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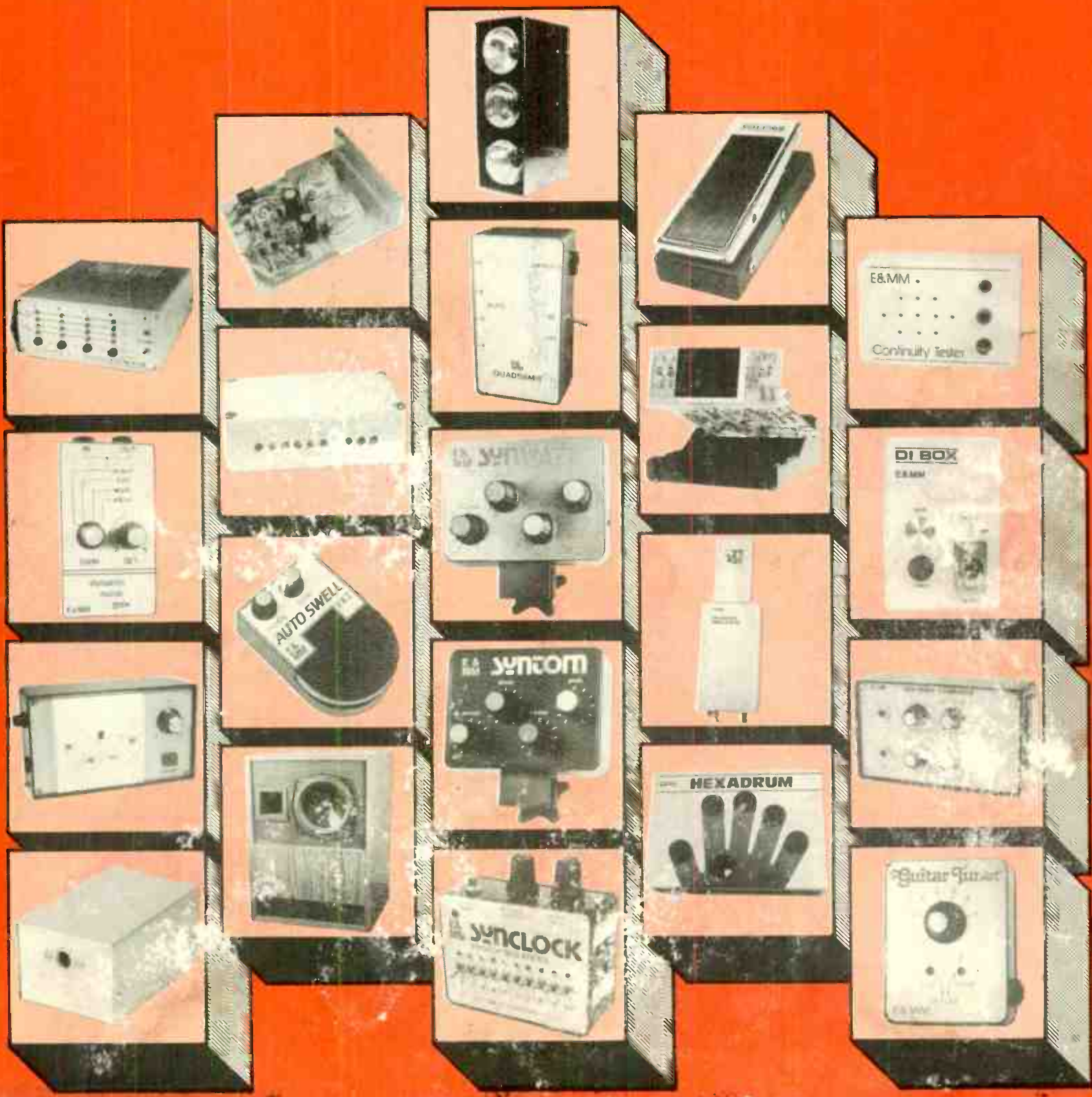
PROJECTS VOLUME 1



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Electronics & Music Maker

Here, at last, we bring together the very best of the smaller projects that appeared during the first year of Electronics & Music Maker. Although E&MM now almost exclusively publish music projects, during their first year the projects covered a much wider sphere and we have selected here a representative range of those projects.
The complexity of the projects ranges from those suitable for the beginner like the Car Battery Monitor, to those suitable for the more experienced constructor, like the superb Noise Reduction Unit. On some of the projects, kits are available which saves listing all the parts separately on your order. Here then is a book of excellent tried and tested projects with all the information you need to build them successfully. We hope you have many happy hours building and using these projects.

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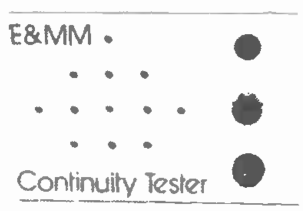

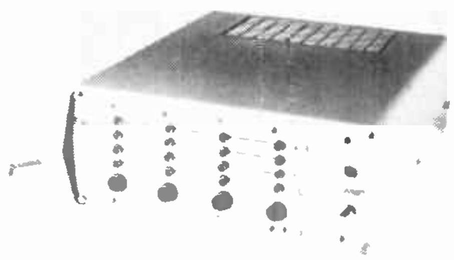


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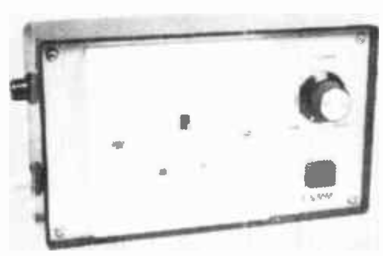
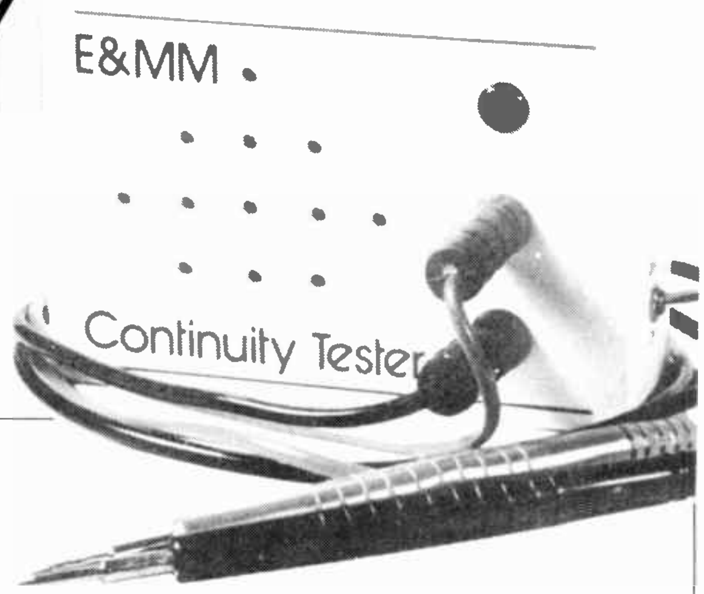


Published by Maplin Electronic Supplies Ltd, P.O. Box 3, Rayleigh, Essex. **Printed by** Mayhew McCrimmon Ltd. **Typeset by** Quillset Typesetting.

