPC

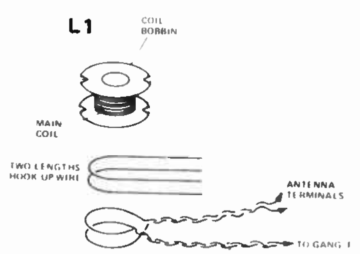
L3

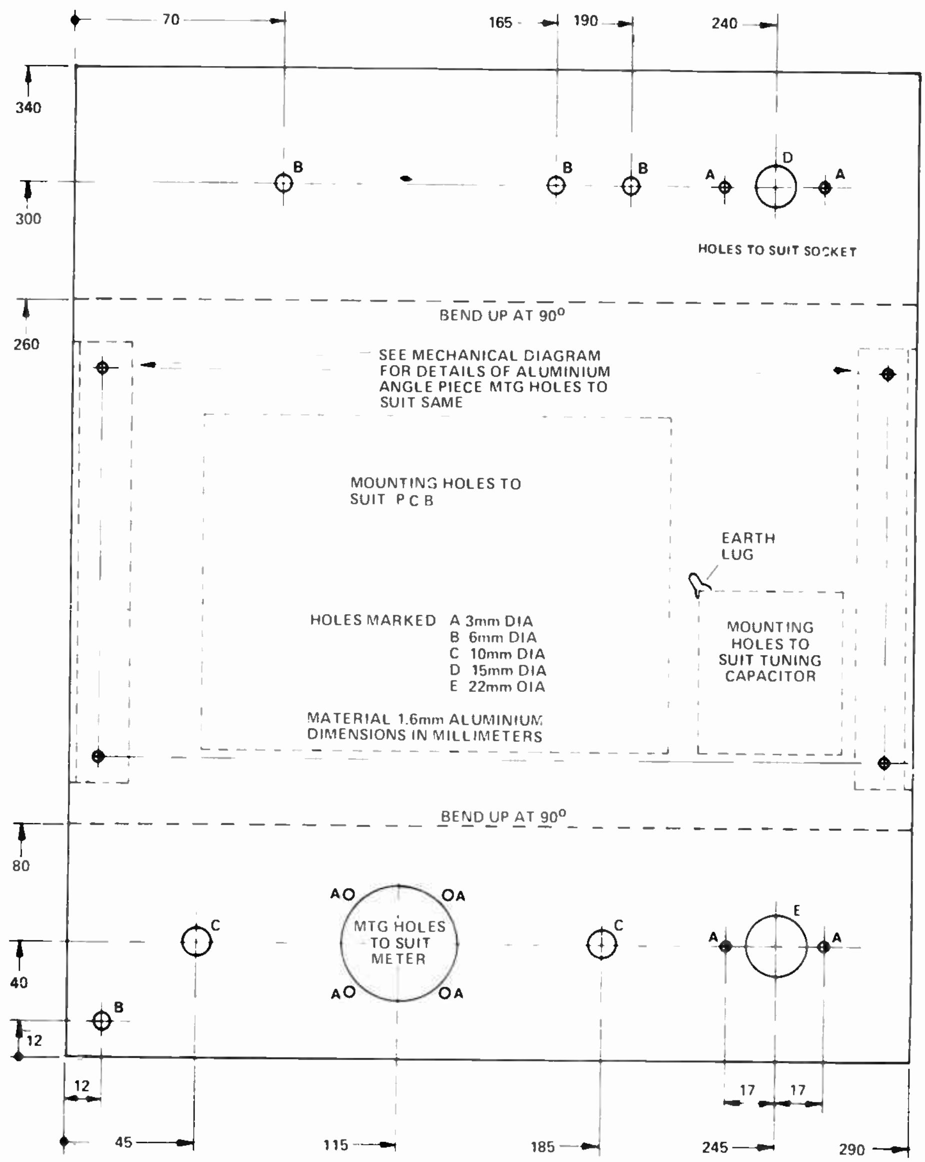
540 turns 36 SWG enamelled wire (0.2 mm).
L3 is wound on Philips P26/16 pot core assemblies, 3B7 material ue = 220, with an adjuster.

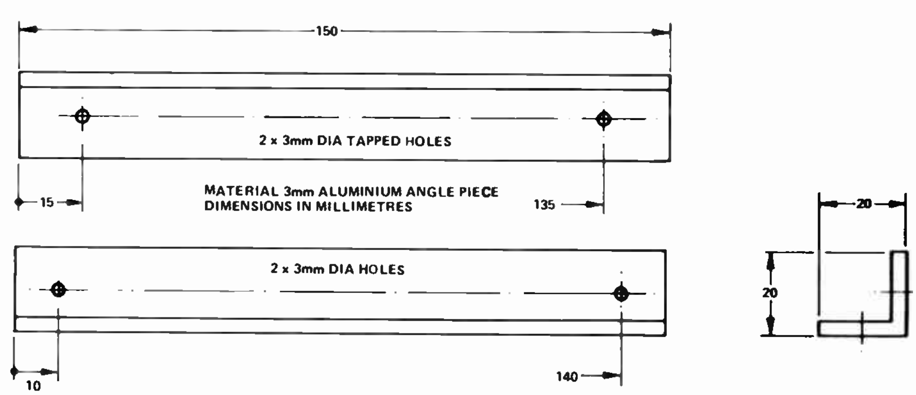
Philips part numbers

Pot core	4322/022/28294
Adjuster	4322/021/30810
Former	4322/021/30330
Clip	2616/HPC

No washer is used with the large pot core.







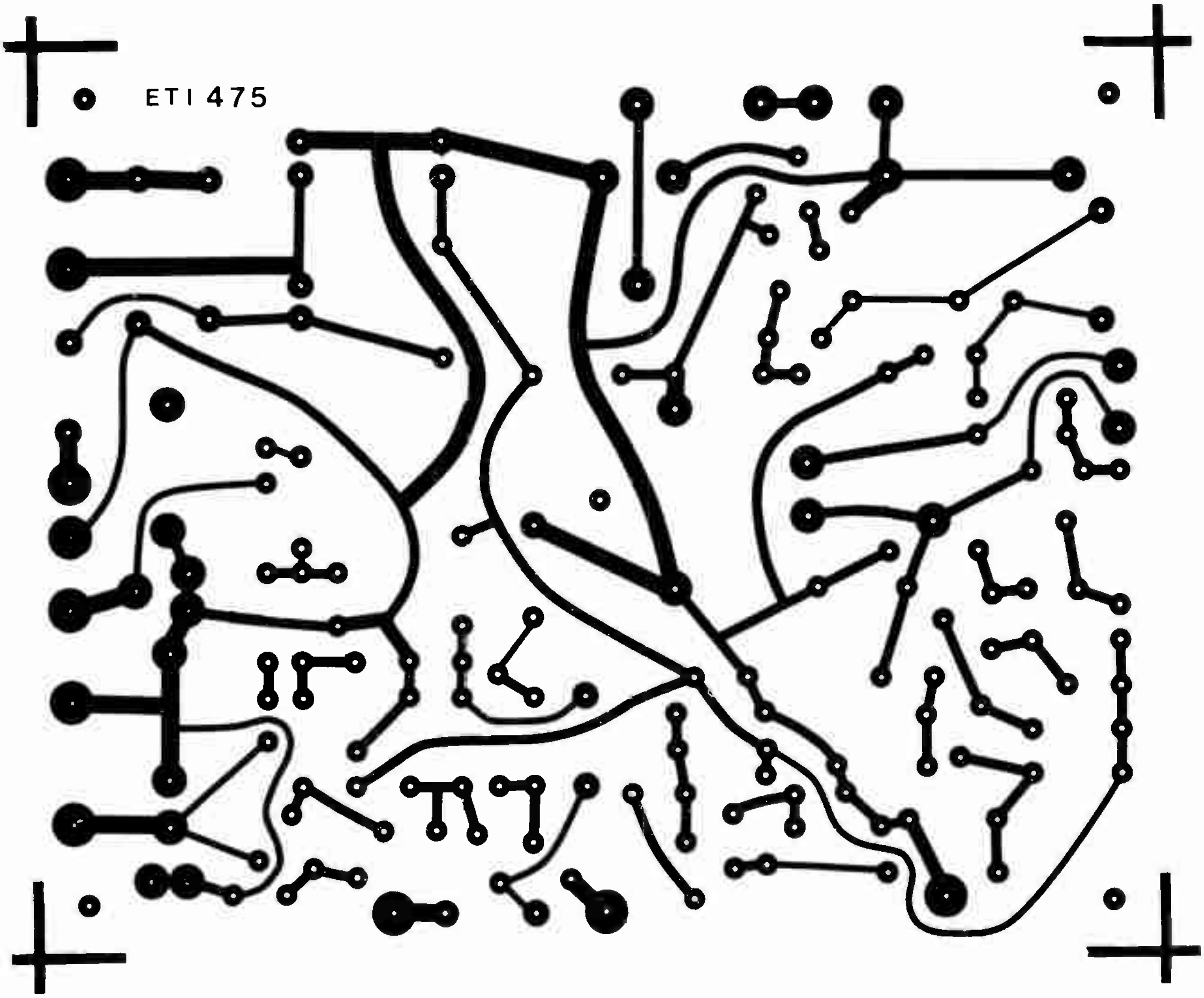
When the chassis is completely prepared, the major components can be mounted on it. The planetary drive should be mounted first. It is best to mount it on adjustable standoff pillars, as shown in the photographs, to allow the position of the dial flange to be adjusted. This should protrude from the front panel about one or two millimetres.

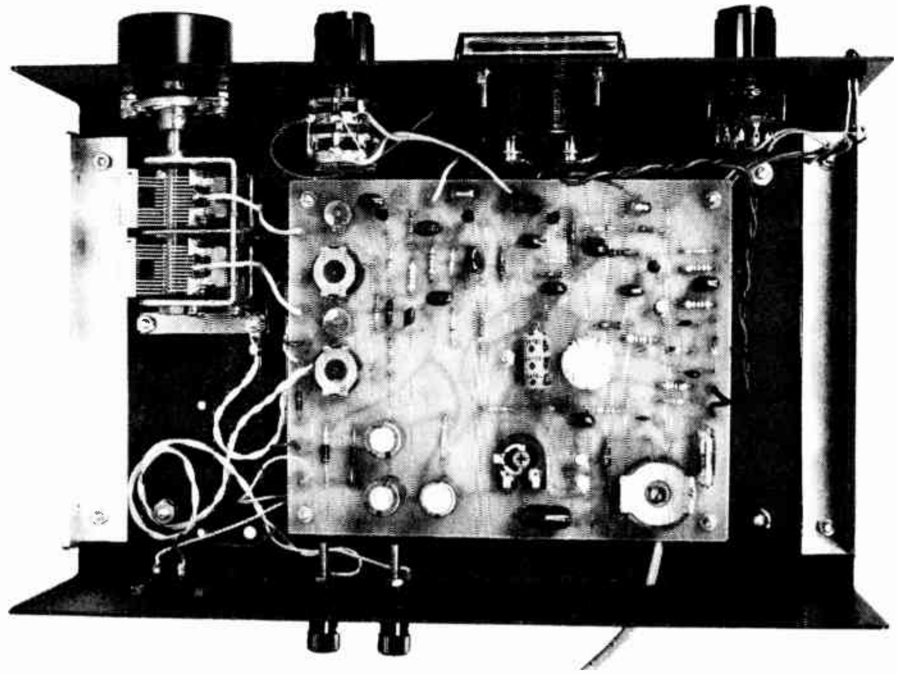
The dial was made up from a disc of 18 gauge aluminium (though 16 gauge or even a thinner gauge would be OK. The local stations were marked on the dial with white rub-down lettering (such as Letraset or Geotype). This disc is attached to the planetary dial flange by two screws supplied with the dial drive. The complete assembly is shown in the exploded-view diagram.

Since it will probably be difficult for most readers to cut a clean circle from aluminium, without the facilities of a machine shop, we have reproduced a suitable dial from which a Scotchcal copy may be made. If you use the metal Scotchcal, you can leave the backing paper on it and attach that as your dial.

The tuning knob on the prototype is a little special. It consists of a large diameter knob with a turned-up aluminium 'cup' pushed over it (the inside diameter forms a snug fit to the knob). We'll leave the knob to your ingenuity. A large knob is recommended as it provides smooth control of the tuning, a good grip and enhances the appearance.

The next step is to mount the tuning gang. It is very important that no strain is placed on the planetary drive from the shaft of the tuning gang. Careful, correct alignment will ensure this. The tuning capacitor is mounted on small standoff nuts and the shaft is carefully aligned, using washers or something ▶


ETI 475



similar, to pack the standoffs so that it mates with the planetary drive properly.

The rest of the chassis-mounted components may now be secured in place.

Assembling the pc board comes next. Mount all the components with the exception of the three pot cores. Take care with the orientation of the electrolytic capacitors, diodes and transistors. The BF494 transistors have an unusual lead configuration — the emitter lead is in the centre. When soldering the trimmer capacitors in place, don't use too much heat or strain the leads while soldering them. This avoids possible distortion of the plastic case and problems when adjusting them.

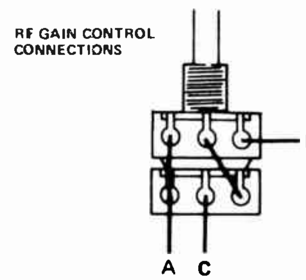
All components should be mounted right down on the pc board using minimum lead length. The transistors should have leads no longer than 5 mm.

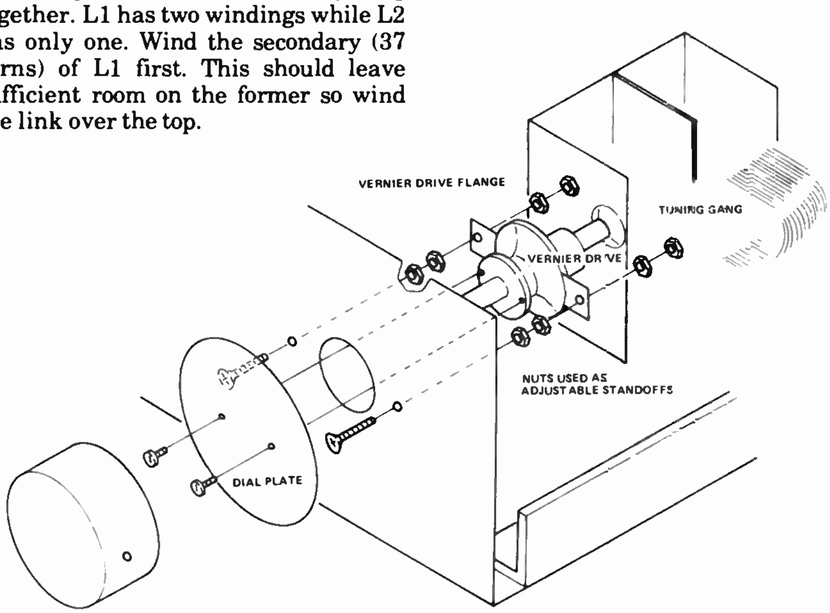
The detector diodes can either be gold-bonded germanium types like the AA119s recommended, or germanium types such as the OA47. The gold-bonded diodes will give lower detector distortion but may be difficult to obtain. We tried both types and could not detect any audible difference.

The resonating capacitor for the whistle filter, C16, should be either a styroseal or mica type to avoid drift in the tuning as this is a high-Q circuit.

The two input tuned circuits use 18 mm diameter pot core assemblies. Each assembly contains two ferrite halves, an adjustor, former, washer and clip. When assembled, the clip solders into the pc board and holds everything together. L1 has two windings while L2 has only one. Wind the secondary (37 turns) of L1 first. This should leave sufficient room on the former so wind the link over the top.

The link is bifilar wound with two lengths of thin, plastic insulated hookup wire. With the two wires parallel, wind a single turn around the former, over the top of the secondary winding. The start lead of one wire is then twisted together with the finish lead of the other. These two leads are twisted together for 150 mm or so and will connect to the antenna terminals after the coil is assembled. The other two leads are also twisted together, for 100 mm or so, and will be joined and soldered to an earth lug under the tuning gang mounting bolt adjacent to the coil locations on the pc board.





EXPLODED DIAGRAM OF VERNIER DRIVE AND DIAL ASSEMBLY

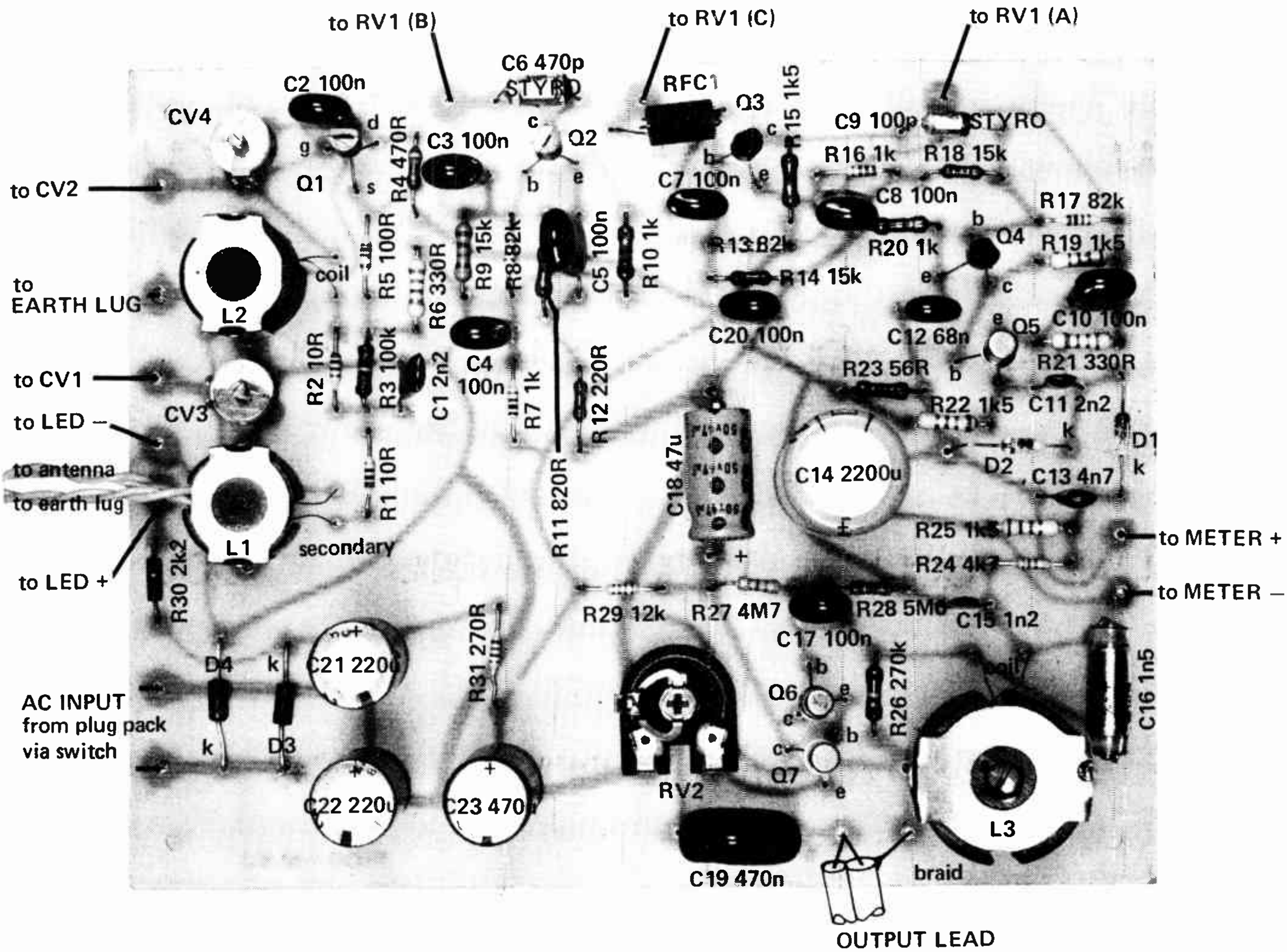
The accompanying exploded diagram should make the assembly of this link winding clear.

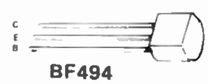
Arrange the wires so that all the link wires come out one side of the bobbin and the secondary wires come out the other.

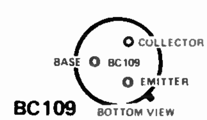
Assemble the two ferrite pot core halves over the former, place the washer on top of the assembly and slip on the clip. The washer ensures an even pressure is transmitted from the clip to the ferrite assembly and should always be used with these small pot cores. Insert the adjuster carefully into the centre hole of the core using a small aligning tool. Take care as it cuts its own thread in the nylon insert and any forcing can damage this.

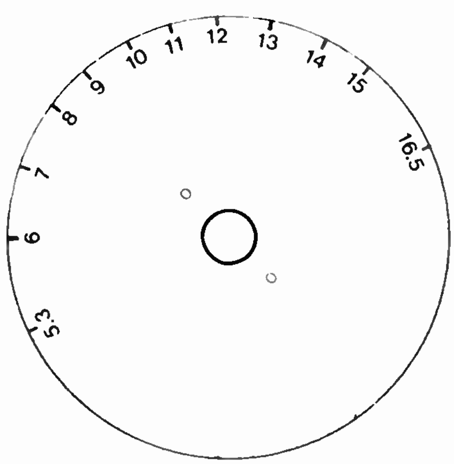
Solder the complete assembly into the pc board (before everything falls apart!) orienting the assembly with the link connections facing the edge of the board and the secondary connections toward the centre.

The second front end pot core, L2, is assembled in a similar way but note









Full-size reproduction (negative) of the dial.

that it has no link.

The whistle filter, L3, uses a 26 mm diameter pot core which is assembled in a similar fashion to the other two except that it does not require a washer under the clip. Wind the wire on the bobbin as detailed in the accompanying box. The wire should almost fill the former, so be careful to wind it firmly, laying the turns neatly on the bobbin.

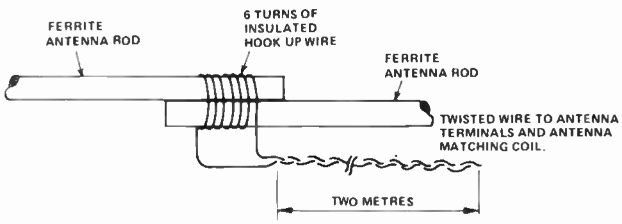
Tuning up

Turn the unit on and connect it to your stereo amp. Turn trimpot RV2 on the pc board fully clockwise. Connect an antenna and turn the RF gain control fully clockwise. Set the two trimmer capacitors at mid range (you can see the plates), and the ferrite adjusters in L1 and L2 at half depth. Tune over the range and you should hear some stations. Select a station at the high frequency end of the range. By adjusting the two trimmers in turn, and the tuning capacitor, bring the station to its correct position on the dial. This requires a little juggling with the three adjustments. Next, find a station at the low frequency end of the dial and repeat the procedure, this time tuning in the station using the two ferrite pot core adjusters and the tuning capacitor. You now have the receiver roughly aligned.

Repeat the process but this time you can set the dial to where a station should be located on the dial (according to the markings) and tune the two trimmers for maximum signal on the meter. Repeat for a low frequency station, adjusting the pot cores.

Repeat once more, just to make sure, ▶

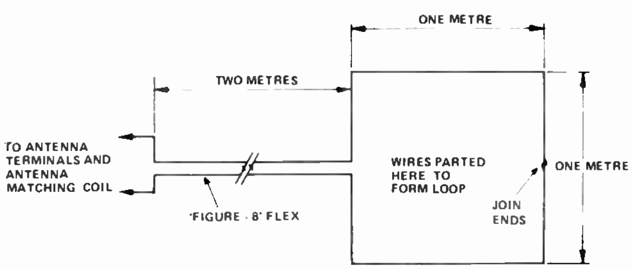
SUGGESTED ANTENNAS



For strong signal areas

This antenna is constructed of two 13 mm diameter ferrite rods. We suggest Neosid types, of F8 material, 12.7 mm diameter by 100 mm long. Most ferrite rods intended for broadcast band 'loopstick' antenna applications will probably suffice though, as construction is not all that critical; performance may vary though.

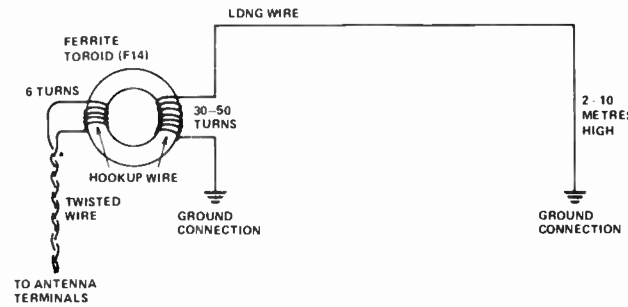
Six turns of hookup wire are wound firmly around the two rods and twisted leads two metres long connect to the tuner's antenna terminals. The two ferrite rods may be extended as illustrated or pushed together so that they overlap over more of their length. As shown, the antenna provides maximum sensitivity and least directivity. The rods should be oriented for best reception for the station or stations of interest.



For medium to weak signal areas

A loop antenna can provide very good results where signals are not too strong. The loop illustrated is made by taking a length of 'figure-8' flex, parting the wires over a length of two metres, joining the free ends and forming a loop of one metre per side. The feedline should be about two metres long. It can be longer but performance may deteriorate. The plane of the loop should be oriented towards the transmitters for best results.

A larger loop may be constructed to improve pickup. Note that a rectangular loop may also be used, if more convenient. Experimentation will indicate which arrangement provides satisfactory results. A matching coil, as shown, may improve results.



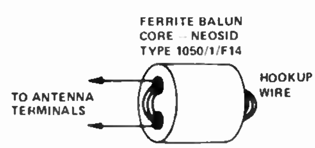
For weak signal areas

If you live in a weak signal area, or want to 'chase the DX', this antenna should provide good results. Run a long, straight wire as high above the ground as you can reasonably manage and as long as will fit in your property (but less than 5 km !). Connect the furthest end to a ground stake. The opposite end connects to the primary of a 'matching' transformer wound on a ferrite toroid. The illustration shows the main details.

The toroid should be of a material having an initial permeability of about 200 to 300, t least, and an A_L factor of around 100 to 150. A Neosid toroid of F14 material, 25.4 mm outside diameter, 19.05 mm inside diameter and 9.5 mm high should do nicely. It's not too critical, and some experimentation may be in order.

Note

The impedance of the antenna will have some effect on the tuning of L1. This may necessitate minor re-alignment of the adjuster in L1 if you change antennas or change the position of the antenna. Check the alignment at the low frequency end of the band.



ANTENNA MATCHING COIL

For use with small loops 6-8 turns
For use with large loops 2 turns

and you should notice that all the stations are in their correct positions on the dial. If you change antennas you may need to make a slight re-adjustment to L1.

The whistle filter can be adjusted by tuning across the dial until you find a 9 kHz whistle between two stations. Wind the ferrite adjuster on L3 in until the tone disappears. If no whistles are found, wind the adjuster all the way out.

An alternative method is to use a signal generator with external AM modulation. Set the modulation to 9 kHz, at about 80%, and tune in the signal. Use the ferrite adjuster on L3 to null the audio from the speaker.

Always use proper adjusting tools (these are available from most suppliers) to avoid breaking the adjusters or affecting their correct operation. The pot core adjusters are delicate and should be treated with kindness. Overzealously screwing them in and out will almost certainly result in permanent damage.

This fairly simple alignment technique yielded an overall bandwidth, at the −3 dB points, extending from 15 Hz to 12 kHz. For those readers with a little more perseverance, this can be improved with judicious adjustment of the tuned circuits.

Operation

With the unit aligned you can connect an antenna and enjoy sounds from an AM tuner you never thought possible!

The output level to your stereo amplifier may be set by adjusting RV2, a trimpot on the pc board. The setting will depend on the signal strengths of the different stations at your location and the tuner input sensitivity of your amplifier. It is best set by experiment.

The antenna required will depend, again, on the signal strengths of the various stations at your location. It is a wise move to spend a bit of effort here as it pays off. The accompanying box shows a variety of antennas that will generally provide more than satisfactory performance under different conditions.

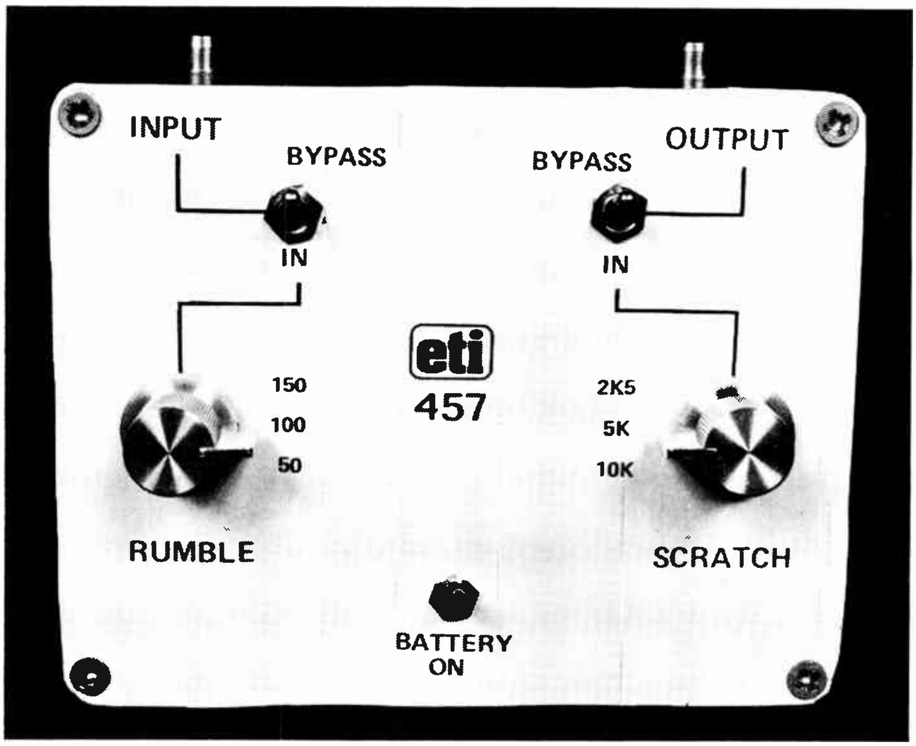
We tried the tuner in different areas of metropolitan Sydney and were quite impressed with the performance. At a location on the north side, where local stations are quite strong, we used a simple ferrite rod antenna with good results. At our offices in the eastern suburbs, where some stations are relatively weak we used a small loop antenna with excellent results. Sound quality is remarkable — you have to hear it to believe it! ●

Cut the crackle and get rid of the rumble with our

Scratch and rumble filter

Staff

If many of your cherished older recordings are showing signs of old age and long use, or if you wish to transcribe your collection of 78s onto tape, this scratch and rumble filter project should help improve the sound quality.



SO YOU SAY you look after your records, but you probably have some that are scratched? Perhaps you lent them to someone who didn't care for them or they might be cherished rarities you picked up in a secondhand shop. Some of the best 'original' recordings are still on 78s especially if you're into jazz, bluegrass, blues or country music. Whatever the cause, clicks, pops and severe surface noise can spoil your listening pleasure, no matter how enthusiastic you are about the artist or the item recorded.

One other source of unwanted sound comes from the turntable in the form of rumble — a low frequency sound which can make your teeth grind in sympathy! If you look at the speaker cone whilst playing a turntable suffering from rumble, you may see it move in and out, although you may not hear anything. This is subsonic rumble and can be detrimental to the performance of the speaker system. The main cause of rumble is a less than perfect turntable transmitting vibration from the motor and bearings to the stylus. Rumble has almost been cured with the introduction of belt drive and good direct-drive turntables but these can suffer from wow and flutter. That's another story, though. Low frequency acoustic feedback from the speakers to the turntable can also occur if the acoustic mounting of the turntable is not up to scratch.

The high frequency surface noise on a recording can be removed with a 'high cut' filter. This will also cut the highs on the recording but on old records this will not be so noticeable. Likewise, low frequency noise can be removed with a 'low cut' filter and again, some of the low frequency information is lost.

It is desirable to only modify enough of the amplifier's frequency response to reduce the problems, therefore we have included switchable cutoff frequencies for each filter. High frequency hiss can usually be removed with the 10 kHz filter while cracks will probably need a lower frequency cutoff.

The unit uses two active filters in series, one a low cut for the rumble filter, the other a high cut for the scratch filter. The filters provide an attenuation of 12 dB per octave at frequencies past the cutoff point and can be switched in and out independently. The unit is battery operated and designed to go between the turntable and the preamplifier on older stereo systems, or between the preamplifier and the main amplifier on modern systems. We have built each channel on a separate pc board to allow the unit to be used for either mono or stereo systems. We have shown only one channel for simplicity. If you wish to build a stereo version you will need to duplicate all components except the switches and batteries, and of course, the box. ▶

PARTS LIST — ETI 457

Resistors all ½W, 5%

- R1 .... 27k
- R2 .... 12k
- R3 .... 2k7
- R4,R5 .... 15k
- R6 .... 220k
- R7,R8 .... 4k7
- R9 .... 10k
- R10 .... 820R

Capacitors

- C1,C4 .... 68n greencap
- C2,C5 .... 100n greencap
- C3,C6,C9 .... 220n greencap
- C7,C8,C16 .... 1u tantalum
- C10,C13 .... 10n greencap
- C11 .... 22n greencap
- C12, C15 .... 4n7 greencap
- C14 .... 2n2 greencap

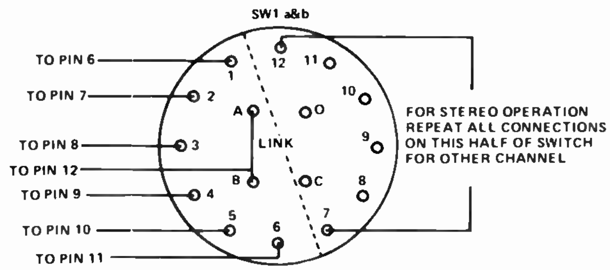
Semiconductors

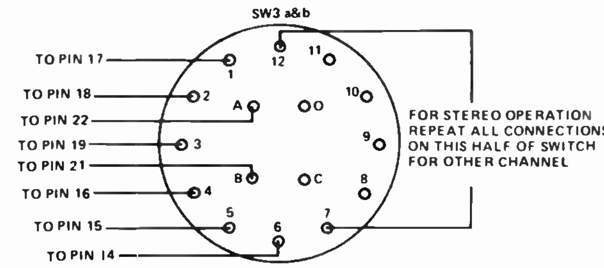
- Q1,Q2 .... BC549, BC109 or sim.
- LED1 .... red LED TIL220R or sim.

Miscellaneous

- SW1,SW3 .... four pole, three-way wafer switches
- SW2,SW4,SW5 ... DPDT miniature toggle switches

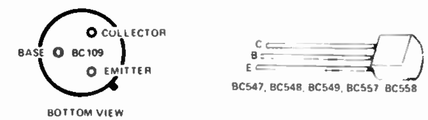
ETI-457 pc board; two RCA phono sockets; box to suit (120 mm × 95 mm × 55 mm); knobs; 9V No. 216 battery and battery clip.





SW1 LOW CUT RANGES	
1.	150 Hz
2.	100 Hz
3.	50 Hz

SW3 HIGH CUT RANGES	
1.	10 kHz
2.	5 kHz
3.	2.5 kHz



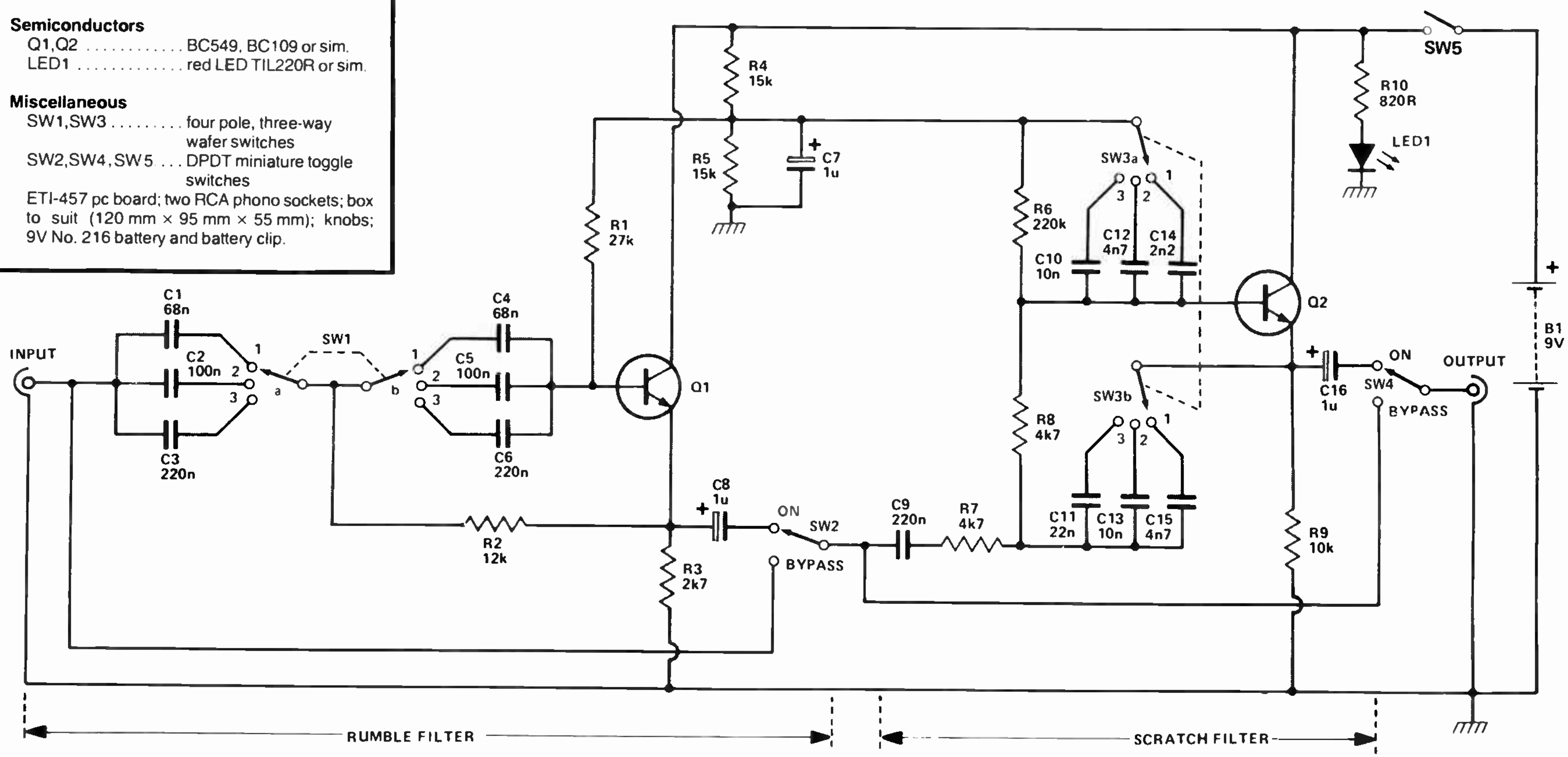


Figure 1 shows the block diagram of the mono version of the scratch and rumble filter. The input signal (from the turntable pick-up) is first fed through a high-pass filter, which rejects unwanted low-frequency rumble signals, and is then fed through a low-pass filter which rejects unwanted high-frequency scratch signals. Each filter can be bypassed via a simple switch if required, so the input signal can be passed through either one, both, or neither of the filters.

Figure 2 shows the circuit (a) and performance graph (b) of a simple single-stage passive high-pass filter. At low frequencies, capacitor C1 presents an impedance that is high relative to R1 so a lot of signal attenuation occurs between the input and output terminals. At high frequencies C1 presents an impedance that is low relative to R1, so negligible signal attenuation occurs between input and output.

The frequency at which the output signal is 3 dB down on the input signal is conventionally known as the BREAK frequency.

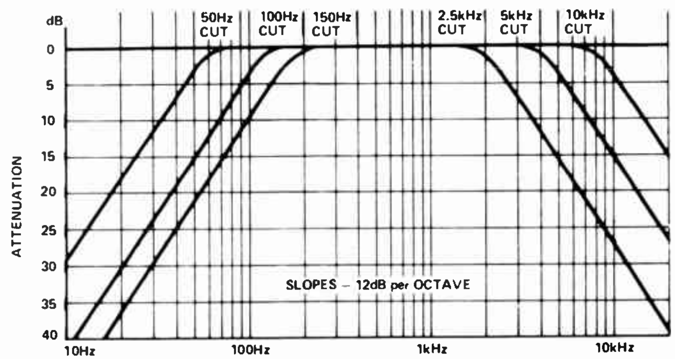
Note in Figure 2(b) that the graph shows a smooth roll-off or slope up to the break frequency point: a single stage filter has a slope or roll off of 6 dB/octave, i.e: the signal output level doubles if the input frequency is doubled.

A number of filter stages can be cascaded to give a roll-off of greater than the basic 6 dB/octave: usually, some kind of electronic buffering or feedback is used between the individual sections of a multi-stage pass filter system.

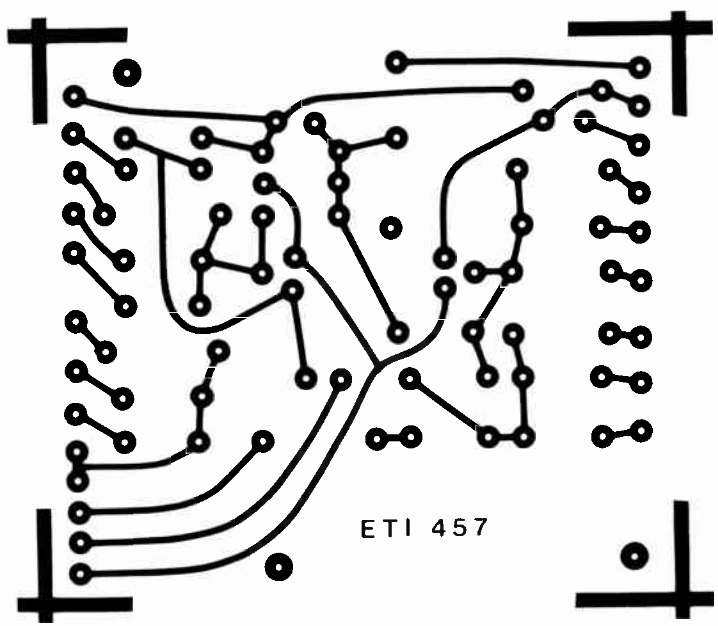
Figure 3 shows the circuit (a) and performance graph (b) of a two-stage high-pass filter. This design is known as a Butterworth filter, and is the type used as the rumble filter section of our project: It has a sharp break frequency, and gives a slope or roll-off of 12 dB/octave.

The basic high-pass filter of Figure 2 can be made to act as a low-pass type by simply transposing the positions of C1 and R1, as shown in Figure 4. Figure 5 shows the two-stage (second order) Butterworth version of the low-pass filter. This is the design that is used as the scratch filter in our project.

In the complete project (see main diagram) the high-pass or rumble filter is designed around Q1 and R1, R2 and C1 - C6, and the low-pass or scratch filter is designed around Q2 and R6 - R8 and C10 - C15. Resistors R4, R5 and bypass capacitor C7 provide the low-Z bias point for the two transistors. The low-frequency break point of the rumble filter can be varied via three-way switch S1, and the high-frequency break point of the scratch filter can be varied via S3.