CONTINUITY TESTER

by Robert Penfold



A continuity tester is a very useful gadget around the home, workshop and on stage – it is invaluable for testing fuses, jack leads, speakers, semiconductor junctions, printed circuit boards, power transistor/heatsink insulation and a multitude of other potential sites of trouble. Like most testers, this design gives an audio indication of continuity, and has the advantage of two modes of operation, giving increased versatility particularly for printed circuit board checking.

A problem that is often encountered when testing for short circuits on component boards is that a semiconductor junction (which can be a diode or part of a transistor or integrated circuit) connected across tracks to be tested could give a false alarm. When forward biased there is a voltage drop of about 0.6 volts across the junction, but this drop is not normally sufficient to prevent the tester from operating and indicating continuity. Though false alarms of this type can often be checked by reversing the test probes (ineffective in circuits where there are two junctions connected 'back to back') this tester can operate such that continuity will only be indicated if the voltage drop across the test probes is less than about 0.5 volts, avoiding misleading results due to forward biased semiconductor junctions.

The Circuit

The circuit diagram of the continuity tester appears in Fig. 1; it is basically just an audio oscillator feeding a loudspeaker.

IC1 is an audio power amplifier device, and it is made to oscillate at a frequency of several hundred Hertz by applying positive feedback through R3, C1 and R1. R2 reduces the amount of feedback somewhat, and prevents the oscillations from becoming unstable. C2 couples the output signal to a high impedance loudspeaker.

With S1 in the 'off' (open) position the unit can be used as a simple continuity tester for leads, fuses, semiconductors etc. and shorting the two test probes together will connect the supply to the circuit producing an audio tone. L.E.D. indicator D1 will also turn on giving a visual indication of continuity. When using the unit for checking semiconductors, connecting the junction one way round will result in it blocking the supply and preventing oscillation, while connecting it the other way round produces conduction and an audio output tone from the unit. The device is open circuit if oscillation cannot be obtained, or closed circuit if both methods of connection produce oscillation.

If S1 is switched to the 'on' position, power will be applied to the oscillator circuit via the base -

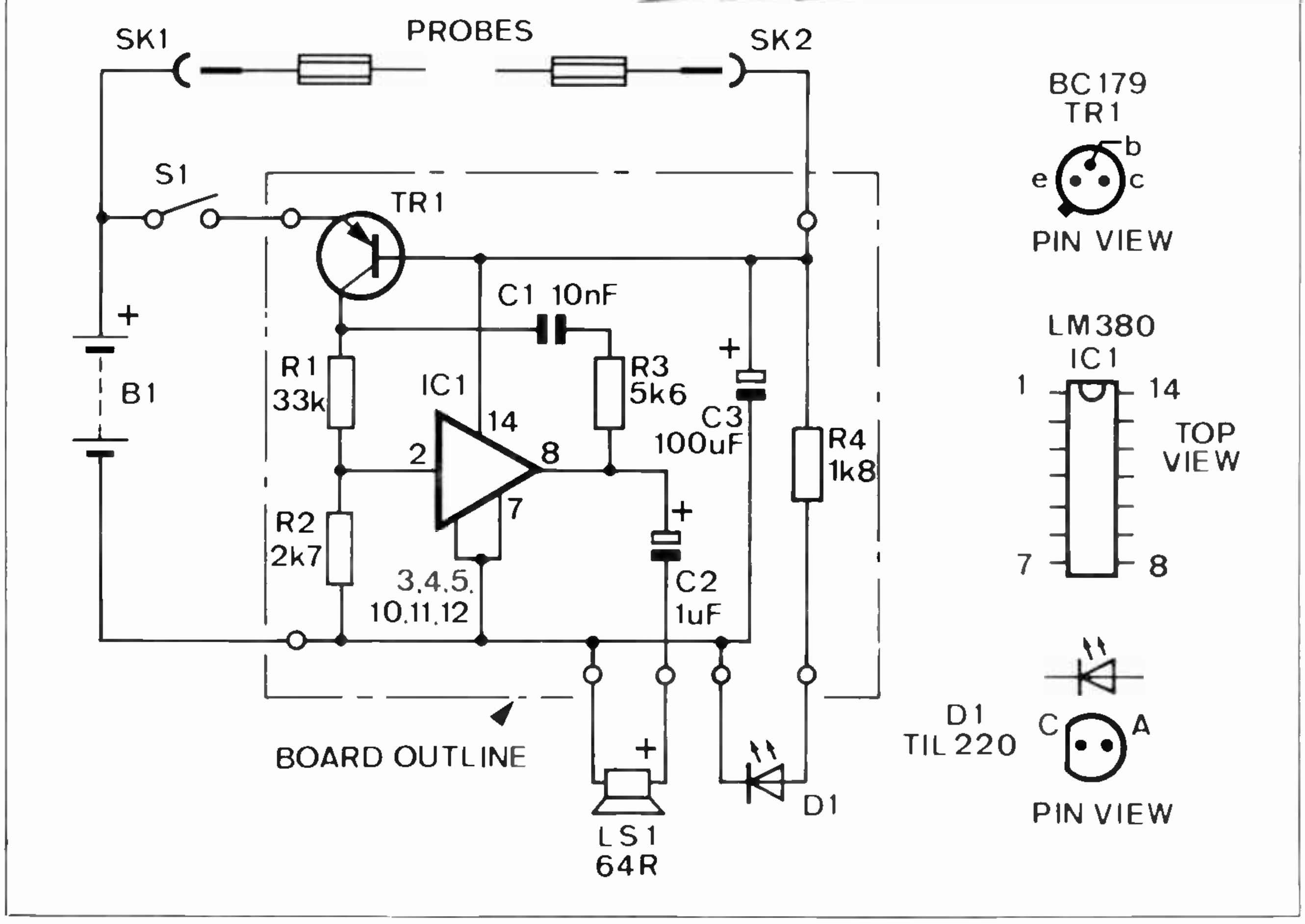
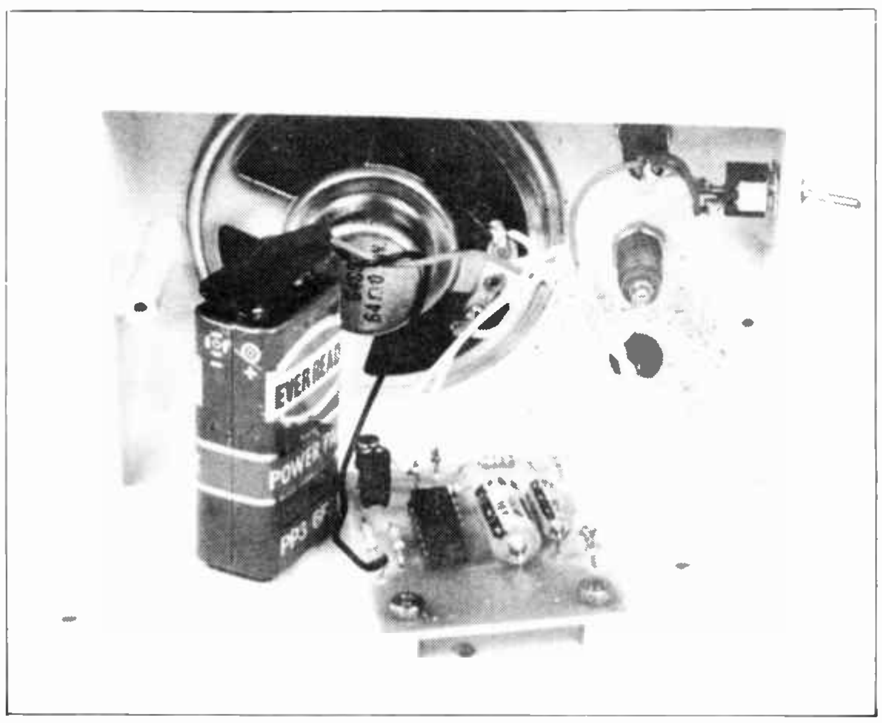


Figure 1. Circuit Diagram of the Continuity Tester



Internal view of Continuity Tester.

emitter junction of Tr1. However, the circuit will not oscillate as the current flowing in the base - emitter circuit of Tr1 switches this device hard on. It therefore obstructs the feedback, and also feeds a small D.C. potential to the input of IC1 so that it is not biased correctly. D1 now operates as an on/off indicator and helps to prevent the unit being inadvertently left switched on.

If the test probes are short circuited, Tr1 becomes switched off by the short circuit across its base-emitter terminals, and the oscillator is able to function normally. A certain amount of resistance across the test probes is also sufficient to switch off Tr1 and produce oscillation. A forward biased silicon junction will give a voltage of between about 0.5 volts and 0.7 volts (depending upon the exact type) across the base-emitter terminals of Tr1. As Tr1 is a high gain device, even the lower figure is sufficient to maintain the device in a state of conduction and block the oscillator. Thus the unit will not respond to forward biased silicon junctions.

Germanium semiconductor junctions have a lower forward voltage drop than silicon types, so a forward biased germanium junction connected across the test probes will switch off Tr1 and give an audio tone from the unit. This is not too important as most germanium devices are now obsolete and little used, but Tr1 can be replaced by a germanium p.n.p. device (OC72, OC81, OC81D, AC128, etc.) if it is likely that the unit will be used to test equipment employing germanium devices.

The current consumption of the circuit is only about 10 to 15mA, and this is provided by a small (PP3 size 9 volt battery.

Construction

A plastic box measuring about 114 x 76 x 38mm (type PB1) makes an ideal housing for the unit. A speaker grille must be made in the left hand side of the front panel, and this can merely consist of a pattern of holes about 4mm. in diameter. D1, SK1, and SK2 are mounted in a vertical line down the right hand side of the panel, and S1 is fitted on the right hand side of the case. The speaker can then be fitted in place, using epoxy-type glue as there is no provision for screw fixing on miniature speakers. Fit the loudspeaker in a position leaving sufficient space for the PP3 battery to fit next to it.

The other components are fitted onto a small printed circuit board which is detailed in Fig. 2, and construction of this is quite straight forward. The completed board is wired to the rest of the unit before being mounted in the case, and Fig. 2 also shows this wiring. 6BA or M3 fixings are used to mount the board on the rear panel of the case so that it fits in the space to the right of the speaker and battery. The printed circuit board can be used as a template when marking the positions of the two 3.3mm diameter mounting holes in the rear panel.

After giving all the wiring a couple of thorough final checks, the unit is then ready for testing and use