FREQUENCY RESPONSE OF FILTER



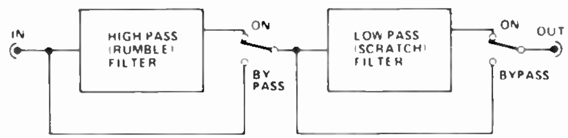


Figure 1. **Block diagram, scratch and rumble filter.**

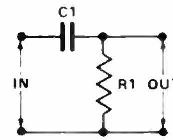


Figure 2 (a). **Simple passive high-pass filter.**

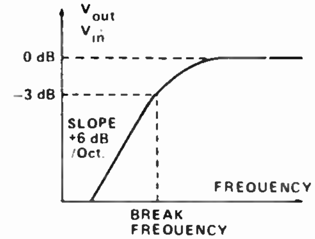


Figure 2 (b). **Frequency response.**

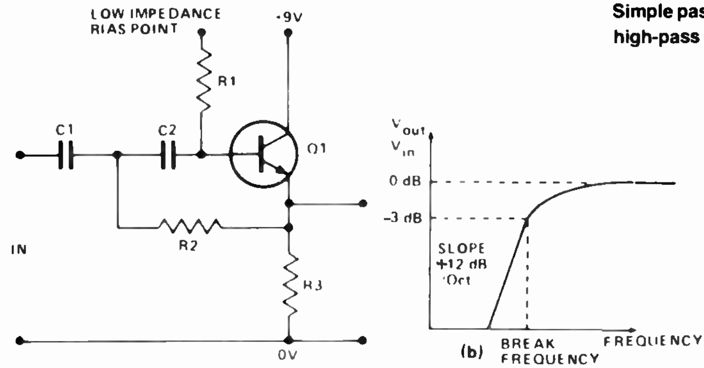


Figure 3 (a). **Two-stage active high-pass filter.** *Figure 3 (b).* **Frequency response.**

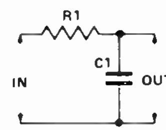


Figure 4 (a). **Simple passive low-pass filter.**

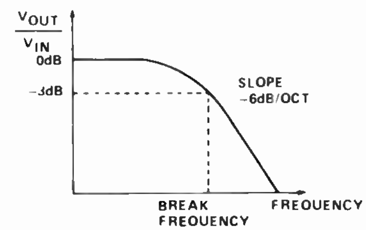


Figure 4 (b). **Frequency response.**

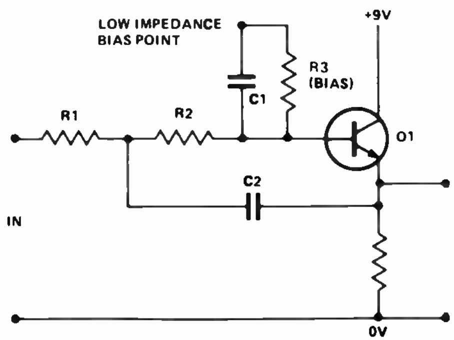


Figure 5 (a). **Two-stage low-pass filter.**

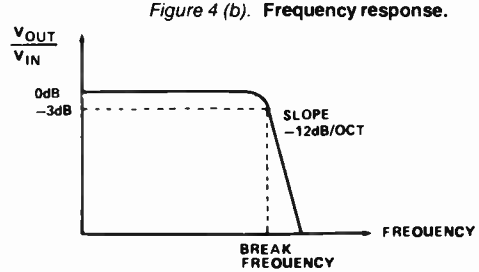


Figure 5 (b). **Frequency response.**

Construction

This description is confined to the mono version. A stereo version is readily assembled from two pc boards. As the switches are all available with a complete extra set of poles and contacts, these components need not be duplicated in a stereo version. Wiring will follow much the same course as described here.

We built our filter into a diecast box, but you may have something else in mind. A diecast box is very robust and provides generally good shielding, although a steel box would further reduce possible hum pickup.

All the switches are mounted on the lid of the diecast box. The pc board is 'hung' off the rotary switches and supported by tinned copper wire from the switch tags. This makes quite a rigid assembly and ensures short wiring to the switches. For a stereo version, the second channel pc board may be mounted behind the first, wired to the switches in a similar fashion.

The input and output sockets are mounted on one wall of the box and wired to the pc board with shielded cable. The bypass toggle switches are wired with hookup wire.

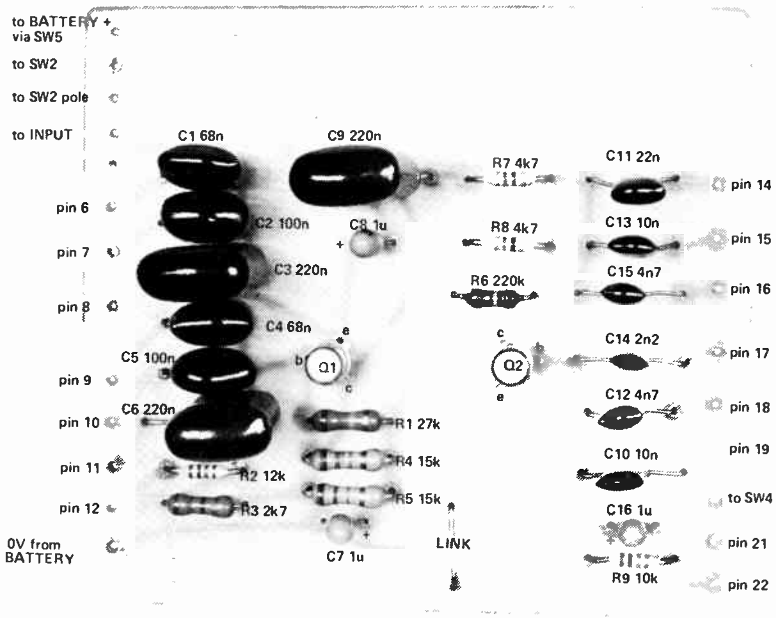
When assembling the pc board, watch out for the polarity of the tantalum capacitors and the orientation of the transistors, otherwise assembly is quite easy. With the components mounted in the board, the next step is to solder 50 mm lengths (longer for the second channel in a stereo version) of tinned copper wire to the lugs on the rotary switches.

Solder suitable lengths of insulated hookup wire to the points on the pc board that lead out to the toggle switches SW2, SW4 and SW5.

Carefully insert the tinned copper wires into their respective holes on the pc board and push the board up the wires to within about 15 mm or so of the switches. Take care not to bend any of the wires. Solder all the wires in place and cut off the excess. If building a stereo version, repeat this, taking care not to get the two channels' switch wiring tangled, pushing the second channel board to within 15 mm of the first.

Wire all the toggle switches input and output leads and you're ready to try it out.

We used a No.216 9 V battery. This is quite sufficient for the mono or stereo versions as current drain is only two milliamps per channel.



Component overlay for the pc board. Note that pin numbers 1 to 5, plus 13 and 20 , are not used. Resistor R10 and the indicator LED1 are mounted off the pc board (we have not used these).

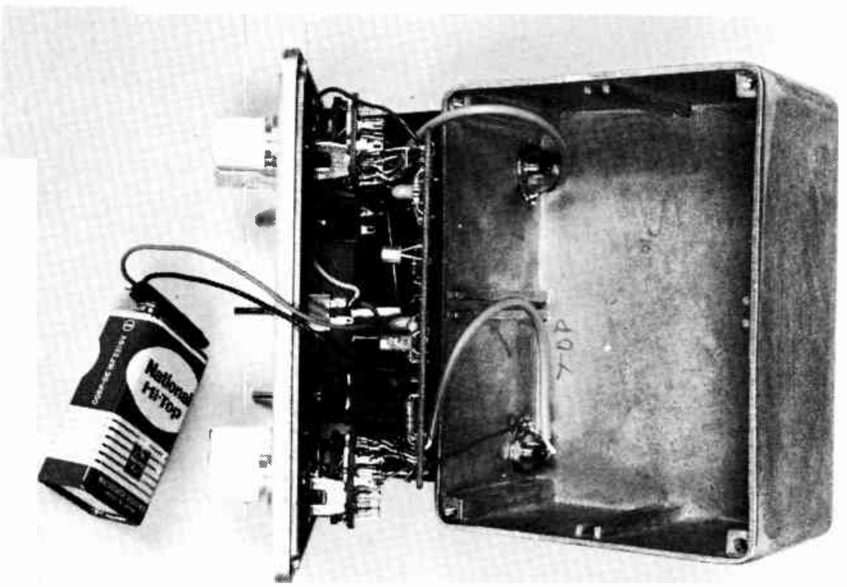
Operation

Operation is quite simple. With the unit's switches set for bypass, put on a record known to suffer from surface noise problems. Switch the unit on and put the scratch filter in circuit. Adjust the rotary switch and note the effect of the different filter frequencies. Do the same for the rumble filter.

A little experimentation should show up the best setting for each recording. It's worthwhile keeping a note of the setting with each record. The Scratch and Rumble filter is also a great aid when making tape recordings of old discs, particularly 78s.

With this unit, those old discs will find a new lease of life! ●

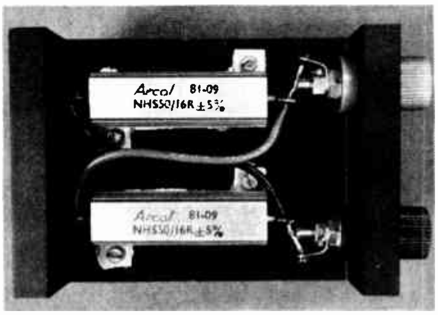
Internal view of the completed project showing the location of the input and output RCA sockets and the pc board 'hung' from the rear of the switches. Note that this is a mono version.



High power 'dummy loads' for audio amplifier testing

Apart from a multimeter and perhaps an oscilloscope, a resistive dummy load of 4, 8 or 16 ohms impedance capable of dissipating up to 100 watts is just about the most useful item of test equipment the audio enthusiast could have. Here are several ways to build one.

Andrew Kay
Roger Harrison



ETI-155b 8 ohm, 100 W dummy load using non-inductive resistors.

WHEN IT COMES to designing electronic equipment — from the very simple to the very complex — if one asked several designers how they would go about a certain design problem undoubtedly you'd get a different answer from each. Here again, we see a fine example illustrating that old saying — ". . . there's more than one way to skin a cat."*

The project staff at ETI have spent some considerable time over the past two years developing a variety of amplifiers. The fruits of these labours have been duly published and enjoyed by many readers. However, we've always lacked a *decent* dummy load for such work and have sort of *made do* with such contraptions as a string of one ohm 5W resistors dangled in a tub (plastic!) of water, lengths of electric jug element, etc, etc. Whilst jury-rigging such things is in the finest traditions of electronic design and development, the (more than) occasional mishap is not just a frustrating interruption but often a decided nuisance giving rise to dark mutterings, steam from the ears and shouts of "we'll have to get a *decent* dummy load ? ! !"

As no doubt many of our more intrepid readers, and/or do-it-yourself audio fanatics, have discovered, such things are hard/difficult/impossible (. . . delete whichever not applicable) to come by.

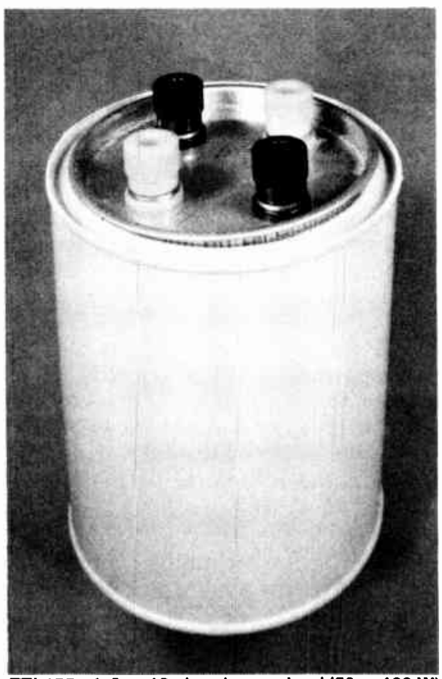
Then, Everest Electronics came to the rescue. Eagle-eyed readers will have seen the item we ran in News Digest in the March 1981 issue concerning the *Arcol* range of metal-clad power resistors carried by Everest. When the information arrived, quick as a flash we organised some non-inductive types for a dummy load. Several weeks later two 16 ohm 50 W non-inductive Arcol resistors arrived on the Editor's desk. An hour later we had a working dummy load! Naturally, everybody thought it would make a good project . . .

In the meantime, a freelance associate of ours, Andrew Kay, had desired exactly the same thing. Andrew, however, went about solving the problem a different way. He purchased a batch of one watt 1% resistors and made a 50 W dummy load. But, he figured, why not have a little more versatility and make two the same, allowing parallel and series connection to obtain a 4 ohm, 100 W dummy load or a 16 ohm 50 W dummy load as well as a twin 8 ohm 50 W dummy load enabling testing of both channels of a stereo amplifier at the same time! Frankly, we don't know why we didn't think of it earlier ourselves.

So — here follows the description of several ways to skin a cat/rockmelon/whatever, or build some high power audio dummy loads.

Multi-resistor method (ETI-155a)

By parallelling resistors of an appropriate value, one can obtain an effective resistance of the wanted value and ▶



ETI-155a 4, 8 or 16 ohm dummy load (50 or 100 W) using 98 390 ohm 1 W resistors.

* *For cat lovers we'll modify that to ". . . skin a rockmelon", or something similar!*

wattage rating. Now, the cheapest, most common power rating for carbon film resistors is one watt (1 W). To obtain a 50 watt resistor, 50 would need to be paralleled. To obtain an effective resistance of eight ohms, each 1 W resistor would have to have a value of 400 ohms. The nearest preferred value is 390 ohms. Fifty in parallel would give an effective resistance of 7.8 ohms which is about 2½% lower than the ideal eight ohms. However, 49 in parallel gives an effective resistance of 7.959 ohms — less than ½% out. If you require the tolerance of your load to be within 1% or better, then you'll have to use 1%, 1 W resistors. If you only require a tolerance of +/−5%, then the common 5%, 1 W variety will do the job. Either way, you're better off using 49 resistors so that the effective resistance of the load comes closer to the ideal eight ohms.

The dummy load described here consists of two eight ohm loads, which enables the testing of both channels of a stereo amplifier.

The idea itself is not at all new or original, having been used by radio amateurs for years to obtain resistive dummy loads for terminating radio transmitters while they are on test. The advantages of a dummy load for any kind of power source are:

- the power source (in this case an AF power amplifier) is presented with an ideal resistive load of the correct value,
- the chances of damaging expensive loudspeakers during experimental phases of construction are eliminated, and
- completely silent "full power" testing is made possible even for extended periods of time; which is great for public relations and your ears.

Essentially each dummy load consists of 49 high stability 1% metal film resistors connected in parallel to give a terminal resistance of 8 ohms. The author used cheap, readily available Beyschlag type MBE 0414 1 W series. Since the tolerance rating of the resistors is 1%, the upper tolerance limit for the combination is 8.04 ohms and the lower limit is 7.88 ohms. The number of resistors to be bought was a compromise between the desire for a result of exactly 8 ohms and the need to keep the cost to a minimum. Obviously, larger numbers of resistors could be used (say 70 x 560 ohms in parallel) and the reader can easily vary the circuit to suit the pocket and availability of the resistors. The resistors used in this project can be obtained from Crusader Electronic Components at 81 Princes Highway, St. Peters NSW, for about 6¢ each. This price is for quantities of 100 up, but since the dual circuit uses 98 resistors there is no difficulty here.

Separate terminal posts are provided for each load so that two separate 50 W sources can be terminated in 8 ohms each or the two halves may be connected in series to give a single 16 ohm 50 W load; and last but not least parallel connection of the two halves will result in a 4 ohm 100 W load. Because metal film resistors are used there are no inductive effects to worry about such as could occur if wirewound units were employed. The stray capacitances present are so low as to be insignificant.

Construction is simple, if somewhat tedious. Lots of soldering is involved! The author used two ordinary household tin cans; one can has a lid (e.g: a coffee tin) the other is a smaller one of the throw-away type (baked beans etc!). The top and bottom of the smaller can were used as soldering planes for terminating the ends of the resistors while the larger can was used to house the project with the lid carrying the terminal posts. Since the coffee tin is virtually leak proof you could fill it with some kind of insulating fluid such as transformer oil and thereby increase the dissipation capability of the dummy loads.

Tin-plated steel is very easy to solder but the sharp edges are dangerous to careless fingers. Blank copper clad printed circuit board could be used instead but does not withstand heat as well as the plain metal sheet.

The arrangement of tin cans may not seem very glamorous but it is highly effective and very cheap — the whole cost of the project comprises about $7 for the resistors and about $2 for the terminal posts. The tin can housing can be spray-painted and the terminal posts labelled and marked to suit individual needs.

Before starting choose a medium sized coffee tin with a resealable lid for the case and select a tin can of smaller diameter which will fit easily into the coffee tin. About eight or nine centimetres in diameter should be fine for the smaller tin can. Using a can opener remove the top and bottom of the smaller can and discard the contents (maybe you should eat the contents — but that's really outside the scope of ETI!). Also, discard the remaining cylindrical portion of the can! Mark up one of the tin-plated discs so obtained with a grid of ten by ten lines as shown in Figure 1 to give 100 intersections.

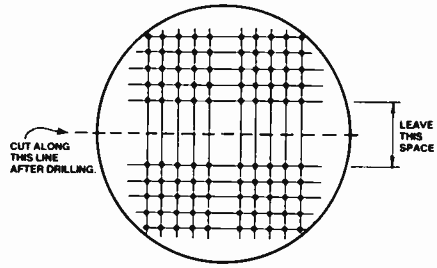


Figure 1. **Drilling and cutting details for the tin-plated discs obtained from a small can.**

Allow a space of about 10 mm along one diameter as shown. This will allow the discs to be cut in half later. Clamp both discs together onto a drill bench or a block of wood, ensuring that they are exactly superimposed. Drill a hole on each intersection of the previously marked lines. Make the holes slightly larger than twice the diameter of the resistor leads; this will assist assembly later on. Take care that your hands are kept clear during drilling since if the drill bit grabs, the two tin discs will whirl around very much like a meat slicer, and almost as sharp! Only 98 holes are needed so don't get carried away.

When the holes are drilled, cut the two discs along the middle space left along one diameter so that you end up with four half discs each with 49 holes. Tin the area around each hole with solder and proceed with assembly.

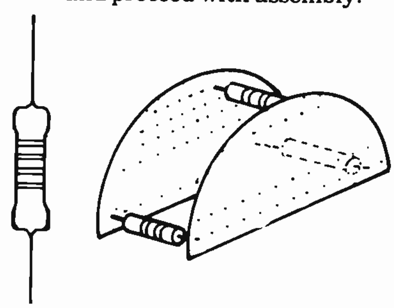


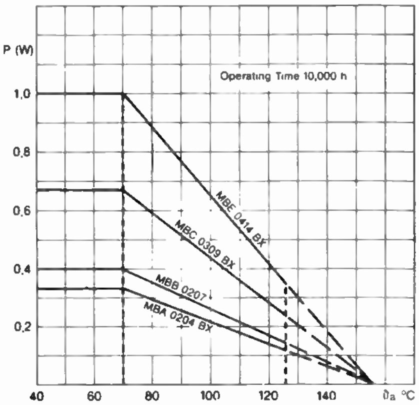
Figure 2. **Cut the resistor leads as shown at left and then solder three resistors to two half-discs as shown to make a rigid assembly.**

Trim all the resistor leads as shown in Figure 2 so that one lead is longer than the other on each component. Take two matching half-discs and using three resistors assemble a rigid structure as shown in Figure 2. Insert the resistors, one row at a time, in between the two tin plates with the leads poking through the holes. If you insert the longer lead of each resistor into its hole first, the other end should be short enough to allow manoeuvring into the hole in the second plate. After one row of resistors is in place solder all the leads of that row on both plates, then proceed with the next row.

Repeat the assembly for the second half of the unit then trim all excess leads flush with the surface of the tin plate. Using a stiff brush (e.g: an old toothbrush) scrub the soldered surface with methylated spirits to remove deposits of flux.

Connect an ohmmeter between the plates of each load — the reading, believe it or not, should be pretty close to 8 ohms. Inspect all solder joints and re-solder if the reading is not correct. Install the four terminals in the lid of the coffee tin using one red and one black terminal for each half of the unit. Lay the two assembled resistor pads side by side as shown in Figure 3. Using fairly stiff copper wire connect the upper plates to one terminal each. Use the same colour terminal for both plates as this will be important later if the loads are to be connected in series or parallel. Using the same sort of wire, but insulated, connect the lower sides of the resistor assemblies to the other two terminals. You should finish up with an assembly which will be supported under the lid of the coffee tin and which is so positioned as to allow it to be inserted into the container and for the lid to be sealed.

To prevent the two halves of the load from shorting together, install an insulating spacer between them using a scrap piece of copper-clad board or matrix board. If using the pcb material, ensure that enough copper is removed to insulate the two halves from each other. If using the matrix board, you will have to drill a couple of additional holes and use small screws to attach the spacer to ▶



The upper curve in this graph shows the typical dissipation characteristics of the Beyschlag resistors used in the ETI-155a dummy load. Power dissipation is derated at operating temperatures above 70°C.

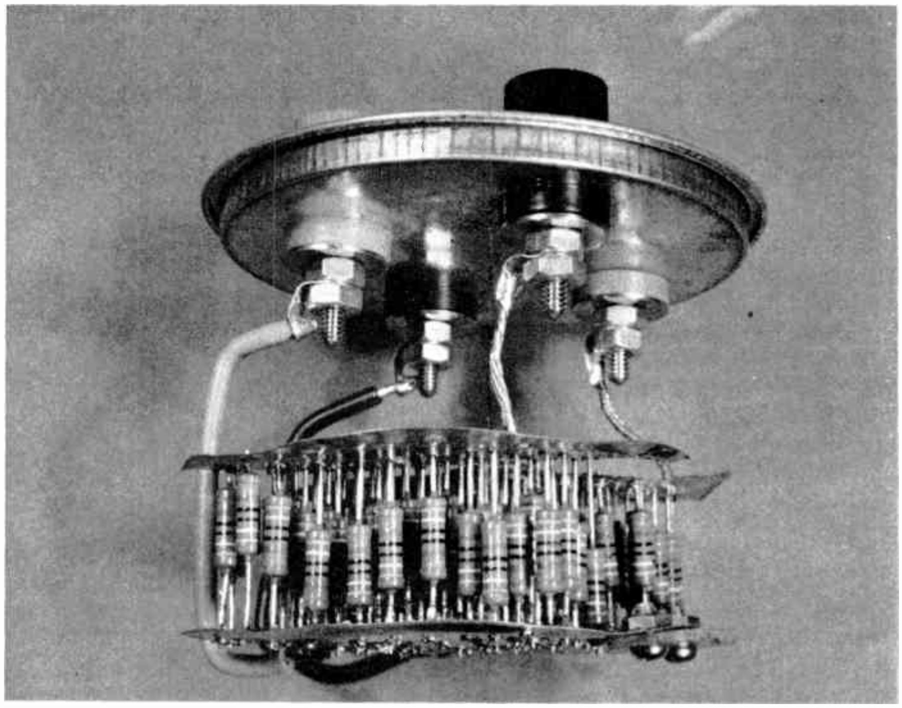


Figure 3. **The two half-disc pairs are assembled side-by-side and heavy gauge wire soldered between the discs and the terminals. An insulating spacer of matrix or pc board holds the two assemblies apart.**

the resistor assemblies.

Before inserting the assembled unit into the coffee container, mark the lid to indicate which terminals are connected by the resistive pads.

To test the unit, connect each load across a known working amplifier or if this is not convenient, use a car battery (not more than 12 V) as the driving source. If using an amplifier, connect an ac voltmeter across the load under test. If you can use a sinewave generator to drive the amplifier, all the better. Adjust the amplifier volume control to give about 10 to 15 volts across the loads. Check by feeling the resistors with your hand that they are in fact warming up. Increase the output of the amplifier until the voltage across the loads is about 20 volts. This should result in the resistors getting quite hot after a couple of minutes.

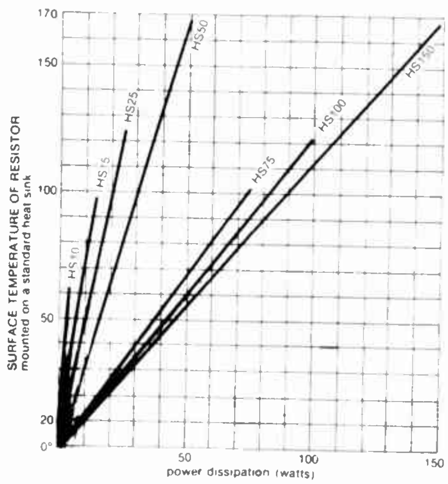
If using a car battery, connect the two loads in parallel and connect the battery across them. Check the current drawn; it should be approximately 3 A with a 12 volt battery.

When testing is satisfactorily completed, install the whole assembly into the coffee container and press down the lid. If you plan to use the loads continuously, fill the container with insulating oil before assembling.

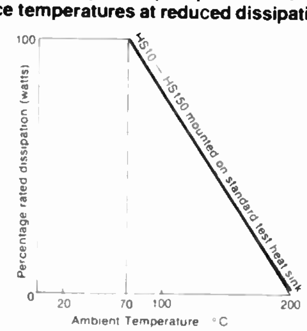
Metal-clad resistor method (ETI-155b)

This has to be just about the world's quickest project! Two 50 W, 16 ohm Arcol resistors connected in parallel were used, as a single 8 ohm, 10 W resistor is more expensive. The Arcol resistors have two diametrically opposed mounting tags drilled to take 6 BA bolts. We mounted them on a short (71 mm) length of heatsink obtained from Autotron Australia, of P.O. Box 202, Glen Waverley 3150 Victoria. We understand a number of component suppliers keep stocks of this product. We chose it because its shape is very convenient for this application but almost any suitable heatsink on which the Arcol resistors can be comfortably mounted will suit.

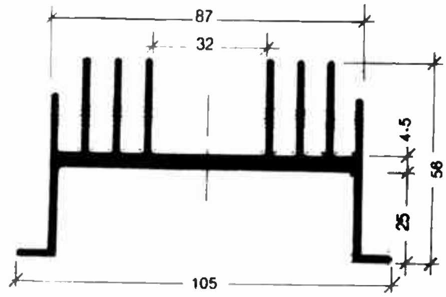
The two resistors are mounted in the 'well' beneath the fins, positioned such that the securing bolt holes do not foul any of the fins. The photograph makes this clear. Two large terminal posts are mounted on one side to provide convenient connections and the resistors are wired in parallel in the manner shown in the photograph.



Temperature-dissipation characteristics of the Arcol metal-clad resistors. The curve marked 'HS50' applies to the types specified for the ETI-155b dummy load. (Temperatures shown are surface temperatures at reduced dissipation).

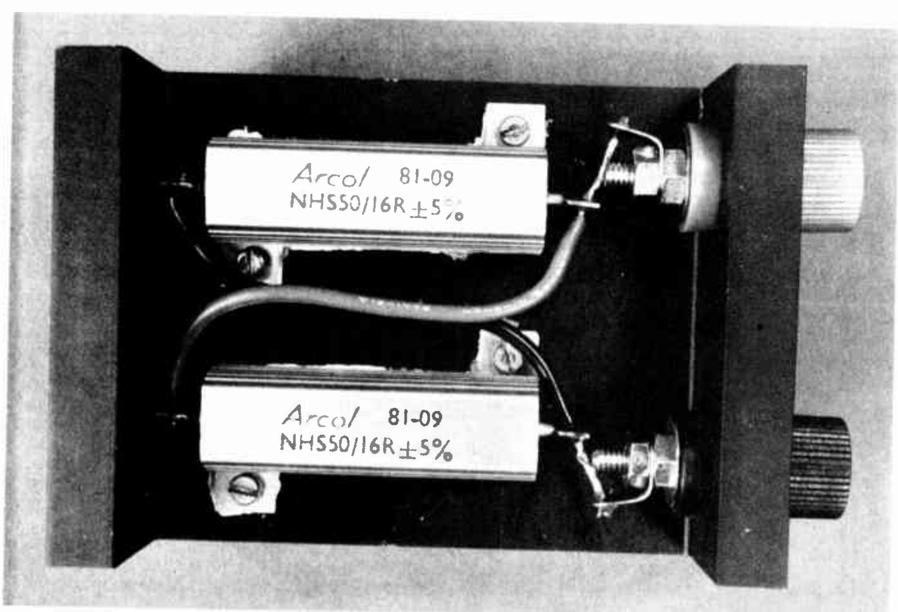


Power derating graph for the Arcol resistors.



Cross-sectional profile of the Autotron type XA heatsink used for the ETI-155b dummy load. This style of heatsink is obtainable in a variety of lengths.

With the heatsink used, at a dissipation of 100 watts, the heatsink temperature rapidly rises and will reach 150°C after some minutes! Fan cooling keeps it within bounds, but if you expect to use the dummy load for lengthy periods then a larger section of the Autotron heatsink or whatever you wish to use is recommended. To keep the resistors at 70°C or below, (their maximum temperature at full dissipation before derating) we suggest either a single 500 mm length (standard size, natural finish) or two 200 mm lengths (standard size, black anodised). ●



The ETI-155b assembled — the world's quickest project!

PARTS LIST — ETI 155

ETI-155a

98 x 390R, 1 W, 1% or 5% carbon resistors.
4 x large binding posts, two black, two red.
Tin cans to suit — see text; high current wire (see text and pics).

ETI-155b

2 x Arcol 16R, 50 W, 5% non-inductive resistors, type NHS50.
2 x large binding posts, one black, one red.
Heatsink (see text); high current wire (see text and pics).

Price estimate

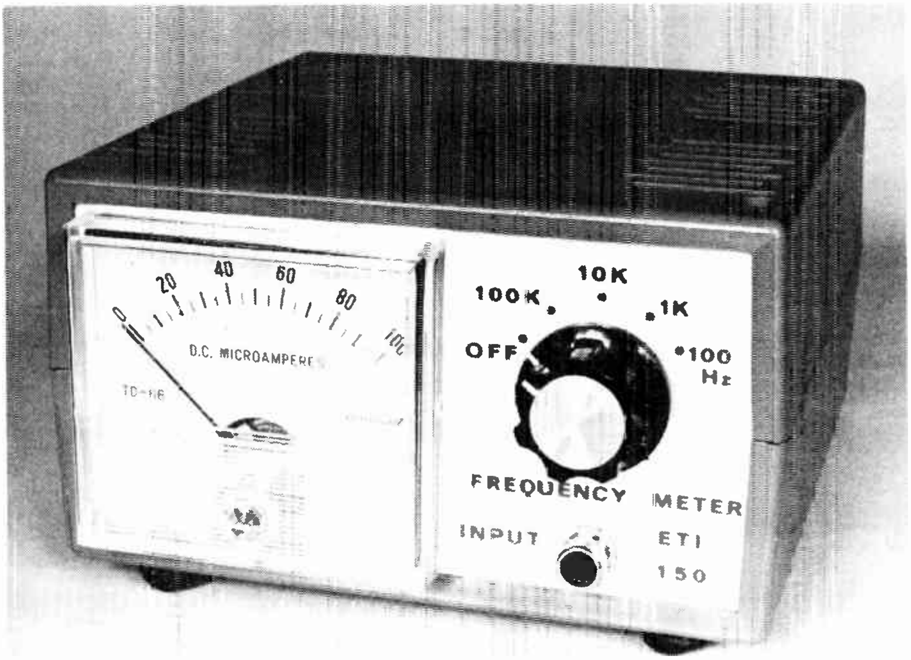
ETI-155a	**$7 - $9**
ETI-155b	**$18 - $22**

Note that these are **estimated** prices only and **not** recommended prices. A variety of factors may affect the price — cost price movements, whether you use 1% or 5% resistors, type of heatsink employed, etc, etc.

Simple analogue frequency meter features linear scale

This simple project is easy to build, inexpensive and should find many uses in the hobby workshop.

Phil Wait



THERE ARE MANY applications in the home workshop where simple audio frequency measurements are required. When experimenting with oscillators, building or repairing function generators etc, it is often handy to have some means of measuring frequency – accuracy to the last Hertz is not always required and thus a full-blown digital counter is not warranted.

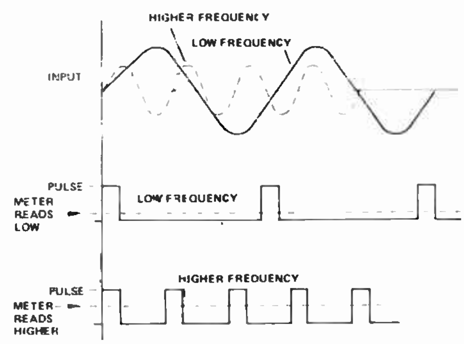
This project will enable you to measure frequency from around 100 Hz right up to 100 kHz with an accuracy of a few percent. It is inexpensive to build but performance is quite adequate to meet a large number of needs in any hobbyist's workshop. Accuracy is unaffected by the waveshape of the signal being measured and the unit will accept signal levels as low as 200 mV. The input is fully protected against high signal levels and against dc voltages up to the rating of the input capacitor, C1. The input is also fully floating above earth – a useful feature.

The frequency meter may be powered from an internal No. 216, 9 V battery or from a Plugpack battery eliminator. A suitable dc socket may be installed on the rear of the cabinet.

All components are readily obtainable, the moulded plastic case in which we housed the prototype is an item supplied by A & R Soanar and is available from many suppliers.

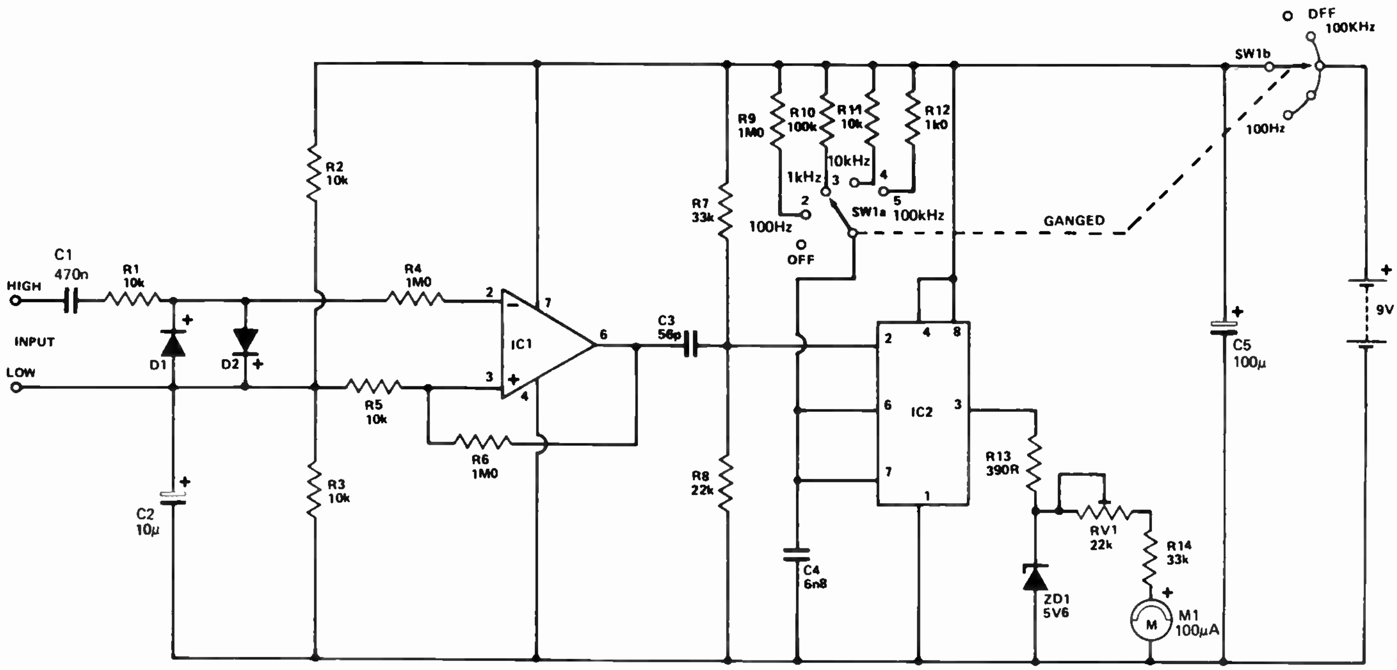
Circuit features

The circuit generates a series of short pulses at the same frequency as the input. These pulses drive a moving-coil meter the current through which will be the average amplitude of the pulse waveform; that is, it will integrate the pulses. This average will be proportional to the ratio of time the pulse is on to the time it is off. The time the pulse is on, that is – the pulse width, is fixed. At low frequencies, the time the pulse is off will be much, much longer than the time the pulse is on. Thus, the average current through the meter will be quite low. At higher frequencies, the time between pulses will be quite short and the average current through the meter will be quite a bit higher (As shown in the diagram). Thus, as the frequency of the pulses is proportional to the input frequency, the pulse on/off ratio, and therefore the meter current, will be proportional to the input frequency. The meter can be calibrated directly in frequency as the relationship is a linear one. We have used a 100 ▶



SPECIFICATIONS ETI 150

Frequency	10 Hz to 100 kHz in four decade ranges
Minimum input	200 mV RMS
Maximum input	250 V peak AC or DC (dependent on voltage rating of C1)
Supply voltage	9 Vdc battery or Plugpack battery eliminator



microamp movement for convenience as it does not have to be re-scaled. The lowest range is 100 Hz full-scale deflection, the highest, 100 kHz.

Only two cheap IC's are used in the whole design a 3140 op-amp and a 555 timer. The 3140 amplifies and squares the input signal and was selected for its high slew rate, wide frequency response and high input impedance. The output of this stage will be a square wave of the same level for all input signal levels and waveforms.

The pulses are generated by a 555 timer connected as a one-shot monostable giving a single pulse output for each input cycle. The monostable has four ranges giving decade scales on the meter. A fifth position on the switch is used as a power switch.

Regulation of the output pulses by a zener diode preserves the accuracy of the unit with falling battery voltage.

HOW IT WORKS – ETI 150

The circuit consists of an op-amp operated as a Schmitt trigger to amplify and square the input signal, followed by a 555 timer wired as a monostable, giving a short output pulse of fixed width for each cycle of input signal. This pulse drives a moving-coil meter, the reading being an average of the pulse amplitude, which is proportional to the pulse frequency. As the pulse frequency is directly related to the input frequency, the meter reading is directly proportional to the input frequency.

The input signal is coupled into IC1 via C1, which provides dc blocking. Protection from overload caused by high amplitude input signals is provided by a diode clipper consisting of D1, D2 and R1. The diodes are connected in an inverse-parallel arrangement so that both positive and negative peaks, above the diode forward conduction voltage, are clipped.

IC1 is a fast op-amp connected as a Schmitt trigger with amplification, as mentioned above. Resistors R5 and R6 provide hysterisis, a 'dead band' in the action of the Schmitt, centred on zero input level. This dead band ensures that the Schmitt ignores noise pulses.

As the unit is required to operate from a single supply, for convenience, R2 and R3 bias the input of IC1 at half the supply voltage.

The output of IC1 is a train of square waves at the same frequency as the input. The output of IC1 is differentiated to provide short trigger pulses for the 555 timer, IC2. The differentiating network consists of C3, R7 and R8. This network is arranged to provide a trigger pulse that is always shorter than the output pulse of the 555. Capacitor C3 is selected to give the shortest possible pulse to the 555 consistent with reliable triggering.

The output of the 555 monostable will be a pulse of fixed width, determined by the range resistors, R9 to R12, and capacitor C4. The ranges are arranged to give a 75% output duty cycle at frequencies of 100 Hz, 1 kHz, 10 kHz and 100 kHz on the input.

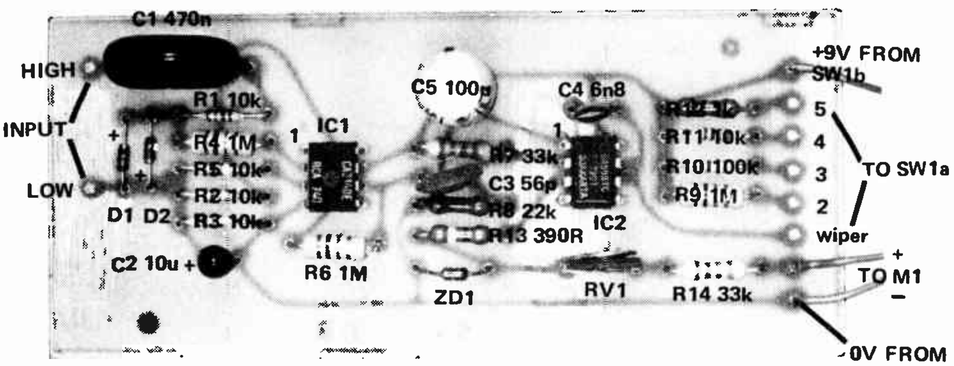
The output pulse from the 555 is clipped at 5.6 V by a zener diode, ZD1, to avoid inaccuracies caused by falling battery voltage (as the battery ages). The meter responds to the average value of the clipped pulses. As the frequency increases, the duty cycle (on/off ratio) of the pulse train increases, increasing the average voltage and thus the meter current in direct proportion. Thus the reading on the meter will be linearly related to frequency.

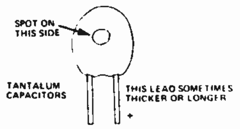
Construction

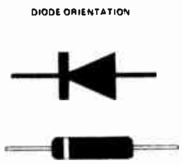
Even though this project is relatively simple, we strongly recommend you use the pc board – saves possible hassles!

As mentioned previously, we constructed our prototype in a commonly available plastic box. This has the advantage that the unit can be operated fully floating from earth – handy in some situations. Check placement of components on the front panel and the positioning of the pc board inside before commencing major assembly. It's probably best to assemble the components on the pc board first. Take care with the orientation of the ICs, diodes and tantalum capacitor.

The input capacitor, C1, can be obtained in several voltage ratings. Greencaps are available in ratings of 100 V, 250 V and 630 V. If all your work is with solid-state circuitry, a 100 V type will be more than adequate. If you anticipate using your unit with say, valve equipment, the highest rating type for C1 is recommended. The rating applies to the combined







The pc board pattern is on page 74.

dc voltage that may be present on the input, *plus* the possible peak value of the input signal.

A 630 V rated capacitor will be physically larger than a 100 V type and the leads may have to be shaped to fit the capacitor on the board.

Once the board is assembled, the major components can be assembled onto the front panel of the case. We made up a Scotchcal overlay for the front panel, to dress it up and give it a bit of a 'professional' look. Kit suppliers will probably have these available shortly after this book goes on sale. Radio Despatch Service in Broadway, Sydney offer a special Scotchcal front panel service for projects so, if you are using a similar case you may have on hand, then they will be able to supply a front panel.

The meter (we used a University TD66 – but many other types are suitable), was mounted in a circular cutout on the left hand side of the panel. The range switch should be mounted next, followed by the input socket. After much discussion around the office ("A jack socket!", "No, screw terminals", "Rubbish! RCA socket" . . .), we settled on an RCA socket. It's a common item on audio equipment, inexpensive and coax cables terminated in RCA plugs, for input leads, are cheap and readily available.

However, any type of socket to suit your individual requirement will do equally well. If you use a metal box, the input connector earth must be the *only* connection from the circuitry to the case, as the negative rail from the battery is not at earth potential.

The pc board may be mounted anywhere convenient in the case and wires run to the front panel for the input and switch connections. Make sure the board does not get in the way of the meter when the front panel is in place.

The unit may be powered from an internal battery, which makes it a handy portable unit. If you wish to operate the unit from a plugpack battery eliminator, then we recommend you purchase a unit giving a nominal 6 Vdc output. The current requirement for the project is quite modest and the output of these small battery eliminators is dependent on the load. A 6 V unit will typically deliver 9 V or so under a light load.

If you do decide to use one of these units, a socket matching the unit's plug will have to be mounted on the rear panel and leads run to the supply rail pads on the pc board. If you wish to have the option of both battery and mains operation, then a small SPDT toggle switch should be mounted on the rear panel also and wired into the circuit.