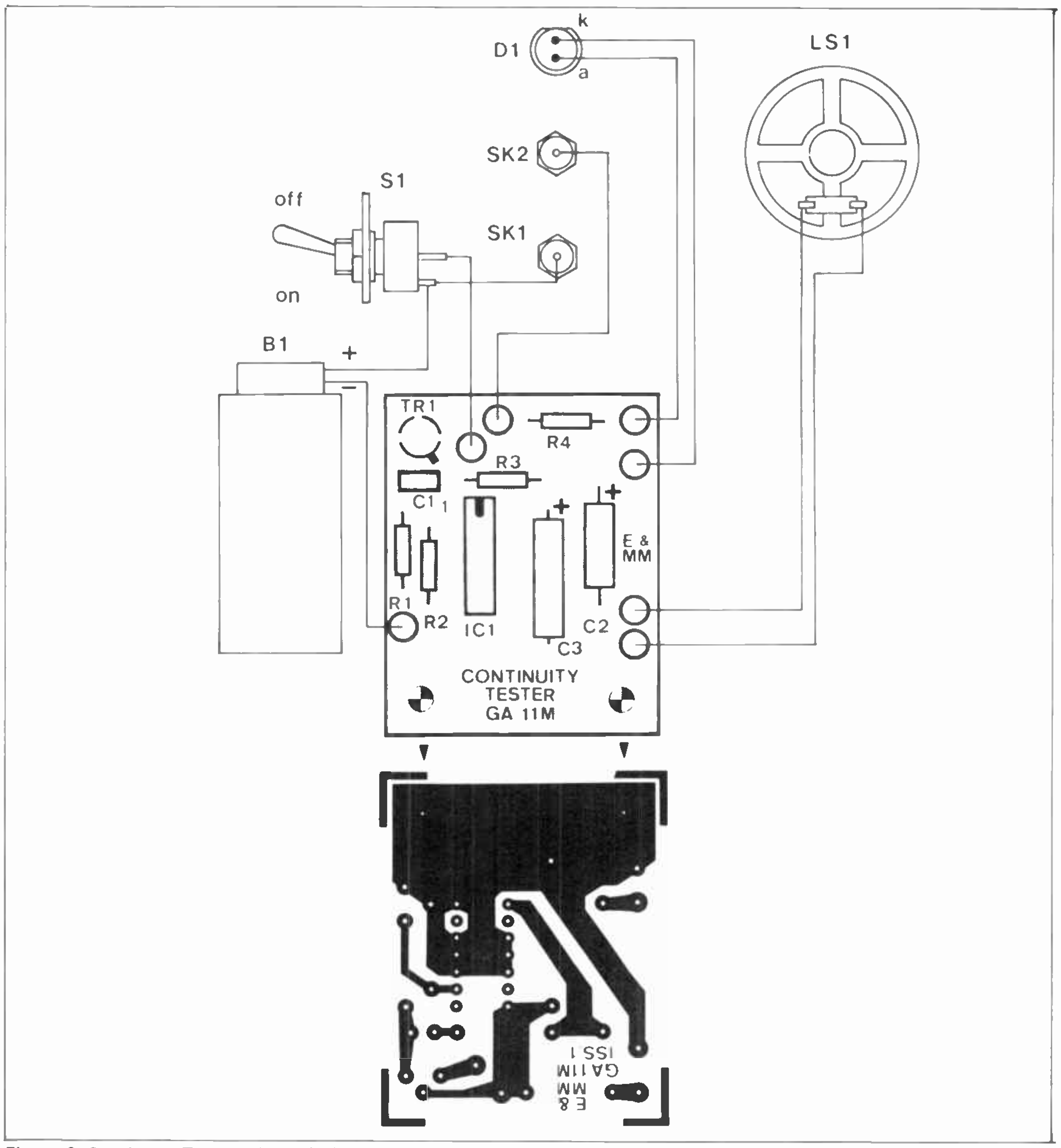
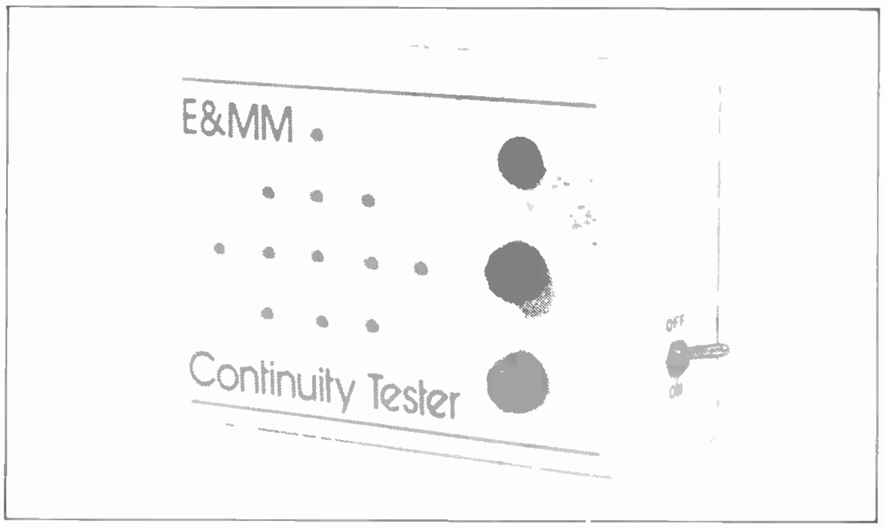


Figure 2. Continuity Tester PCB & Wiring

PARTS LIST

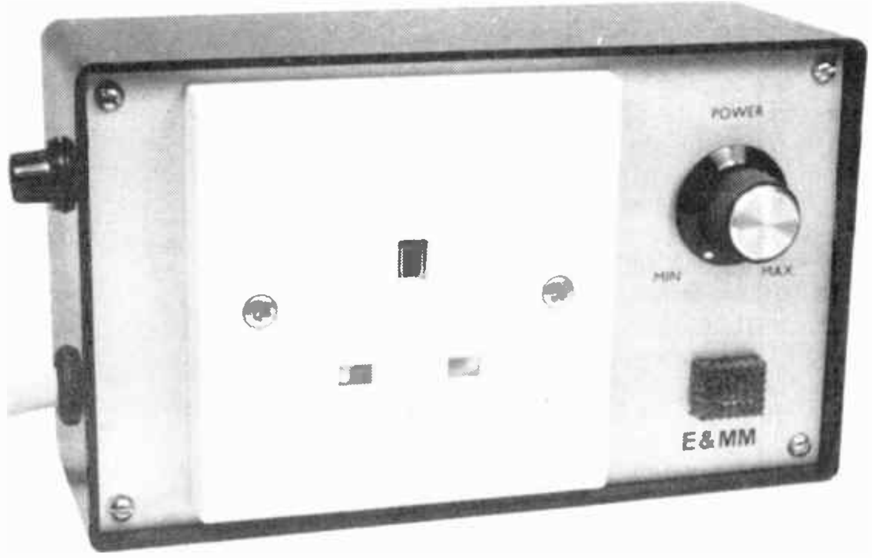
Resistors - all 1% 0.4W metal film		
R1	33k	(M33K)
R2	2k7	(M2K7)
R3	5k6	(M5K6)
R4	1k8	(M1K8)
Capacitors		
C1	10nF mylar	(WW18U)
C2	1uF 63V electrolytic	(FB12N)
C3	100uF 10V electrolytic	(FB48C)
Semiconductors		
IC1	LM380	(QH40T)
TR1	BC179	(QB54J)
D1	TIL220 (0.2 in. red LED)	(WL27E)
Miscellaneous		
S1	Ultra-miniature SPST toggle switch	(FH97F)
LS1	Loudspeaker 66mm diameter 64R impedance	(WF57M)
SK1	Red 4mm socket	(HF73Q)
SK2	Black 4mm socket	(HF69A)
	Test probe pair	(HF33L)
B1	PP3 battery	
	PP3 connector	(HF28F)
	Printed circuit board	(GA11M)
	LED Cover	(YH54J)
	Case PB1 (or similar)	(LF01B)
	Insulated hookup wire	(BL00A)
	6 BA ½" bolts	(BF06G)
	6 BA nuts	(BF18U)



POWER CONTROLLER

by R. A. Penfold

- **Dual Purpose**
- **Smooth Action Lighting Control**
- **Useful for Electric Drills and Soldering Irons**
- **Neon Indicator**



This versatile power controller is suitable for use as a lamp dimmer with a standard or table lamp, and can also be used as a drill speed controller. It can handle loads of up to 720 watts, and this is more than sufficient for any normal domestic lamp or electric drill. The controller is easy and convenient to use since there is a mains outlet on top of the unit and the controlled equipment is merely plugged into this.

It is possible to use other items of equipment with the unit, and it could be used as a soldering iron temperature controller for example. It is quite common for soldering irons to have a short bit life due to an excessive operating temperature, and this can be corrected by slightly reducing the power fed to the iron. This gives a bit temperature that is still adequate for efficient soldering, and substantially fewer replacement bits are needed.

An unusual feature of the unit is the precise control it provides at low output power levels. When an ordinary power controller is advanced from zero it tends to give no output until the power control has been advanced some way, and then suddenly operates normally. This controller has additional circuitry which eliminates this effect.

Fluorescent lamps require a special type of dimmer circuit incidentally, and cannot be used with normal power controllers such as the one described here.

Operating Principle

A power controller can simply consist of a high power variable resistor (rheostat) connected in series with the supply, but this results in a lot of wasted power in the resistor which is converted into unwanted heat. A more efficient method is to use a switching circuit where the power fed to the load is controlled by pulsing it full on for a certain proportion of the time.

This method is illustrated by the output waveforms of Figure 2. The waveform shown in Figure 2a is that obtained with the controller set for maximum power. Here power is applied to the load almost at the beginning of each mains half cycle, and is maintained practically until the end of each half cycle. While there is obviously a small loss of power because the very beginning of each mains half cycle has been cut off, this loss is far too small to be of practical significance.

In order to reduce power the mains is not switched through to the load until later in each half cycle. In the waveform of Figure 2b only the second half of each mains pulse is applied to the load, thus giving half power. In the waveform of Figure 2c only the final part of each half cycle is present at the output, giving practically zero output power.

The Circuit

The circuit diagram of the controller is shown in Figure 1 and the circuit is basically a conventional triac-diac type.

The triac (CSR2) is connected in series with the supply to the output, and this device will normally be switched off. It can be made to conduct between the MT1 and MT2 terminals by applying a brief trigger current of around 30mA to the gate terminal, and it will continue to conduct until the end of the mains half cycle when the current through the device falls to zero, and it switches off. CSR1 is a diac connected in the gate circuit of the triac. A diac has similar characteristics to a triac, but it has no gate terminal, and triggers automatically if it is subjected to a potential of more than about 32 volts.

With RV1 set for minimum resistance C2 will charge rapidly via R3 at the beginning of each half cycle, and the voltage across C2 will be virtually equal to the mains voltage. When the charge voltage on C2 reaches the trigger potential of the diac the latter 'fires' and discharges C2 into the triac's gate. This triggers the triac very early in each half cycle, giving virtually full power at the output. Once triggered, the triac short circuits R3, RV1, and C2 so that C2 remains uncharged until the end of the half cycle, and is ready to start from the beginning at the start of the subsequent one.

If RV1 is set for a higher resistance this increases the lag between the charge on C2 and the mains voltage so that the circuit triggers later in each half cycle and gives reduced output power. The circuit will not trigger at all with RV1 at maximum resistance, and it is therefore possible to vary the output power from zero to maximum by means of RV1.

If the resistance of RV1 is such that the diac does not trigger, there is a residual charge left on C2 at the end of the mains half cycle. C2 is subjected to a charge of opposite polarity on the next half cycle, and the residual charge on C2 obviously counteracts the new charge to some extent. It is this that would give poor control when RV1 is advanced from zero power, with RV1 needing to give a fairly low resistance in order to overcome the reverse charge on C2 and charge the latter to the trigger voltage of the diac. Once this happens, C2 is uncharged at the end of each half cycle, and the unit functions normally.

In this circuit the problem is overcome by the inclusion of R2, D1, and D2. On half cycles where the MT2 terminal of CSR2 is

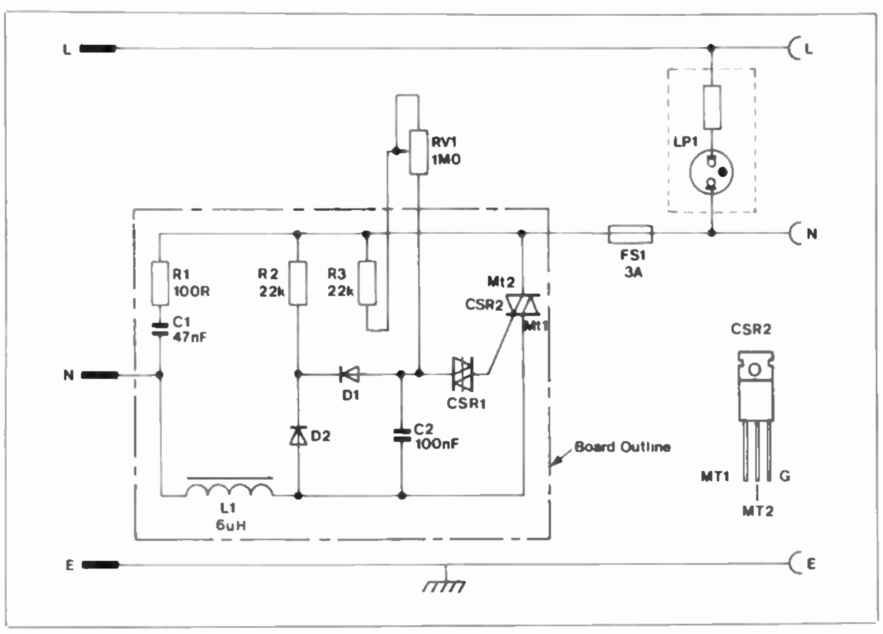


Figure 1. The Circuit Diagram of the Power Controller

negative of its MT1 terminal the additional components have no real effect. A small current will flow through R2 and D2, but this is not sufficient to have any significant affect. About 0.6 volts is produced at the junction of R2, D1, and D2, and the charge on C2 reverse biases D1 so that this has no effect either.