Calibrating it

Calibration of the frequency meter is very easy, aided by the fact that it has a very high input impedance.

With the unit switched to the 100 Hz range, touch your finger to the input. There will usually be enough 50 Hz field from the electrical wiring in a building to drive the input. This will cause a deflection on the meter and RV1 should then be adjusted to give a meter reading of 50 (half scale). Move the unit near house wiring to increase the amount of signal to the input if a reading cannot be obtained.

If a signal generator of known accuracy is available the instrument can be calibrated on any range. Only one range need be calibrated as the others will automatically fall into line.

If it is impossible to obtain any reading on the meter, the coupling capacitor (C3) may have to be increased in value to say 100p or 150p. This component has been selected to give a very short trigger pulse into the 555 and has been found to work correctly, using the value shown in the circuit, with several different ICs.

Using your meter

Selecting the 100 kHz range will connect power to the unit and the unknown signal can then be applied to the input. Note the reading and switch to a lower range if required. This procedure avoids the possibility of spurious readings that may be obtained on lower ranges due to re-triggering of the 555 by high frequency signals. There are no other adjustments, so all you need is something to measure.

This is the sort of instrument that, once you have it, seems to find a great many uses for itself! ●

PARTS LIST - ETI 150

Resistors all ½W, 5%

R1–R3	10k
R4	1M
R5	10k
R6	1M
R7	33k
R8	22k
R9	1M
R10	100k
R11	10k
R12	1k
R13	390R
R14	33k

Capacitors

C1	470n greencap
C2	10µ tantalum
C3	56p ceramic
C4	6n8 greencap
C5	100µ 25V electrolytic

Semiconductors

D1, D2	1N914 or similar
ZD1	5V6,400mW Zener diode
IC1	3140 op amp
IC2	555 timer

Miscellaneous

M1	100µA meter, University TD - 66 or similar
RV1	22k min vert mounting trim pot
SW1	two pole five pos wafer switch

Plastic box to suit (approx. 75 mm x 135 mm x 130 mm); input connector chassis mounting RCA socket or similar; knob, ETI 150 pc board.

General purpose 150 W MOSFET power amp module

Here's a high power, general purpose power amplifier module for guitar and PA applications employing rugged, reliable MOSFETs in the output.

David Tilbrook

Geoff Nicholls

WE PUBLISHED the ETI-477 MOSFET amp in Electronics Today (Jan, Feb and Mar '81) and have received a continued demand for a general-purpose MOSFET power amp module suitable for guitar or PA applications. The ETI-477 can of course be used but is unnecessarily complex for this purpose. To fill this demand we have developed the ETI-499 module. It produces similar power levels to the ETI-477 (i.e: around 100 W RMS into an 8 ohm load or around 150 W into a 4 ohm load when used with the Ferguson PF4361/1 transformer), but has been designed with constructional simplicity as the foremost consideration. The power supply circuitry is included on the pc board so that the only connections to the board are from the power transformer, input sockets and output terminals. This greatly simplifies construction and installation and ensures that all power supply wiring is done with the necessary high current handling capability. In all transistor amplifiers, but especially with MOSFETs, the resistance between the main filter capacitors and the output devices must be kept as low as possible if low distortion and stability are to be ensured.

The circuit used in the ETI-499 is a development from one published in the Hitachi application notes for these MOSFETs. The original circuit used very high-gain bipolar driver transistors developed especially by Hitachi for use as MOSFET drivers. Unfortunately these devices are at present unavailable in Australia. Since these are an extremely fast device, replacement by more common bipolars limits the open loop bandwidth and causes the amplifier to be unstable. The main departures from the Hitachi circuit are therefore to ensure a stable design with common transistors.

We used the BF469 and BF470 as drivers. These are a complementary video output pair supplying good slew rate and V_{ceo} figures at a reasonable price. The resulting power amp module is fast and stable, with distortion figures completely adequate even for many high fidelity applications. The module is easy to construct and capable of withstanding continued clipping or full-power operation for extended periods when provided with a suitable heatsink.

SPECIFICATIONS — ETI-499

Power output
150 W RMS into 4 ohms
100 W RMS into 8 ohms
(at onset of clipping)

Frequency response
20 Hz to 20 kHz, +0 −0.5 dB
10 Hz to 60 kHz, +0 −3 dB
(measured at 1 W and 100 W levels)

Input sensitivity
1 V RMS for full output

Hum
−98 dB below full output

Noise
−114 dB below full output

Total Harmonic Distortion
0.006% at 1 kHz
0.03% at 10 kHz
(measured at 12 W level)

Stability
Unconditional — tested to full output driving 3.5 uF short circuit at 10 kHz.

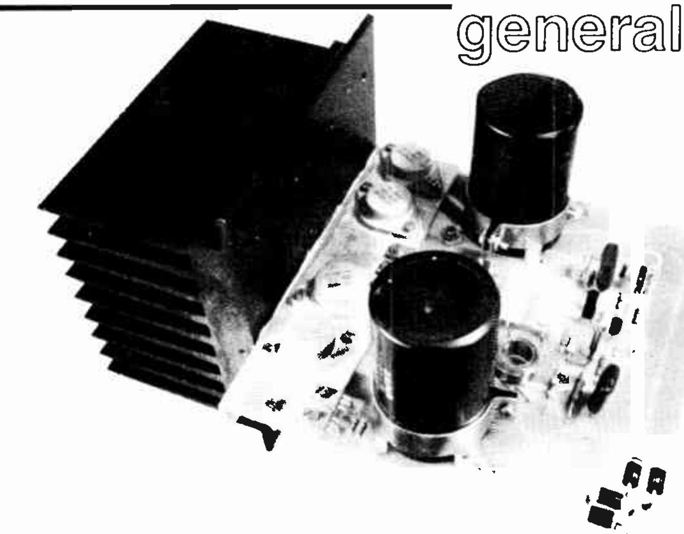
Why MOSFETs?

The power MOSFET is a relatively recent development and offers several distinct advantages over the more common bipolar transistor. To understand these differences it is helpful to look at some of the characteristics of bipolar output transistors.

Most power amplifiers employ bipolar transistors in a common-collector or emitter-follower configuration. The relationship between the output signal voltage and the input signal voltage is a function of the load impedance and the forward transfer admittance of the particular device. Forward transfer admittance is commonly given the symbol y_{fs} and its non-linear characteristic gives rise to distortion in the output stage. With bipolar transistors, the greatest non-linearity occurs for low input voltages, typically between 0 V and 0.6 V. Once outside this voltage range the forward transfer admittance is high and quite linear. So most of the distortion generated in a bipolar output stage occurs at low signal voltages and is called *crossover distortion* (for a more detailed explanation of crossover distortion refer to the article on the ETI-477 power amp module published in January 1981).

The most common method used to overcome this problem is to make use of bias current. A fixed voltage of around 0.6 V is applied to the bases of the output transistors so that the applied signal voltage does not have to operate the transistor over the most non-linear region. However, a problem arises with this technique because this voltage must be controlled extremely accurately. Even 0.5 V in excess of the correct voltage will saturate the output devices, probably destroying them. Furthermore, as the output devices heat up due to normal operation, the bias voltage must be decreased to maintain the same operating conditions. This is very difficult to do accurately enough, so the power amp is often running either with insufficient bias current or is dangerously close to destruction.

The problem occurs because the bipolar transistor has a positive temperature coefficient. This means that as the



temperature of the device is increased the collector-emitter current will increase if the base-emitter voltage is held constant. The increased current causes further heating and a further increase in current. This condition is called *thermal runaway* and results in the destruction of the output device.

Another problem with conventional bipolar output transistors is speed. The techniques used in the construction of these devices to ensure broad SOAR characteristics (SOAR stands for Safe Operating ARea) usually conflict with those to ensure high speed. Since the output transistors must handle the largest currents they are usually the slowest devices in the amplifier and determine the maximum signal slope that can be handled by the amplifier before distortion results. Distortion generated by this mechanism is called *slew-induced distortion* and *transient intermodulation distortion.* Once unnecessarily high signal slopes have been removed by a suitable filter at the input of the power amp the only solution is to increase the slew rate of the output devices.

One of the major advantages of power MOSFETs is their extremely high speed. When driven correctly the MOSFETs used in this project can switch a current of around 2 A in 30 nanoseconds! This is roughly *100 times* the speed of commonly available bipolars. Another advantage of MOSFETs is their very high input impedance. Unlike the bipolar transistor, they are a voltage-controlled device and require only enough drive current to overcome their input capacitance. Probably their most important advantage over bipolar transistors, however, is that *they have a negative temperature coefficient.* Heating causes an increase in the resistance of the device, so MOSFETs are inherently self-protecting. If one part of the device attempts to conduct more current it heats up more than the surrounding region, increasing its resistance, which distributes current over the rest of the device. Similarly if several devices are used in parallel, the negative temperature coefficient will ensure that all devices share current equally. In guitar and PA applications the negative temperature coefficient of MOSFETs provides the amplifier with unprecedented reliability, and the high speed helps to eliminate the problem of slew-induced distortion.

On the other hand a disadvantage with MOSFETs arises from their relatively low forward transconductance when compared to a good bipolar transistor. Although the transconductance of bipolars is highly non-linear when the base emitter voltage is below 0.6 V, it increases dramatically once outside this region. The MOSFET, although not as non-linear for small voltages, never achieves the forward transconductance of the bipolar transistor. The distortion generated by the power MOSFETs is therefore higher than that of bipolar transistors and must be reduced to acceptable limits through the use of negative feedback. This is not a real problem, however, since the high input impedance eliminates at least one stage of a conventional bipolar amplifier design. This allows a simpler circuit with fewer active devices and consequently improved stability margins, allowing greater levels of overall negative feedback before oscillation results.

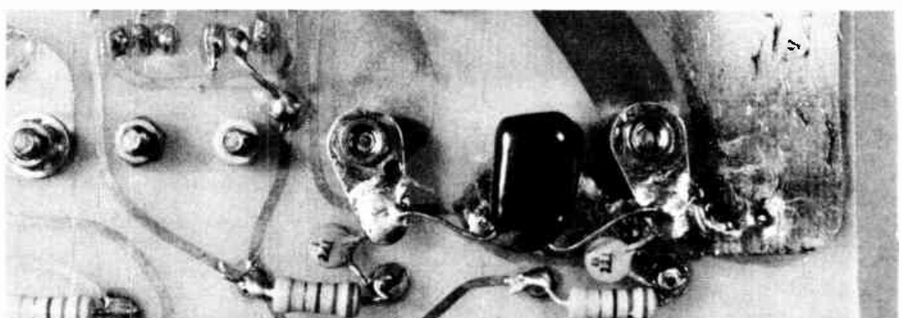
Construction

Construction of the ETI-499 is relatively simple, since all the components mount on the pc board, including the output transistors and power supply components. The design of a good pc board pattern is often as difficult as the design of the original circuit! This is especially true for power amplifiers or any circuit in which both large and small currents are involved. The problem of large currents occurs because of voltage drops across earth return paths, destroying the integrity of earth reference points for small signal currents. To overcome this problem, the pc board must be designed to ensure the ▶

HEATSINKING

The heatsink will need to dissipate around 100 W when the module is run at full output for lengthy periods. A heatsink with a thermal capacity of around 0.65°C/watt is recommended if free-air cooling is contemplated. A 152 mm length of Philips 65D6CB will do nicely (cost — around $30). Alternatively, the module may be mounted on one of the ETI-designed Series 5000 heatsink panels. In fact, two modules may be mounted on a Series 5000 heatsink panel. The panels are available from some kit and component suppliers.

If fan-forced cooling is contemplated, then a heatsink rated at 1.2 to 1.5°C/watt should be used. A 225 mm length of commonly available extruded 'fan' type heatsink will do the job. This type of heatsink is flat on one side, the other side having two sets of fins fanning out from a central channel. A suitable length will set you back about $10. A fan will set you back around $20 to $30, unless you have one lying around.



Compensation capacitors are required for the two 2SK134 output MOSFETs (Q8 and Q9) to equalise the input capacitances between the n-channel and p-channel output devices. They are mounted under the board as shown here. Solder lugs are placed on top of the mounting nuts and held with another nut each. C6 and C7 mount from these to the pads shown, while C7 mounts between them. Note the resistors mounted under the board also.

validity of the earthing arrangement. If at all possible, the pc board published should be used, as departures from this design could seriously affect amplifier performance.

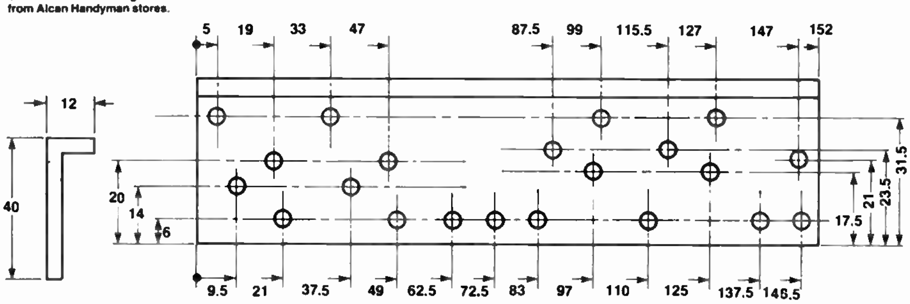
Commence construction by soldering all the resistors onto the circuit board with the exception of the four 0R22 output resistors. These effectively connect all the sources of the MOSFETs together and make it difficult to locate faults in the mounting of the MOSFETs. Solder the 1 W resistors slightly above the circuit board since these can become hot under certain conditions. The components marked with an asterisk on the circuit diagram are mounted on the rear of the pc board. They should be mounted close to the MOSFETs. Do not solder the resistors to the rear of the circuit board at this stage. These are best left until after the MOSFETs have been mounted.

Solder the capacitors onto the circuit board with the exception of those on the rear of the board and the two large electrolytics. The 100u capacitor C3 is the only other electrolytic, so be careful with the orientation of this component. The capacitor is marked to indicate which of its leads are to be connected to a positive or negative voltage. Check the correct orientation on the overlay diagram. This also applies to the diodes and zener diodes used in the circuit, which can be mounted next.

Both the driver and power transistors are mounted on a length of aluminium angle extrusion, which is bolted to the pc board by bolts through the transistor mounting holes. This is shown in the accompanying diagrams. The extrusion is used to conduct the heat generated by the output and driver transistors to the heatsink, which will also be bolted to the extrusion. If you purchase the mod-

ALL 4 mm DIA.
MATERIAL 40 x 12 x 3 ALUMINIUM ANGLE EXTRUSION
Drilling details for the heatsink bracket assembly. All dimensions are in millimetres.
Suitable aluminium angle stock is available from Alcan Handyman stores.

BRACKET DRILLING DETAILS



PARTS LIST — ETI-499

Resistors .... all ½ W, 5% unless stated
R1,R2 .... 100k
R3,R11 .... 1k
R4,R5,R18-R21 .... 220R
R6,R7 .... 3k9
R8 .... 22k
R9 .... 680R
R10 .... 10k
R12,R15,R16,R17 .... 100R
R13 .... 33k
R14 .... 10k 1 W
R22-R25 .... 0R22 W
R26 .... 4R7 1 W
R27 .... 1R 1 W
RV1 .... 100R preset
RV2 .... 250R preset

Capacitors
C1,C9 .... 220n greencap
C2 .... 2n2 greencap
C3 .... 100u/25 V electrolytic
C4 .... 33p ceramic
C5 .... 6n8 greencap
C6,C8 .... 330p ceramic
C7 .... 47n greencap
C10,C11 .... 100n greencap
C12,C13 .... 8000u/75 V electrolytic

Semiconductors
Q1,Q2,Q3 .... BC546
Q4,Q5 .... BF470
Q6,Q7 .... BF469
Q8,Q9 .... 2SK134 Hitachi MOSFET
Q10,Q11 .... 2SJ49 Hitachi MOSFET
D1-D4 .... 1N914
D5-D8 .... 1N5404
ZD1,ZD2 .... 12 V 400 mW zener

Miscellaneous
ETI-499 pc board; plastic bobbin (from P26/16 potcore or similar); 5 A fuse (speaker fuse, not mounted on pc board); fuse holder; 1 m of 0.8 mm enamel-covered copper wire; 155 mm length of aluminium extrusion, 40 mm x 12 mm, for use as the heatsink bracket; assorted nuts and bolts, hookup wire, etc; two solder lugs.

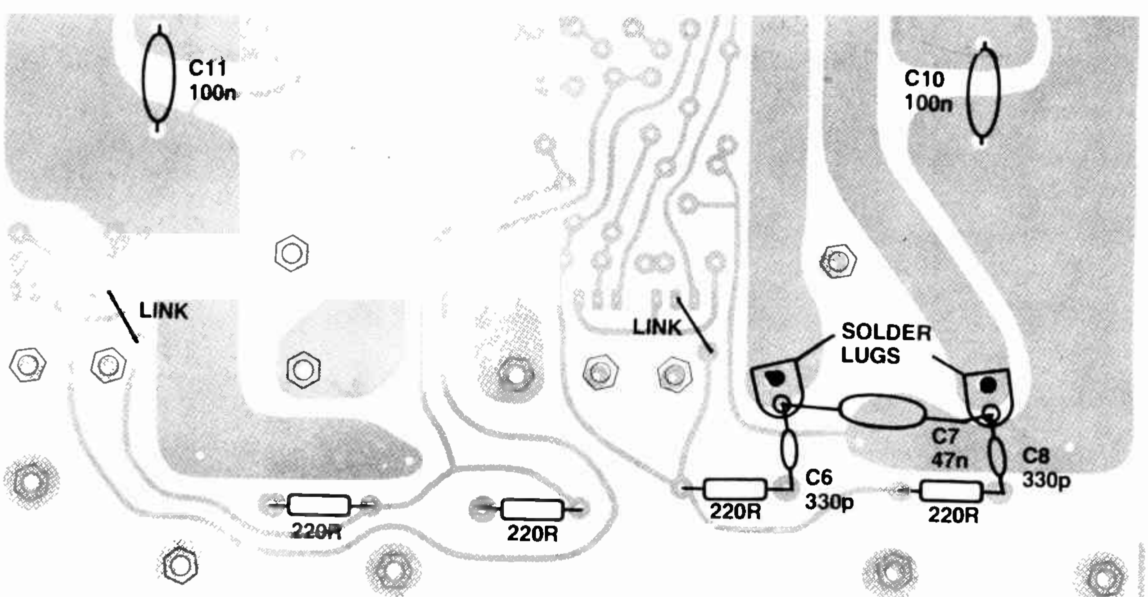
Price estimate

We estimate the cost of purchasing all the components for this project will be in the range:

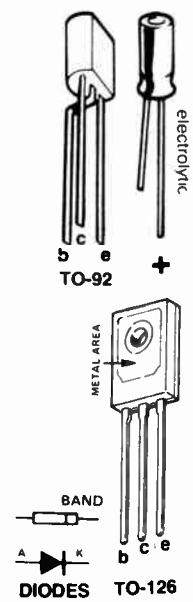
$75-$85

(heatsink & transformer extra)

Note that this is an **estimate** only and **not** a recommended price. A variety of factors may affect the price of a project, such as — quality of components purchased, type of pc board (fibreglass or phenolic base), type of front panel supplied (if used), etc — whether bought as separate components or made up as a kit.



TRACK SIDE OVERLAY DIAGRAM

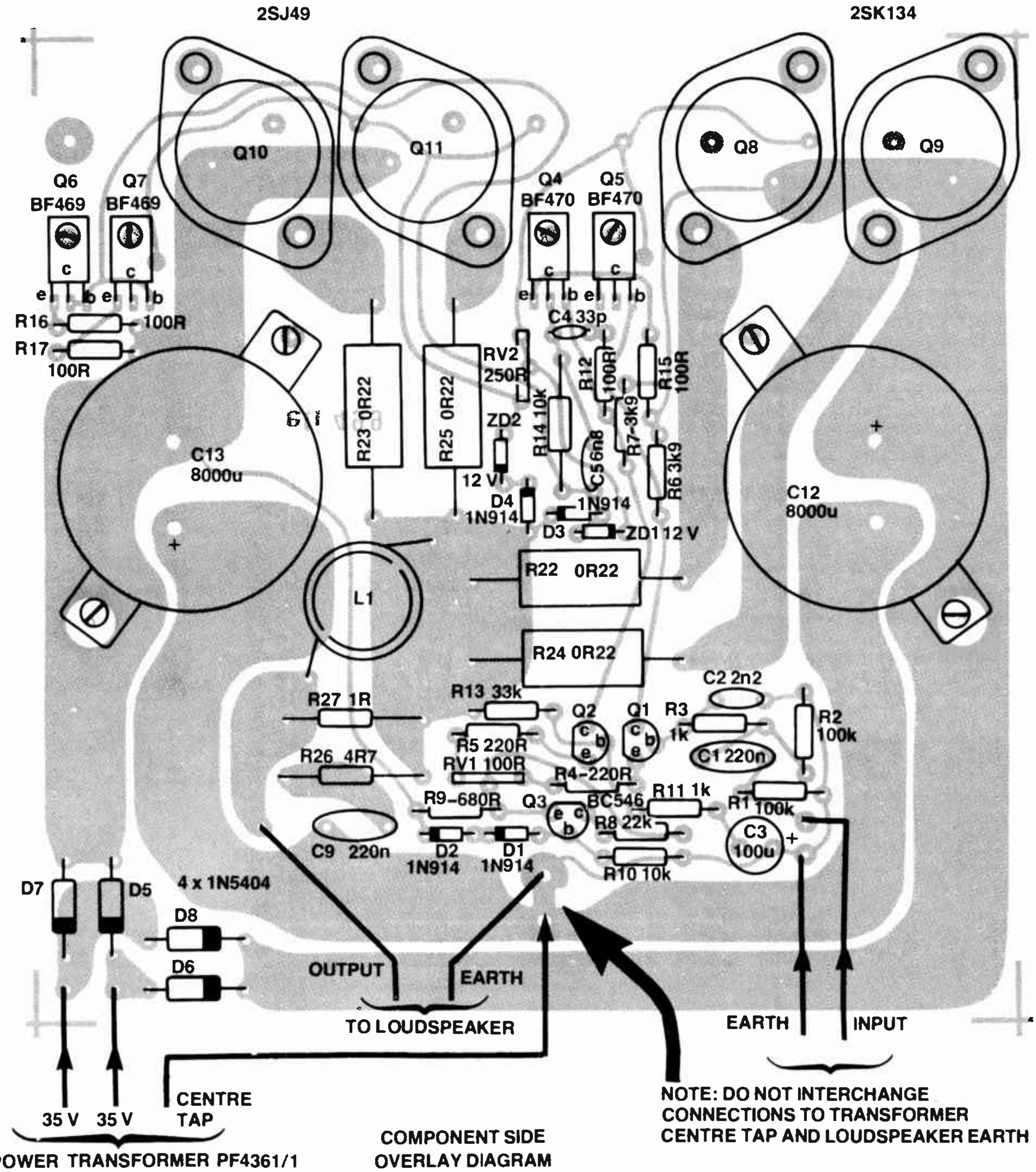


ule in kit form from one of the kit suppliers who support our projects, this bracket should be supplied drilled, ready to mount the transistors. If not, drill all the necessary holes before proceeding further. Make certain the holes are free of burrs or shavings that might otherwise cut through the transistor insulating washers. This is best done with a couple of twists of an oversize drill (i.e: around 13 mm diameter).

The bolts holding the MOSFETs in place also serve to make electrical connections to the cases of the devices. These bolts must be insulated from the heatsink bracket, which will be at earth potential. This is done with the use of short insulating sleeves cut from a length of 'spaghetti' insulation. Use a small quantity of heatsink compound on both sides of the transistor insulating washers to ensure good thermal contact. Insert the sleeves in the holes of the heatsink bracket and mount the four MOSFETs as shown in the accompanying diagram.

The four driver transistors can now be mounted. Again, use transistor insulating washers between the metal sides of the transistors and the heatsink bracket, although insulating sleeves are not necessary.

Once all the transistors have been mounted on the heatsink bracket use a multimeter to check for any short circuits to the heatsink bracket by measuring the resistance from the case of each MOSFET, and from the centre lead of each driver transistor, to the bracket. The measurements should show open circuit on all transistors. If a short does exist the transistor should be ▶

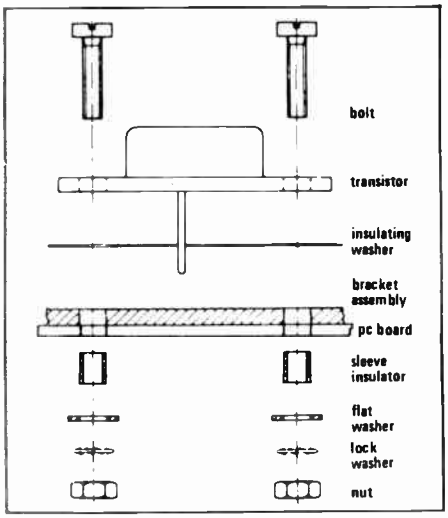


COMPONENT SIDE OVERLAY DIAGRAM

removed and remounted, possibly with a new insulating washer. Finally, solder the leads to the transistors.

Once the MOSFETs and drivers have been mounted, the remainder of the components can be mounted on the pc board, including the small signal transistors and the components on the rear of the pc board. Mount the two 8000u electrolytic capacitors last; be sure to bolt the capacitors down, however, before soldering the lugs. Mount the four 0R22 resistors now, leaving around 5 mm between the resistor and the board. Ensure that all components mounted on the rear of the pc board are mounted close to the board with their leads cut as short as possible.

The output inductor, L1, is formed by winding 20 turns of 0.8 mm enamel wire around a 14 mm former. The plastic bobbins supplied for use with the P26/16 potcores are ideal for this purpose.



Exploded view of how to mount the output devices to the bracket and pc board.

Powering up

Supply fuses have not been included on the pc board because the resulting resistance necessitates the use of a second set of electrolytic capacitors close to the output devices. To protect the loudspeakers in the case of failure of the power amp a fuse should be used in series with the loudspeaker cable. We will shortly be publishing a loudspeaker protector, with the emphasis on PA applications, for use with the ETI-499. In the meantime, however, use a fuse as specified in the parts list.

Before powering up check all stages of construction, including the orientation of all polarised components. Check that no shorts exist between the cases of the output devices and the heatsink bracket. Mount the heatsink bracket to a suitable heatsink, again using heatsink compound to ensure good thermal contact. Do not connect a loudspeaker at this time. Adjust RV1 to centre and RV2 fully counterclockwise, as viewed from the positive rail side of the pc board. If all is in order, connect the module to the power transformer and switch on. Using a multimeter on the 1 V range, adjust RV2 so that the voltage between the ends of RV2 reads 0.8 V. Now adjust RV1 so that the voltage between the output terminal and ground is as close to zero as possible. Ideally, a digital multimeter should be used for this measurement since most analogue meters do not have the necessary resolution. Adjust RV1 to achieve a dc voltage on the output of less than 10 mV, if possible. If your multimeter does not allow measurement of voltages this small, leave RV1 set at the centre position. When both of these adjustments have been made, the module is ready for operation.

Performance

We have tested the prototype into both inductive and capacitive loads and at all times *it performed impeccably.* The sound is clean and smooth with no sign of the harshness sometimes experienced with transistor power amps. The high speed of MOSFETs helps to ensure freedom from slew-induced distortions and the amp clips cleanly with no sign of instability.

Later in this book you will find articles on a loudspeaker protector board and a preamplifier which form a complete PA or guitar amplifier.

SOME CAPACITORS AREN'T . . .

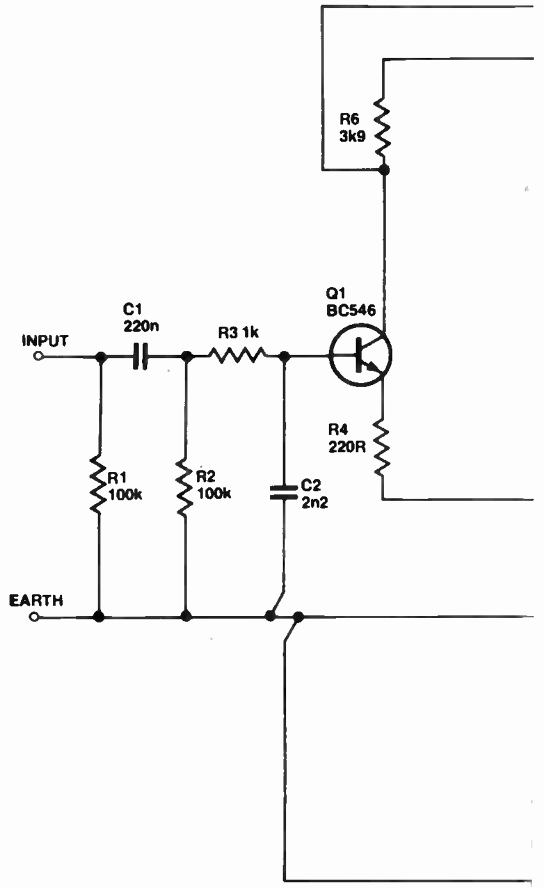
For the R26-C29 network to provide an effective high frequency load to the output stage it is imperative that C9 (220n greencap) have low self inductance. From experience, we have found Elna type greencaps and Philips polycarbonates meet this requirement. High frequency instability, if not outright oscillation, may result if this requirement is not met.

To a lesser extent, the same applies to C7, C10 and C11. Note that C7 ac-couples the sources of Q8 and Q9 together, so that the self inductance of the source ballast resistors R22 and R24 is no longer important, preventing high frequency instability in this section of the output stage brought about by the inductance of the wirewound ballast resistors.

HOW IT WORKS — ETI-499

The circuit is a development from one published in Hitachi's application notes for these MOSFETs. The original circuit uses driver transistors designed by Hitachi for use as MOSFET drivers. Unfortunately these devices are not available in Australia at the present time, so most of the differences are to ensure stability and low distortion with a more readily available driver. We have used the BF469, BF470 complementary video output pair as used in the 477 module. These transistors provide the necessary speed so as not to degrade the performance of the output transistors.

One of the most difficult stages in the development of an amplifier module of this type is the pc board design. Separation of the large currents flowing to the electrolytic capacitors from signal earth is absolutely imperative if low distortion is to be obtained. An earlier pc board using exactly the same circuit gave distortion figures as high as 1% when driven into 8 ohms at around 10 W RMS! The problem was simply interaction between charging currents to the electrolytic capacitors and the earth reference to the input differential pair. For best performance use the pc board design published with this article and pay special attention to all earth and supply connections. In particular ensure that the connections to the



centre point of the transformer and the loudspeaker earth are soldered into the correct positions on the pc board. Although these two points are immediately adjacent on the pc board they are *not* equivalent electrically due to the slight resistance of the board. If these wires are connected the wrong way around the distortion will be increased possibly by as much as 20-30 dB!

Transistors Q1 and Q2 form an input differential pair. Their function is to compare the output signal with the input signal and drive the voltage amplifier transistors in the driver stage with the necessary correction signal, sometimes called the error voltage or error signal. The base of Q1 is held at ground potential by resistor R2. Capacitor C1 in conjunction with R2, R3 and C2 forms an input filter, which defines the upper and lower 3 dB points of the amplifier. This filter therefore restricts the maximum possible signal slope capable of being driven to the input of the differential pair. This is an essential function since it eliminates slew-induced distortions such as TIM, provided that the rest of the power amp has a slew rate in excess of this limit. For a more detailed description of slew-induced distortions and their remedies see the articles on the 477 MOSFET module published in Electronics Today, January, February and March 1981.

The gain of the differential pair is around 17, so most of the open loop gain is done by the driver transistors Q4 and Q5, and their associated current mirror formed by Q6 and Q7. The series RC network C4, R12 ensures stability of the amplifier by decreasing the gain of the driver stage at very high frequencies, while keeping the phase shift produced within 90°.

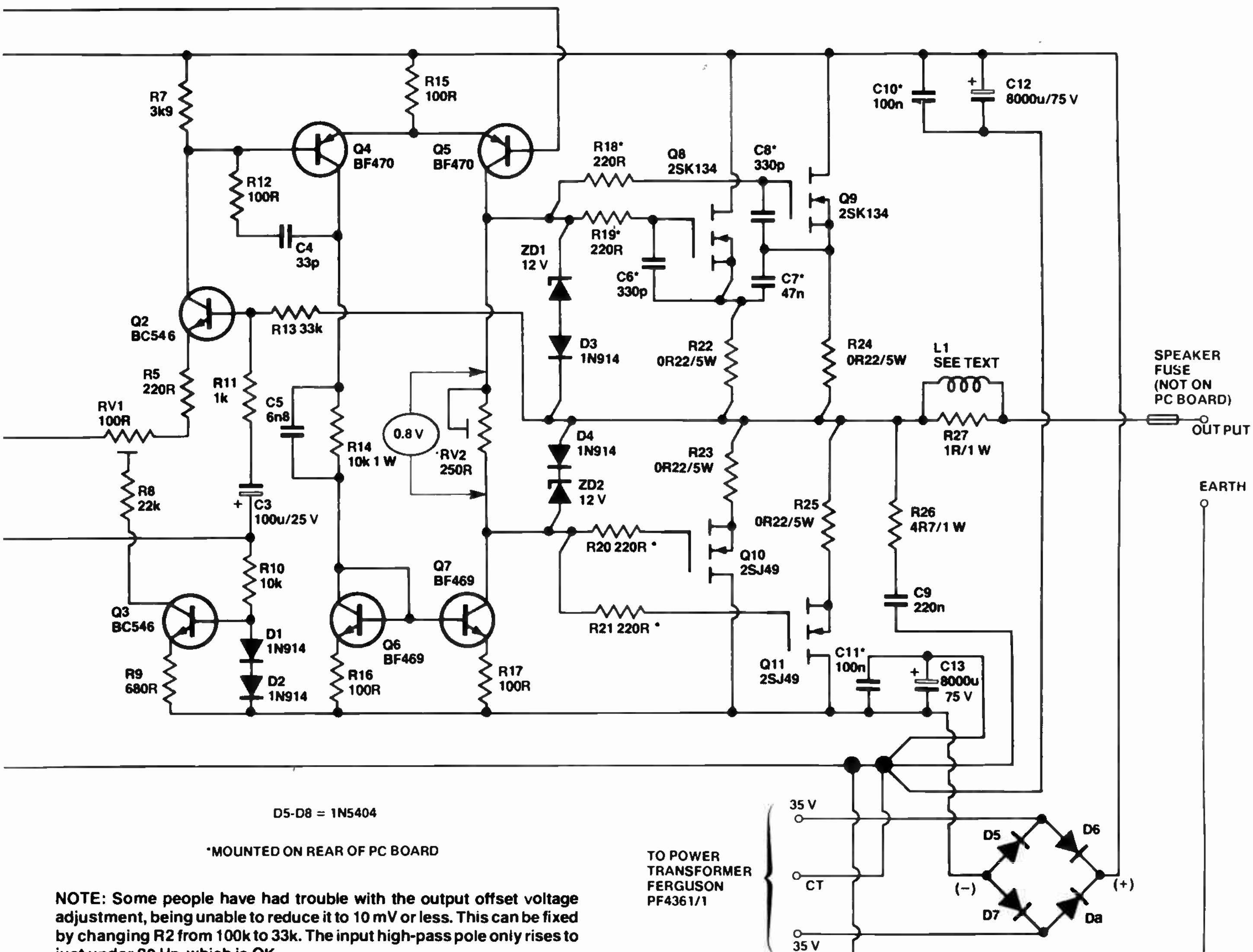
As stated above, transistors Q6 and Q7 form a current mirror. The purpose of these devices is to ensure the current through the two driver transistors remains identical. At the same time the very high impedance represented by Q7 on the collector of Q5 ensures high open loop gain, and consequently low distortion through the relatively large amount of negative feedback available. RV2 varies the voltage between the gates of the output MOSFETs and therefore the amount of bias current through the output transistors. If the voltage across this preset is set to around 0.8 V the bias current will be approximately 80 mA, which is about right. If the bias current is decreased completely by turning RV2 fully away from the MOSFET end of the board, the MOSFFTs will remain off until a signal is fed to the input. This is pure class B operation and results in the coolest operation of the power amplifier. The disadvantage, however, is that a slight increase in distortion, called crossover distortion, will result. In PA or guitar applications this is not a problem, so the amplifier can be used in this mode without hesitation.

The diodes D3, D4 and the zener diodes ZD1 and ZD2 ensure that the voltage between the gates of the FETs and their sources never exceeds 12.6 V, the most common cause of MOSFET failure.

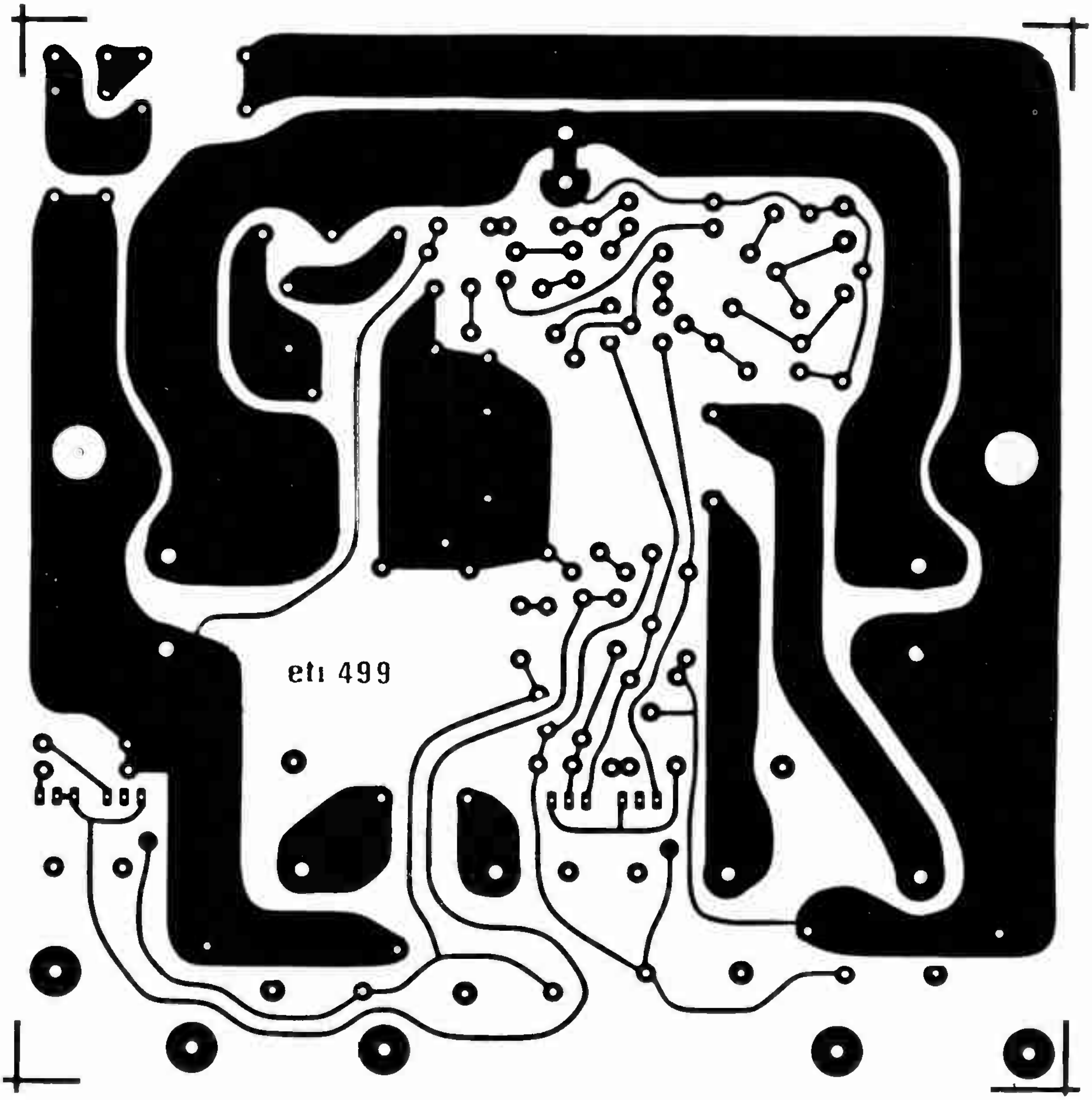
Capacitors C6 and C8 equalise the capacitive input characteristics of the MOSFETs and make it considerably easier to correctly stabilise the output stage. Capacitor C7 brings the sources of the two 2SK134 MOSFETs to the same potential at high frequencies, and overcomes possible problems that might otherwise be caused by inductance in the source resistors R22 and R24.

The four resistors R22-R25 help to match the differences between the characteristics of the different output devices.

The passive filter network formed by R26, C9 ensures that the module always has a load at high frequencies. If the amplifier is tested with large high frequency sinewaves this resistor will become extremely hot, but this does not indicate a fault condition. The inductor L1 and the resistor R27 help to ensure total stability into capacitive loads, such as when driving extremely long loudspeaker leads.



NOTE: Some people have had trouble with the output offset voltage adjustment, being unable to reduce it to 10 mV or less. This can be fixed by changing R2 from 100k to 33k. The input high-pass pole only rises to just under 20 Hz, which is OK.



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

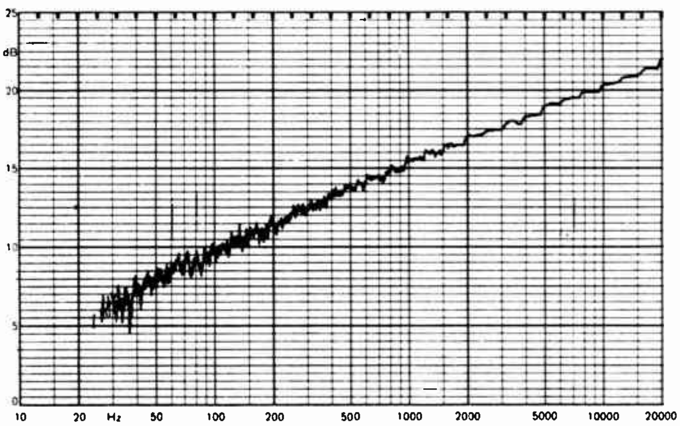
3 Ag fuse ratings in amperes. (Do not use slow-blow-fuses).

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

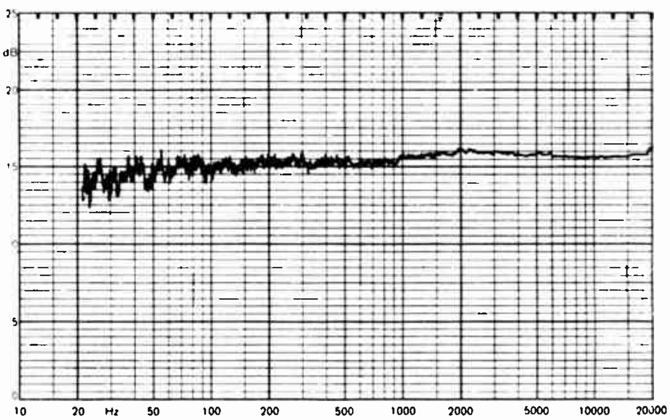
Project 596

Audio 'white noise' generator employs digital technique

This project has many and varied applications, from a test signal source for audio systems through to a noise source for a 'sound effects' unit.



Measurement of the amplitude of white noise versus frequency as measured with a one-third octave filter set.



Measurement of the amplitude of pink noise versus frequency as measured with a one-third octave filter set. To the ear, pink noise appears to have more bass content.

WIDEBAND, or 'white', noise is a curious phenomenon, as it is a signal that contains a very wide spectrum of randomly distributed frequencies, all with random amplitudes, but it has equal amounts of energy in equal bandwidths over the total frequency range of interest. If you measured the noise energy in the band between 100 Hz and 200 Hz, for example, it would be the same as the energy between 3000 Hz and 3100 Hz, or 10 000 Hz and 10 100 Hz. That is, when the energy in a particular band is averaged over a reasonable unit of time it will have the same power as that in the same bandwidth anywhere else in the spectrum.

The basic sound of white noise resembles that of steam escaping from a valve. Electronic 'steam train' effects employ a white noise source to generate the basic sound. Electronic music synthesisers include a white noise generator that can be employed to generate a variety of effects with suitable modification.

White noise and 'pink' noise are used for audio system testing in a variety of ways, which we don't have space to go into here. Pink noise is a modified version of white noise, where the signal is filtered to produce noise that has equal energy per percentage change in bandwidth. For example, the energy in the band 100 Hz to 200 Hz would be equal to that between 3 kHz and 6 kHz — a 100% change in bandwidth in each case.

To hear it, pink noise appears to have more bass content than white noise and also 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 rolls off with decreasing frequency at 3 dB per octave (10 dB per decade).

As this project has application in a variety of areas we have not provided details of housing the board as it may be built into some existing or planned equipment, used on its own, or used with filters and attenuators to suit your particular application.

Making noise

There are several ways to generate white noise. Mother nature provides us with many sources in the electronic world, and an often-used source is the noise generated by a zener diode. We used this technique in the ETI-441 Audio Noise Generator (Jan. '76). However, it has the drawback that the output level is low and results vary widely from diode to diode.

This unit employs a 'digital' method of generation and is similar to the noise generator employed in the ETI-601 4600 Synthesiser (Oct. '73 — July '74). Operation is explained in 'How It Works'.

Construction

Whilst we have provided a pc board design, the project may be constructed on Veroboard, Uniboard or matrix board as layout is not particularly critical.

No particular order of construction is necessary, but take due care with the ICs, as they are both CMOS types. Note that there is a link on the pc board, adjacent to R1. Watch the orientation of the semiconductors.

When you have the board assembled you can test if it's working quite easily. Connect the output to the input of an audio amplifier of some sort via shielded cable. Set RV1 *fully anti-clockwise* and apply a supply of between 5 and 15 volts (the unit draws only a few milliamps of current). Carefully advance RV1 and you should hear a hissing sound not unlike frying or that of escaping steam.

PARTS LIST — ETI 596

Resistors all ½W, 5%
- R1 .... 4k7
- R2 .... 10k
- R3 .... 2k2
- R4 .... 27k

Potentiometer
- RV1 .... 10k

Capacitor
- C1 .... 470p ceramic

Semiconductors
- IC1 .... 4070B
- IC2 .... 4006B
- Q1 .... BC547, BC107 etc.

Miscellaneous
- ETI-596 pc board.

Pink noise

Modifying the output of the generator to produce pink noise simply requires the addition of a suitable filter. A circuit to do this is shown in Figure 3. A filter network with the appropriate characteristics (roll-off of 3 dB per octave above 20 kHz) is connected in the negative feedback circuit of a simple common emitter amplifier stage employing a BC548. Tantalum capacitors are recommended for the two polarised capacitors shown at the input (25u) and the output (1u). They should be rated at 16 V or more. The white noise input should be taken from the wiper of RV1 in the noise generator. Layout is not critical. A 220u/16 V electrolytic bypass capacitor is recommended for the supply rail. ●

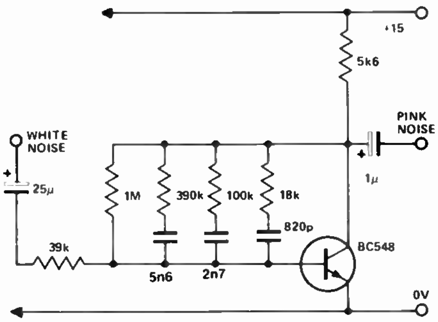
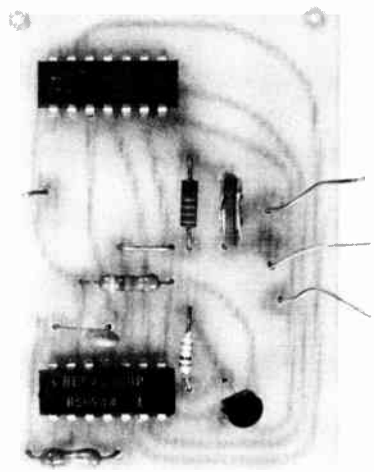
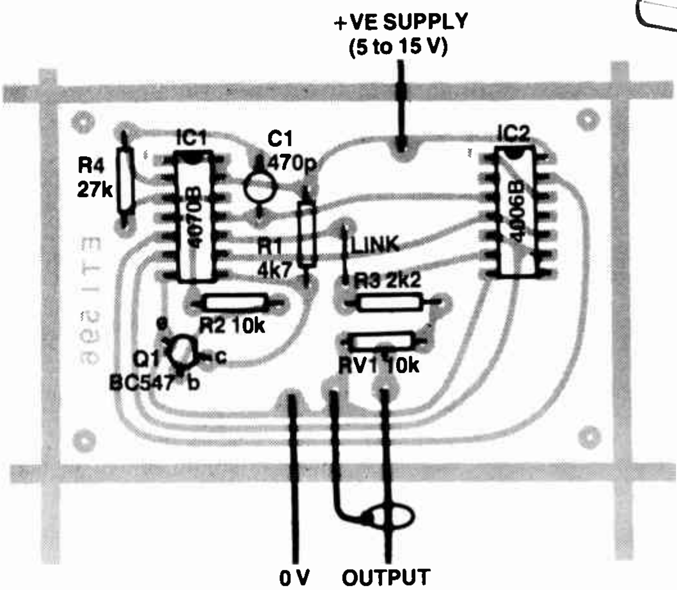
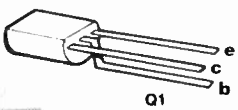


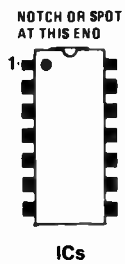
Figure 3. **A filter circuit to produce pink noise from the output of the white noise generator.**

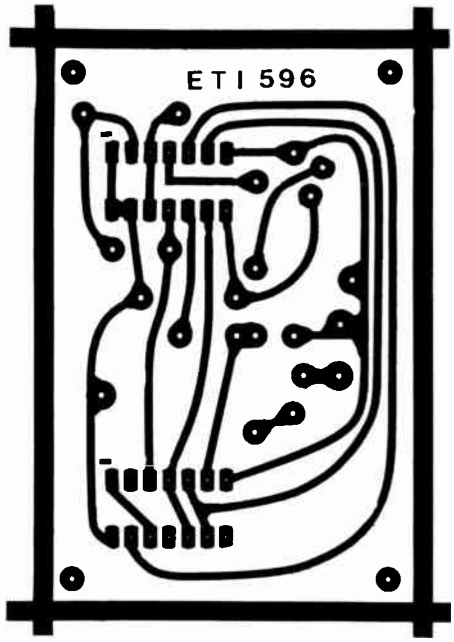




Component overlay.







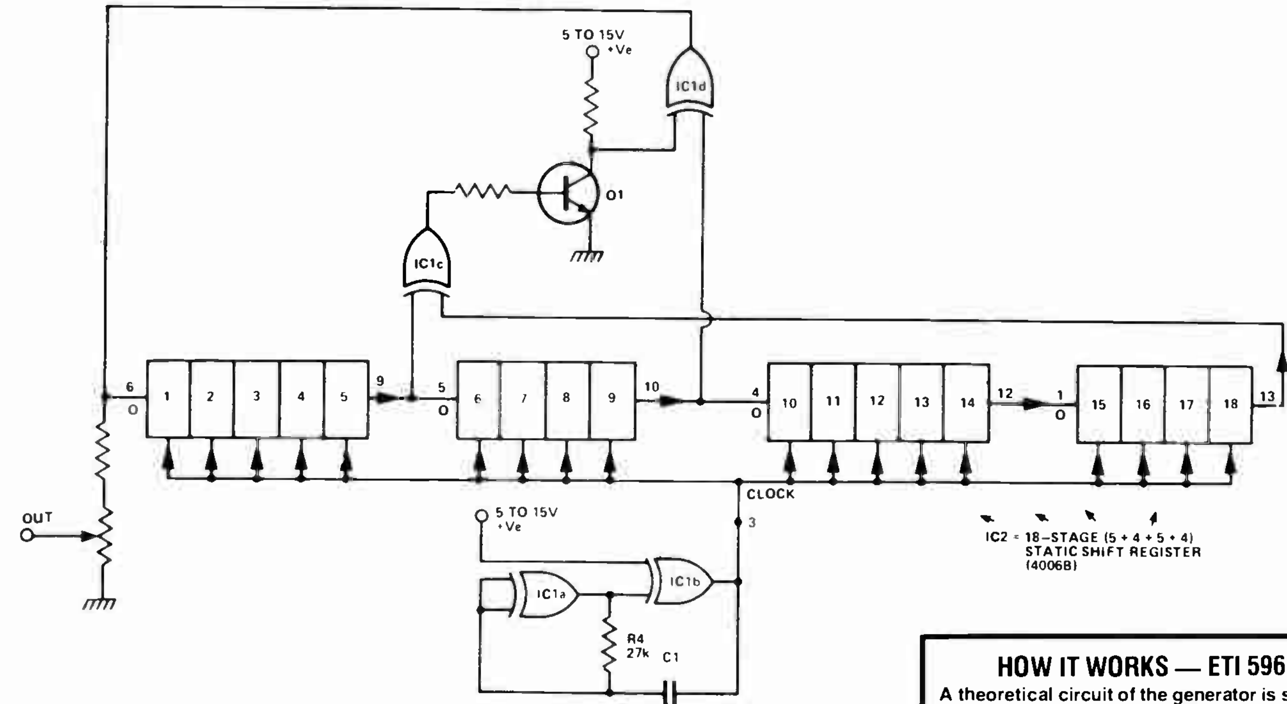


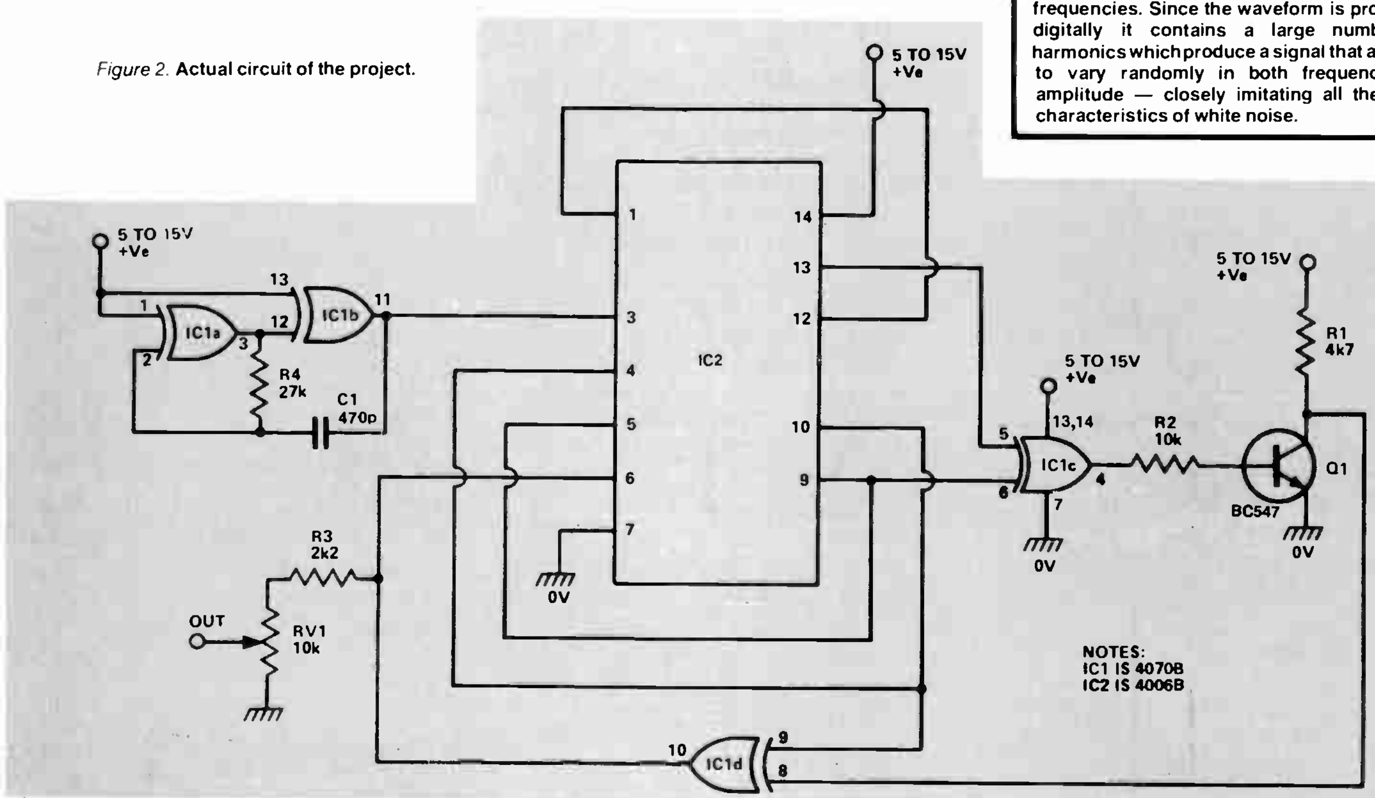
Figure 1. Theoretical circuit, illustrating the operation of the shift register.

HOW IT WORKS — ETI 596

A theoretical circuit of the generator is shown in Figure 1. IC2 is an 18 stage (5+4+5+4) static shift register, in which the logic (0 or 1) information on the data (D) terminal is fed forward one step on the arrival of each pulse from a 60 kHz oscillator, IC1a-IC1b. Exclusive-OR gates IC1c-IC1d are used in conjunction with an inverter, Q1, to feed various outputs of IC2 back to the first data terminal in such a way that the data feeds through the register in an apparently random or jumbled fashion.

In reality, a complex sequence of 0s and 1s flows through the register, repeating once every few seconds and producing the apparently random jumble of fundamental frequencies. Since the waveform is produced digitally it contains a large number of harmonics which produce a signal that appears to vary randomly in both frequency and amplitude — closely imitating all the basic characteristics of white noise.

Figure 2. Actual circuit of the project.



Versatile public address amplifier

Featuring 'speech filtering' and ALC on the mic inputs to improve intelligibility, an 'insert' input and a power MOSFET output stage delivering 150 W RMS maximum output to 100 V, 70 V or low impedance lines, this PA amplifier has much to offer.

Geoff Nicholls

THIS PROJECT has been designed specifically for use in large open areas where ambient noise levels are often high, such as outdoor sporting events. It features *two microphone inputs,* which can be configured for *high or low impedance* microphones, as desired, when the project is constructed. The microphone signal path incorporates *speech filtering* to increase the intelligibility of the voice signal under 'difficult' conditions where other noises compete for the audience's attention. *Automatic level control* (ALC) is included to decrease the variation in voice level between different commentators using the PA and, again, improve 'penetration' where extraneous noise is present. This feature may be switched in or out, as desired.

An *auxiliary input* is provided, which can be used to connect a line level signal or a speaker output from a radio, etc. A *preamp output* enables several power amplifiers to be 'slaved' from the one preamp, and provides a high-level signal for output to a tape recorder, for example.

The *insert* socket allows further signal processing devices to be 'inserted' in series with the signal path from the output of the preamp to the input of the power amp.

The completed amplifier has actually been used at a large, noisy outdoor venue driving eight horn speakers. It gave an excellent account of itself, especially when compared directly against a system using two of our old ETI-480 100 W modules (circa 1976) driven by an ETI-419 preamp (circa 1973!).

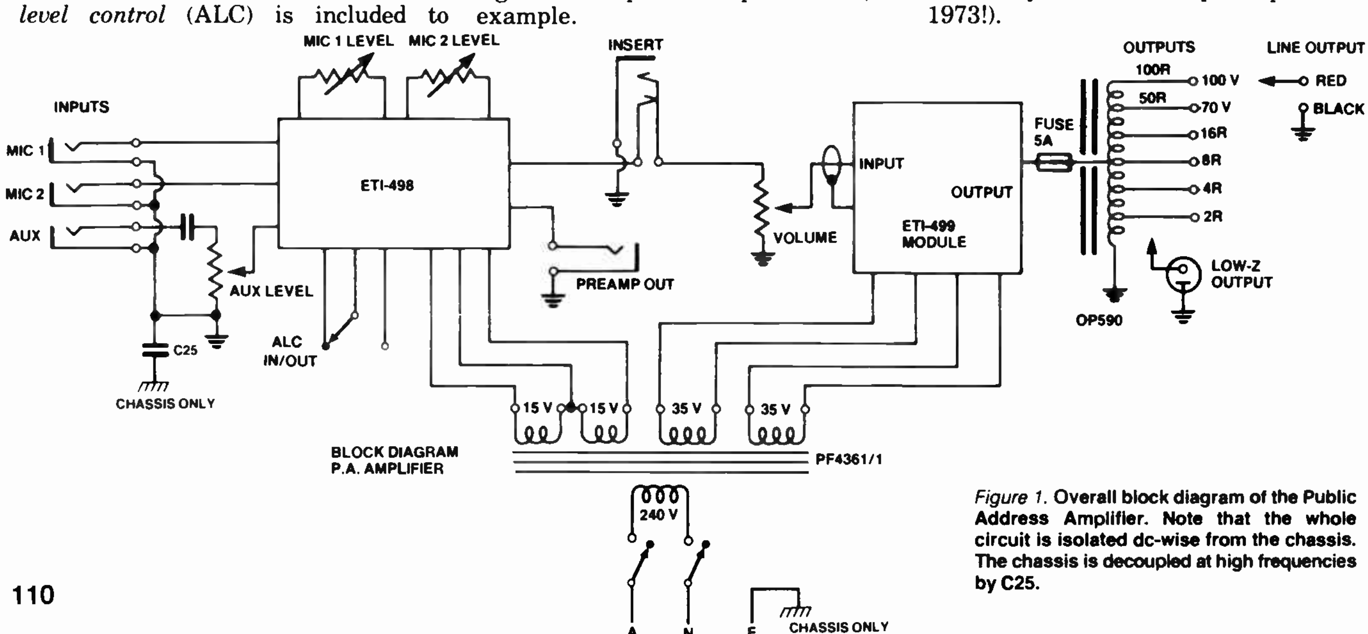


Figure 1. **Overall block diagram of the Public Address Amplifier. Note that the whole circuit is isolated dc-wise from the chassis. The chassis is decoupled at high frequencies by C25.**

Design notes

Most public address set-ups, for all but the largest outdoor events, generally use a single announcer or commentator, occasionally two. Thus as more than two microphones are rarely required, only two microphone inputs have been provided. This allows the constructor either to provide two low impedance inputs, two high impedance inputs or one of each. In a pinch, a low impedance input could be used with a high impedance, high output mic, provided the level control is near minimum. The auxiliary input is shunted by a low value resistor to provide loading for the earphone or external speaker output of radios, cassette decks, etc. It also terminates a line level output correctly.

The signal 'earth' is isolated at low frequencies from the chassis and mains earth to eliminate the possibility of hum loops brought about by connecting external mains powered equipment to the amplifier. The chassis is ac-coupled to the signal earth to higher frequencies as the impedance of capacitor C25 (which couples the chassis to the signal earth) decreases.

The frequency response of the microphone preamps is rolled off rapidly below 150 Hz. This allows the output transformer to deliver more useful power than would be the case if a flat frequency response were employed. The distortion is also reduced by rolling off the low frequencies. Distortion and frequency response curves for the OP590 are illustrated in the accompanying graphs (courtesy Ferguson Transformers).

Automatic Level control (ALC) is included, the purpose of which is to maintain a nearly constant output level with large excursions in signal level. For a signal input level range of greater than 20 dB, the output level will only vary by 3 dB or less for typical microphone input levels. This greatly improves intelligibility and 'punch' of the sound produced, particularly where all sorts of extraneous sounds are about, interfering with the audience's ability to hear the PA.

To provide ALC, I have used half of an NE570/571 audio compressor IC. The basic circuit employed is shown in Figure 2. Inside the NE570/571 are a precision rectifier (the block shown with the diode inside), a variable gain cell (△ G) and an op-amp. Resistors R1, R2 and R3 are inside the chip. The rectifier and variable gain cell operate in the feedback circuit of the op-amp, the gain of which is varied with signal level. At high input levels, the op-amp gain is reduced, at low input levels the gain is increased. Resistor R_x is used to set the maximum gain. The gain of this circuit is given by:

$$K = \frac{R1\,R2\,I_B}{2R3V_{IN\,(avg)}}$$

where I_B = 140 uA

and $\dfrac{V_{IN}}{V_{IN\,(avg)}} = \dfrac{\pi}{2\sqrt{2}}$

for sine waves

The maximum gain is limited (by R_x) to prevent high output levels occurring at low input levels (such as background noise).

The maximum gain is given by:

$$\frac{\frac{R1 + R_x}{1.8}\,.\,R2.\,I_B}{2R3}$$

The output voltage is determined by:

$$V_{out} = \frac{R1R2I_B}{2R3} \cdot \frac{V_{IN}}{V_{IN\,(avg)}}$$

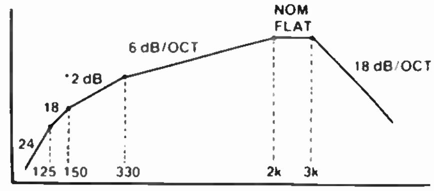
The time constant is important as the circuit needs to react quickly to plosive sounds, yet 'hang on' following peaks. The ALC time constant is determined by:

$$\tau = R1.C_{RECT}$$

The value of C_{RECT} affects the distortion, so its choice is a compromise between providing an effective time constant and keeping the distortion within bounds. Reducing C_{RECT} improves the time constant but increases the distortion; increasing it reduces the distortion but reduces the ALC effectiveness. Distortion is determined by:

$$\text{Distortion} = \frac{100n}{C_{RECT}} \cdot \frac{1\text{ kHz}}{\text{freq.}} \cdot 2\ (\%)$$

The frequency response of the mic signal path is 'tailored' to improve speech intelligibility. There is much redundant energy in the spectrum produced by the voice. Reducing those components below 300 Hz and rapidly attenuating components above 3 kHz removes the redundancy and subjectively provides improved intelligibility for the listener — particularly where extraneous noise is present. Speech 'weighting' filters are used to great effect in communications equipment. In the preamp, the response below 330 Hz is rolled off at 12 dB per octave down to 150 Hz, where an extra filter section provides 18 dB/octave roll-off down to 125 Hz, where the roll-off is further increased to 24 dB/octave. Between 330 Hz and about 2 kHz, the preamp has a response rising gently at 6 dB/octave. Between 2 kHz and 3 kHz, the response is essentially flat. At 3 kHz the response rolls off steeply at 18 dB/octave.



Generalised shape of the preamp mic signal path response showing the rolloff frequencies and attenuation rates. This response provides an effective 'speech weighting' filter to improve 'punch' under noisy conditions.

The steep roll-off below 150 Hz reduces distortion contributed by the output transformer at low frequencies, as mentioned earlier.

The combination of ALC and speech filtering has an additional advantage in that it permits greater sound levels to be achieved before howl-round feedback becomes a problem.

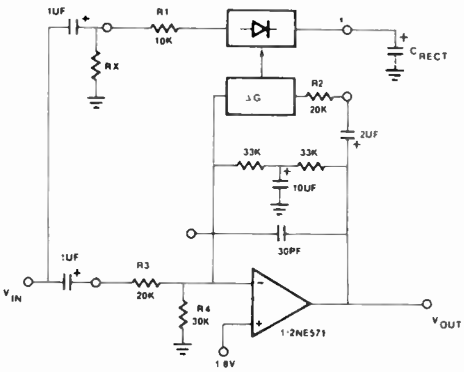
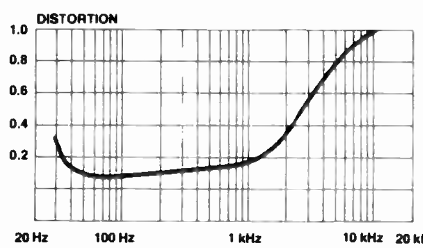
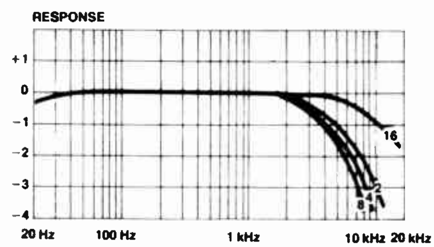


Figure 2. **General circuit of the NE570/571 audio compressor chip connected as an automatic level control (ALC). Note that R1, R2, R3 and R4 are inside the chip.**





Distortion and frequency response characteristics of the Ferguson OP590 line output autotransformer (courtesy Ferguson Transformers). ▶

SPECIFICATIONS ETI-498/499 PUBLIC ADDRESS AMPLIFIER
(as measured on prototype)

Maximum power output	150 W RMS (at onset of clipping)		
Sensitivity	(RMS input for full output)		
	1 kHz	**2 kHz**	
MIC 1	50 mV 1 mV	15 mV (level at min.) 0.3 mV (level at max.)	
MIC 2	100 mV 30 mV	35 mV (level at min.) 10 mV (level at max.)	
AUX.	60 mV	60 mV (level at max.)	
Signal/noise ratios			
(level controls set to provide rated output for the quoted input levels at 2 kHz — ALC off)		**MIC 1** **MIC 2** **AUX.**	−71 dB re 1 mV −73 dB re 50 mV −74 dB re 100 mV
Outputs	2 ohms, 4 ohms, 8 ohms, 16 ohms		
(selectable)	50 ohms (70 V nom.), 100 ohms (100 V nom.)		

For the majority of applications I recommend you drive the output transformer at the 8 ohm tap. The ETI-499 will then deliver 100 W RMS maximum output to the 100 V or 70 V line. On the occasions you may need the greater output, drive the transformer at the 4 ohm tap. However, note that the 100 ohm output will now deliver 123 V, but the 50 ohm tap will deliver 108 V. Change the taps on the OP590 accordingly.

Construction

This article covers construction of the ETI-498 preamp board and assembly of the preamp and ETI-499 power amp module into the case. For details on constructing the ETI-499 150 W MOSFET power amp module refer to page 100 of this book.

If you are constructing this project from a kit, then all the hardware will probably have been pre-drilled or punched. This can certainly save a lot of dreary work. However, if you have the facilities and are tackling the complete construction yourself, the place to start is with the hardware. The case should be cut and drilled according to the dimensions given in our metalwork diagrams.

Note that all the potentiometer shafts should be cut to leave about 12 mm for the knobs to fit flush against the panel when assembled. The solder lugs on each pot should be modified by cutting off the eyelets so that the lug can pass through the holes provided in the pc board for soldering directly to the relevant pad. Pads for the pot lugs should be drilled with a 2 mm diameter hole. When cutting the eyelets from the lugs, make sure you leave as much of the stem of the lug as you can, otherwise they may not reach through the pc board. Note that the volume control pot, RV4, has only one lug soldered to the pc board — check the photo of the assembled pc board and only modify the relevant lug. The auxiliary level pot, RV3, has only two lugs modified.

Bend the modified lugs down *toward the pot shaft* and install each pot on the pc board. Pass the pots through the board from the *non-copper* side and secure them with *one nut* on the copper side.

Next mount the jack sockets. Orientate them so that their lugs are adjacent to the relevant holes in the pc board, and tighten the nuts firmly.

Mount R1, R6 and R15 between the switch and tip lugs on the two MIC jacks and AUX jack, respectively. Extend the switch lug end of the resistor pigtails to the sleeve lug in each case, as shown in the accompanying drawing.

General assembly of the pc board-mounted components follows. Note that *all soldered joints on the copper side of the board should be 'clinched' or cut short* as the board is mounted close behind the front panel of the case.

Mount R21 and R22 between the pc board and the lugs of the PREAMP OUT and INSERT jack sockets, as shown on the component overlay.

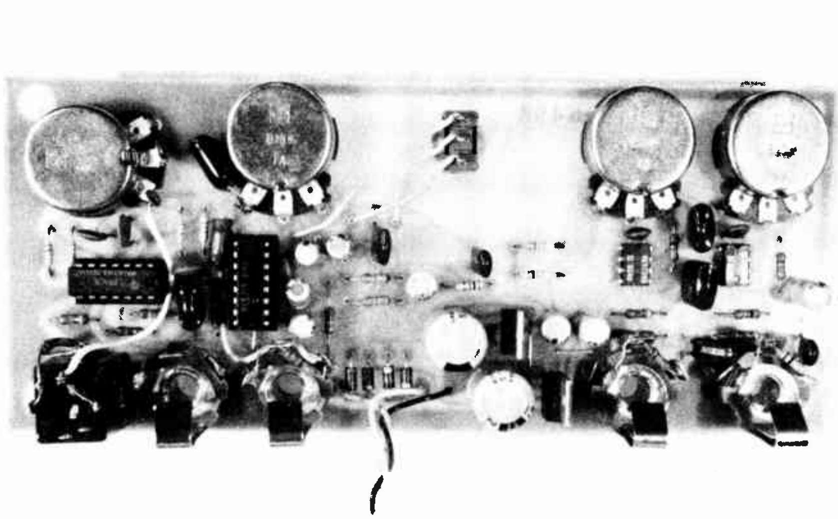
Probably the easiest order of construction is to first install all the resistors, then the ICs and diodes, followed by the capacitors. Watch the orientation of the ICs, the four diodes and the tantalum and electrolytic capacitors. The component overlay shows the appropriate orientation of all these components. Note there are a number of links on the board. Use insulated hookup wire for these.

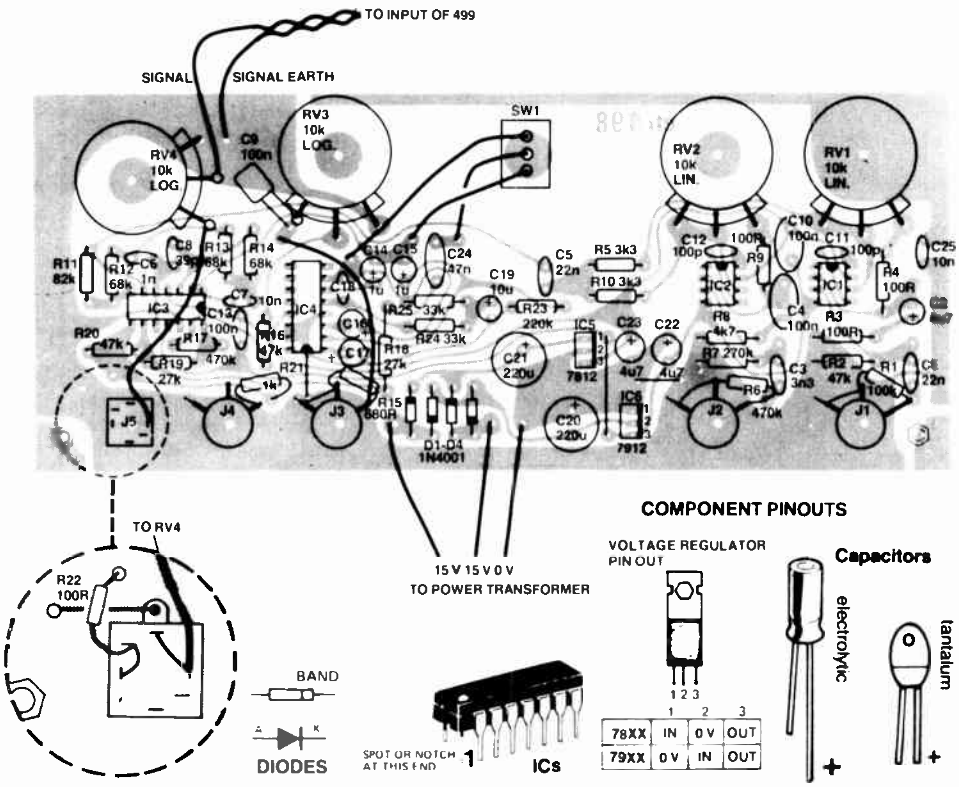
Having finished the assembly of the components to the board, check it carefully to see that you have everything in its correct place and that there are no suspect joints, missed joints or solder bridges on the copper side of the board.

Now solder the two 15 V windings of the PF4361/1 power transformer to the preamp board (make sure they're correctly phased) and attach about 100 mm of twisted-pair hookup wire from the preamp output. Use hookup wire at least as big as 24 x 0.2 mm for this twisted-pair lead.

Mount two countersunk 8 BA x 12 mm screws in the appropriate front panel holes and secure them with *one nut and*

The completed preamp board (but I hadn't put C25 on yet . . .).



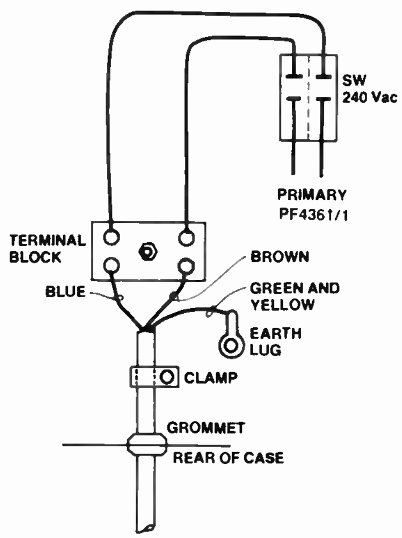


two flat washers each. Cut a piece of thin cardboard to fit between the rear of the pc board and the rear of the case front panel. Cut out clearance holes for the pot shafts and jack sockets. Mount the assembled preamp board to the front panel, not forgetting the cardboard cut-out, which prevents any possibility of shorts between the rear of the pc board and the front panel. Sit the power transformer inside the case as you do this. Check that all sockets clear the case. Use a shakeproof washer (8 BA) and nut on the *component* side of the pc board when securing it to the two screws previously mounted to the case front panel. We used a metal Scotchcal label to 'dress up' the front panel and indicate what controls and sockets were what. This should be attached next. Take care aligning it and smoothing it in place. When you've got it right, install a nut on each pot shaft and all will be secured. Put insulation tape on the case bottom beneath each jack socket to preclude the possibility of the jack contacts touching the bottom of the case.

Next, install the low-Z output socket on the rear of the case. This is a polarised DIN-type socket and installs from the outside. Secure it with two 8 BA x 12 mm screws — nuts on the inside of the case. Install a grommet in the rear panel of the case for the mains cord. We used a Heyco nylon insulated bushing type A2030.

Now for the heatsink, which I presume has already been drilled to take the power amp mounting bracket. There are six other holes in the heatsink: four for the mounting bolts, one to allow access to the low-Z output socket and one for the mains cord. Another grommet should be installed in the latter. Then feed the mains cord through this and through the mains-entry grommet in the rear of the chassis. Just let the mains cable 'hang loose' for the moment.

Mount the heatsink using 6 BA x 20 mm long screws and 6 mm long brass spacers to hold off the rear panel.



Mains cable wiring. Be sure to sleeve all exposed connections for your own protection.

PARTS LIST — ETI 498

Resistors — all ½W, 5% unless noted

R1	100k
R2, 16, 20	47k
R3, 4, 9, 22	100R
R5, R10	3k3
R6, R17	470k
R7	270k
R8	4k7
R11	82k
R12, 13, 14	68k
R18, R19	27k
R21	1k
R23	220k
R24, R25	33k

Potentiometers

RV1, RV2	10k lin.
RV3, RV4	10k log.

Capacitors

C1, C5	22n greencap
C2, 22, 23	4u7/50 V RBLL
C3	3n3 greencap
C4, 9, 10, 13	100n greencap
C6	1n greencap
C7 C25	10n greencap
C8	39p ceramic
C11, C12	100p ceramic
C14, C15	1u/25 V electro.
C16	2u2/25 V electro.
C17	470n/20 V tant.
C18	33p ceramic
C19	10u/25 V electro.
C20, C21	220u/35 V electro
C24	47n greencap

Semiconductors

IC1, IC2	NE5534, NE5534A
IC3	TL074, uA774
IC4	NE570, NE571
IC5	7812
IC6	7912

Miscellaneous

ETI-498 pc board; 4 x 6.5 mm mono phono jack sockets — shorting type; 1 x 6.5 mm stereo phono jack socket with switch (closed when empty); SW1 — SPDT miniature toggle switch; etc.

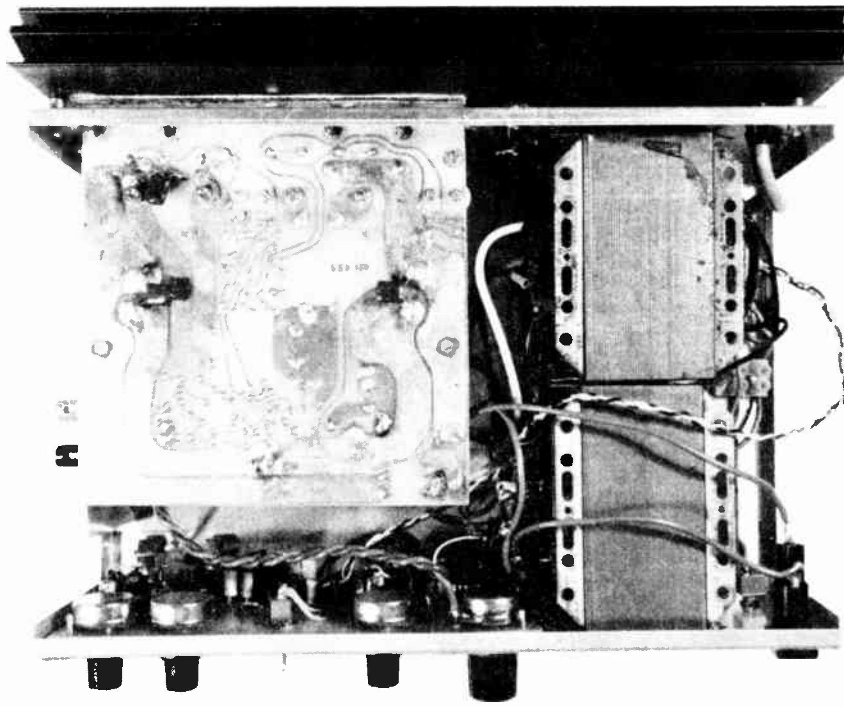
Other components to complete PA amplifier

ETI-499 150 W MOSFET Module (page 100); 1 x PF4361/1 Ferguson power transformer, 1 x OP590 Ferguson line output transformer; 1 x DIN-type polarised speaker connector; 1 x front-loading fuse holder and 5 A fuse; spring terminal type speaker connector strip; K&W case, model C1284; Scotchcal or silk-screened front panel to suit; heatsink — e.g: Rod Irving type HS5 300 mm length black anodised flat-sided, fan-finned type or similar; 4 x 6.5 mm long stand-off pillars; four knobs to suit; mains cable, cable clamp and plug; hookup wire, nuts, bolts & etc.

Price Estimate $230–$240 inclusive

Use lock washers and nuts on the inside of the case, passing the screws through the heatsink and spacers.

Now you can mount the mains terminal block; expose the three wires at the end of the cable inside the case, remove about 20 mm of the mains cable outer sheath and clamp the cable securely to the bottom of the case with a cable clamp. Make sure the mains wires can ▶



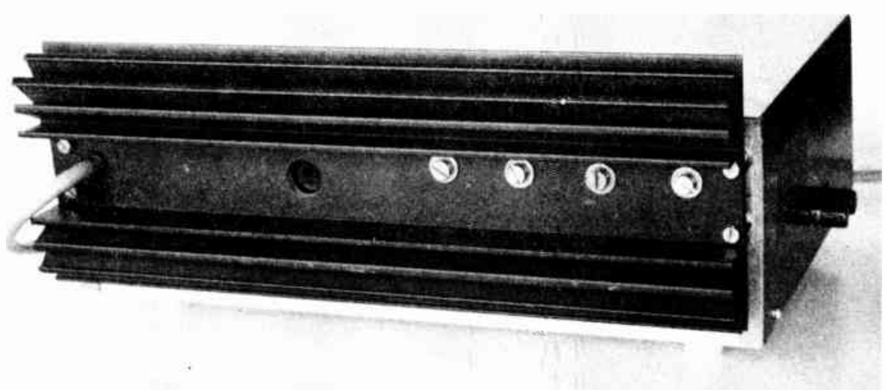
General view inside the case showing location of components. Mount the mains transformer with the 15-0-15 V winding tags uppermost for easiest access.

be easily terminated in the terminal block.

Now install the mains on/off switch on the front panel and wire it up to the terminal block *exactly as shown*. Sleeve all exposed mains connections on the rear of the switch.

Connect the two 35 V windings on the PF4361/1 power transformer to the ETI-499 module (which I presume is ready assembled and waiting). Make sure they're correctly phased. Bolt the transformer in position using 4 BA x 6 mm bolts and lock washers under the nuts. Solder its primary leads to the power on/off switch (sleeve exposed connections, as before).

Install the line output terminals next. I used a spring terminal pair mounted on a strip. These are common speaker connectors, mounted on an angle bracket at the left hand side of the case (when viewed from the front). If you have to fabricate the bracket yourself, do this now. We'll leave the exact details to you as you may have different terminals from the ones I used. The angle bracket is mounted to the bottom of the case with a couple of 8 BA x 6 mm screws. Make sure the bracket clears the filter capacitors on the ETI-499 module. This module mounts upside down from the heatsink and the capacitors come close to the line output bracket. The terminal



Rear view of the amplifier. Note the access hole for the Lo-Z output socket in the heatsink, just left of centre. and the line output terminals protruding from the right hand side of the case.

THE '100 V LINE'

The 100 V line voltage system is widely used for public address loudspeaker reticulation because it simplifies the interconnection of a number of loudspeakers to a single amplifier. It is analagous to the 240 Vac mains voltage system, which allows appliances of vastly different power consumption to be operated from a single supply. For example, a toaster designed to consume 1 kW is made with an impedance of $(240)^2/1000$, or 58 ohms. An electric clock can be run from the same supply, but it only draws 3/5 of 5/8 of 30% of half of . . . and probably has an impedance of 10 000 ohms or more. The ac mains supply can drive many different loads to their designed power rating because it can maintain the voltage supply substantially constant. This is another way of saying the supply has a low source impedance.

Now, public address set-ups often require many loudspeakers of varying power ratings. In order to drive every loudspeaker to its maximum rating at the same time, a constant line voltage is assumed (for a given amplifier power output) and individual step-down transformers (usually with tappings for different power/impedance ratings) are used at each loudspeaker. The most common line voltage employed is 100 V. This means that each loudspeaker will deliver its rated power when fed with a 100 V signal.