When the supply is of the opposite polarity D2 is reverse biased, and so is D1 initially since the voltage on C2 lags behind the mains voltage. The situation is different towards the end of the half cycle when the mains voltage drops towards zero. C2 then rapidly discharges through D1 and R2 so that the unwanted residual charge is lost. The circuit therefore functions normally on the subsequent half cycle, and the diac will trigger if RV1 has a suitable setting.

Circuits of this type inevitably generate some radio frequency interference due to the rapid rise in the output voltage when the triac triggers. R1, C1, and L1 are included to minimise this interference.

LP1 is a neon indicator lamp with integral resistor which varies in brightness according to the power level fed to the load, and is useful when the unit is used with something like a soldering iron which gives no obvious or immediate indication of the power level it is receiving. Fuse FS1 protects the unit if the output is loaded excessively.

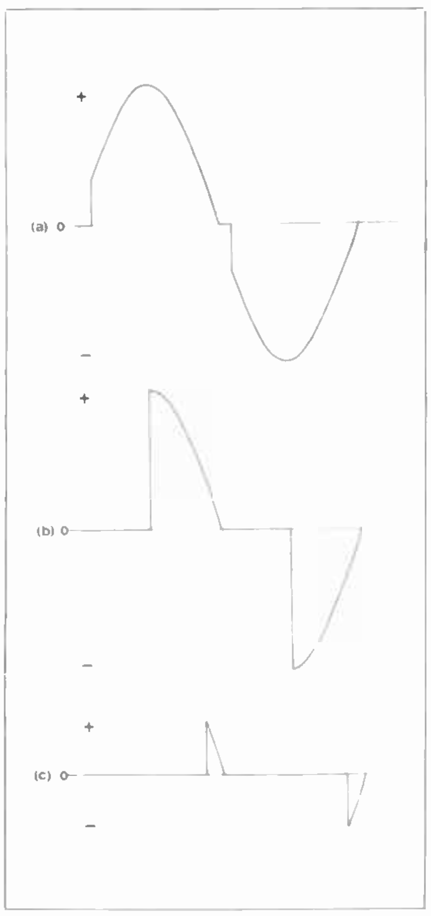


Figure 2. Output Waveforms:
(a) at maximum power
(b) at half power
(c) at virtually minimum power

PARTS LIST

Resistors — all ½W 5% unless specified

R1	100R		(S100R)
R2	22k 3W W/W min.		(W22K)
R3	22k		(S22K)
RV1	1M0 lin. pot.		(FW08J)

Capacitors

C1	47n polyester		(BX74R)
C2	100n carbonate		(WW41U)

Semiconductors

CSR1	ST2 Diac		(QL08J)
CSR2	SC146D Triac		(QL05F)
D1,2	1N4004	2 off	(QL76H)

Miscellaneous

L1	6uH 3 Amp suppression choke		(HW06G)
	3 AMP 20mm quick-blow fuse		(WR06G)
	Plastic case with aluminium panel		(WY02C)
	Printed circuit board		(GA25C)
	Square panel neon (red)		(RX81C)
	Unswitched mains socket		(HL68Y)
	Knob M2		(RW89W)
	20mm panel fuseholder		(RX96E)
	Mains cable, 6A, 2m		(XR04E)
	Vaned heatsink, plastic power		(FL58N)

Construction

A suitable case for the unit is the Metal Panel Box (type M4005) which has outside dimensions of 161 x 96 x 59mm, but any case of about the same size should be satisfactory. RV1 and LP1 are mounted on the right hand side of the top panel, with the mains outlet on the left. A large cutout is required for the outlet, and this can be made by drilling a string of very closely spaced holes around the inside of the perimeter of the cutout. A miniature round file can then be used to join the holes and complete the cutout. The outlet is bolted in place using M4 fixings. The right hand side of the case is drilled to take the mains input lead and the fuseholder.

All the other components are fitted onto a small printed circuit which is detailed in Figure 3. This diagram also shows the other wiring of the unit. The printed circuit is completed in the normal way, and is bolted to the base panel of the case, beneath RV1 once it has been wired to the rest of the unit. Note that the triac must be fitted with a small finned heatsink or the unit will only be able to safely handle loads of up to about 300 watts.

In the interest of safety it is essential that the metal front panel is earthed, and the connection is made by the earth connector and one of the mounting bolts of the mains outlet. The case must be a type that bolts together, and should not be one that could simply be unclipped to reveal mains wiring. RV1 should be a type having a plastic spindle and it should be fitted with a plastic control knob. Do not touch any of the wiring while the unit is plugged into the mains supply.