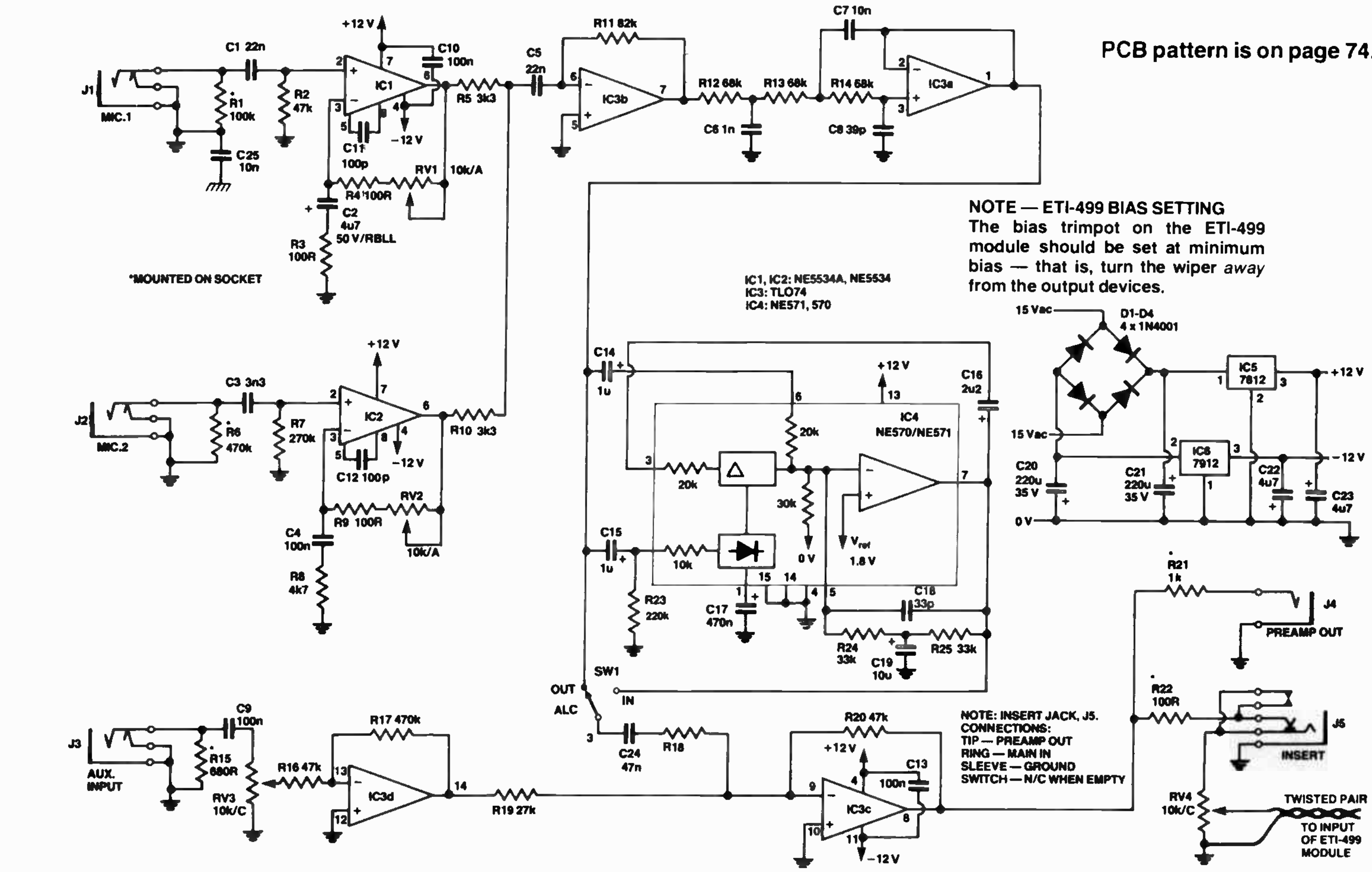
For example, a loudspeaker with an 8 ohm voice coil impedance and a 30 W maximum rating will require a drive voltage of $\sqrt{(8 \times 30)} = 15.5$ V to achieve full output. Thus it will require a step-down transformer having a turns ratio of 100:15.5. With tappings at greater ratios, less power will be delivered to the speaker; i.e: at a tapping providing a ratio of 100:11, 15 W will be delivered to the speaker; at a ratio of 100:9, 10 W will be delivered, and so on.

Any other loudspeaker power/impedance combination with a suitable transformer ratio can be connected to the same 100 V line and driven to the power rating selected, provided that the total loudspeaker power requirement can be supplied by the amplifier driving the line. That is, if ten 10 W-rated loudspeakers are connected to the line, the amplifier must be rated to deliver 100 W. Alternatively, a 100 W amplifier may be employed to drive four 10 W-rated loudspeakers and two 30 W-rated loudspeakers, all connected to the same line.

In practice, horn loudspeakers — which are commonly employed in PA applications — are usually operated at about ¼ to ½ of their maximum rated power, so that a 150 W amplifier with a 100 V line output (such as the one described in this article) will happily drive 15 or 20 horns with maximum ratings of 20 to 30 watts.

Note that a 70 V line system is sometimes employed, but this is less common than 100 V line systems. It works in the same way.

PCB pattern is on page 74.



HOW IT WORKS — ETI 498

The overall circuit arrangement and design features are explained in the general text, so this explanation will be confined to specific circuit description.

MIC INPUTS

We have designated MIC 1 as a low-impedance input and MIC 2 as a high-impedance input. Signals from the MIC 1 input are amplified by IC1. The gain of this stage can be varied by varying RV1, a potentiometer connected in the feedback loop of IC1. The gain can be varied between about 2 and 102. Signals from the MIC 2 input are amplified by IC2, the gain of which can be varied in the same way as IC1. RV2 here can vary the gain between 1 (unity gain) and 3. Resistors R1 and R6 provide dc return for each input capacitor, preventing 'clicks' or 'pops' when inserting or unplugging a mic.

An RC network on each of the MIC 1 and MIC 2 inputs forms single-pole high-pass filters with a breakpoint set at about 150 Hz to reduce low frequency output (the reason for this is explained in the main text). C1-R2 are the relevant components on the MIC 1 input, C3-R7 on the MIC 2 input. A further single-pole high-pass filter is introduced in the feedback network of each mic amp stage: C2-R3 for IC1 and C4-R8 for IC2. The breakpoint is set at 340 Hz and is the lower roll-off point for the speech filtering.

MIC SUMMING

The amplified microphone signals are summed at the outputs of IC1 and IC2, at the junctions of R5 and R10. Another high-pass pole is introduced by these two resistors, in conjunction with C5. This too, is part of the speech filtering.

IC3b buffers the summed mic signals and provides a further amount of gain. The output of this stage, pin 7, drives a low-pass filter stage comprising R12, 13 and 14, capacitors C6, 7 and 8 and IC3a. This filter has a breakpoint set at 3 kHz, providing a sharp roll-off above this frequency.

The net effect of the high and low-pass filters up to this stage provides filtering of the voice frequency spectrum to improve 'intelligibility' where listeners to the PA have to contend with a variety of interfering noises from numerous sources at open-air events.

ALC

The 'automatic level control' (ALC) circuit centres on IC4, an NE570 or 571 'compander' IC. This circuitry attempts to maintain a near-constant output, provided the input from the microphone stages exceeds a threshold level determined by the value of R23. Decreasing R23 *increases* the threshold, reducing the effect of the ALC. The ALC prevents 'soft' sounds from being lost in external noise while helping prevent clipping from plosive sounds ('p' and 't' for example) in speech.

The 'attack' time of the ALC is determined by C17. Decreasing the value of this capacitor improves the transient response of the stage (helping it cope with plosives) but has the drawback of increasing the distortion. We chose the value shown as a compromise between these two parameters and it seems to work well in practice.

The input to the ALC circuit (from pin 1, IC3a) and the output (from pin 7, IC4) go to SW1, a SPDT switch, which selects ALC IN or ALC OUT.

AUX INPUT

The auxiliary input is meant for low impedance, 'line level' signals, such as from the output of a tape recorder. Input impedance is determined largely by R15 and is around 600 ohms or so. IC3d provides amplification, having a gain of 10. The input level may be attenuated by RV3. This, in conjunction with C9, provides a single-pole high-pass filter.

INPUT MIXING

The MIC and AUX inputs are summed at the input of IC3c (pin 9). C24 and R18 provide a further high-pass filter for the MIC-ALC stages with a breakpoint at 125 Hz, further increasing the attenuation below this region.

OUTPUTS

The output of IC3c (pin 8) passes to the input of the ETI-499 power amp module via the INSERT jack (J5) and the volume control, RV4. A PREAMP OUTPUT is provided too, at J4.

The INSERT jack is a stereo/switched type and provides a point where a graphic equaliser or howl-round stabiliser can be introduced (see the ETI-485 Graphic Equaliser and the ETI-486 Howl-Round Stabiliser in our publication '30 Audio Projects').

Resistors R21 and R22 provide isolation between the outputs and short-circuit protection for the output of IC3c.

POWER SUPPLY

The on-board power supply derives its input from the two 15 Vac windings on the PF4361/1 power transformer used to supply the ETI-499 module. These two windings are connected to provide a centre-tapped 30 Vac supply (15-0-15). Two full-wave capacitor input rectifiers, comprising D1 to D4 and C20-C21, then provide about ±20 Vdc input to a pair of three-terminal regulators: a 7812 which provides a +12 V rail, and a 7912 which provides a −12 V rail.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strip should be mounted to the bracket with countersunk 8 BA x 6 mm screws — secure them well so that the screw heads are flush. A cutout is necessary in the case lid to clear these terminals.

Mounting of the OP590 output transformer comes next. First, solder three lengths of heavy gauge hookup wire to the appropriate lugs on the transformer. Use 24 x 0.2 mm insulated hookup wire, at least. Bolt the OP590 in place using 4 BA x 6 mm bolts and lock washers under the nuts. Wire the output of the ETI-499 module to the fuseholder. Then wire up the OP590, the low-Z output socket and line output terminals.

The ETI-499 power amp module can now be mounted. First, however, wire up the twisted-pair input from the preamp. Apply thermal compound to the heatsink bracket on the power amp module and then bolt the module to the heatsink using 2 BA x 12 mm screws and lock washers under the nuts.

Now do a double check of all your interwiring. Using an ohmmeter, ensure there is no dc path between the preamp-power amp signal earth and the case. If all seems well, install a three-pin on the mains cable and you're ready to power up for a test run.

Testing it

Set all level controls to the minimum position (fully anti-clockwise). Connect a loudspeaker to the appropriate output (either low-Z or line output). Turn the ALC off. Insert a microphone in the appropriate jack — according to what mic you have and how you've configured the inputs. Hold your breath . . . and turn on. Advance the mic level control and the volume control and talk into the mic. The sound should be clear and undistorted. If you have any problems, switch off and trace the fault before continuing.

Assuming all is well at this stage, obtain a cassette deck or tape recorder and plug it into the auxiliary input. Play a pre-recorded tape and see that the sound is clear and that the gain controls have plenty of 'room to move'. Next, try recording on tape, taking the recorder input from the preamp output jack while speaking into the mic.

Try out the effect of the ALC. There should be a dramatic difference in the dynamic range when speaking, without noticeable distortion.

Note that the speech filtering in the mic circuits reduces the bass and 'softens' plosive sounds like 'p' and 't'. The steep roll-off above 3 kHz contributes to making the voice sound less 'natural', but dramatically improves intelligibility when extraneous noise is present. There is no filtering on the auxiliary input and the bandwidth is only limited by the output transformer.

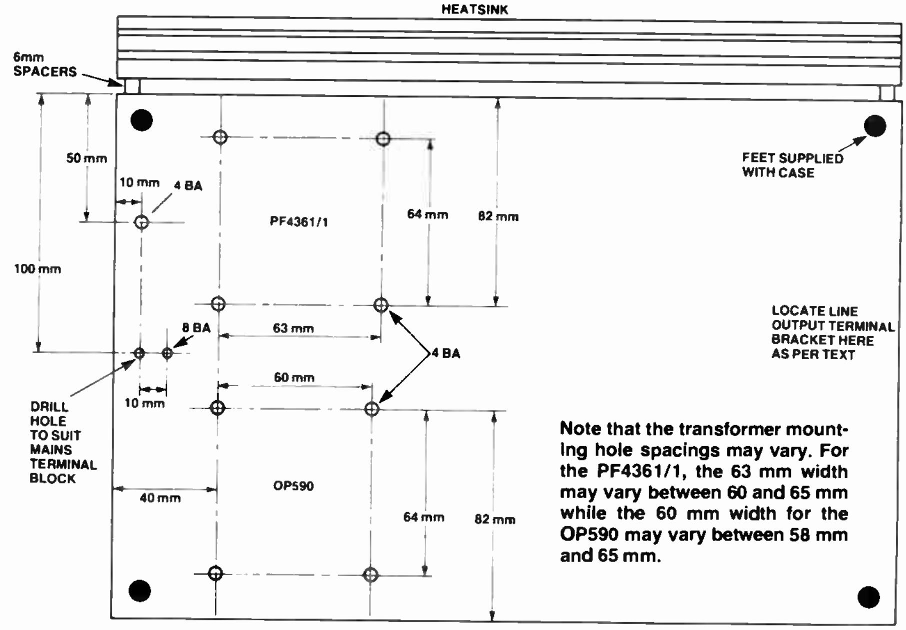
Conclusion

In use, the ALC needs to be employed with care — it is not necessary on all occasions. It is generally most effective when other sources of extraneous noise are present in the area.

The heatsink proved more than adequate for the job, from experience, barely getting warm to the touch.

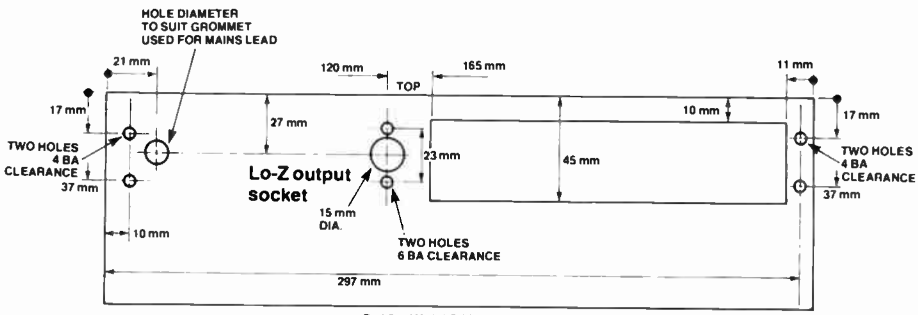
I think you'll find the ETI-488/499 PA very effective — may it 'cut through the mush' for you every time! ●

ARTWORK: Full-size reproductions of the front panel artwork are available from Electronics Today International. Send an A4 stamped, self-addressed envelope to ETI-498/499 Artwork, Electronics Today International, 4th Floor, 15 Boundary St, Rushcutters Bay NSW 2011.

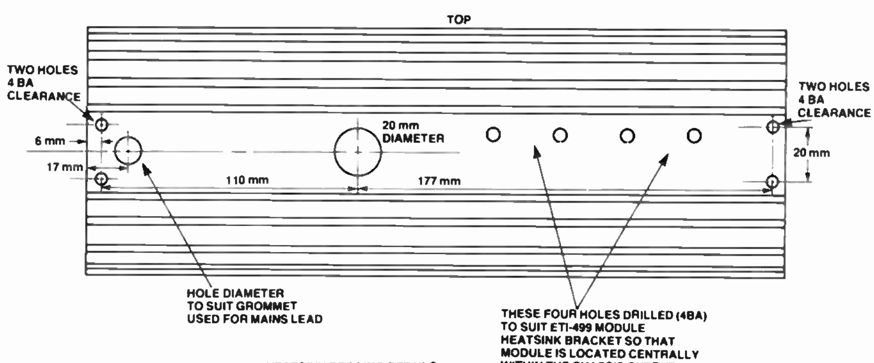


Note that the transformer mounting hole spacings may vary. For the PF4361/1, the 63 mm width may vary between 60 and 65 mm while the 60 mm width for the OP590 may vary between 58 mm and 65 mm.

DRILLING DETAILS — BOTTOM OF CASE



REAR PANEL DRILLING DETAILS



HEATSINK DRILLING DETAILS

NOTE: The front panel is best marked out and drilled using the preamp pc board as a template. The output fuseholder and mains power switch are located at the extreme right of the front panel, at top and bottom, respectively. The front panel artwork or Scotchcal panel may be used to locate hole centres for these, and they mount directly on the panel. Drill clearance holes for the preamp controls and input jacks, 2 — 3 mm oversize.

Universal dc-dc inverter

This inverter can be configured to suit a wide variety of applications — powering our audio amplifier modules from a 12 V battery, powering 12 V equipment from a 24 V or 32 V battery, deriving a high voltage supply from a low voltage dc source etc, etc.

David Tilbrook
Roger Harrison

WE'VE CALLED this a 'universal' dc-dc inverter as it has been designed so that, simply by winding the appropriate secondary, or secondaries, on the output transformer you can derive almost any dc output voltage(s) you want. Thus, this project can be used to power any of our audio power amplifier modules (ETI-470, ETI-477, ETI-480, ETI-499), with perhaps the exception of the ETI-466 300 W amp. You can power the ETI-565 HeNe laser and the ETI-452 Guitar Practice Amp, or any other project or device you desire, providing it falls within the power rating of the inverter.

We described a dc-dc inverter power supply to power a PA/guitar amp employing one of our 100 W '480 modules way back in May 1977. This was the ETI-481PS which provided ±40 V rails from a 12 V battery. It ran at 20 kHz and required special rectifier diodes, a pot-core and a ferrite transformer assembly — all of which are now very difficult, if not impossible, to obtain. Since we described the Series 5000 stereo power amplifier in the January-February-March '81 issues of ETI, many readers have sought to adapt it for use in their vehicles (car/truck/sin bin . . .). Parts for the ETI-481PS have now virtually 'dried up' and we have been pressed to do a 'replacement'. Well, this is it, albeit with some refinements.

We decided to make this inverter a 'universal' project as it struck us there are wider applications than was first envisaged. Besides, we've had a number of requests for a 12 V inverter to power the ETI-565 HeNe laser, both to provide portability and to free it from mains operation for improved safety. It seems that many schools have built the laser for use in their science labs.

Design considerations

A number of factors were considered of prime importance when we tackled the design of this inverter. First came the frequency of oscillation. Would we have a low frequency design, which eases component selection and ensures their availability, or set the oscillation frequency above the audio band? A third option was to do something between those two extremes. Cost, size and component availability were also important.

The problem with a low frequency inverter, operating at — say — 2 kHz or less, is suppressing the switching 'spikes' that appear on the power supply rails. This can be difficult and these spikes almost inevitably create interference in low-level input stages. As the spikes contain predominantly odd harmonics the result is a cacophony of buzzes that is constantly present. Rectifier filter capacitors at the lower frequencies are, by necessity, large and we didn't want a bulky project.

Setting the inverter oscillation frequency above the audio band, at 20 kHz for example, gets rid of the above problem but introduces several others. Circuit techniques that work at low frequencies require specialised components at 20 kHz. Hence, a different inverter technique is necessary, and this inevitably increases costs and specialised components often prove hard to get — which became the major problem with the ETI-481PS which ran at about 25 kHz.

We chose the median course. Setting the oscillation frequency at around 6-7 kHz puts the odd harmonics where they (mostly) won't be heard. Filtering is easy and suitable capacitors are compact.

We wanted a design that used a minimum number of components, so that the project would be compact, but consistent with the other restraints. There are three common techniques employed in transistor dc-dc inverters these days — the self-excited single transformer circuit, the self-excited dual transformer circuit and the driver inverter.

The self-excited single transformer inverter is by far the simplest. The general form of this inverter is shown in Figure 1. The transistors operate in push-pull and feedback is taken from a winding on the output transformer. It

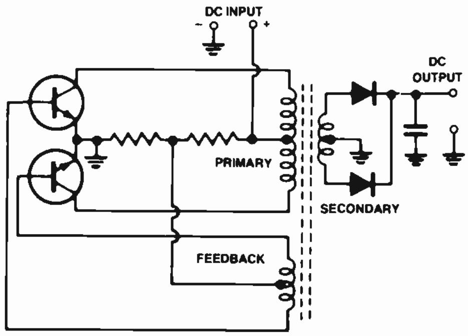


Figure 1. **Typical circuit of a self-excited single transformer dc-dc inverter. Efficiency of this type of circuit often exceeds 90%, with correct choice of transistors.**

has two great advantages — simplicity and high efficiency. With proper choice of transistors, efficiency can be in excess of 90%. However, at the sort of powers we envisaged the inverter would have to deliver — around 200 W or so — switching transistors with suitable current ratings and low saturation voltages (for that's where you lose efficiency) are not cheap, or readily available. Germanium switching transistors are the best. Tried to buy a 20 A germanium switching transistor lately? Some MOS switching devices are also suitable, but still hard to get. You could parallel transistors of lower current rating but the traditional method of using emitter 'ballast' resistors severely affects efficiency. By using a special primary winding on the transformer, as shown in Figure 2, the devices are essentially in parallel but collector-emitter current sharing is done in the transformer primary. Base current sharing is effected by the series base resistors.

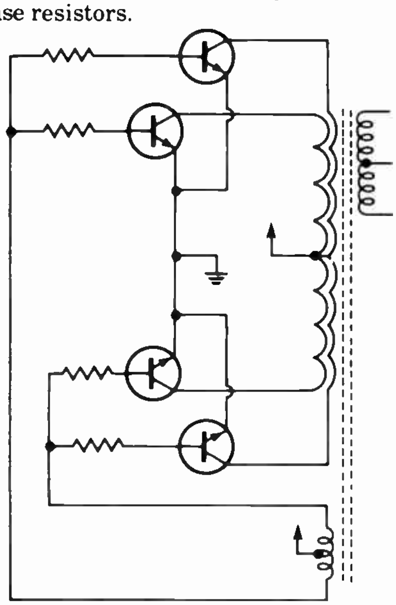


Figure 2. **Obtaining more power output from the simple self-excited inverter by means of 'paralleling' transistors with a special, quadrifilar-wound, primary. For high power use, lower cost transistors can be used.**

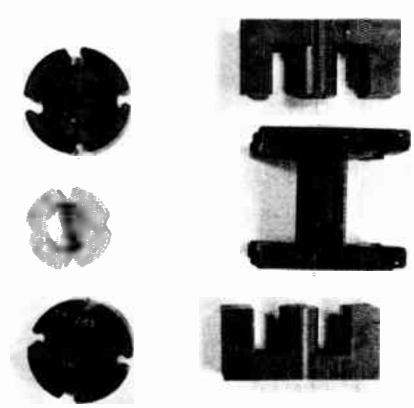
Thus, common-or-garden transistors, like 2N3055s, can be used and little power is lost here. The major drawback is attaining the required oscillation frequency. For this sort of inverter, the oscillation frequency is given by:

$$f \cong \frac{V_{supply}}{4N\delta_m}$$

where: f is frequency of oscillation
V_{supply} is dc input
N is primary turns, collector to centre tap
δ_m is magnetic properties of transformer core

Thus, for a given voltage and core properties, N must be relatively small to achieve the required frequency of oscillation. But here comes a catch. The feedback winding must develop enough voltage to drive the base such that the transistor(s) saturate properly. In practise, this means about 3 V. If the dc input is 12 V, you need a turns ratio of 4:1 between the primary and feedback windings. If you make the feedback winding (centre tap to one set of bases) one turn, then the primary has to be four turns. With available cores, the oscillation frequency did not even approach what we wanted.

A driven inverter is more complex, but it overcomes the problem just outlined. A typical arrangement is shown in Figure 3. This employs a low power, self-excited push-pull oscillator driving a set of push-pull output transistor switches which drive the output transformer. The old ETI-481PS inverter was of this type. Efficiency is the greatest drawback of this type of circuit. The driving oscillator always draws significant power. You can achieve efficiencies



The transformer assemblies we used. At left is the FX2242 potcore assembly; right is the Philips EC52/24/12 assembly. Note the ferrite 'Es' of this assembly have round centre legs. The bobbins are shown in the middle.

of 80%, typically, but at 200 W output, you're losing 40 W and in a battery system, this is not good.

We settled on the self-excited dual transformer technique, illustrated in Figure 4. Here, a separate feedback transformer is used and it is this which controls the frequency of oscillation. This enables the choice of the right sort of core to obtain realistic turns ratios and the desired frequency of oscillation. We managed to use a common potcore for the feedback transformer (a 36 mm diameter FX2242 type) and an EC-core for the output transformer of a type we have used previously (in the ETI-1505 fluorescent light inverter). Supply rail filtering can be done quite effectively with common greencaps. Ordinary 2N3055 transistors — which cost less than a dollar these days — can be employed, thus keeping the cost down. ▶

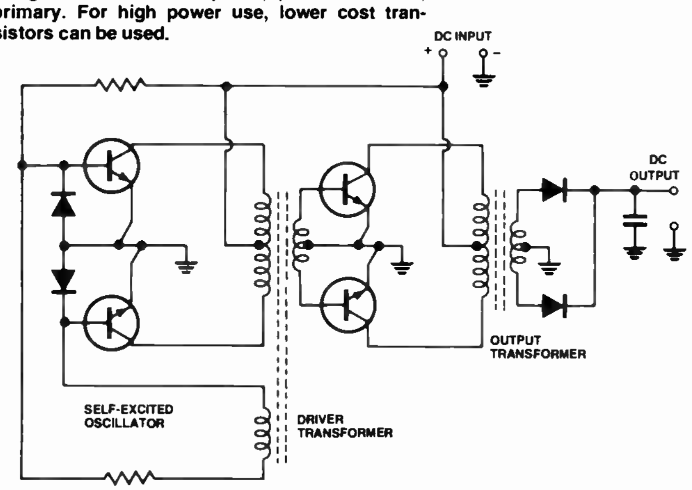


Figure 3. **Typical circuit of a driven dc-dc inverter where a low power self-excited oscillator drives a set of transistor switches which drive the output transformer. Of the three types, this technique has the poorest efficiency.**

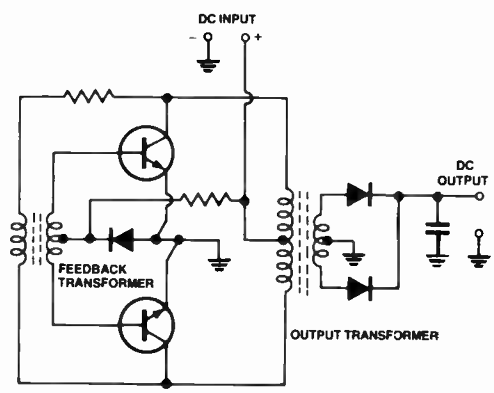


Figure 4. **Circuit of a self-excited dual transformer dc-dc inverter where the feedback is separated from the output transformer. In this circuit, the oscillation frequency is determined by the feedback transformer. Efficiencies similar to the Figure 1 circuit can be achieved.**

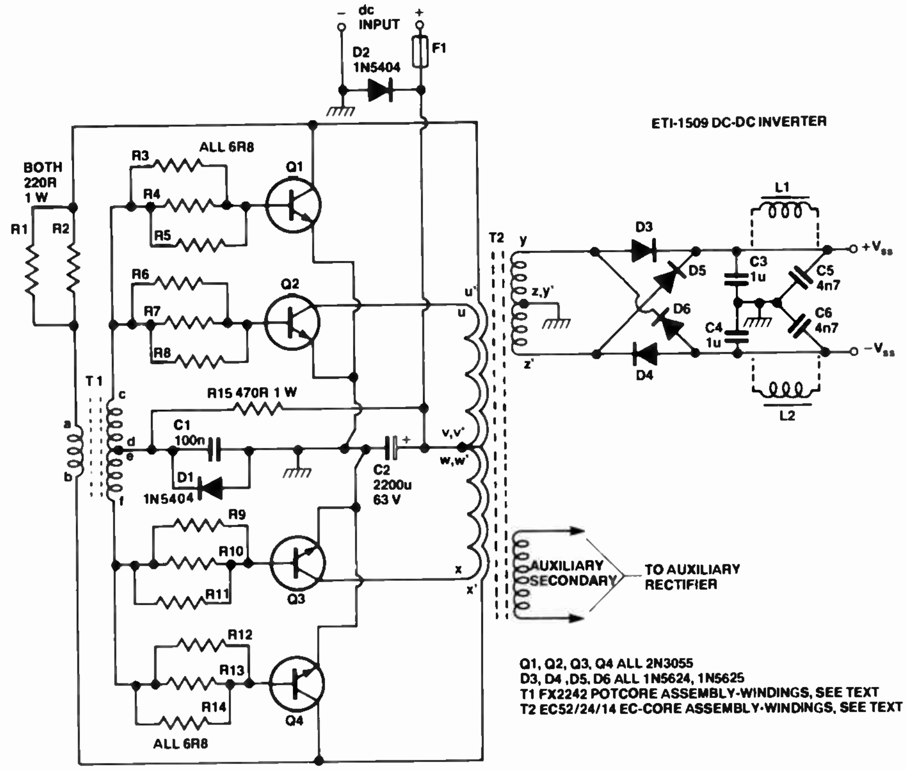


Figure 5. **Circuit of the universal dc-dc inverter. The two inductors shown at the rectifier output, L1 and L2, are only necessary if extra filtering of the dc output is necessary. These can be any value between about 1 mH and 10 mH. Speaker crossover inductors are ideal for this application. Choose inductors rated to carry the dc output current. At low currents (200 mA or less, say) RF chokes will suffice.**

HOW IT WORKS — ETI-1509

The circuit is a push-pull self-oscillating inverter with feedback provided by transformer T1, output being taken from T2. The moment the supply voltage is connected, current flows through the 470 ohm resistor R15, through the secondary of T1 (c-d-e-f) and the base current limiting resistors of the transistors Q1 to Q4 One of these transistors will turn on as circuit balance is not perfect and device characteristics are not matched. If, for example, Q1 commences to turn on, current will flow through the u'-v' primary of T2. This causes a magnetic field to build up in the transformer core creating a positive voltage on x' (x goes positive too). This puts a positive voltage on b of T1's primary which is wound and connected so that a positive voltage here causes a positive voltage on c of the secondary. Thus, the bases of Q1 and Q2 are driven positive, turning them hard on and the bases of Q3 and Q4 are driven negative, turning them hard off.

Since the primary of T1 acts as an inductor, the current flowing in it increases linearly for as long as the voltage is applied until finally the magnetic field intensity reaches a maximum, where the transformer core saturates. At this moment, the impedance of the core drops since the saturated core cannot maintain the relatively high inductance of the primary. The decreased impedance causes an increase in current flowing in the primary, driving the core even further into saturation until most of the coupling between the primary and secondary is lost. This causes the drive voltage to the bases to disappear. Current stops flowing in the transformer and the magnetic field starts to collapse. This causes the voltage sense of each winding to reverse. Thus, u and u' on T2 go positive and x-x' go negative. This causes b on T1 to go negative and thus c goes negative, removing charge from the bases of Q1 and Q2 which turn off. At the same time, f goes positive, turning Q3 and Q4 on. The whole sequence of events then repeats for the opposite 'side' of the oscillator, until once again, Q1 and Q2 are driven on and oscillation results.

The frequency of oscillation depends primarily on the core material of T1, the turns on its primary winding (a-b) and the applied voltage. In this case the frequency is around 6 - 7 kHz. Resistors R1 and R2 provide control of the feedback and diode D1 provides a return path for the base emitter current to the transistors when the secondary voltage of T1 reverses direction.

The output transformer, T2 has been arranged to provide a simple volt/turn ratio. Common, low cost 2N3055 transistors have been used and the special, quadrifilar-wound, primary is a means of effectively connecting pairs in parallel, as discussed in the main text.

Although a full wave bridge rectifier is shown here, any rectifier circuit may be used on the output. Filtering is readily provided by low value non-polarised capacitors, such as greencaps.

This inverter circuit, like the simple self-excited inverter shown in Figure 1, has the advantage that, when an overload or short circuit is applied to the output, oscillation cannot be maintained and it stops, which is not the case with a driven inverter. Thus, some protection is afforded both the inverter and the equipment connected to it.

Diode D2 prevents damage by blowing the fuse should the dc supply input be connected in reverse.

The circuit of the inverter is shown in Figure 5.

As it was to be a 'universal' inverter, the output transformer was designed to provide a simple volts/turn ratio for the secondary, making it easy to calculate and wind the secondary for the required output. The ratio is two volts per turn. A 'Table of Suggested Outputs' has been drawn up for a variety of applications. Note that more than one secondary can be wound on the output transformer if required.

Construction

The inverter can be housed in any convenient enclosure and we have shown only general construction as layout is not particularly critical. A suggested layout is shown in the wiring diagram. A metal case is assumed. Four tagstrips are used as tiepoints to terminate the two transformers and to mount the resistors, capacitors and diodes. The four transistors can be mounted to the case. No heatsink is necessary as little power is dissipated, but the transistor cases need to be insulated so use insulating washers and bushes as shown in the accompanying assembly diagram.

The primary windings of T1 are terminated to a 7-lug tagstrip. The collector of each transistor is connected to the appropriate lug on the tagstrip using heavy duty insulated hookup wire, preferably 32 x 0.2 mm or heavier, as switching currents can be in excess of 20 amps. The mounting lug of the 7-lug tagstrip is used as a ground tiepoint for the transistor emitters. Connect them up with heavy duty hookup wire too. Capacitor C2, the input protection diode D2 and the bias resistor R15 also mount on this tagstrip.

Two 5-lug tagstrips are used to terminate the feedback transformer, T1, and to support the base current limiting resistors and several other components.

The rectifier components are mounted on another tagstrip. The high frequency bypassing components should be mounted on a tagstrip away from the rectifier and very close to the hole where the supply output wires exit from the case. If extra filtering is necessary (by the addition of L1 and L2) then it is easily inserted between the rectifier tagstrip and the 3-lug tagstrip containing C5 and C6.

When laying out the components to suit your case, keep the two 5-lug tagstrips fairly close to the four transistors so that the base current limiting resistors can be easily mounted between the tagstrips and the base pins of the transistors. Also, mount the 7-lug tagstrip that terminates the primaries of T1, close to the transistors so that high

PARTS LIST — ETI-1509

Resistors .... all ½ W, 5% unless noted
R1, R2 .... 220R, 1 W
R3 — R14 .... 6R8
R15 .... 470R

Capacitors
C1 .... 100n greencap
C2 .... 2200u/63 V axial electro. (or 2500u/50 V)
C3, C4 .... 1u greencap
C5, C6 .... 4n7 ceramic

Semiconductors
D1, D2 .... 1N5404
D3, 4, 5, 6 .... 1N5624, 1N5625
Q1, 2, 3, 4 .... 2N3055

Miscellaneous
T1 .... FX2242 potcore assembly (windings, see text).
T2 .... Philips EC-core assembly (windings, see text).
2 x EC52/24/14 cores (4322-020-52520)
1 x former, no tags (4322-021-33020)
clamp assembly:
1 x 52PLATE
1 x 55UBOLT
2 x 632NC2A

Two 7-lug tagstrips; two 5-lug tagstrips; one 3-lug tagstrip; chassis or box as required; winding wire — 1 mm and 0.4 mm enamelled copper wire; 32 x 0.2 mm hookup wire; 10 x 0.2 mm hookup wire; nuts, bolts etc.

Price estimate $30 — $35

current carrying leads are kept short.

Make sure you have enough room to mount both T1 and T2. A single bolt through the centre hole of the T1 pot-cores will secure it but use a fibre washer under either the head of the bolt or under the nut to prevent cracking one of the potcore halves.

If you use a chassis that comes in two halves (like we did on our prototype) mount all the tagstrips and components on the one half so that all the ground tie points are connected together whether the case is split in half or not. If you use a box with a lid (like a diecast box, for example) mount all the components and tagstrips either on the lid or in the box, for the same reason.

When you have a layout finalised for the housing you're using, it's best to assemble all the electronics first, leaving the transformers till last. *Then* wind the transformers. Use our Table of Suggested Outputs and the Transformer Winding Details as a guide to assemble the two transformers. When you've done these and checked that all is correct, mount the transformers and wire them up.

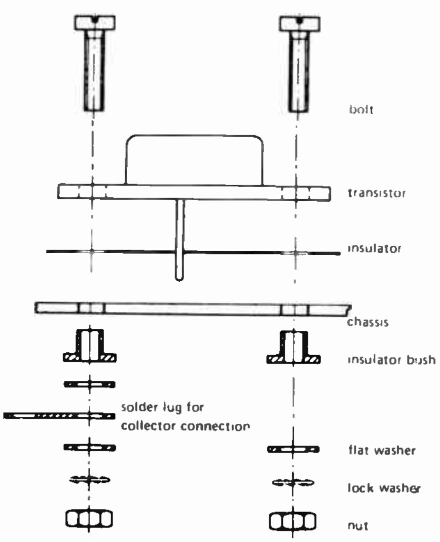
For the dc input leads, use heavy duty cable or hookup wire, remembering that 20 A or so may be passing through it. Don't forget the line fuse.

Firing it up

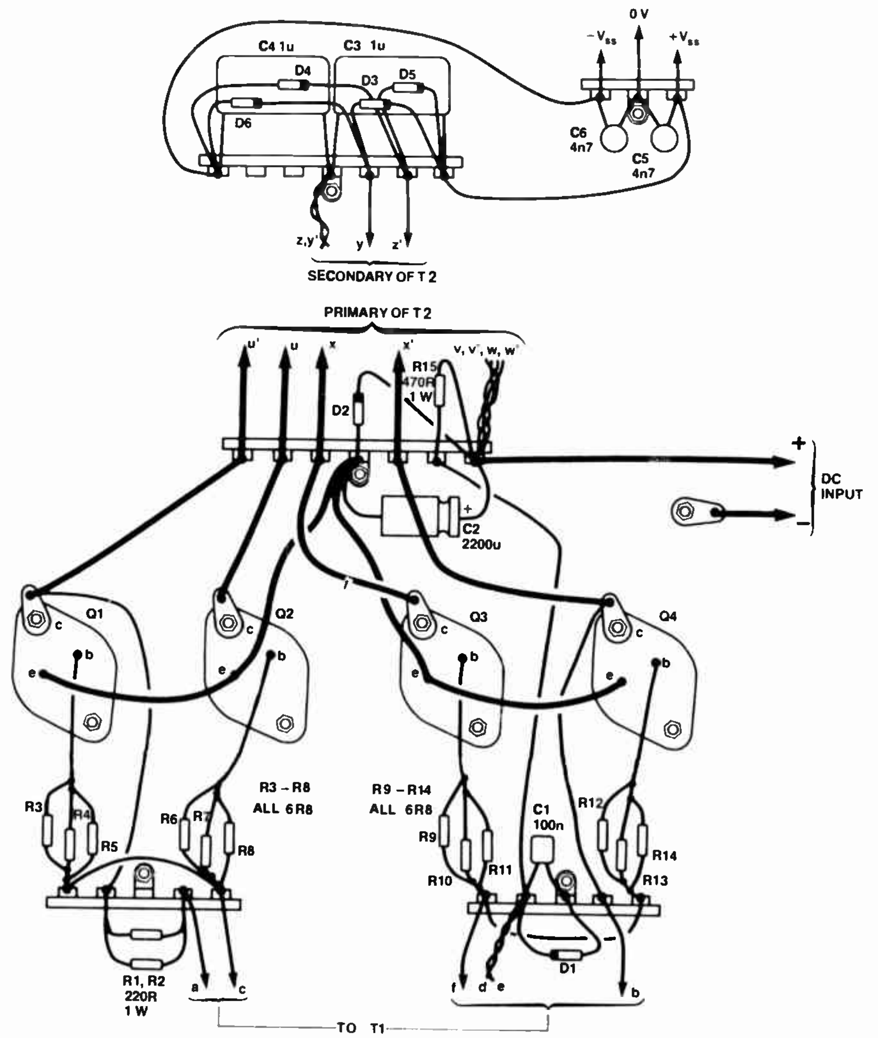
Simple. If you're confident you've wired it up correctly (*and* checked it), hook a multimeter to the dc output and connect power to the dc input. You should hear the transformers 'sing' immediately at quite a high pitch (around 6 kHz if your pitch sense is that good). The output voltage should rise to what you require, also. If you don't hear the transformers sing, then switch off and reverse the primary (a-b) connections of T1. Switch on and the inverter should burst into life. If it doesn't — or worse, bursts into flames! — switch off and take a look at your wiring. Correct any faults before trying again. In particular, make sure you have D1 the right way round.

If all is working as it should, you could try and assemble a 'dummy' load for the output. A suitable set of power resistors will do. The exact resistance will depend on the supply and the load current it has to supply. We'll have to leave this to you.

Under load, the output voltage should be within a few per cent of what you require and power dissipation of the



Mounting the power transistors.



General wiring diagram of the inverter. Layout is not critical. However, use heavy duty (32 x 0.2 mm) hookup wire for the heavy leads shown here for wiring the collectors and emitters of the four power transistors. Capacitors C5 and C6 provide high frequency bypassing of the output rails. Mount them close to where the supply output leads exit the case.

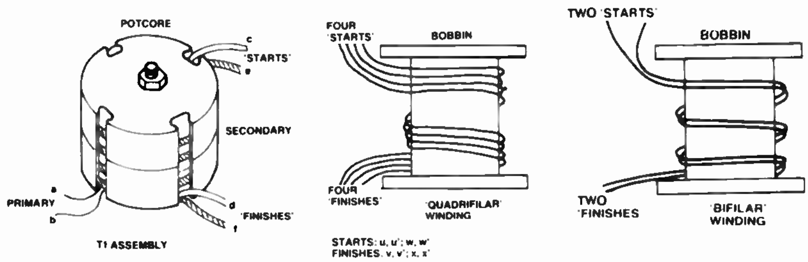
TRANSFORMER WINDING DETAILS

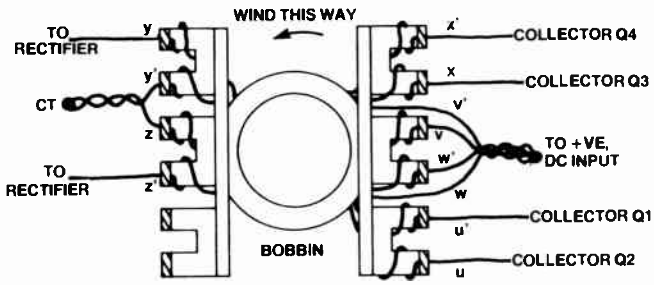
T1

Core	FX2242, 36 mm dia. potcore, two halves with single section bobbin.
Wire	enamelled copper wire, about one metre of 1 mm diameter wire and about 1.5 metres of 0.4 mm.
Primary (a b)	20 turns of 0.4 mm wire wound evenly on the bobbin. Cover with a layer of insulation tape. Note that b is the start.
Secondary (c-d, e-f)	4 turns, bifilar wound (see diagram), of 1 mm wire spread over bobbin. Cover with a layer of insulation tape. Bring out the starts at one end of the bobbin, finishes at the other. Starts are c and e.
Notes	The above refers to T1 wound for a **12 V (nominal) supply.** On a **24 V (nominal) supply,** the primary (a-b) and secondary (c-d, e-f) should be doubled (i.e: 40 turns and 8 turns, bifilar, respectively). On a **32 V (nominal) supply,** the primary turns (a-b) should be increased to 50, the secondary (c-d, e-f) to 10 turns, bifilar.
Winding order	Wind the primary (a-b) **first.**

T2

Core	Philips EC-core assembly, as per the parts list.
Wire	enamelled copper wire, 0.4 mm dia. — length to suit application, and about two metres of 1 mm dia.
Primaries (u-v, u'-v', w-x, w'-x')	use 1 mm wire wound **quadrifilar** (see diagram), two volts per turn. i.e: for 12 V (nominal) supply — six turns; for **24 V (nominal) supply** — 12 turns; for **32 V (nominal) supply** — 16 turns.
Secondary (y-z, y'-z')	0.4 mm or 1 mm wire (to suit current) bifilar wound, two volts per turn. i.e: for **±40 V supply rails** — 20 turns; for **±50 V rails** — 25 turns, etc. See table of suggested outputs.
Winding order	Wind the quadrifilar primaries (u-v, u'-v', w-x, w'-x') **first.** Cover with two layers of insulation tape.





WINDING T2

TABLE OF SUGGESTED OUTPUTS

OUTPUT	T2 SECONDARY	RECTIFIER	APPLICATION
±50 V	25 turns bifilar	2 x fullwave	Power 2 x ETI-477 (Series 5000) or 1 x ETI-498 module.
±40 V	20 turns bifilar	2 x fullwave	Power 2 x ETI-480 (50 W or 100 W versions) or 2 x ETI-470 modules.
±15 V	8 turns bifilar (Note 1)	2 x fullwave	Auxiliary secondary to power a preamp. e.g: ETI-481M guitar mixer/preamp or ETI-498 PA preamp.
1400 V	700 turns (Note 2)	voltage doubler (Note 2)	Power ETI-565 laser
12 V (from 24 V or higher dc input)	7 turns (Note 1)	fullwave bridge	Power 12 Vdc equipment from 24 V or higher dc input.

Note 1. This takes rectifier voltage drop into account.
Note 2. The secondary can drive the ETI-565 laser power supply directly, replacing the original power transformer.

transistors should be quite low. They should only get warm to the touch. (But don't touch them while the inverter's operating — you can get a 'belt'!) You will notice that the operating pitch of the inverter drops when a load is applied.

If all tests out well, hook up the inverter to the unit it is intended to power and try it out. With audio amp modules the 6 kHz oscillation frequency of the inverter should not be audible in the loudspeakers or should be a very, very long way 'down'. Where it is used in conjunction with a sensitive preamp, earthing loops and supply line induction into input leads and earths can cause 'break through' of the 6 kHz oscillation frequency. Take care with the routing of supply leads from the inverter. Keep them away from input leads and make sure the audio equipment is earthed at a single point, either at the power supply chassis or at the dc input common (negative).

Avoid radiation from the inverter inducing 6 kHz breakthrough into any equipment by keeping the inverter physically separate from such equipment. Both transformers have very little external field, but the wiring of the inverter carries considerable switching currents and can induce small, but significant, signals into sensitive audio or RF input leads. If you intend mounting it inside the equipment case, build it in a separate, shielded (i.e: all metal) enclosure and mount that inside the equipment but away from input circuitry.

Performance

This project was hastily built up one Saturday, from a 'lash-up' prototype, to power the ETI-498/499 PA amp for a function the following day! It worked first off. We didn't even have to reverse the primary of T1 to get the feedback phase correct!

Performance was faultless — for both the inverter and the PA.

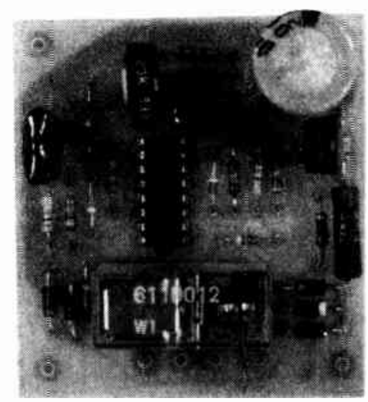
Occasionally you win some! Breakthrough of the 6 kHz oscillation frequency was only evident with the low level mic gain and the volume pot on the PA amp set full up. Two secondaries were wound on the inverter output transformer (T2), one to provide the power amp with ±50 V and the other to provide the preamp with ±15 V. The breakthrough was subsequently found to be primarily due to a double-earthing problem.

The PA amp ran from a 12 V battery faultlessly for an all-day event. The inverter ran cool — admittedly, it was midwinter and quite cold, but efficiency approaches or exceeds 90% and in this sort of application, average dissipation is very low. ●

Project 494

Signal powered loudspeaker protector

David Tilbrook



This unit affords both dc and over-power protection of loudspeakers or loudspeaker systems rated at up to 1500 watts! The unit requires no power supply and has no discernible audible effect on sound quality making it suitable for both hi-fi and sound reinforcement applications.

THE ETI-455 loudspeaker protector has proved to be a very popular project. It was published in March 1980 and since then we have had numerous phone calls from readers with stories of how the unit had saved their loudspeakers from almost certain disaster. Usually the power amplifier had gone faulty and applied the full dc supply rail to the loudspeaker terminals. Without the loudspeaker protector in circuit the result would be at least an open circuit bass driver and probably worse. The protector prevents this by monitoring the loudspeaker lines for the presence of dc, opening a set of relay contacts if this occurs, disconnecting the loudspeaker from the faulty amplifier.

The ETI-455 works well but requires its own power supply, either batteries or a small regulated mains supply. Another disadvantage results from the type of filter used to distinguish between dc and the audio signal. This was a conventional passive filter set to around 10 Hz. The problem is that it is still possible with very large amplifiers to trigger the protector on low frequency audio content. So the circuit, although perfectly satisfactory for its quoted maximum power of around 100 watts or so, is unsuitable for very high powered amplifiers.

We decided to overcome these limitations in this new loudspeaker protector, the ETI-494. Since the old one was published we have had numerous requests for a circuit that could be mounted inside a loudspeaker enclosure. These requests have come largely from the sound reinforcement industry although the unit would obviously be applicable to all loudspeakers. The protector would not be able to be powered from a mains supply since it is not always desirable or even possible to run mains to the loudspeakers. This is particularly true in a sound reinforcement or public address system. Similarly, batteries are unsuitable since access would have to be provided to facilitate testing and changing them when required. In addition, when we first published the ETI-499 MOSFET PA Module (re-printed on page 100), we promised to follow up with a loudspeaker protector. This is it. The solution, used in this project, is to power the unit from the audio signal itself.

This is done in this case by placing a fullwave rectifier across the speaker lines and charging a 1000u capacitor through a 47 ohm resistor. The worst possible load presented to the speaker line is therefore 47 ohms and this is only while charging the capacitor and for signal voltages in excess of 12 V. This ensures that the unit has no discernible effect on audio quality but makes ▶

possible a truly 'set-and-forget' loudspeaker protector that can be mounted inside the loudspeaker enclosure if desired.

The ETI-494 tests for both dc and over-power, which can be adjusted by a preset on the board to suit a particular loudspeaker or application. The circuit also uses a new filter design with an almost 'brick wall' response enabling it to be connected to very high power amps. This is discussed in more detail in the 'How it Works' section.

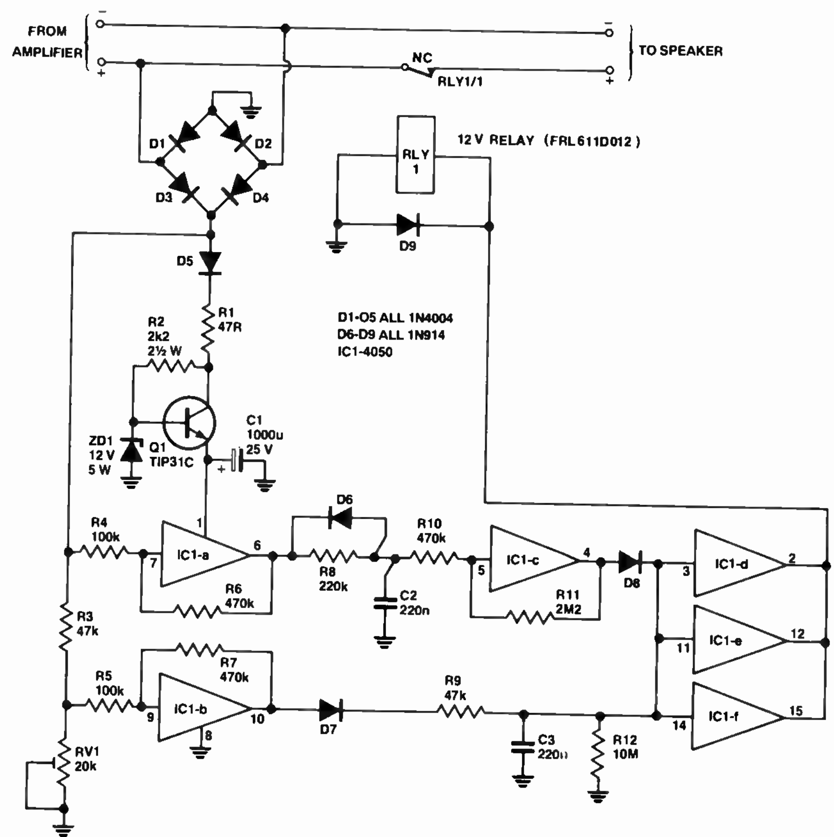
The maximum power that can be applied to the unit is determined by the type of regulator transistor (Q1) used. We have specified a TIP31C for this device which has a 100 V collector-to-emitter breakdown voltage. Since the emitter is at 12 V, the maximum voltage that can be applied to the unit is 112 V. This is equivalent to an amp capable of

HOW IT WORKS — ETI 494

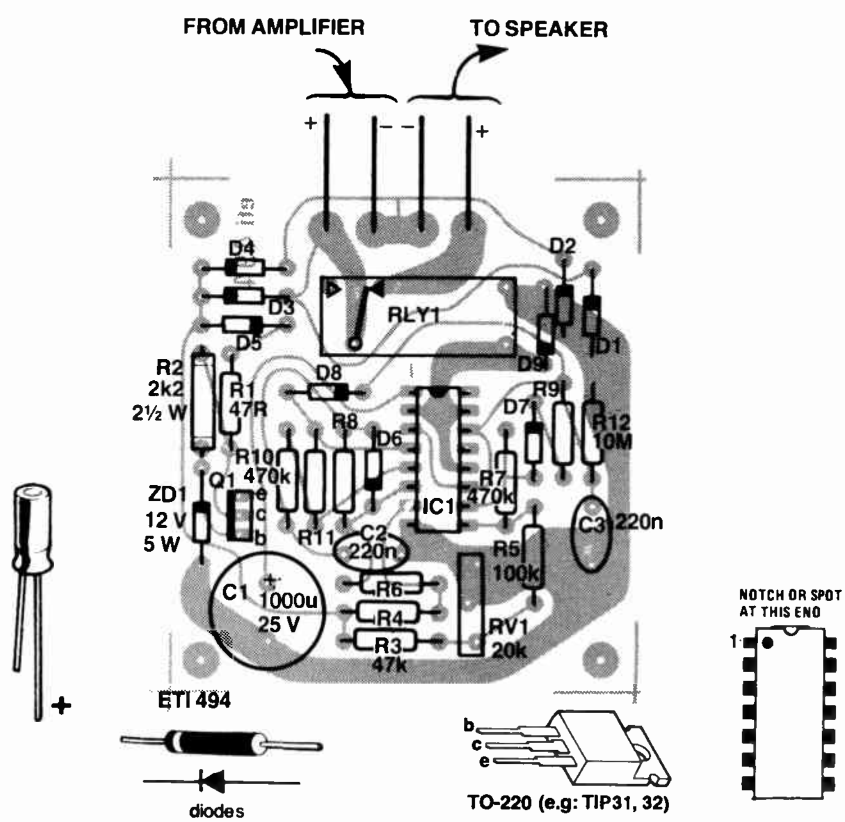
The signal from the power amp is rectified by the fullwave rectifier formed by D1-D4. The output of this is fed through a 12 V regulator circuit formed by Q1 and its associated resistors and zener diode, and charges the electrolytic capacitor, C1. The output of the rectifier is also fed to the input of the dc sense and over-power detection circuitry.

IC1 gates a and c form the dc filter. Resistors R4 and R6 form a Schmitt trigger with a small deadband. When the signal goes above the trigger voltage the output of the trigger swings hard to the positive supply rail of the IC, charging C2 through the 220k resistor, R8. Resistors R10 and R11 with gate c form a second Schmitt trigger monitoring the voltage across C2. If the voltage across C2 reaches the trigger voltage of this second Schmitt, gates d, e and f are activated, pulling in the relay contacts and disconnecting the loudspeaker. It takes about 100 ms to charge C2 through R8, and on normal audio content the output of gate 'a' will be driven low before this occurs, discharging C2 rapidly through D6. Only signals which do not have a zero crossing for longer than 100 ms will trigger the protector.

The over-power protector consists simply of a voltage divider feeding a third Schmitt trigger. Whenever the signal voltage exceeds the trigger voltage the output of gate 'b' is driven high and C3 starts to charge. If this condition persists for long enough the output gates are turned on and the relay pulls in. Note that both the dc and over-power sense circuits charge C3 when activated. The circuits are decoupled from this capacitor by diodes so that, once charged, C3 can only be discharged by the parallel resistor R12 (the effect of the input impedance of the gates is negligible). Since it takes about one second to discharge this capacitor, the relay will hold in for this time. The protector therefore reconnects the loudspeaker approximately one second after the fault condition has been removed.



The pcb pattern is on page 74.



PARTS LIST — ETI-494

Resistors .... all ½ W, 5% unless notec
R1 .... 47R
R2 .... 2k2, 2½ W
R3, R9 .... 47k
R4, R5 .... 100k
R6, 7, 10 .... 470k
R8 .... 220k
R11 .... 2M2
R12 .... 10M
RV1 .... 20k min. trimpot

Capacitors
C1 .... 1000u/25 V RB electro.
C2, C3 .... 220n greencap

Semiconductors
D1-D5 .... 1N4004, EM404
D6-D9 .... 1N914, 1N4148
IC1 4050
Q1 .... TIP31C
ZD1 .... 12 V, 5 W zener

Miscellaneous
ETI-494 pc board; RL1 — Fjuitsu FRL611D012, 12 volt SPDT 10 A contacts or similar relay (pc mount type).

Price Estimate
$20-$24

supplying approximately 784 watts into an 8 ohm load or 1568 watts into a 4 ohm load. If the amplifier to be used is capable of powers greater than these the regulator transistor should be substituted for a device with a higher V_{ceo} rating. The relay pulls around 40 mA when operated, so power dissipation in the regulator transistor will be around 4 watts when dropping 100 volts. Although this is not a particularly high dissipation it is high enough to lie outside the SOAR rating of many high voltage transistors, so be careful when choosing an alternate regulator transistor.

Construction

Construction is straightforward since all of the components are mounted on the pc board. The usual precautions should be taken to ensure that all polarised components have been mounted with the correct orientation. The IC used is a CMOS type and is therefore static sensitive. Solder this last and preferably using an earthed soldering iron. It is a wise precaution to discharge yourself before handling the device by first touching an earthed metal appliance. For a more detailed description of precautions when handling CMOS refer to our article 'Electrostatic discharge — nemesis of electronic systems' in the June edition, 1981.

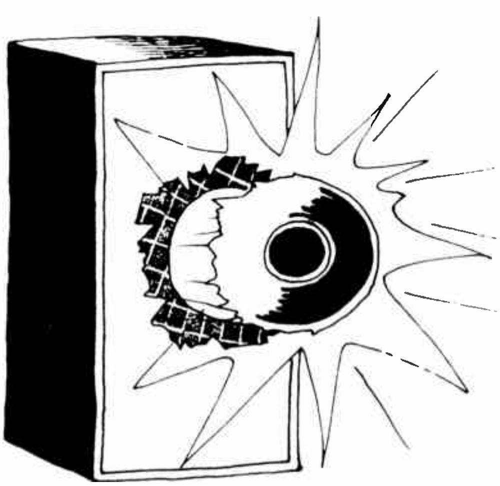
It is a wise precaution to space the 2.5 W resistor, R2, off the pc board slightly. In the case of a high powered loudspeaker going faulty with dc this component will get quite hot and spacing improves ventilation around the component and prevents the possibility of charring the pc board. If you can't obtain a 2.5 watt type (e.g. Philips PR52), then a 5 W type may be substituted.

Before mounting the unit check operation by connecting around 20 V dc across the speaker input terminals on the pc board. The relay should cut in after about one tenth of a second. If the protector passes this test connect the speaker wiring. If the preset is turned fully down (turn it anticlockwise when viewing the board with the components on top and the relay to the right) the relay will cut in when the power exceeds around 20 watts for an extended period. The protector allows transients to the full supply rail to pass but will prevent a continuous 20 W from being applied to the loudspeaker. To increase this, turn the preset clockwise until the desired response is achieved.

Performance

We tested the loudspeaker protector for its effect on audio performance as well as its reliability. A variety of power amps were used to ensure that the load represented by the protector would not affect audio performance. Even a very low power amplifier, with a comparatively small damping factor (high output impedance) could drive the unit with no degradation to the sound quality. During every test the protector worked well and cut in at the correct time to prevent damage to the loudspeakers.

NOTE. *Some amplifiers are unstable when driven into an open circuit. This is particularly true of valve power amplifiers some of which destroy themselves the moment the speaker is disconnected. Loudspeaker protectors are however, not usually required for use with valve power amps since the possibility of dc on the speaker lines is remote, but over-power protection may be required.* ●



OPERATING NOTES — PROJECT 494

A loudspeaker protector such as this is basically meant for a *once-only* operation. It is not meant for successive cut-in, drop-out operation, particularly in the case of a dc fault. This project should not be used if you're trying to trace an intermittent dc fault. If the relay operates, it's time to switch off the amplifier, disconnect the load and find the fault — preferably using a dummy load. For amplifiers having supply rails in excess of about 50 V, it is advisable to add dc-quenching across the relay contacts. A series RC network is all that's necessary. Use a 100 ohm, 1 W resistor and a 630 V rated capacitor of about 100 n or greater value.

For really high power amplifiers having supply rails of 100 V or more, it is probably best to use two relays with their coils connected in parallel and their contacts (with de-quenching) connected in series.

Setting up an outdoor PA system

Geoff Nicholls

This article covers the background theory and practical techniques you need to know to successfully set up and operate an outdoor PA system — with particular reference to the ETI-498/499 150 W PA project.

SETTING UP an outdoor public address system correctly can mean the difference between effective audience communication and totally indifferent results. And guess whose can gets kicked when the system doesn't work as expected?

Before starting out, it is wise to know a little background theory to the various parts of the system. A little theory is introduced at each stage, to provide the appropriate background, so let's start off with sound propagation.

Sound propagation

Sound propagates from a vibrating source in the form of longitudinal mechanical waves, which oscillate the particles in the medium along an axis in the direction of sound propagation.

The velocity of sound in still air is temperature dependent, and is approximated by the formula:

$$v = 20\sqrt{273 + T}$$

where v = velocity of propagation in m/s
T = air temperature in C

Logic would suggest that the sound pressure level should fall off with increasing distance from the sound source by an inverse square law, because of the expanding area of the sound wavefront. In fact, additional losses are present due to dissipation of the sound energy by mechanisms too complex to discuss in this article.

These loss processes are frequency dependent, and lead to increasing attenuation of high frequencies with distance, but fortunately they can be ignored for speech frequencies up to distances of about 100 m. The inverse square law is therefore adequate for *general* outdoor PA calculations. The decibel SPL formula is given by:

$$\text{dB SPL(X)} = \text{dB SPL(R)} + 20 \log \frac{D_R}{D_X}$$

where dB SPL(X) is the SPL in decibels at point X
dB SPL(R) is the SPL in decibels at the rise to refraction or bending of the sound reference point R
D_R is the reference distance from the sound source
D_X is the reference to point C from the sound source

Temperature gradients in the air give rise to refraction or bending of the sound from its original direction. When the sound is refracted it bends towards the coolest region because the sound travels faster through the warmer region. This is analagous to a bimetallic strip which bends because of differential expansion.

Most outdoor venues are warmest near the ground during the day, and so the sound tends to bend upwards. One notable exception is over a large water surface, which during the day tends to be cooler than the air, and so causes sound to bend down towards the surface. This can cause sound to carry long distances over water.

Windy conditions cause sound to be refracted because of gradients in wind speed in a similar manner to temperature gradients. In general winds are slower near the ground, and this causes an upward bend when the sound is into the wind and a downward bend when the sound is with the wind. Transverse winds have little effect on refraction, although irregularities in all winds cause scattering of the sound.

The ground will reflect a certain amount of sound and absorb the rest. The reflected part can be utilised to reinforce the direct sound and increase the overall level by up to 3 dB, depending on the ground surface.

Setting up a PA

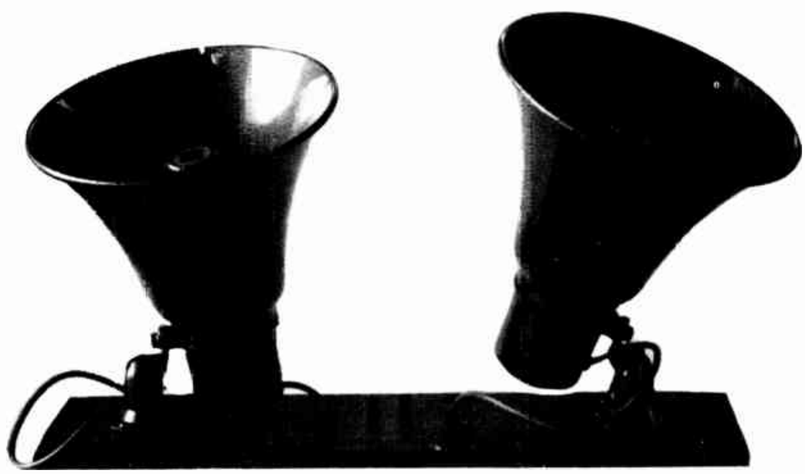
A PA system will be satisfactory if all the listeners can understand what is being announced without concentrated effort. The following criteria will generally allow this:

- The SPL at the listener is below the tolerable limit.
- The articulation of consonants is acceptable.
- The PA SPL at the listener exceeds the ambient noise SPL by at least 10 dB SPL.
- The sound at the listener does not contain annoying echo.
- The system is not 'howling'.

It is obvious that no one will remain in an area where the sound is so loud it is uncomfortable. Certain outdoor events involving high-powered motors (such as drag boat racing) can have an ambient level over 120 dB SPL, but only for a short period of time. It is impractical to have the PA loud enough to override such ambient levels. ▶

The articulation of consonants depends primarily on the voice characteristic of the announcer. Successful announcers usually have good consonant articulation. It is possible to improve this factor by using a shaped filter response, such as the speech filter employed in the ETI-498.

The public address system sound must obviously be perceived as louder than the ambient noise, or it will be drowned out. An increase of SPL by 10 dB subjectively sounds twice as loud, and for outdoor set-ups forms a good signal-to-noise ratio to aim for at the limit of the PA coverage area.



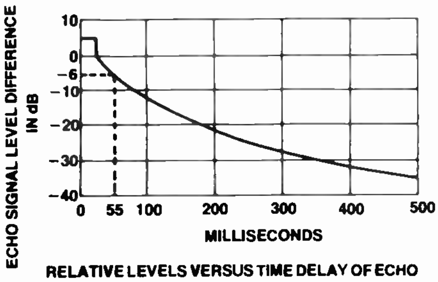


Figure 1. **Echo level versus time delay for 10% audience annoyance, produced by Doak and Bolt (see References at end of article).**

An echo will arise when there are unequal distances between the listener and two (or more) loudspeakers being driven by the same signal. An investigation by Doak and Bolt resulted in the compiling of a chart which allows us to estimate when an echo will become annoying to 10% of the audience. The chart plots the difference in SPL between the main signal and the echo against the time delay of the echo. An echo can also arise due to reflection off a hillside or building.

Acoustic feedback

Nearly everyone will have experienced the howling that occurs when a microphone is placed too near a loudspeaker it is driving. This phenomena is acoustic feedback and arises when the total gain of a sound system from the microphone through the amplifier to the speakers and back to the microphone exceeds unity. This usually occurs at a single frequency or a few dominant frequencies, because of peaks in the system response.

The problem of acoustic feedback is complicated when public address systems are used indoors because of the room shape which gives rise to many resonances. Complex equalisers are employed to smooth out the overall response and therefore allow the sound level to be increased before feedback occurs. Indoor public address techniques will be the subject of a future article in ETI.

Acoustic feedback is less of a problem in open spaces since there is usually only direct sound present — little or no reverberation from reflecting surfaces. Correct system layout should avoid feedback problems.

Speakers

The horn loudspeaker is by far the best type for outdoor use. Horns can be made weatherproof and have an efficiency of better than 20% compared to a few per cent for ordinary speakers. This allows an amplifier of lower power to be used, with consequent savings in electricity, physical size and weight. Horn speakers are available with inbuilt 100 V line transformers, usually with several taps to select different power levels. This allows some speakers to be placed closer to the audience and their output reduced to compensate without affecting other speakers on the 100 V line.

Horns are intrinsically limited in their frequency response, and their efficiency is inversely proportional to their bandwidth. PA horns are designed to operate over the voice band at maximum efficiency. The horn itself is esentially an impedance transforming device which increases the acoustic loading on the driving diaphragm to allow better matching to the air. The shape of the horn is usually based on the exponential function and provides a cross sectional area which is dependent on distance along the horn by the formula:

$$A = A_0 E^{mx}$$

where A = area of cross-section at distance 'x' from throat
A_0 = throat area
E = Naperian base (2.718128)
m = 'flaring' constant.

The horn may be straight, as shown in Figure 2, or folded, as shown in Figure 3. The folded horn is physically smaller and is the most common type in low cost PA systems. Folding the horn reduces the efficiency slightly but increases the coverage or dispersion, which is usually an advantage. The straight horn has a long 'throw' and is useful for narrow sound coverage at greater distances but is more cumbersome, especially when you are 8 m up a ladder! ▶

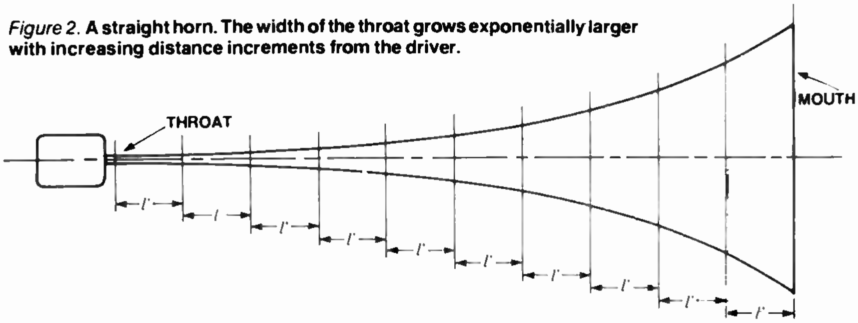


Figure 2. **A straight horn. The width of the throat grows exponentially larger with increasing distance increments from the driver.**

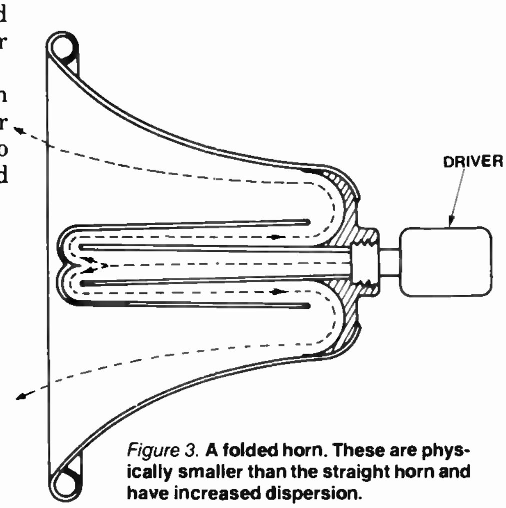


Figure 3. **A folded horn. These are physically smaller than the straight horn and have increased dispersion.**

Microphones

A good microphone forms the heart of a good PA system and vice versa. The most suitable type for outdoor use is the unidirectional, low impedance dynamic microphone. This type is rugged and can withstand the abuse an outdoor setup will inflict. The directional characteristic is extremely important when the announcer is within the range of the speakers, and can make a big difference to the sound level attainable before feedback occurs. The low impedance microphone can also be used with a longer cable than the high impedance types, and will not pick up as much interference.

System layout

The overall performance of an outdoor PA is dependent on the location and type of loudspeakers and microphones. The placement of the amplifier and microphone is usually dictated by the facilities at the venue, i.e: commentators are found in control towers, dias' etc., which are fixed structures. If mains power is needed, the amplifier position is limited by the length of available extension cords.

WARNING! *Check the integrity of extension cords before allowing anyone to use the system.*

Assuming that the amplifier location is determined, the next job is to arrange the horn speakers to cover the listening area.

The simplest layout is the centralised cluster, where all speakers are together. This eliminates any time delay effects and simplifies wiring. To be effective, the cluster must distribute the sound so that nearby listeners are not deafened and distant listeners are able to hear the PA. This will require a high mount and possibly the use of straight horns to reach the furthest listeners.

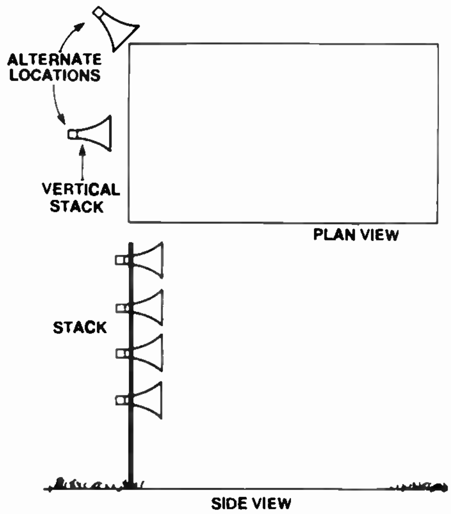


Figure 4. **The centralised cluster of speakers. Note the speakers should point slightly downwards.**

PERCEIVED SOUND PRESSURE LEVELS

The human sense of hearing is stimulated by small variations in the air pressure at the ear. In order to perceive a sound the local pressure variations must conform to a limited range of frequencies and a minimum amplitude of vibration.

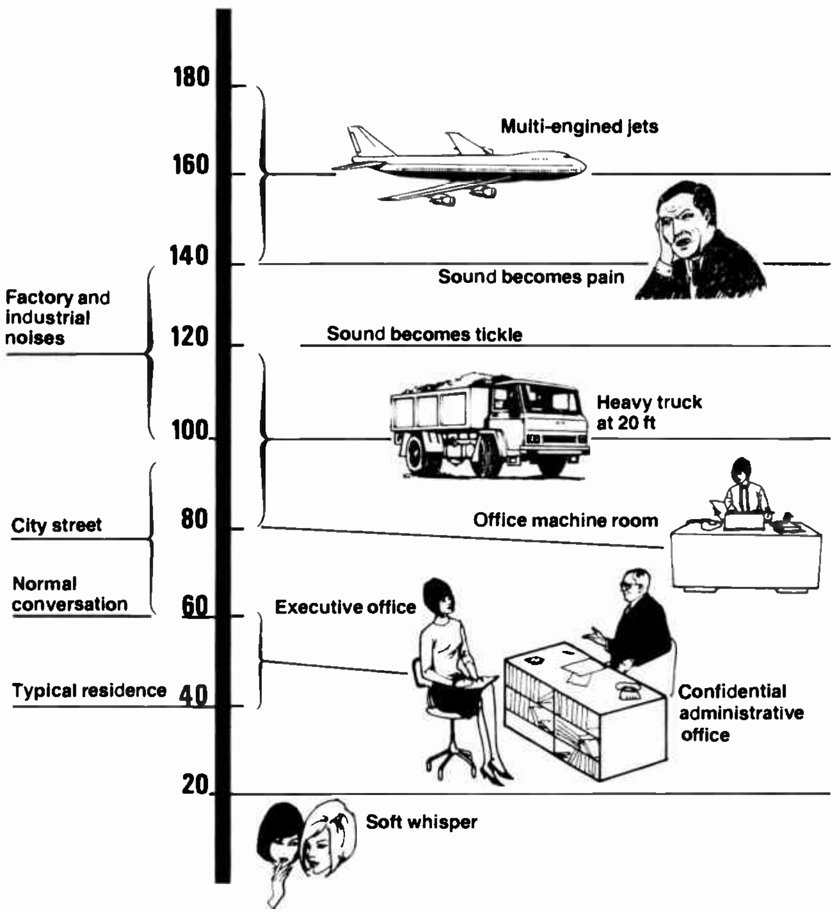
The amplitude frequency range is usually quoted as 20 Hz to 20 kHz, although the precise range depends on individual characteristics, particularly age and when the ears were last cleaned out! The minimum amplitude of pressure variations required for the perception of a 1 kHz sound in a young person's ear is about 20 uN/m^2 (or 20 uPa). Since this level represents a lower limit it is used as a reference for sound pressure levels and is given the value 0 dB SPL.

The largest sound pressure variation that can be tolerated without pain is about 100 N/m^2, which can be expressed as:

$$20\log_{10}\frac{100}{20\times10^{-6}}\,\mathrm{dB}$$

or about 134 dB SPL.

It is interesting to compare this to the variation in air pressure that often accompanies the approach of a storm, when a drop of 10 000 N/m^2 can occur in a few hours! Such slow changes in pressure cause our ears no distress because the inner ear is vented to the atmosphere through the Eustachian tube, which equalises the pressure on the eardrum.



- For a sound to be perceptibly louder or softer, it must be changed by three decibels.
- A noise twice as loud or half as loud is a change of ten decibels.
- A reduction in noise of a few decibels in the low noise region (administrative office) is not significant. The same change at high sound levels (office machine room) is significant.

Venues suitable for a centralised cluster are ovals and parks where the length-to-width ratio is less than about 2 to 1. The speakers should be positioned away from the commentary along the short end position and should point slightly downwards from the horizontal in a vertical stack. If they must be sited along the long side of a rectangular area then an additional vertical stack should be added and splayed about 75° apart.

It may be necessary to use long-throw straight horns to reach the furthest areas, these should be mounted on the top of the stack.

Additional 'side fill' horns are used to service listeners behind the main coverage area.

The vertical stacking results in a horizontal 'fan' of sound and reduces wasted acoustic energy upwards and downwards.

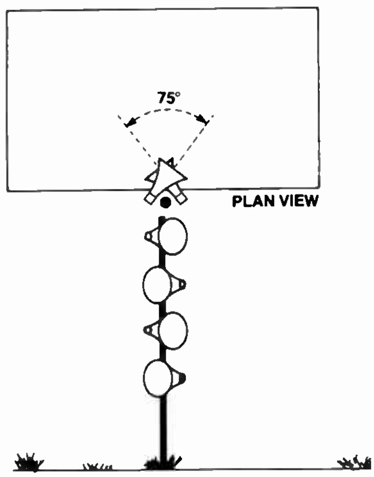


Figure 5. **Mounting a cluster along the side of the area to be covered requires the horn throats to be angled at about 75° and they must overlap.**

Some venues are not suited to a centralised cluster. For instance, riverside events tend to concentrate the crowd in a thin rectangle along the bank. Such cases require multiple loud speakers, and care must be used in planning the sound sources to avoid annoying echo effects.

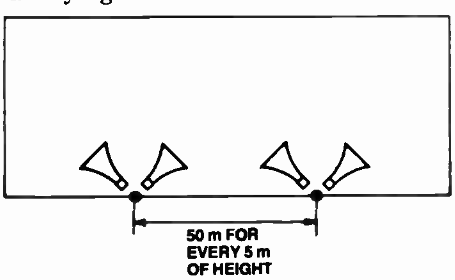


Figure 6. **Horn positioning for covering a long, narrow area.**

The best results are obtained by using a large number of speakers evenly spaced along the long axis operating at fairly low levels, but this is expensive. The higher the speakers can be mounted the further apart they may be spaced. As a rule of thumb, each 5 m of height allows a speaker spacing of 50 m.

Wiring

I have found that figure-8 lighting cable is well suited for wiring 100 V loudspeaker systems. Although somewhat overrated for audio power levels, the cable is durable and cheap, and is easy to strip. Many connections are made with the strip-twist-tape technique atop ladders, and a light fiddly cable is a hassle to use. It is convenient to have fixed lengths of pre-cut cable to avoid constantly breaking and rejoining a single cable.

I have mounted pairs of horn speakers on wooden battens with spring connectors mounted on them. Holes in the battens allow the speakers to be tied to various poles or trees with rope or spare figure-8 cable. The spring connectors prevent progressive shortening of the cable from the speaker due to repeated cutting and stripping of the end of the cable.

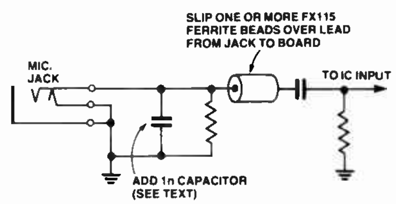
All budding sound or PA engineers should get a copy of what is alm Bible' — "Sound System Enginee by Don and Carolyn Davis, publish Howard Sams (USA). This is curre available through ETI Book Sales.

RF INTERFERENCE SUPPRESSION — TECHNIQUES

Interference pickup on speaker leads may be cut by winding part of the lead, nearest the amplifier terminals, on a ferrite rod — available at many parts suppliers.

Public address amplifier systems may be prone to RF interference from a variety of sources — and the source may be unknown or hard to track down. Sometimes the source is well known but impossible to eliminate — a nearby AM broadcast transmitter, for example. CB or marine transceivers in the vicinity of a PA system are notorious sources of annoying intermittent interference. But it's not the fault of the 'offending' transmission; the characteristics of modern solid state devices are the major culprits.

A number of techniques can be employed to protect a PA amp from interference. As it will depend on the individual application, we leave it to the constructor how much, or how little, interference protection to incorporate.



Adding RF suppression to the low level inputs.

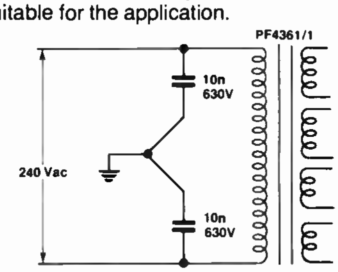
THE 'FRONT END'

The low-level input stages are particularly prone to RF pick-up. There are two components you can add quite simply to protect each low-level input. Firstly, a ferrite bead, such as the commonly available FX115 type, can be slipped over the lead running between the jack socket and the pc board. Secondly, a 1n 'greencap' capacitor can be soldered directly across the input jack socket terminals. If the leads of this capacitor are cut to a length of 25 mm, the capacitor will have a broad series resonance around 27 MHz, greatly aiding suppression of CB and marine radio interference. These components may be added to both MIC 1 and MIC 2 inputs.

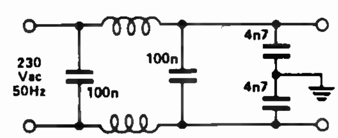
For the AUX input, a greencap with a value between 2n7 and 10n should be used.

THE 'BACK END'

Long runs of loudspeaker cable have the annoying tendency to act as antennas. 'Choking off' the RF once it gets on a cable run can be problematical. One of the most effective methods is to wind that part of the cable nearest the amplifier speaker terminals on a ferrite rod — such as is used for transistor radio loopstick antennas. This makes a very good broadband RF choke, but it *must* be installed as close to the amplifier output terminals as possible. There's nothing critical about it, but the ferrite rod should be at least 100 mm long, preferably longer. Ferrite rod in 200 mm lengths, 9.5 in diameter, is commonly available and quite suitable for the application.



Adding interference suppression on the mains input. The value of each capacitor may be anything between about 4n7 and 100n. They should be rated at 630 V or 1 kV.



Circuit of a mains input filter. The chokes should have an inductance between 5 mH and 50 mH and be capable of carrying up to 2 A. The capacitors may be greencaps or ceramic types rated at 630 V or 1 kV.

MAINS-BORNE INTERFERENCE

Apart from radio interference coupled into mains cables, light dimmers, motor controllers and switch contacts on mains equipment connected to the same line as the PA amp can cause a variety of clicks, pops and buzzes to be heard on the system. Proprietary mains filters can be obtained and often prove very effective. Alternatively, you can build a filter into the PA amp.

One of the simplest suppression methods is to connect a 10n/630 V greencap or ceramic capacitor from each side of the mains transformer primary to the chassis — at the same point. Three-pin mains plugs can be obtained with capacitors installed and may be quite effective. A 'pi' filter can be built up, as shown in the accompanying circuit, and installed in the amp's chassis.

Principles and problems in loudspeaker design

If you ever wanted to know, in a practical fashion, what is involved in the design of a loudspeaker system, then here's a down-to-earth insight — ending in a real design.

David Tilbrook

MORE MONEY can be saved by the construction of a pair of loudspeakers than any other single component of the hi-fi system. Unfortunately, they are also the most important hi-fi component! Unless the turntable or amplifier is particularly poor, the loudspeaker will undoubtedly determine the overall sound of the system. For this reason it is disappointing there are so few really good kit loudspeakers.

The fact that a "correct" loudspeaker doesn't exist is to be expected, since the principles of loudspeaker operation are enormously complex. Every loudspeaker model makes certain assumptions to simplify the mathematics and to make the model manageable. If these assumptions are overdone the model rapidly loses relevance, becoming incapable of making worthwhile predictions about the *real* loudspeaker. While it is true that a detailed understanding of loudspeaker operation is not necessary to enable a kit loudspeaker to be built, some understanding will enable the optimum to be obtained from the loudspeaker and is essential for those brave experimenters who would like to get involved in modifying the loudspeaker drivers. When designing a loudspeaker it is necessary to understand the mechanism of operation of the drivers. Only then can the best choice of driver, enclosure type and crossover be established. Although loudspeaker design is as much an art as it is a science, the loudspeaker that has been created with a motley assortment of drivers placed in a box with some "general purpose crossover" is more likely to sound like a dropped saucepan than a good loudspeaker!

The most common loudspeaker consists of several moving-coil direct-radiating drivers mounted in an enclosure. These cover different frequency bands within the audio spectrum. A crossover is used to separate these frequency bands and feed them to the appropriate driver.

If the drivers used had perfectly flat frequency responses, were constant eight ohm loads with infinite power handling, and the crossovers represented lossless transfer characteristics with no untoward interactions with the drivers, and if nature did not object to the reproduction of low frequencies in confined volumes (i.e. if the speed of sound was one tenth the speed it is) loudspeaker design would be a simple matter.

Most of these problems can be summarised with one word . . . inertia. This is that property of nature whereby things resist change. We can't really complain too strongly about inertia since it is responsible for much of the order that exists in the universe. Nevertheless, in loudspeaker design it causes real problems. The signal voltage from the power amp, the magnetic field

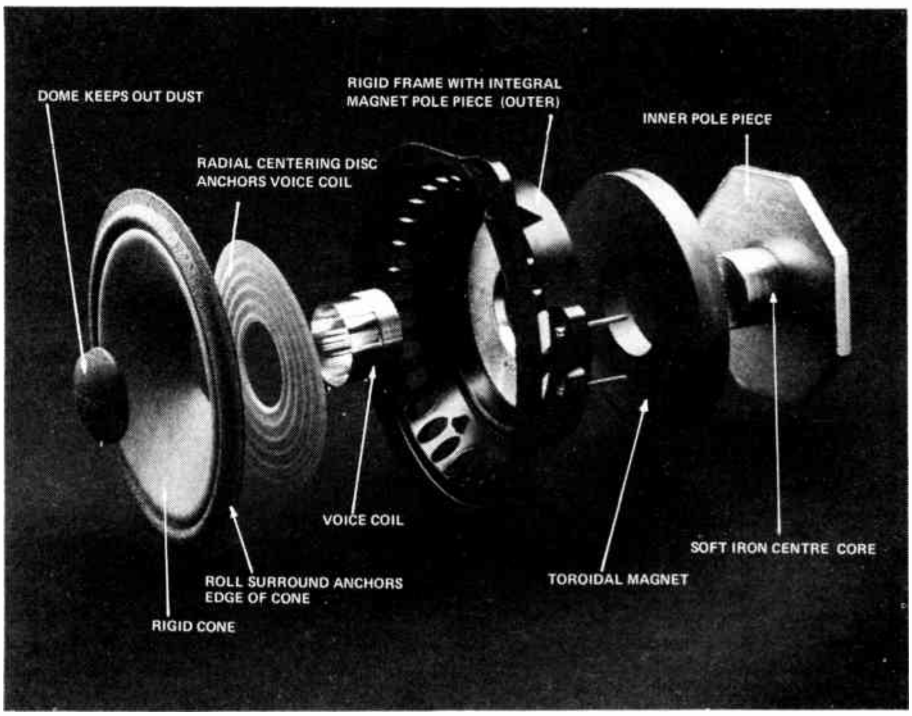
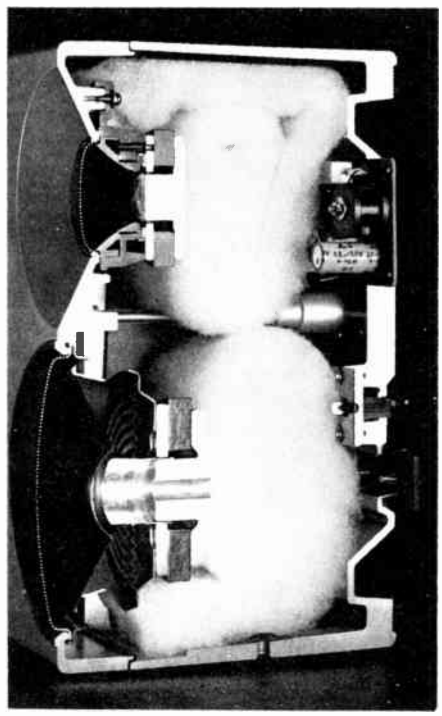


Figure 1. **An exploded view of a moving-coil loudspeaker showing the various components in its construction. Compare this with the cutaway view of a speaker at right (Pic: courtesy Bose).**



Cutaway view of a speaker unit showing internal construction (Pic: National).

around the voice coil, the movement of the coil and loudspeaker cone, all resist change. Since the objective of loudspeaker design is to convert an electrical signal into its exact acoustic counterpart, these sources of inertia cause errors resulting in distortion. The effects of inertia don't stop at just slowing down the system. The resistance to change of motion by the cone for example results in some parts of the cone moving before others. Sound waves start to travel along the cone itself, travelling radially out from the voice coil. Depending on the nature of the flexible surround between the cone and the chassis this sound wave will be partially reflected back down the cone. This causes constructive and destructive interference with the original sound wave propagating up the cone resulting in colouration. Clearly, this is not something the home constructor can do much about, since it depends on the manufacture of the particular driver concerned, but it indicates the sorts of problems that will be encountered.