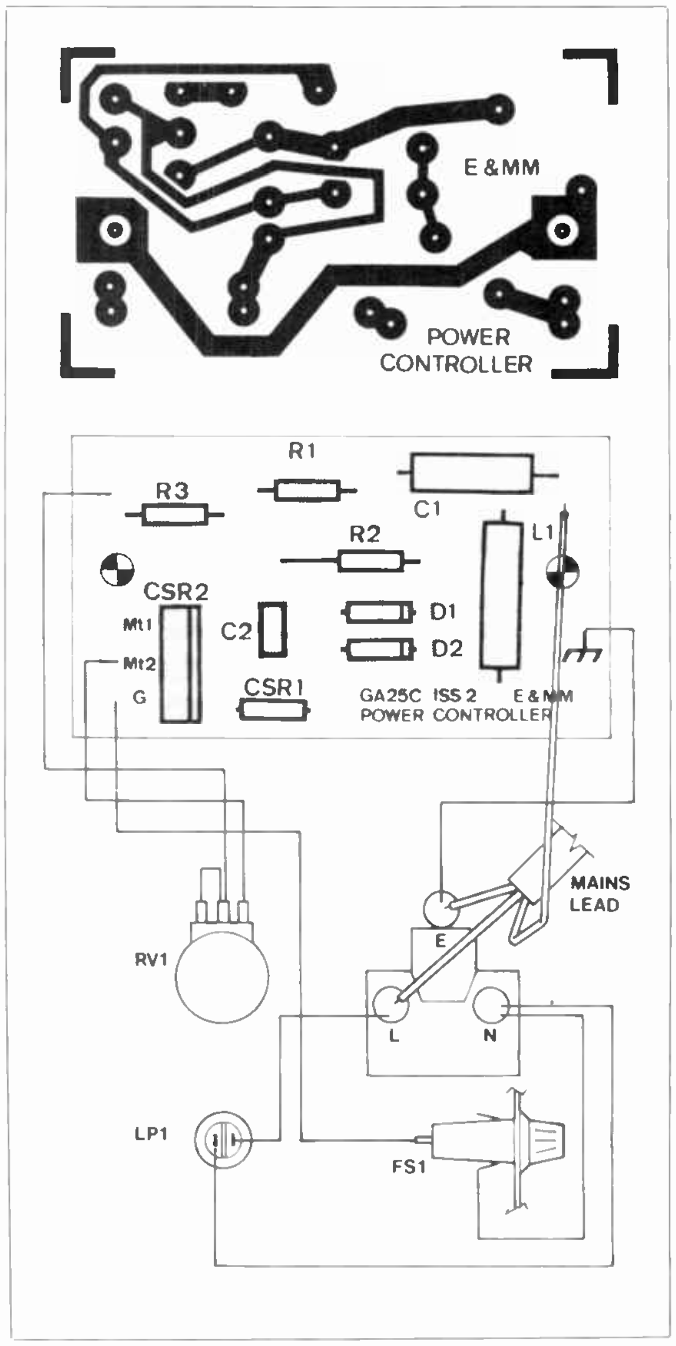
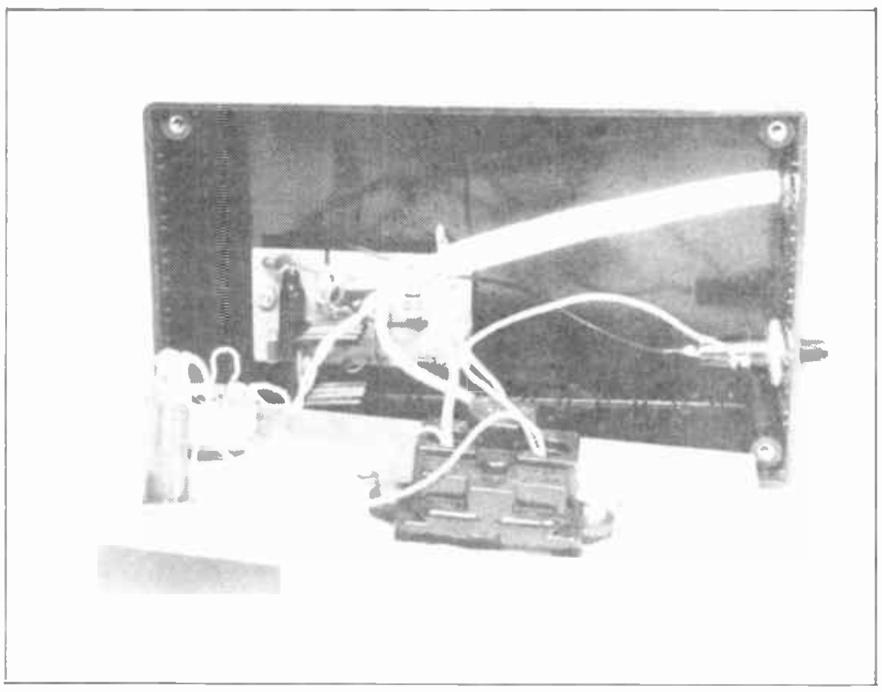


Figure 3. The Power Controller Printed Circuit Board

If the unit is working correctly LP1 will light up at full brightness when the unit is connected to the mains supply (due to the current it receives via R3 and D2), and it will vary in brightness in sympathy with the setting of RV1 when a load is connected to the unit. It may be found that due to component tolerances zero power is achieved some way before RV1 is fully backed-off. If necessary, this can be remedied by adding a 2.7 megohm ⅓ watt resistor in parallel with RV1.



CAR IGNITION TIMING STROBE

by Michael Maurice

An ideal unit for improving your car's engine performance

E&MM

CAR IGNITION TIMING STROBE

In order to gain the maximum efficiency, economy and performance from a petrol engine, the ignition must be set to fire at a certain point in the firing sequence; this is normally just before the piston reaches the top of the compression stroke (known as top dead centre). A few degrees out and the fuel consumption and performance will suffer with possible damage to the engine. An engine in which the ignition is retarded will suffer from lack of power and possibly overheat, while an engine in which ignition is advanced will give off a metallic knocking sound (PINKING). There will also be undue strain put on the pistons and crankshaft bearings, which could eventually lead to expensive engine damage.

One way of setting the ignition timing is to rotate the engine either by hand or on the starter, until the timing marks on the crankshaft pulley or the engine flywheel line up with the corresponding marks on the engine; then rotate the distributor body until the points just open. This is not an accurate method, although useful for initially setting the timing. A more accurate setting can be obtained using the unit described.