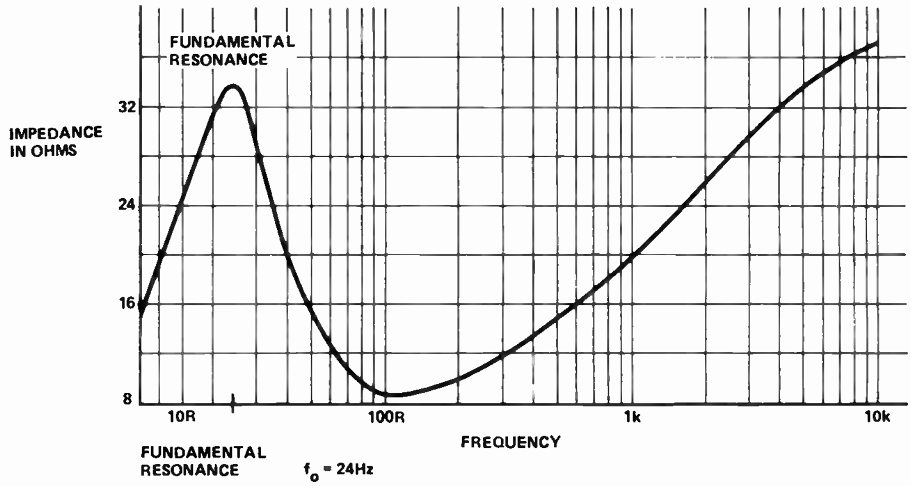


Figure 2. **Typical impedance versus frequency characteristics of a moving-coil loudspeaker**

The moving-coil direct-radiating speaker

The vast majority of drivers used in loudspeakers are of the moving coil type and as such all operate in a very similar way. Figure 1 shows a typical moving coil loudspeaker. Signal voltages from the power amp give rise to signal currents that flow through the voice coil. This is simply a coil of wire wound on a hollow circular former. In normal 8 ohm drivers the dc resistance of the voice coil is around 8 ohms, but the driver will only represent this resistance to the power amp at one specific frequency, the actual impedance of the driver varying widely as the frequency is varied (see Figure 2). A given signal voltage level will therefore produce different signal currents for different frequencies. The signal current causes a varying magnetic field to be produced around the voice coil. This field interacts with an intense magnetic field from the drivers' pole piece and magnet assembly causing a force to be exerted on the voice coil and loudspeaker cone.

As the cone moves it will compress or rarify the air immediately in front of it, creating an area of either increased or decreased pressure. These pressure variations comprise a sound wave that travels from the driver to our ears.

The electrical impedance of the driver is caused by several phenomena each one dominating in a specific frequency band. One of the most significant mechanisms is the back EMF (EMF stands for electromotive force, i.e: voltage) of the driver. The movement of the voice coil in the magnetic field acts as a generator causing a current to flow in the voice coil. This current is of opposite polarity to the applied signal current (another natural application of the principle of inertia) causing decreased current flow in the voice coil for a given signal voltage. This is seen by the amplifier as an increase in the drivers' impedance.

EMF is given by the simple equation:

EQUATION 1

$e = Blv$ where 'e' is the back EMF in volts
'B' is the magnetic flux
'l' is the length of wire in the magnetic field
and 'v' is the velocity of the cone

Since the magnetic flux and the length of wire in the magnetic field can be considered as constants, the equation shows that the amount of EMF generated is directly proportional to the velocity of the cone.

So the electrical impedance is a secondary phenomenon, is certainly not constant, and does not relate directly to the radiated acoustic power. The amount of back EMF will be determined by the velocity of the cone, and this is a function of nearly every major parameter of the loudspeaker box.

The force exerted by the voice coil on the loudspeaker cone is given by the equation:

EQUATION 2

$F = Bil$ where 'F' = force on the voice coil
'B' = Magnetic field intensity
'i' = current in the voice coil
and 'l' = length of wire in the field

Again, regarding 'B' and 'l' as constants, the equation shows that it is current and not voltage that determines the force on the voice coil. Since the voltage contains the signal information from the power amp, it would be necessary for a perfectly linear relationship to exist between applied voltage and resulting signal current flow if a distortionless signal is to be produced. The impedance would have to be a constant and this is not the case. Fortunately the movement of the cone is not directly related to the current in the voice coil in the simple way shown above or the frequency response of a loudspeaker would simply be the inverse of its rather lumpy impedance curve.

In order to understand the parameters that determine the acoustic power actually radiated, it is necessary to look at the sources of mechanical rather than electrical impedance.

Converting energy

In the operation of a moving-coil direct-radiating driver there are really two energy conversions going on simultaneously. First the electrical energy is converted into mechanical energy of the voice coil and cone. Secondly this mechanical energy is converted into acoustic energy by the interaction of the cone with the neighbouring air molecules. Both these conversions must be accurate if the final result is to be a low distortion replica of the input voltage waveform.

The laws that apply to mechanical and acoustic forces are directly analogous to those of electrical forces and for this reason we can represent what happens in any acoustic or mechanical problem by a circuit diagram. In mechanics and acoustics there are direct and simple relationships like Ohm's law in electronics. It is only the complex arrangement of mechanical or acoustic circuit elements that makes the picture look complicated.

FIGURE 3A

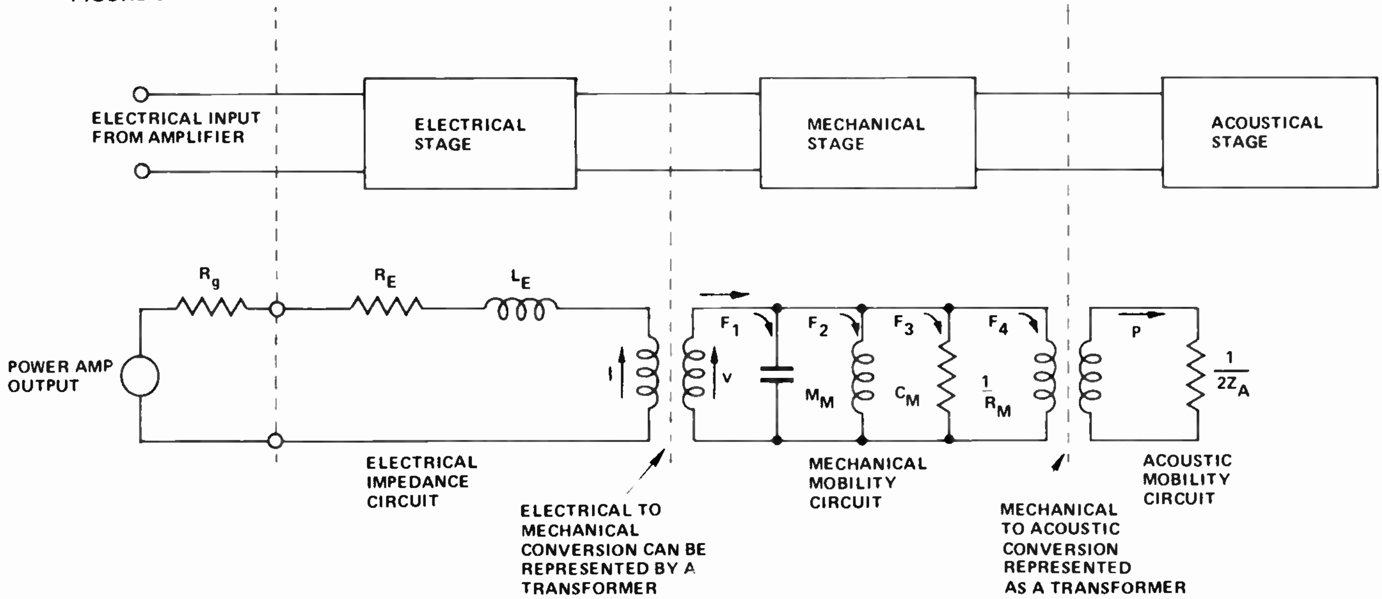


FIGURE 3B

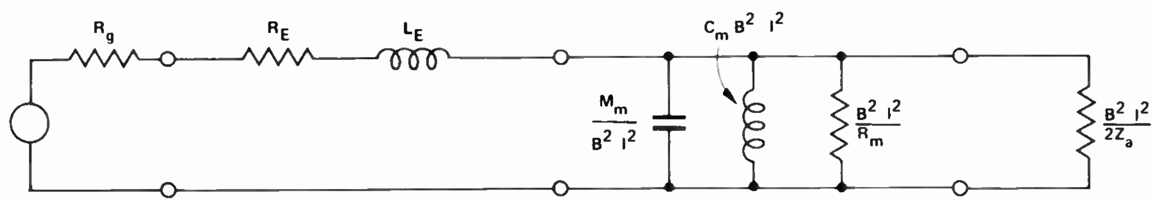


Figure 3. **Equivalent circuit for a typical moving-coil direct-radiating driver mounted in an infinite baffle.**

Just as an electronic circuit can look complex, but can be broken down into smaller and simpler circuits, so too can any acoustic or mechanical problem.

We can represent a complete picture of a dynamic loudspeaker by a circuit diagram showing electrical/mechanical and mechanical/acoustic conversions (see Figure 3a).

The power amplifier is connected via a net series resistance Rg, to the terminals of the loudspeakers. This resistance is the result of the internal resistance of the power amplifier and connecting cables. Since the voice coil is a coil of wire it possesses both inductance and resistance. The applied electrical signal sees these two in series and we represent this by the resistance R_E and the inductance L_E. The "E" simply implies that these are electrical quantities. Current flowing in the voice coil gives rise to the magnetic field that causes mechanical movement of the voice coil and cone assembly. This conversion of electrical to mechanical energy is represented in the circuit diagram as a transformer. Voltage across the primary is represented by the letter "e" and gives rise to velocity 'v', of the voice coil and cone assembly at the secondary of the transformer.

The total force applied by the voice coil ("F") is shown in the mechanical stage as "flowing" through the "wires" just as current would flow through the wires of an electrical circuit. This force sees three mechanical components in parallel, a mechanical capacitance M_M, a mechanical inductance C_M and a mechanical resistance, $1/R_M$. The mechanical capacitance M_M is caused by the mass of the cone. As frequency rises inertia comes into play and it becomes increasingly difficult for the cone to follow the input voltage waveform. The mass of the cone causes a frequency response roll-off at higher frequencies. This could be represented either by an inductance in series or a capacitance in parallel with the load. In Figure 3 this has been shown as the parallel capacitance, M_M.

A loudspeaker cone has a certain springiness, due to the nature of the cone's suspension and the overall construction of the particular driver. We specifiy this springiness by a spring constant, which is simply a number, represented by the letter 'k'. In loudspeaker technology we more often use the term compliance rather than spring constant. Compliance C_M, is defined as:

$C_M = \frac{1}{k}$ where 'k' is the spring constant.

The compliance impedes large movement of the cone. Since bass frequencies require longer cone excursions the compliance of the driver causes a frequency response that falls as frequency decreases. This can be represented as a capacitance in series or an inductance in parallel with the load. In Figure 3 the compliance C_M is represented as an inductor in parallel with the load.

The remaining term in the mechanical part of the loudspeaker circuit diagram is the mechanical resistance. Just as all circuit elements in an electronic circuit have resistance, so to does the mechanical circuit. The resistance is seen in series with the whole mechanical circuit and could be represented as a series resistor or a parallel inverse resistance. If R_M is the mechanical resistance of the circuit, an inverse resistance is defined as:

$$\frac{1}{R_M}$$

In Figure 3a force is shown as 'flowing' in the mechanical 'wires'. The total available force is shared into four major parts; the forces needed for the mass M_M, the compliance C_M, the mechanical resistance R_M and the load. If we define these four forces as F_1, F_2, F_3 and F_4 respectively, we can say that

F (the total force available) = $F_1+F_2+F_3+F_4$, and this has been shown in the mechanical circuit diagram. We have to represent the series resistance as an inverse resistance, $\frac{1}{R_M}$ and place it in parallel with the load, to illustrate the way it obtains its part of the total available force (F).

In this case the load is the primary of the mechanical/acoustical transformer. Of course this transformer doesn't actually exist. It is merely a way of representing the conversion of mechanical energy to acoustic energy by the interaction of air molecules with the surface of the loudspeaker cone. Mechanical force in the primary of the transformer is converted into sound pressure 'p', in the acoustic circuit.

In Figure 3a it is assumed that the loudspeaker is mounted in an infinite baffle. This is a partition that extends to infinity in all directions, cutting the universe into two halves, with a hole in which the loudspeaker is mounted. This is just a little impractical, but the only important thing is that no sound produced by the back of the speaker cone can interact with the sound from the front.

In order to move air molecules, the cone must do work so the air impedes movement of the cone. This impedance is called the acoustic impedance and is represented in the circuit diagram by Z_A. Since the loudspeaker is mounted in an infinite baffle the acoustic impedance is the same on both sides of the cone and becomes $2Z_A$.

We are now in a position to understand the causes of variations in the electrical impedance and acoustic radiated power. As was shown earlier, the back EMF is one of the dominant forces acting to increase the driver's impedance. It is related to the velocity of the loudspeaker cone as was indicated by Equation 2. If the motion of the cone is impeded, i.e: if the cone is held, the velocity must decrease, causing a decrease in the amount of back EMF. The decreased back EMF will cause a drop in loudspeaker impedance. So an *increase* in mechanical impedance causes a *decrease* in electrical impedance. With this in mind the electrical/mechanical/acoustic circuit diagram of Figure 3a can be converted into the all electrical circuit diagram of Figure 3b.

This circuit predicts the impedance characteristics of the driver. A generally increasing impedance with frequency is caused by L_E, while M_M, C_M, and R_M form a damped parallel resonant circuit. We would expect a sharp increase in impedance at one frequency, dropping to the dc resistance of R_E and R_G and then slowly rising as frequency increases. This is exactly the response as shown in Figure 2 which is the measured impedance response of a typical 12 inch (300 mm) woofer. This resonance point is called the fundamental resonance of the driver, and being a function of the compliance of the driver, can be expected to decrease in frequency a little as the driver wears in. This is the reason some loudspeaker experimenters "run in" the driver before measuring resonant frequency.

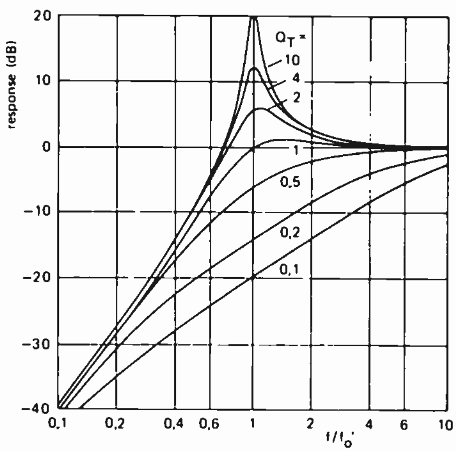


Figure 4. **Normalized frequency response of a typical low frequency loudspeaker for different values of Q_T (after Hermans & Hull, Electronic Applications Bulletin, Vol. 35, No.2., Feb. 1978, Philips).**

A more accurate model

The model of the loudspeaker developed so far has assumed that the shape of the loudspeaker cone remains unchanged and moves as a "rigid piston", following the input signal. This rigid piston theory works well at predicting the characteristics of drivers at low frequencies. At higher frequencies inertia again comes into play and the cone can no longer be considered as a rigid piston. If the driver remained a rigid piston throughout the audio spectrum its frequency response would fall off at a rate of 12 dB/octave at higher frequencies, limiting its useful frequency range.

The equation showing the relationship between the frequency of a sound and its associated wavelength is

$\lambda \nu = V_A$ where V_A is the velocity of sound in air
λ is the wavelength in meters
and ν is the frequency in Hertz.

The equation shows that the wavelength of sound decreases as frequency is increased. It should be noted that the velocity of sound depends on the medium in which the wave is propagating. The velocity of sound in the loudspeaker cone will be substantially different to that in air. Using this equation we can calculate the frequency at which the wavelength of sound approaches the radius of the loudspeaker cone. For a 300 mm (12 inch) loudspeaker this frequency is around 400 Hz and it is at this frequency that the rigid piston theory starts to come unstuck. Above this frequency the sound wave propagates up the cone, hopefully to be damped in the rubber surround. The sound wave is attenuated as it moves through the cone, and this attenuation effect increases with increasing frequency, causing a decrease in the effective cone diameter. This is the effect that enables a single cone loudspeaker to operate over a wide frequency range, since the decreasing effective cone diameter decreases the inertia presented to the coil assembly at higher frequencies. It should be noted that, in this range of the frequency spectrum, the rim and the cone will be radiating in anti-phase with the coil assembly. The way the cone material and suspension react to this multiple wave propagation is one of the biggest differences between a good driver and a poor one. It is for this reason that metal cones for instance are so often unsuccessful. Their ability to damp multiple resonances is generally poor in comparison to materials like paper or plastics.

Damping and Q-factor

In midranges and tweeters the drivers can be operated in frequency ranges that exclude their fundamental resonances. The crossover points are usually chosen so that at least one full octave exists between the crossover point and the fundamental resonance. In the case of bass drivers however it is necessary to operate the driver at and below the resonance of the woofer.

This is the main reason so many different bass loading principles have been developed. The fundamental resonance of the bass driver must be damped so that an acceptably flat frequency response can be established. If the resonance is not damped adequately, the all too common 'one note bass' sound results. This is a particularly noticeable and fatiguing loudspeaker fault and considerable effort must be spent on obtaining a smooth bass end response.

Since the loudspeaker is a resonant circuit the amount of damping can be specified by quoting the Q or quality factor. Q is defined by:

$Q = \frac{f_0}{f_1 - f_2}$ where f_0 is the frequency of the fundamental resonance. and f_1, f_2 are the 3 dB points.

Figure 4 shows a graph of bass-end frequency responses at a variety of Qs. Although the flattest response appears to be given by the case when the Q=1, this is not the optimally damped case and some boomy bass often occurs in bass systems with Qs around unity. The best Q is probably about 0.5. The bass is not boomy but is also not over-restricted which can happen if the Q drops to around 0.2 or 0.3. The best damping for any specific case needs to be established by experiment and ultimately, as always, the ear must be the final test.

Loudspeaker compliance, the total mass of the cone and the net series resistance with the voice coil, all determine the response of any loudspeaker system and any or all of these can be adjusted in order to achieve the optimum damping and frequency response. In practice, adjustments to the Q of the system are done by modifying the compliance and acoustic mass and resistances caused by the loudspeaker enclosure rather than modification of the driver itself.

The enclosure

The circuit in Figure 3 has been developed assuming that the driver is mounted in an infinite baffle. The air load on the cone of the loudspeaker is represented by an impedance of value:

$$\frac{1}{2Z_A}$$

When the driver is mounted in a practical loudspeaker enclosure, this acoustic impedance becomes a little more complicated and the circuit in Figure 5 replaces the simple resistor of Figure 3. This new impedance is made up of two major components. The radiation impedance from the front of the box (M_{AR}, R_{AR}) is related to the size of the baffle and is independent of the volume of the box.

The volume of air that the driver has access to is related to the effective radiating area of the cone and to the size of the baffle. Since bass frequencies have a greater dispersion than higher frequencies the volume of air accepting the radiated sound increases at lower frequencies. So the impedance on the front of the cone will be greatest at higher frequencies. If the size of the baffle is large in comparison to the radiating size of the driver the box approximates an infinite baffle down to a lower frequency than it would otherwise, and the frequency at which the driver has access to a 360^o radiation pattern is decreased. This is represented by the series combination of the inductance M_{AR} and R_{AR}, which gives an impedance characteristic that increases with frequency like the front radiation.

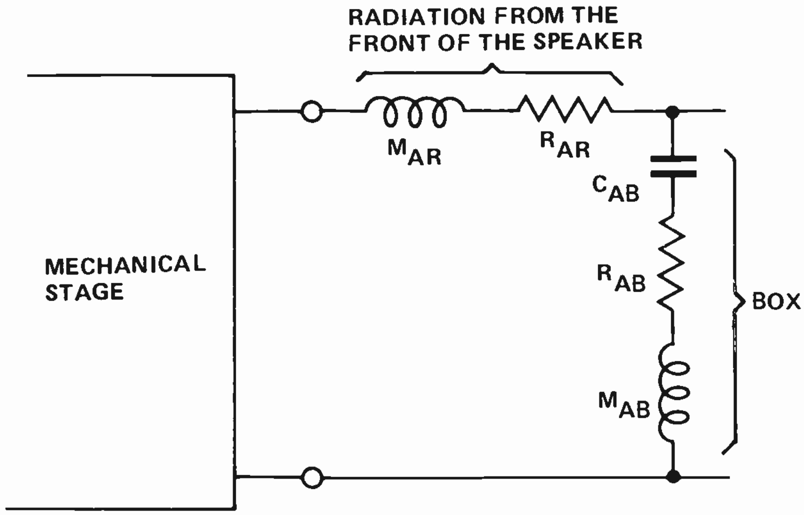


Figure 5. **Equivalent circuit diagram for the acoustic stage of a direct-radiating moving-coil driver mounted in a sealed enclosure.**

The second component of the radiation impedance is caused by the enclosed volume of air within the box. If we consider a sealed enclosure the volume of air within the box will be compressed by the driver. So the enclosure volume will affect the overall compliance of the loudspeaker system. This acoustic compliance is represented in Figure 5 by the capacitance C_{AB}. The effect of this is to increase the stiffness of the loudspeaker cone resulting in an increase in the fundamental resonance of the enclosure.

The volume of air in the box will also have an equivalent mass represented by the inductance M_{AB}. This mass will also affect the resonance of the system by increasing the overall mass of the cone. The acoustic resistance with the enclosure is shown in Figure 5 as R_{AB}.

The final resonant frequency of the driver in the box is a function of the total effect of all the compliances and masses in the system. If the total mass is represented by M_a and the total effect of the compliances is C_a then the resonant frequency of the system will be given by the equation

$$f_o = \frac{1}{2\pi\sqrt{M_a C_a}}$$

The equation shows that a decrease in either the total mass or compliance will cause an increase in the resonant frequency of the loudspeaker system.

Resonances

The acoustic circuit in Figure 5 represents the reactances caused by the enclosure around the resonant frequency, but as usual in loudspeaker science things get more complicated as frequency increases. As the wavelength of the sound wave produced inside the enclosure becomes shorter the box no longer acts as a simple spring. The produced wave travels from the back of the driver towards the rear and sides of the cabinet, where it is reflected back towards the driver. This sound wave will interact with the cone, either reinforcing or impeding the motion of the cone depending on the particular frequency. This results in successive rises and dips in the frequency response and for this reason it is important that this reflected wave is damped as much as possible. In order to absorb this unwanted energy the enclosure is usually lined with an absorptive material such as bonded mineral wool, acetate fibre or bonded hair felt.

The most important parameter of any of these materials is that they are reasonably open. If the material is too dense it will not only have little absorption but it will decrease the total available volume within the box. ▶

Generally a layer of 25 mm speaker innerbond on the back, sides, top and bottom is about right.

Lining the box also has the effect of altering the acoustic resistance on the back of the cone. This effect is often used to increase the damping and thereby decrease the Q of the system resonance. Usually the necessary damping can be only partially achieved and it is necessary to partially fill the box with an absorptive material as well. If the enclosure is completely filled sound waves within the enclosure are converted into heat in the filler material. Normally this overdamps the box resulting in a Q sometimes as low as 0.1.

Filling the box has one other major effect. Owing to the heating of the material inside the box and its different density to that of air, the velocity of sound within the enclosure is reduced dramatically. The speed of sound in air is around 344 m/s and this could drop to as low as 292 m/s. This has the effect of increasing the compliance of the enclosure and thereby decreasing resonant frequency in the same way as increasing the box volume would. An optimally filled box could appear some 30% to 40% bigger than it really is.

Throughout this article we have discussed the sealed enclosure, leaving explanations of other bass loading techniques to a later time. One of the most common enclosures is the bass reflex, which uses a port cut in the baffle to augment the bass radiation of the driver. The acoustic circuit diagram must show the effects of the mass, compliance and resistance of the port in addition to those shown for the rest of the box, making the loudspeaker equivalent circuit even more complicated. Both the bass reflex and the sealed enclosure are capable of very good results and it is not possible to state simply which is better. We have omitted a detailed discussion of the principles of the bass reflex loudspeaker simply for the sake of simplicity.

The next article deals with the problems of mating a collection of drivers to form a completed loudspeaker — another project in our 'Series 4000' line-up of hi-fi components.

●

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Hi-Fi speaker systems; D. Hermans and M.D. Hull, Phillips applications bulletin 35/1 and 35/2.

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Loudspeakers in vented boxes: A.N. Thiele.

Principles and problems in loudspeaker design

David Tilbrook

Our second article on this subject concludes with a discussion on crossovers. To put all the theory into practice, turn to page 56 where you'll find the first of the Series 4000 speaker designs.

THE LAST ARTICLE dealt with the characteristics of a typical moving-coil direct-radiating loudspeaker and the interactions that are likely to occur with the loudspeaker enclosure. Once these problems are understood and the bass performance has been optimised, we are in a position to finish the design. I have discussed the bass end of the audio spectrum first not because it is the most important, but simply because it is the most difficult to optimise. The mid-range is by far the most critical part of the audio spectrum since it is midrange distortion that the ear objects to more than any other.

It was shown before that drivers have limited frequency responses and that it is therefore necessary to use several drivers, each covering its own frequency range. By far the most common arrangement is the three-way, so called because it uses three drivers to cover the audio range. A woofer covers the bass end, crossing over to a midrange driver somewhere between 400 Hz and 1 kHz. The midrange driver, sometimes called a 'squawker', carries the frequency range from this crossover point up to where the tweeter takes over, usually around 3 kHz to 5 kHz. The tweeter covers the remainder of the audio spectrum up to around 18 kHz, about the limit of human hearing. A crossover is used to separate the input signal from the output of the power amplifier into the three frequency bands.

Passive loudspeaker crossovers

The design of the crossover for any particular group of drivers must be done only after a thorough investigation into the characteristics of the drivers has been carried out. It is essential to choose drivers with an adequate overlap in their frequency responses, or a 'hole' will result in the response of the final loudspeaker. The amount of overlap needed depends on the slope of the filters used in the crossover. If a fast slope is used a smaller overlap is required, but filters with very fast slopes are complicated and expensive.

The basic crossover filter consists of a low pass and a high pass section. In a two way loudspeaker only one of these sections would be used, while in a three way loudspeaker two sections are used, one for the bass-mid crossover and the other for the mid-treble crossover. The simplest crossover is called a first-order crossover and has a slope in its attenuating region of 6 dB/octave. An octave is a range of frequency such that the highest frequency in the band is double the lowest frequency; for instance, an octave above 50 Hz is the frequency range 50 Hz to 100 Hz, while an octave above 5 kHz is 5 kHz to 10 kHz. This is not a precise definition of an octave but is essentially correct and is adequate for loudspeaker analysis.

Figures 2 and 3 show circuit diagrams for series and parallel first-order crossovers. The series configuration is less commonly used since it is only applicable to two-way loudspeakers and has no advantages over the parallel type. If the first-order crossover is terminated with ideal resistive loads, the two slopes will add to give a linear response with the phase response perfectly preserved. In this respect it is fairly unusual since no other simple passive crossover will give a response that is linear in both frequency and phase. Unfortunately 6 dB/octave slopes require a choice of drivers with very broad overlapping frequency responses. Generally it is necessary to have a usable response from a driver to a frequency where the crossover is giving about 12 dB of attenuation. For a 6 dB/octave low pass crossover this would be two octaves above the crossover point. A woofer crossing out at 500 Hz would be very unlikely to have a response to 2 kHz, so a 6 dB/octave filter could not be used.

The most common crossover is the

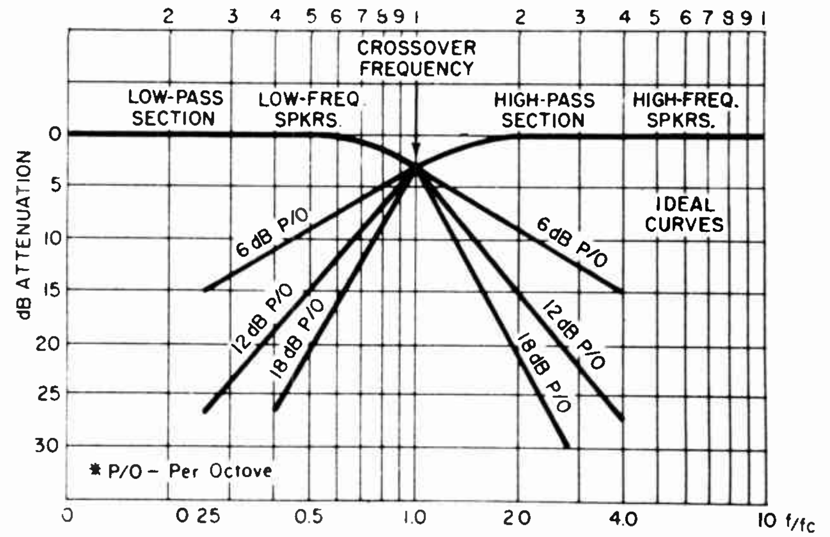


Figure 1. **Typical frequency response curves of a two-way crossover network. The graph shows three different pairs of filters, each having a different rate of attenuation.**

CONSTANT K CROSSOVER NETWORKS

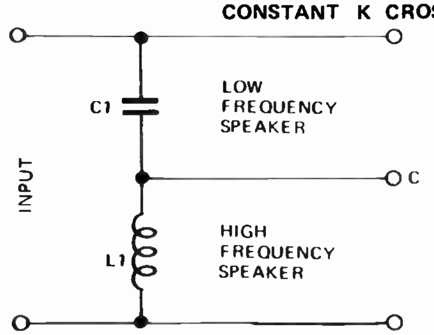


Figure 2 – **series type, 6 dB/octave.**

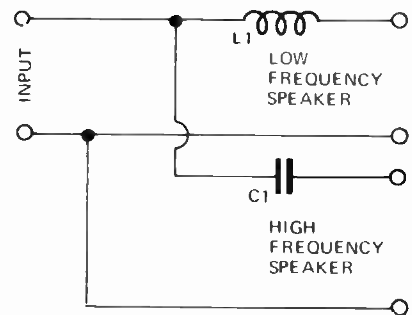


Figure 3 – **parallel type, 6 dB/octave.**

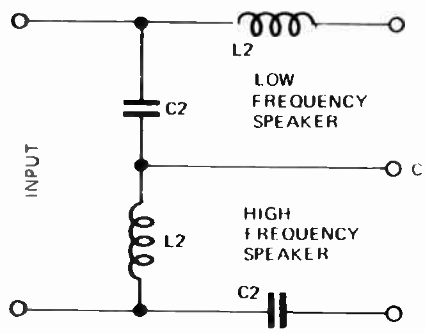


Figure 4 – **series type, 12 dB/octave.**

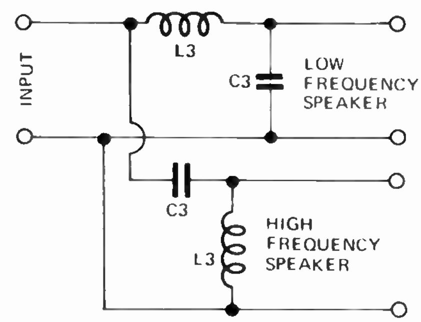


Figure 5 – **parallel type, 12 dB/octave.**

second order crossover, having a slope of 12 dB/octave. Figures 4 and 5 show the circuit diagrams for series and parallel second-order crossovers. Once again the series configuration is less commonly used. When terminated with ideal resistive loads the second-order crossover does not give an overall flat response. The phase characteristic causes the outputs of each half of the crossover to approach a 180 degree phase difference at the crossover point. The two outputs cancel each other, leaving a massive hole in the frequency response of the system. The 'cure' is to invert one of the drivers so that it is driven out of phase normally. The phase inversion around the crossover point brings the two drivers in phase again and the two outputs add, instead of cancelling. Unfortunately they still don't add perfectly and the result is an overall response that has a slight hump in the frequency response of around 2 dB. This is not really noticable, as few drivers have responses that are flat to this degree.

At the present time there is a great deal of discussion as to whether this non-idealistic phase response is audible. Some manufacturers insist that it is audible and design their loudspeakers accordingly, while others are most emphatic that it is not audible. The first work that I know of that was done on the subject was by Helmholtz, in his "Sensations of Tone". The quality of any sound was said to be a result only of the relative intensities of the component sine waves and not their phase relationships. The waveshape could therefore be totally different but they would sound the same.

There is another source of phase error caused by the misalignment of the acoustic centres of the drivers. The conventional way to mount the drivers ▶

FREQ.	C1	C2	C3	L1	L2	L3
100	199	281	141	12·7	9	18
150	133	188	93·8	8·49	6	12
200	99·5	141	70·3	6·37	4·5	9
250	79·6	113	56·3	5·09	3·6	7·2
300	66·3	93·8	46·9	4·24	3	6
350	56·8	80·4	40·2	3·64	2·57	5·14
400	49·7	70·3	35·2	3·18	2·25	4·5
500	39·8	56·3	28·1	2·55	1·8	3·6
600	33·2	46·9	23·4	2·12	1·5	3
750	26·5	37·5	18·8	1·7	1·2	2·4
1000	19·9	28·1	14·1	1·27	·9	1·8
1250	15·9	22·5	11·3	1·02	·72	1·44
1500	13·3	18·8	9·38	·849	·6	1·2
2000	9·95	14·1	7·03	·637	·45	·9
2500	7·96	11·3	5·63	·509	·36	·72
3000	6·63	9·38	4·69	·424	·3	·6
3500	5·68	8·04	4·02	·364	·257	·514
4000	4·97	7·03	3·52	·318	·225	·45
5000	3·98	5·63	2·81	·255	·18	·36
6000	3·32	4·69	2·34	·212	·15	·3
7500	2·65	3·75	1·88	·17	·12	·24
10000	1·99	2·81	1·41	·127	·09	·18

Table 1: **Component values for constant K loudspeaker crossover networks. Inductance in millihenries, capacitance in microfarads, speaker impedance = 8 ohms.**

is to simply bolt them to the front panel. This lines up the chassis of the drivers, but since different drivers have different depths, the voice coils of the drivers are all at different distance from the listener. If two notes are sent simultaneously to both the woofer and the tweeter for example, the note sent to the tweeter will get to the listener momentarily before the note from the woofer. Furthermore, the woofer cone is heavier than the tweeter or midrange cones and this combined with the effect of the air load on the drivers moves their actual acoustic centres even further away from the chassis. Manufacturers concerned with this effect mount the drivers on a multi-level front panel so that the tweeter is further away from the listener than the midrange. Similarly, the midrange is mounted on a plane that is further away from the listener than the woofer. This gives the sound from the midrange and woofers a head start over the tweeter, and attempts to correct for the differences in their acoustic centres.

Both types of phase errors need to be recognised and dealt with independently if a meaningful analysis of the audibility of phase errors in loudspeakers is to be carried out. Even if phase errors of this magnitude are audible (and only experiment can tell us), an extremely good loudspeaker can still be constructed along the more conventional techniques using second-order crossovers with drivers mounted on a plane baffle.

The 4000 Series of loudspeaker projects commencing in this issue use the more conventional approach to driver mounting and crossover design to simplify construction and decrease cost. If you choose to experiment with the audibility of phase in loudspeakers, and construct a Series 4000 loudspeaker with a stepped front panel, the best way to establish the correct distance between the panels is by experiment. The drivers should be connected to the crossover and mounted in separate enclosures. The size of these enclosures is not critical.

Supply the power amp driving the loudspeaker with a source of low repetition pulses (or a low frequency square wave around 20 Hz). If the loudspeaker is now monitored with a microphone and the output of the mic amplifier fed to an oscilloscope, the transient performance of the loudspeaker can be determined.

When the front baffles of the enclosures are aligned, as would be the case in a conventional loudspeaker, the input pulses will be seen to be converted into a series of pulses. Each pulse corresponds to one of the drivers. If the enclosures are moved slowly back with respect to the woofer enclosure these pulses will merge into a single pulse. This is the correct position for the baffles, and using these measurements the final enclosure can be built. If you have the necessary equipment to do this experiment we would be interested in hearing about your results.

The crossovers described so far belong to a class of filters called constant-K filters. These filters are designed on the assumption that the product of the impedances of the capacitor and the inductor in the stage is equal to the square of its characteristic resistance, i.e.

$$Z_C \times Z_L = R_o^2$$

The characteristic resistance of a filter is that resistance into which there is maximum power transfer. Originally, 'K' was used instead of the now more common symbol R_o for the characteristic resistance, hence the name constant K. Table 1 gives values for constant-K filters for a variety of crossover points assuming an 8 ohm resistive load.

M-derived filter sections

The assumptions made to simplify the design of the constant-K filters lead to some non-ideal characteristics. It is sometimes mistakenly thought that constant-K implies constant impedance. This is not the case and the impedance is a variable with the effect occuring mostly around the crossover point. The other problem with constant-K filters is that the slopes are slowest near the crossover point. The solution to these problems has been known since 1923, when Zobel proposed that other sections could be used to flatten the response within the passband and sharpen the roll-off point. These stages are called M-derived sections, since the values of inductance and capacitance used in the filter are obtained by first deriving them for a constant-K type filter and then converting these values into M-derived values with the use of a mathematical equation that contains the term M. M is simply a number between 0 and 1, usually around 0.6 for crossover applications. Either the phase or frequency characteristics may be optimised but not both at once; 0.6 is a good comprimise. Table 2 and Figures 6 to 9 give values for inductors and capacitors for M-derived crossovers with M = 0.6.

The other major advantage of this filter is that it allows a third-order or 18 dB/octave filter to be built. Third-order filters can be made to have a linear frequency characterisitic when the outputs of the two channels are summed, but like the second-order filters described before, suffer from a very non-linear phase response. Each filter shifts the phase at the crossover point by 180 degrees, so there is a 360 degree phase shift between the two outputs.

Loudspeaker impedance

So far I have assumed that the loudspeakers connected to the crossover are fixed 8 ohm loads, but as was seen in last month's article, this is most definitely not the case. Most drivers have an impedance characteristic that presents maximum impedance at their resonant frequency, dropping to the nominal dc resistance of the driver at a frequency above this, followed by a generally increasing impedance as frequency rises (see Figure 2 in the last article). Provided the driver is not being used near its resonant frequency, which should always be the case with midranges and tweeters, this impedance variation can be corrected by a series capacitor-resistor network placed in parallel with the loudspeaker Figure 10 shows a typical circuit. This network has an impedance that decreases with increasing frequency, tending to cancel the increasing impedance of the driver. The component values shown are applicable to an average woofer, although the actual values in any specific application are best established by experiment. This works very well and it is not difficult to obtain an impedance response that is flat within one ohm over most of the driver's operating range.

Matching sensitivities

Once the crossover points have been established from an analysis of the driver's best operating regions, the final step is to equalise the various sensitivites of the different drivers. This is done by a resistor divider network as shown in Figure 11. A simple resistor placed in series with the driver would of course decrease the power in the loudspeaker for a given signal voltage, but this increases the impedance seen by the rest of the crossover altering the crossover frequency point. The resistive divider network shown in Figure 11 can be set to represent a fairly constant 8 ohm▶

M-DERIVED CROSSOVER NETWORKS

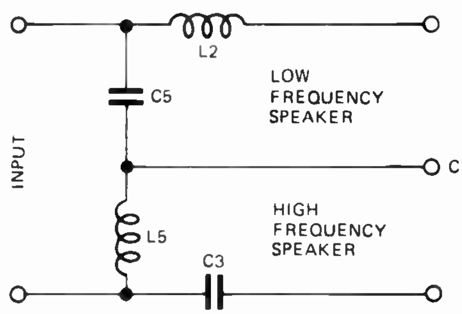


Figure 6 – **series type, 12 dB/octave.**

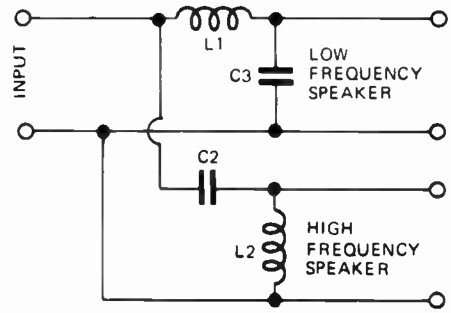


Figure 7 – **parallel type, 12 dB/octave.**

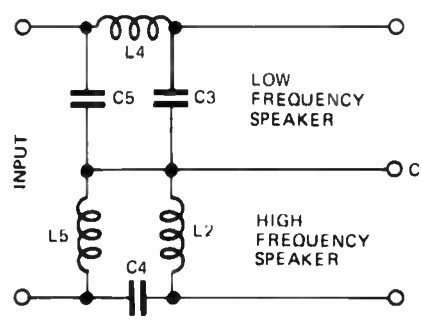


Figure 8 – **series type, 18 dB/octave.**

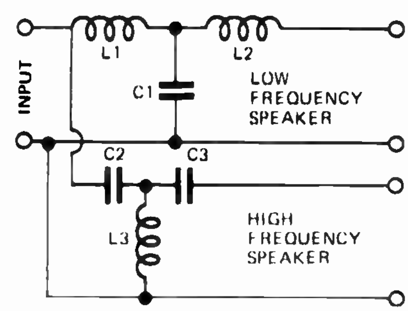


Figure 9 – **parallel type, 18 dB/octave.**

FREQ.	C1	C2	C3	C4	C5	L1	L2	L3	L4	L5
100	398	124	199	99.4	318	20.3	12.7	6.37	25.4	7.96
150	265	82.9	133	66.3	212	13.5	8.49	4.24	16.9	5.31
200	199	62.1	99.4	49.7	159	10.1	6.37	3.18	12.7	3.98
250	159	49.7	79.5	39.7	127	8.15	5.09	2.55	10.1	3.18
300	133	41.4	66.3	33.1	106	6.79	4.24	2.12	8.49	2.65
350	114	35.5	56.8	28.4	90.9	5.82	3.64	1.82	7.28	2.27
400	99.4	31	49.7	24.8	79.5	5.09	3.18	1.59	6.37	1.99
500	79.5	24.8	39.7	19.9	63.6	4.07	2.55	1.27	5.09	1.59
600	66.3	20.7	33.1	16.5	53	3.4	2.12	1.06	4.24	1.33
750	53	16.5	26.5	13.2	42.4	2.72	1.7	.849	3.4	1.06
1000	39.7	12.4	19.9	9.95	31.8	2.04	1.27	.637	2.55	.796
1250	31.8	9.95	15.9	7.96	25.4	1.63	1.02	.509	2.04	.637
1500	26.5	8.29	13.2	6.63	21.2	1.36	.849	.424	1.7	.531
2000	19.9	6.22	9.95	4.97	15.9	1.02	.637	.318	1.27	.398
2500	15.9	4.97	7.96	3.98	12.7	.815	.509	.255	1.02	.318
3000	13.2	4.14	6.63	3.32	10.6	.679	.424	.212	.849	.265
3500	11.3	3.55	5.68	2.84	9.09	.582	.364	.182	.728	.227
4000	9.95	3.11	4.97	2.49	7.96	.509	.318	.159	.637	.199
5000	7.96	2.49	3.98	1.99	6.37	.407	.255	.127	.509	.159
6000	6.63	2.07	3.32	1.66	5.31	.34	.212	.106	.424	.133
7500	5.31	1.66	2.65	1.33	4.24	.272	.17	.085	.34	.106
10000	3.98	1.24	1.99	.995	3.18	.204	.127	.064	.255	.08

Table 2: **Component values for M-derived loudspeaker crossover networks. Inductance in millihenries, capacitance in microfarads, M = 0.6, speaker impedance = 8 ohms.**

A Bruel and Kjaer sound level meter. Instruments like this are used to determine the frequency response of a loudspeaker. This instrument has several weighting curves built in as well as a one octave filter set to allow pink noise analysis.

load. Resistor R2 is placed in parallel with the driver, resulting in a decreased total impedance. This impedance is then brought back up to the desired impedance by placing R1 in series. The correct values for R1 and R2, assuming an 8 ohm loudspeaker system, are given by the following three simple equations.

1. $d = \text{antilog} \frac{-\text{signal drop in dB}}{20}$

2. $R2 = \frac{8d}{1-d}$

3. $R1 = \frac{64}{8 + R2}$

First establish the amount of attenuation that is required in dB. Normally by this stage a frequency response curve has been established by measuring the loudspeaker, and an estimate of the required attenuation can be obtained from this. Now use equation 1 above. The antilog of a number can be found either using log/antilog tables or the inverse log key on any scientific calculator. I have used the symbol 'd' for the result of equation 1 mainly to simplify the written form of equation 2, but in reality 'd' is equal to the voltage across the loudspeaker divided by the voltage from the amplifier. i.e.

$$d = \frac{V_s}{V_i}$$

where 'V ' is the signal voltage across the loudspeaker and 'V_i' is the applied signal voltage from the amplifier.

The result of equation 1 is plugged into equation 2, which yields the correct value for R2. The value of R2 is then used in equation 3 to obtain the value of R1. For example, if a midrange is to be decreased in sensitivity by 3 dB, equation 1 becomes:

$$d = \text{antilog} \frac{-3}{20}$$

$d = \text{antilog} -0.15 = 0.7079$

So a 3 dB drop in output signal level is equivalent to decreasing the signal voltage across the loudspeaker to $0.7079V_i$. Plugging this into equation 2 gives

$$R2 = \frac{8 \times 0.7079}{1 - 0.7079} = 19.4 \text{ ohms.}$$

Using this result in equation 2 gives

$$R1 = \frac{64}{8 + 19.4} = 2.3 \text{ ohms.}$$

The nearest value resistors to these would be 18R and 2.2R, and with these resistors the impedance presented to the remainder of the crossover would be approximately 7.7 ohms which is close enough in practice.

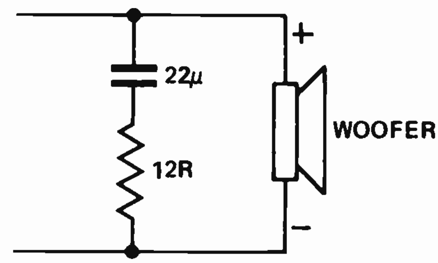


Figure 10. **Circuit to improve the apparent impedance of a loudspeaker.**

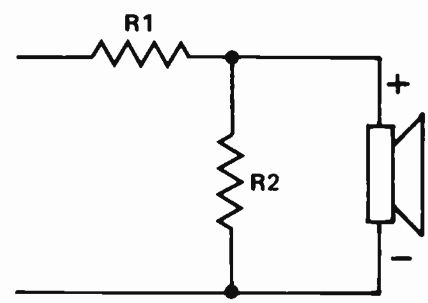


Figure 11. **Potential divider used to compensate for different loudspeaker sensitivities.**

The final test

Now that the bass performance has been optimised through a suitable choice of enclosure size and damping, the crossover points and slopes have been established, and the impedance and sensitivities of the various drivers corrected, the final subjective tests can begin. The importance of good subjective testing in loudspeaker design cannot be overestimated. The most common form of subjective analysis, other than simply listening to some good records, is an A-B test with other loudspeakers. Although this method can give some meaningful results, its validity is generally overestimated in my opinion. The best form of subjective testing is comparison to the original live performance. Simply recording a voice onto high quality recording equipment with a good microphone will tell you more about a loudspeaker than any amount of A-B testing. ●

THD analyser for audio circuits

This article describes a spot frequency audio distortion analyser, designed and built by a reader, Laurie Tunnicliffe of Mulgrave, Victoria. Measurements can be made at 100 Hz, 1 kHz and 10 kHz, and the final resolution of the instrument is 0.01%.

IN RECENT YEARS there has been a trend towards considering the transient behaviour of audio circuits rather than their steady-state behaviour. Although the attention given to this side of circuit design is not unwelcome, total harmonic distortion (THD) analysis is far from redundant.

While the transient behaviour is a go/no go situation, the THD behaviour is a measure of how well a circuit will perform. For instance, it is not a question of to what degree an amplifier will slew limit, or to what extent the internal loop will overload; these transient characteristics are barriers, and until reached will have no effect on the amplifier's performance.

THD measurements are therefore still a valuable analytical tool when developing new circuits or measuring and giving figures of merit to existing circuits.

When a single frequency is passed through an amplifier with a non-linear transfer curve, other frequencies are produced. These other frequencies are integer multiples of the test signal and sum of these is the THD.

There are a number of ways of measuring the harmonic distortion of an amplifier. The most recent is the 'Fourier Analyser', which is a computer-based instrument that samples the output waveform and performs the necessary mathematics (Fourier transform) to break down the waveform into its component parts. These instruments however cost tens of thousands of dollars. A technique becoming popular is the use of a spectrum analyser, which is a swept bandpass filter. The results are displayed on a CRO. THD is then calculated from

$$THD = \sqrt{\left(\frac{F1}{F}\right)^2 + \left(\frac{F2}{F}\right)^2 + \left(\frac{F3}{F}\right)^2 \cdots}$$

N.B. Square root sign in formula applies to sum of all terms.

Another technique used by Quad and described elsewhere[1], compares the input and output of an amplifier (using a differential amp), the difference being distortion. In Quad's experiment, music is used as the source and a monitor amp/speakers are used to listen to the distortion played by itself. This then becomes a 'real-time' distortion analyser, and any non-linearities, whether transient or steady state, are revealed.

The last and most commonly used method is to eliminate the fundamental signal and read the resultant harmonics on a moving coil meter. This is sometimes called 'Noise and THD' measurement, as hum and 'electronic noise' are lumped together with the harmonies. This is the technique used for the instrument presented here.

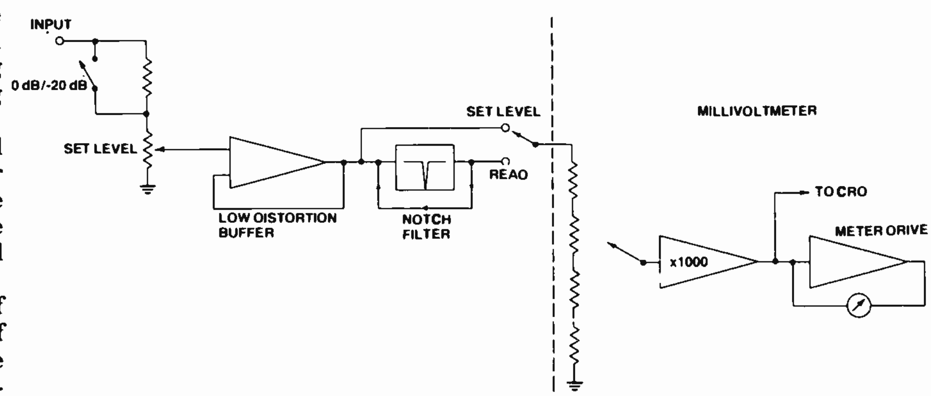


Figure 1. **Block diagram of the THD analyser described in this article.**

Block diagram

Refer to Figure 1 for the instrument's block diagram.

The input is applied to a buffer stage via a 0 dB/−20 dB attenuator. This allows easier control of the 'set level pot' when large signal levels are being measured. The buffer stage provides a low impedance source to drive the notch filter. The design of the buffer is of major consideration, as any non-linearities it contributes will be indistinguishable from the signal being tested.

There are a number of options available when designing a notch filter. A derivative of the Wien bridge was chosen due to its simplicity and the fact that it only requires two variable reactances to balance.

Bootstrapping around the filter is necessary to tighten up the notch width. Without this, the attenuation of the second harmonic would be excessive. With the amount of bootstrapping used the second harmonic is attenuated by less than 1 dB.

The notch filter is followed by a millivoltmeter and reads the average value of the harmonics relative to the fundamental. The meter is calibrated to read full scale for 0.775 V RMS input, and therefore the meter can be used separately to measure dBs into a 600 ohm load, relative dB (dBV) or millivolts, giving the instrument a dual function.

A CRO output is taken after the x1000 amplifier, so that the residual harmonics can be investigated. This will often give considerable insight into the cause of the distortion.

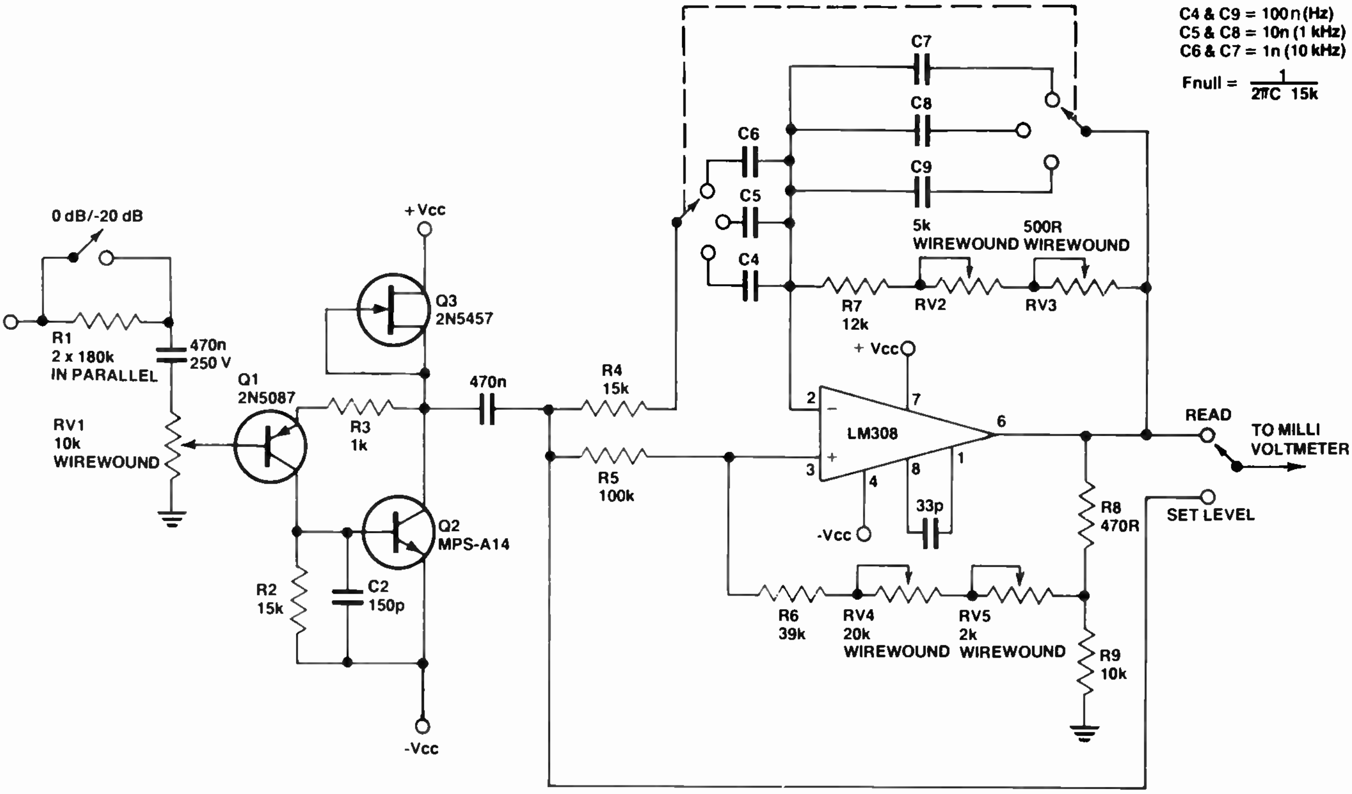


Figure 2. **Circuit diagram of the THD analyser (above and opposite).**

Circuit description

By examining Figure 2 the complete circuit diagram can be understood.

Q1 and Q2 form the non-inverting buffer with Q3 acting as an active load for Q2. The distortion contributed by this stage can be calculated to a first approximation as follows. The input transistor contributes negligible distortion relative to the second stage, due to the small signal levels it handles. The second stage is the prime mover, with a voltage gain of approximately 60 dB. The base drive voltage will therefore be

$$\frac{.775 \times \sqrt{2}}{1000} = 1.1\ \text{mV}$$

peak and the distortion generated is 1.1% second harmonic[2]. Since the buffer stage has 100% feedback (unity gain), the loop gain is also 60 dB and the distortion is reduced to

$$\text{loop gain} = \frac{1.1}{1000} = .0011\%$$

The buffer feeds the notch filter, which may be looked upon as a frequency dependent differential amp. At the notch frequency, the parallel arm and the series arm balance to give the same impedance ratio as the resistive arms. The input then appears as a common mode signal to a differential amp, and the output is zero. The common mode rejection ratio of the op-amp is of particular importance; however, most op-amps have a CMRR of at least 80 dB, which is sufficient to give a 0.01% resolution.

R8 and R9 provide the positive feedback (bootstrap) necessary to tighten the notch width.

Following the notch filter is the millivoltmeter, which consists of a constant gain, x1000 amp, preceded by a step attenuator giving 20 dB steps. The meter ranges will therefore be 100%, 10%, 1%, 0.1%.

The amplifier is followed by a fairly conventional meter-driven circuit, with the four bridge diodes being placed in the feedback loop of the op-amp. The non-linearities of the diodes are rendered insignificant and the meter reads accurately, even at the low end of the scale. Diodes 5 and 6 are used to protect the meter from an overload.

In order to achieve a wide enough bandwidth for the x1000 amp, an externally compensated op-amp was necessary. The op-amp used in the meter circuit is a dual low-noise device, and therefore helps to keep the instrument noise level low and reduces parts count.

C12 rolls off the frequency response above 70 kHz. This helps improve the square wave response by reducing any ringing, and also reduces high frequency noise. This will allow measurement up to the seventh harmonic of 10 kHz, and should prove adequate.

Construction

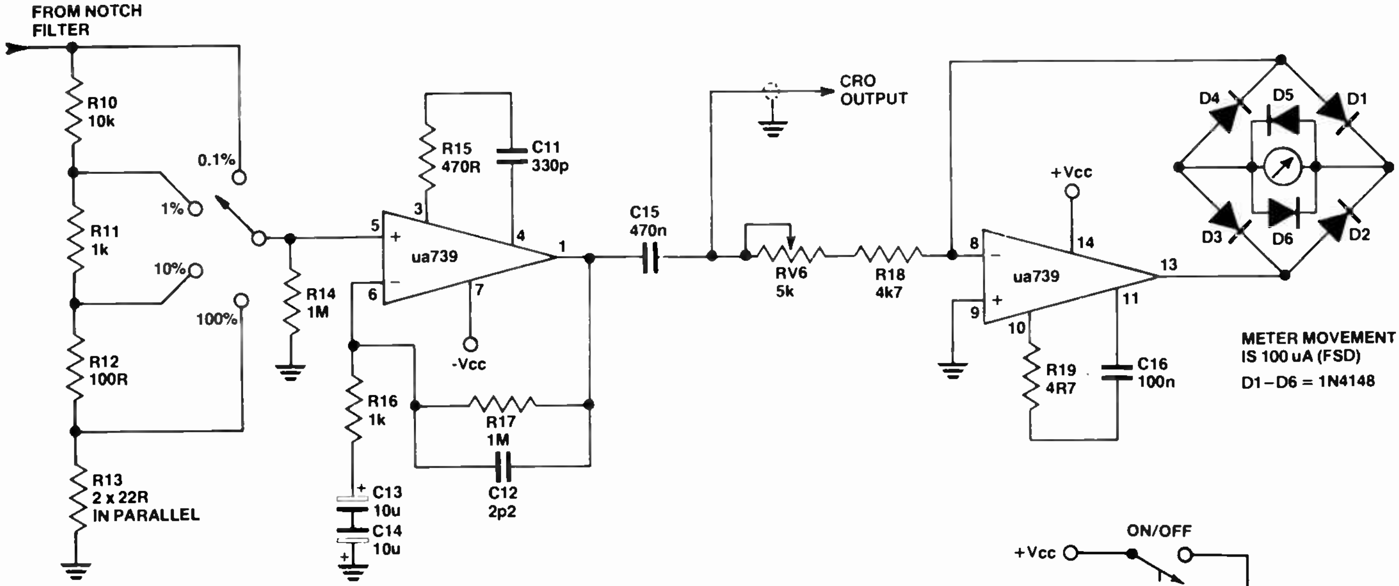
If accuracy and stability are to be reproduced, all components must be close tolerance, high stability. I used ¼-watt metal film resistors, which are available from Dick Smith. The filter capacitors are either styroseal or green caps; ceramic capacitors should be avoided as they are voltage dependent.

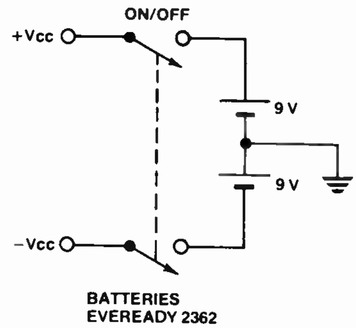
Veroboard should also be avoided, as stray capacitance across the strips causes problems when you are looking for one part in 10 000.

I used tagstrip and wire-wrap sockets for the ICs and found this easy to work with, giving satisfactory results.

My prototype was built in a diecast box; however, I had to rewire it three times, using shielded cable for every connecting wire, before a stable layout was found. For this reason I strongly suggest building the instrument as two separate units – a notch filter and a millivoltmeter. This will also give the versatility of being able to use one or the other independently.

The nulling pots are wirewound, having the advantage of a continuous track. The continuity of a wirewound pot is limited by the fact that a step in resistance equivalent to one wire winding is the smallest change possible. Carbon pots are certainly worse, as they sometimes jump resistance value mid-track. The fine control pots are single turn and provide reasonable ease of nulling at low levels. However, if you





feel it is worth the extra cost, ten-turn (spiral wound) pots will alleviate having to be careful when adjusting the controls.

The only adjustment to be done is the calibration of the meter. This is accomplished by applying a 0.775 V RMS, 1 kHz signal to the input with the unit switched to 'set level', and adjusting RV6 for full-scale deflection of the meter. Alternatively RV6 and R18 can be changed to 1k and 6k8 (fixed values) with less than 1% error in meter reading.

Operation

The instrument is operated as follows:

- Set Level/Read switch to Set Level
- Range switch to 100%
- Adjust Set Level pot for FSD of meter
- Set Level/Read switch to Read
- Adjust the nulling pots for minimum meter reading
- THD is now read from the meter and range switch.

The nulling procedure is accomplished by starting with the coarse pots, and alternately adjusting them for minimum meter deflection until it becomes difficult to proceed. The process is then repeated with the fine pots. If the user has a CRO available, this will assist the nulling procedure.

Figures 3, 4 and 5 show some oscilloscope photographs of the input and output of the meter. Figure 3 is an under-biased class B amplifier and shows spikes in the residual. This is a common waveform from class B amplifiers and the meter can be used to set the bias level for optimum.

Figure 4 shows second harmonic distortion from a voltage-driven common emitter amplifier. The input signal was 1.1 mV peak and the meter reading was 1.1%. This confirms the analysis used for the buffer stage.

Figure 5 is third harmonic caused by a thermistor-stabilised Wien bridge oscillator. THD was 0.02% at 100 Hz. (This is one of the problems encountered when using a thermistor at low oscillator frequencies.)

It should be noted that a low distortion oscillator will need to be used when making measurements. The residual distortion of the oscillator may set the lower limit to the measurements.

Some performance measurements

Final resolution of the meter was 0.005% at 100 Hz and 1 kHz and 0.01% at 10 kHz. Below this, drift in components' values with temperature, circuit noise, distortion introduced by the buffer and the filter stage and CMRR limitations of the filter all take their toll. However, distortion values of less that 0.01% are purely academic in my view, regardless of what hi-fi manufacturers' sales departments would have us believe.

References
1. P.J. Baxandall, Wireless World, November 1977, pp. 63-66.
2. E.F. Taylor, Wireless World, August 1977, pp. 28-32.

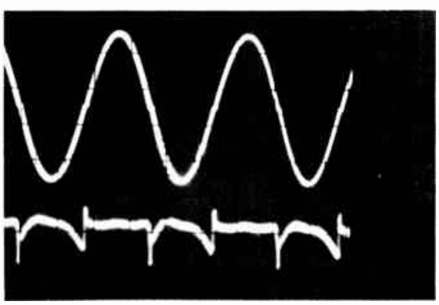


Figure 3. **Class B amp, meter reading 1.5%.**

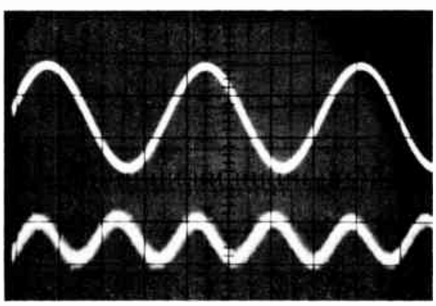


Figure 4. **Second harmonic distortion, meter reading 1.1%.**

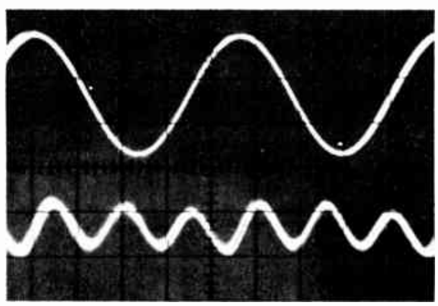


Figure 5. **Third harmonic distortion, THD 0.02%**

High performance audio power amplifier

Department of Aeronautical Engineering, University of Sydney.

This article, which previously appeared in Australian Electronics Engineering, Sept. 1980, describes the design of a very high performance 25 W audio power amplifier featuring low harmonic and crossover distortion, very stable bias current, large open-loop bandwidth to avoid transient intermodulation distortion, very good square-wave response with capacitive loads and relatively simple design.

L. Stellema

THE DESIGN of audio amplifiers is undergoing considerable change, stimulated by requirements for lower distortion and greater stability.

In this article the design of a high performance 25 W power amplifier is described and its performance figures given. The amplifier has extremely low harmonic and crossover distortion, and by a combination of thermal coupling techniques and circuit design, very high stability of bias current is achieved. The circuit also has the advantage of being relatively simple.

To achieve low crossover distortion, two steps are taken. Firstly, open-loop bandwidth is made rather high (15 kHz). Since crossover distortion consists of narrow spikes, feedback is not very effective in reducing these if the open-loop bandwidth is low. Secondly, instead of using conventional Darlington amplifiers in the output stage, the output transistors are driven by emitter followers that have constant current loads. This greatly improves the high frequency characteristics and also provides output current limiting.

Low harmonic distortion of the complete amplifier is achieved by designing each amplifier stage to give low distortion and to have a linear input resistance.

The relatively large open-loop bandwidth should provide low transient intermodulation distortion.

Thermal stability of the output stage bias current is kept very high by thermally coupling each emitter follower with the output transistor that it drives. For that purpose, driver and output transistors should be in identical suitable packages (e.g: TO-126, TO-220, SOT-93, etc.) and be of complementary types.

Circuit description

For convenience, the amplifier is split into two parts: a voltage amplifier and a voltage follower (the output stage). The voltage amplifier is built on a printed circuit board, while all transistors of the voltage follower are mounted on one heatsink, Philips or Mullard type 35D2CB(÷3.4°C/W). Fastened to the heatsink is a small printed circuit board containing the passive components of the voltage follower. Thus the two parts can be tested separately.

A patent covering the circuitry of the amplifier has been applied for.

Voltage amplifier

The circuit of the voltage amplifier is shown in Figure 1.

Transistors Q2 and Q3 are connected as a long tailed pair and are preceded by emitter followers Q1 and Q4. This circuit was chosen because of its inherent low distortion and also because the complete power amplifier is connected as an operational amplifier. This circuit is very suitable as a first stage for such use.

The complete power amplifier is an inverting one, the virtual earth point being at the base of transistor Q1. For this reason the input signal seen by transistor Q1 is extremely small and distortion due to large common mode voltages encountered in some circuits is avoided. Because distortions in transistors Q2 and Q3 cancel to a large extent and the input voltage is very low, the long tailed pair does not need local feedback in the form of emitter resistors.

The input resistance is determined by resistor R1 (22k). Capacitor C1 sets the low frequency 3 dB point at 10.6 Hz. The terminal marked 'feedback' is connected to the similarly marked terminal in Figure 2. Capacitor C4 is chosen to provide a closed-loop high frequency 3 dB point at 48 kHz. Transistor Q1 provides a high input impedance to reduce problems related to offset current and non-linear loading of the virtual earth point. Transistor Q4 serves to compensate for the voltage drift due to temperature variations in transistor Q1. The gain of the first stage is 175.

The emitter followers (Q5 and Q6) prevent the reduction of gain of the first stage due to loading. Of more importance, they reduce the distortion of the voltage amplifier because the load presented by the input resistances of transistors Q7 and Q8 is sufficiently non-linear to cause distortion despite the relatively large emitter resistors R23 and R24 and the push-pull configuration. The emitter followers also

VOLTAGE AMPLIFIER

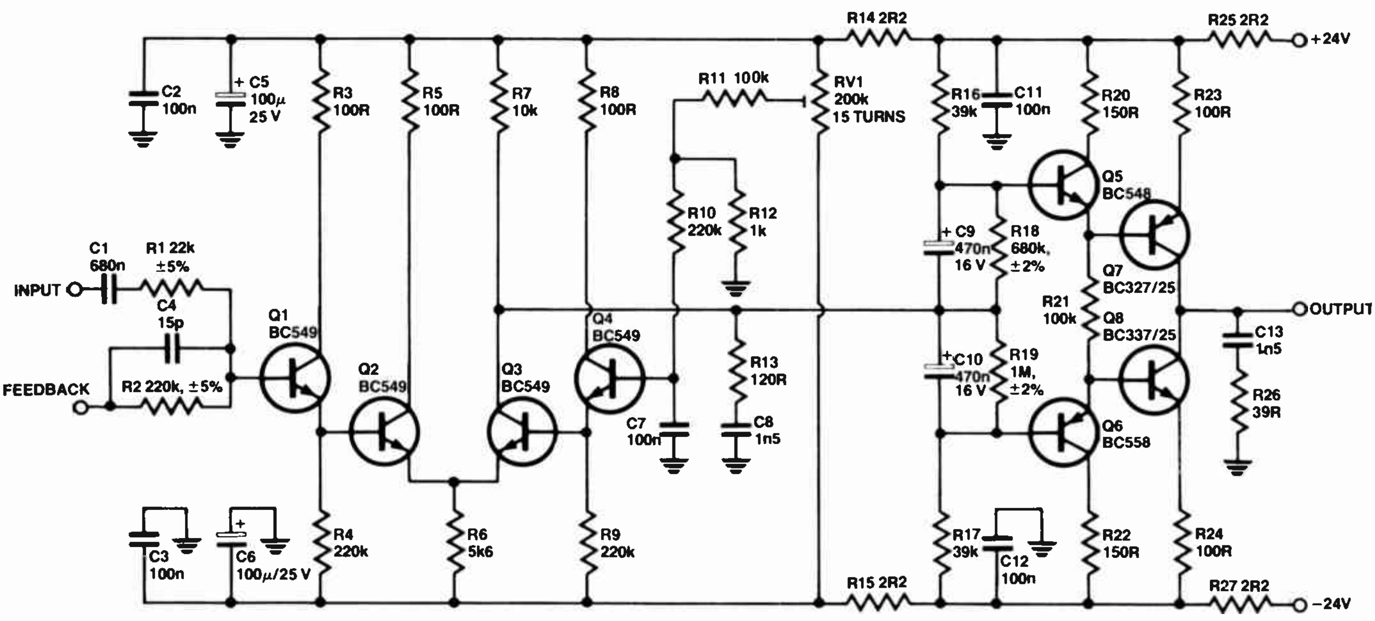


Figure 1. **Voltage amplifier. Note that R1, R2, R18 and R19 are metal film types. Capacitors C9 and C10 are tantalum or plastic foil types.**

PARTS LIST

All resistors used are ±5% ¼W carbon film types unless specified otherwise.

VOLTAGE AMPLIFIER

Resistors

R1	22k
R2,R4,R9,R10	220k
R3,R5,R8, R11, R21	100k
R6	5k6
R7	10k
R12	1k
R13	120R
R14,R15,R25,R27	2R2
R16,R17	39k
R18	680k, ± 2%, metal film
R19	1M, ±2%, metal film
R20,R22	150R
R23, R24	100R
R26	39R
RV1	200k, 15 turns

Capacitors

C1	680n
C2,C3,C7, C11,C12	100n
C4	15p
C5,C6	100u/25 V
C8,C13	1n5
C9,C10	470n/16 V tantalum or plastic foil

Semiconductors

Q1-Q4	BC549
Q5	BC548
Q6	BC558
Q7	BC327/25
Q8	BC337/25

stabilise the quiescent current through transistors Q7 and Q8, because changes of the base to emitter voltages of these transistors, due to variations in ambient temperature, are compensated by the change in the base to emitter voltages of transistors Q5 and Q6. The emitter followers also prevent reduction in bandwidth due to Miller capacitances of transistors Q7 and Q8.

The quiescent current through transistors Q7 and Q8 is approximately 10 mA. At full output, the current in each transistor swings between 5 mA and 15 mA. A higher quiescent current would result in lower distortion; however, as medium-power transistors with a relatively high f_T should be used, a compromise of 10 mA was made.

Harmonic distortion in the second amplifier stage is very low due to the fact that it is a push-pull amplifier and the emitter resistors R23 and R24 are of relatively high value.

Capacitors C8 and C13 are for frequency compensation. With the chosen values, the open-loop high frequency 3 dB point of the complete amplifier is 15 kHz. The resistors R13 and R26 reduce the phase shift at those high frequencies at which the reactance of the capacitors becomes very low, thus increasing the stability. Without these resistors, the capacitors must have a much higher value.

The second stage is coupled to the first via two capacitors C9 and C10 and via the dc attenuator, consisting of resistors R16, R17, R18 and R19. When the zero set control, RV1, is set for zero output voltage, the voltage at the collector of transistor Q3 is approximately 4 V and the collector currents of transistors Q2 and Q3 are approximately equal, due to different values being used for resistors R18 and R19. Owing to the use of the dc attenuator, the zero set control can be used for the whole amplifier and the setting is very uncritical. If the base-emitter voltages of transistors Q1, Q2, Q3 and Q4 are roughly matched, no zero set control is required, except when trouble-shooting with the feedback loop removed.

Components R14, R15, R25, R27, C2, C3, C11 and C12 prevent high frequency oscillation due to parasitic coupling via the supply leads. Electrolytic capacitors C5 and C6 are employed for decoupling purposes.

The gain of the voltage amplifier when loaded with the voltage follower is 5000. With the compensation capacitors C8 and C13 removed, the high frequency 3 dB point of the first stage is 220 kHz and that of the complete voltage amplifier 180 kHz when loaded with the voltage follower.

Resistors R3, R5, R8, R20 and R22 are used to stop any possible oscillation due to lead inductance, to limit collector currents in case of breakdown, and to aid in trouble-shooting.

Voltage follower

The circuit for this stage (output) is shown in Figure 2.

Output transistors Q5 and Q6 are driven by emitter followers Q1 and Q3, which are also loaded by constant current loads (Q2 and Q4). This produces source resistances for the output transistors which at all times (even when the output transistors are switched off) are very low. The constant currents selected by resistors R5, R6, R7, R8 and R9 should be higher than the worst-case peak base currents required ▶

by the output transistors. The low source resistances result in a lower harmonic distortion and less crossover distortion at high frequencies because of reduced switching time.

The emitter follower transistors are complementary to the output transistors they drive to avoid output power reduction due to the base-to-emitter voltage drops and for temperature compensation.

With no input signal, the voltage across the bias trimming potentiometer, RV1, is almost identical to the voltage between the emitters of the output transistors. Transistors Q3 and Q5 are mounted on the heatsink using a single screw. Transistor Q5 is insulated from the heatsink by a mica washer and the tab of transistor Q3 is mounted on the tab of transistor Q5, insulated by a mica washer. All mica washers are smeared with thermally conducting paste. Therefore the junction temperature of transistor Q3 follows that of transistor Q5. Transistors with a hole in them (TO-126) can be mounted together with the metal side of the emitter followers fitted against the insulated side of the output transistor. No mica washer is required, but the use of thermally conducting paste improves the thermal coupling.

The ideal bias current for an output transistor is a current that produces 25 mV across each emitter resistor. Since this emitter resistor is one-quarter ohm, the bias current should be 100 mA. However, since the emitters of the transistors will have some small ohmic resistance, a 90 mA bias current was chosen. The bias current value is not critical at all.

The components L1, C3, R14 and R15 are the normal output stabilising components. The feedback is taken from the junction of resistors R12 and R13. These resistors are used to avoid using an unsuitably high value for resistor R2 in Figure 1. When both resistors are equal (say 4k7) the closed loop gain is 20.

The resistive network consisting of resistors R1, R2, R3 and R4 and trimming potentiometer RV1 provides a load to the voltage amplifier which is much lower than the relatively high but slightly non-linear input resistance of the emitter followers (Q1 and Q3). This keeps the distortion low.

Conclusion

A power amplifier designed according to the circuit as described in this article has an extremely high performance. This was confirmed by the results of tests on two amplifiers as described and two 15 W amplifiers employing TO-126 output transistors BD235 and BD236. The results for each amplifier were similar and agreed with the given test results. ●

ACKNOWLEDGEMENT

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TEST RESULTS

Open-loop gain:	5000
Closed-loop gain:	Minimum 10 (with circuit values given 20)
Open-loop frequency response:	10 Hz to 15 kHz
Closed-loop frequency response:	10 Hz to 48 kHz
Harmonic distortion at 1 kHz, 10x:	0.001%
1 kHz, 20x:	0.002%
1 kHz, 100x:	0.01%
1 kHz, 5000x:	0.5%
10 kHz, 10x:	0.004%
10 kHz, 20x:	0.008%
10 kHz, 100x:	0.04%
10 kHz, 5000x:	1%

All distortion figures were measured at just below the onset of clipping, which was at 25 W or slightly higher, with a load resistance of 8 ohms. As output power is reduced, the distortion goes quickly below measurable levels.

Square-wave response:	Rise and fall times are 7 μs. The shape is perfect with no overshoot or ringing with a capacitive load between 0 and 0.47 μF.
Bias current stability:	Maximum change in bias current is ±1.5 mA or ±1.7%.
Test conditions:	1 hour at zero output power followed by 1 hour at 10 W output power (maximum dissipation of output transistors).

VOLTAGE FOLLOWER

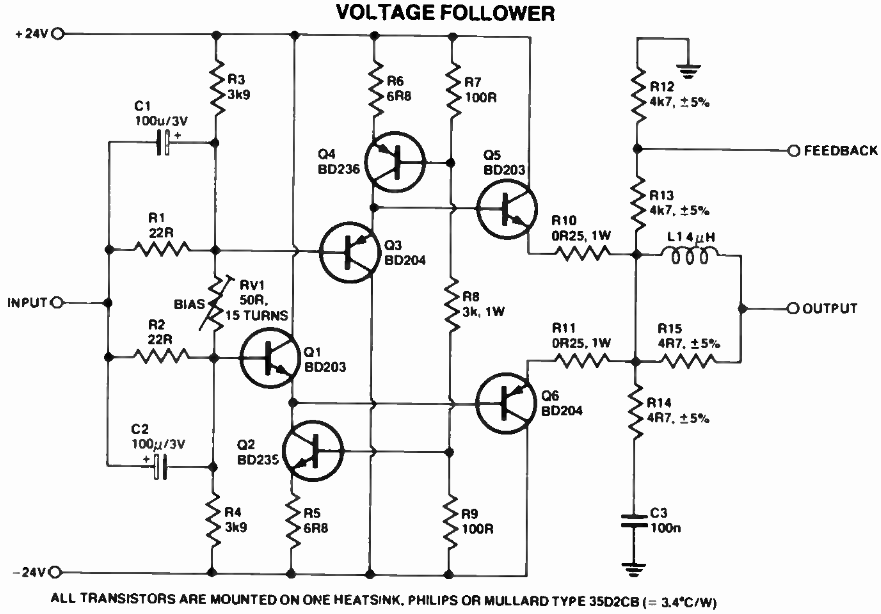

Figure 2. **The voltage follower stage. Note that R12 and R13 are metal film types, while R14 and R15 are carbon film types.**

VOLTAGE FOLLOWER

All resistors used are ±5% ¼W carbon film types unless specified otherwise.

Resistors
- R1,R2 .... 22R
- R3,R4 .... 3k9
- R5,R6 .... 6R8
- R7,R9 .... 100R
- R8 .... 3k, 1 W
- R10,R11 .... 0R25, 1 W
- R12,R13 .... 4k7, ±5%, metal film
- R14,R15 .... 4R7, ±5%, 1 W, carbon film
- RV1 .... 50R, 15 turns

Capacitors
- C1,C2 .... 100u/3 V
- C3 .... 100n

Semiconductors
- Q1,Q5 .... BD203
- Q2 .... BD235
- Q3,Q6 .... BD204
- Q4 .... BD236

All transistors are mounted on one heatsink, Philips or Mullard type 35D2CB (≑ 3.4°C/W).

Miscellaneous
- L1 .... 4 uH

Shoparound

THIS PAGE is to assist readers in the search for components and printed circuit boards for the projects in this book. Where companies are mentioned only by name, we have listed their complete addresses at the end of this section.

ETI-466/467 300 W amp & preamp

These two are stocked as kits by Dick Smith Electronics, Rod Irving Electronics, All Electronic Components and Electronic Agencies.

ETI-476 Series 3000 stereo amp

This project is stocked in kit form by Rod Irving Electronics and All Electronic Components.

Series 4000 stereo amp

This series comprises 470/471/472 and complete kits, as well as kits of the individual items, can be obtained from All Electronic Components, Rod Irving Electronics and Electronic Agencies.

ETI-473/577 MC preamp

This is the Series 4000 moving coil preamp. Kits are obtainable from Jaycar, Rod Irving Electronics and All Electronic Components.

ETI-474 interface board

Kits for this one are available from All Electronic Components.

ETI-496/497 Series 4000 speakers

Kits for these, with boxes available separately, can be obtained from Electronic Agencies, Jaycar, Rod Irving Electronics and All Electronic Components.

ETI-451 hum filter

This one is stocked as a kit by All Electronic Components.

ETI-453 general purpose amp

In kit form, this is stocked by Rod Irving Electronics, All Electronic Components and Jaycar. It also goes under the guise of the Hobby Electronics HE 105 Bench Amplifier.

ETI-492 Sound bender

This popular project is stocked by Dick Smith Electronics stores, Electronic Agencies, Rod Irving Electronics, All Electronics Components and probably quite a few we don't know about.

ETI-475 AM tuner

This project is stocked as a kit by Rod Irving Electronics and All Electronic Components.

ETI-457 Scratch & rumble filter

The only kit supplier for this project is All Electronic Components.

ETI-155 dummy loads

You'll find kits for one form or another of these dummy loads at Rod Irving Electronics and All Electronic Components. The Arcol resistors are available from Everest Electronics, 61 Compass Drive, Seaford SA 5169. Beyschlag 1% resistors are available from Crusader Electronic Components, 81 Princes Hwy, St Peters NSW.

ETI-150 freq. meter

All Electronic Components appear to be the only stockists of a kit for this project.

ETI-498/499 P.A. amp

This comprises the '499 150 W MOSFET amp modules and the '498 preamp. Both are widely stocked as kits in various forms. Try these firms: Altronics, Jaycar, Dick Smith Electronics, Electronic Agencies, Rod Irving Electronics, All Electronic Components.

ETI-596 white noise gen.

It seems only two firms stock this as a kit: All Electronic Components and Rod Irving Electronics.

ETI-1509 dc-dc inverter

Power any of the audio amp modules from a 12-24-32 V source. Kits for this one are available from Dick Smith Electronics, Rod Irving Electronics, All Electronic Components, Electronic Agencies and Altronics.

ETI-494 speaker protector

Kits are available from Altronics, Electronic Agencies, Jaycar, Rod Irving Electronics and All Electronic Components.

PC boards and panels

All the pc boards and sometimes front panels for the projects published in this issue are available from the following suppliers:

All Electronic Components
118 Lonsdale St, Melbourne Vic. 3000

Electronic Agencies
115-117 Parramatta Rd, Concord NSW 2137
and
117 York St, Sydney

James Photronics
522 Grange Rd, Fulham Gardens SA 5025

Jemal Products
P.O. Box 168, Victoria Park WA 6100

Mini Tech, P.O. Box 9094, Auckland NZ

Radio Despatch Service
869 George St, Sydney NSW 2000

Rod Irving Electronics
425 High St, Northcote Vic. 3070

RCS Radio
651 Forest Rd, Bexley NSW 2207

Sunbury Printed Circuits
Lot 14, Factory 3,
McDougall Rd, Sunbury Vic. 3429

Panels are obtainable from:

E.D.C.
17 Elizabeth Ave, Dulwich Hill NSW 2203

Kits & Components

All Electronic Components
118 Lonsdale St
Melbourne Vic. 3000

Altronics
105 Stirling St
Perth WA 6000

David Reid Electronics
127 York St
Sydney NSW 2000

Electronic Agencies
115-117 Parramatta Rd
Concord NSW 2137

Ellistronics
289 Latrobe St
Melbourne Vic. 3000

Dick Smith Electronics
Mail Order
P.O. Box 321
North Ryde NSW 2113

Jaycar
125 York St
Sydney NSW 2000
and
Cnr Carlingford & Pennant Hills Rds
Carlingford NSW

Kalextronics
101 Burgundy St
Heidelberg Vic. 3084

Magraths
208 Little Lonsdale St
Melbourne Vic. 3000

Radio Despatch Service
869 George St
Sydney NSW 2000

Radio Parts
562 Spencer St
West Melbourne Vic. 3003