The unit has three leads — two connect to the battery and one to the spark plug which is to be used for setting the timing. The strobe emits a flash of light when the spark plug fires and this is used as a basis on which to check and set the timing. Commercial units

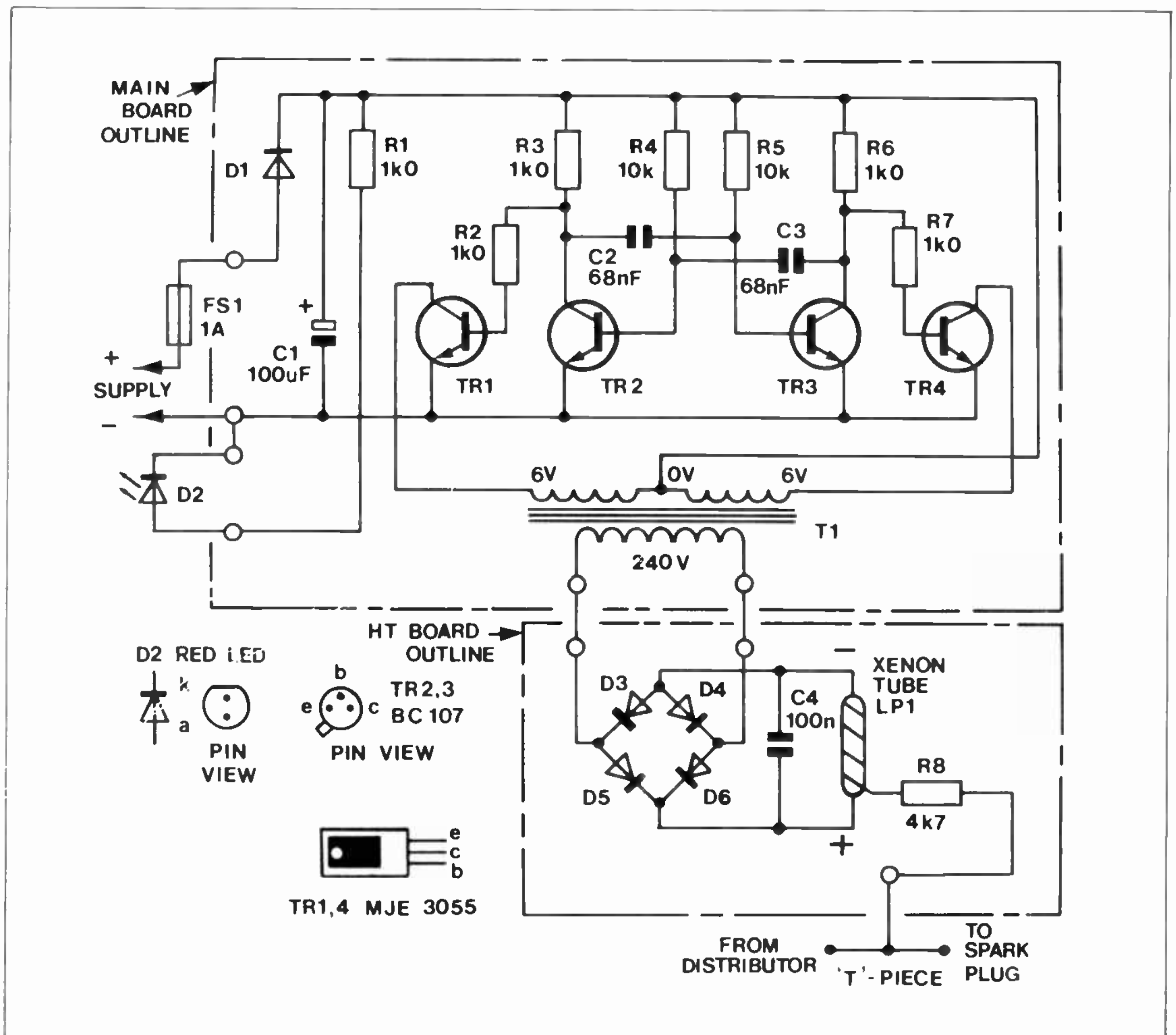


Figure 1. Circuit diagram of the timing strobe.

are available which perform the same function, but they either work off the mains or utilise a neon lamp whose light output is sometimes insufficient. Commercial units which utilise a Xenon strobe tend to be expensive.

Circuit

The unit described is a Xenon tube strobe which runs off the car's 12 volt battery. Figure 1 shows the complete circuit diagram.

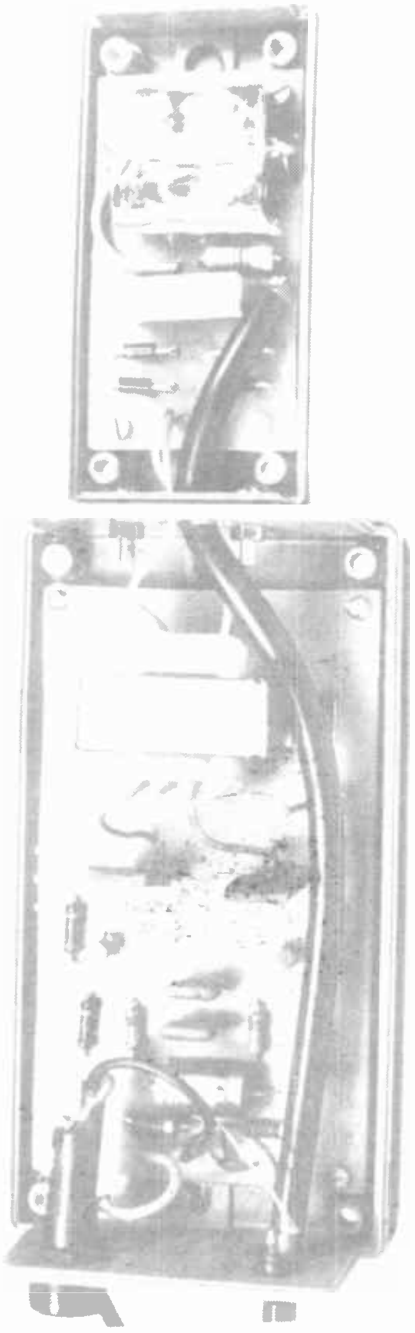
The heart of the circuit is an inverter, this is designed to step up the 12 volts DC from the car battery to approximately 400-500 volts. TR2, TR3, R3, R4, R5, R6, C2 and C3 form a simple multivibrator oscillating at approximately 800Hz. The waveform at the junction of TR2, R3, R2, C2, is 180° out of phase with that at the junction of TR3, R6, R7, C3. TR1 and TR4 are power transistors used in common emitter configuration. The outputs are fed through a 6-0-6V miniature mains transformer wired in reverse. The output from the 240 volt winding is rectified by D3-D6, smoothed by C4, and fed to the Xenon tube. The Xenon tube fires on receiving the spark plug pulse. D1 is used to protect against accidental reverse connection of the supply. D2 is an LED and indicates correct connection to the battery. R8 is used so that in the event of arcing of the HT supply to some other component, the HT to the engine is not interrupted as this would cause the engine to severely misfire and run unevenly.

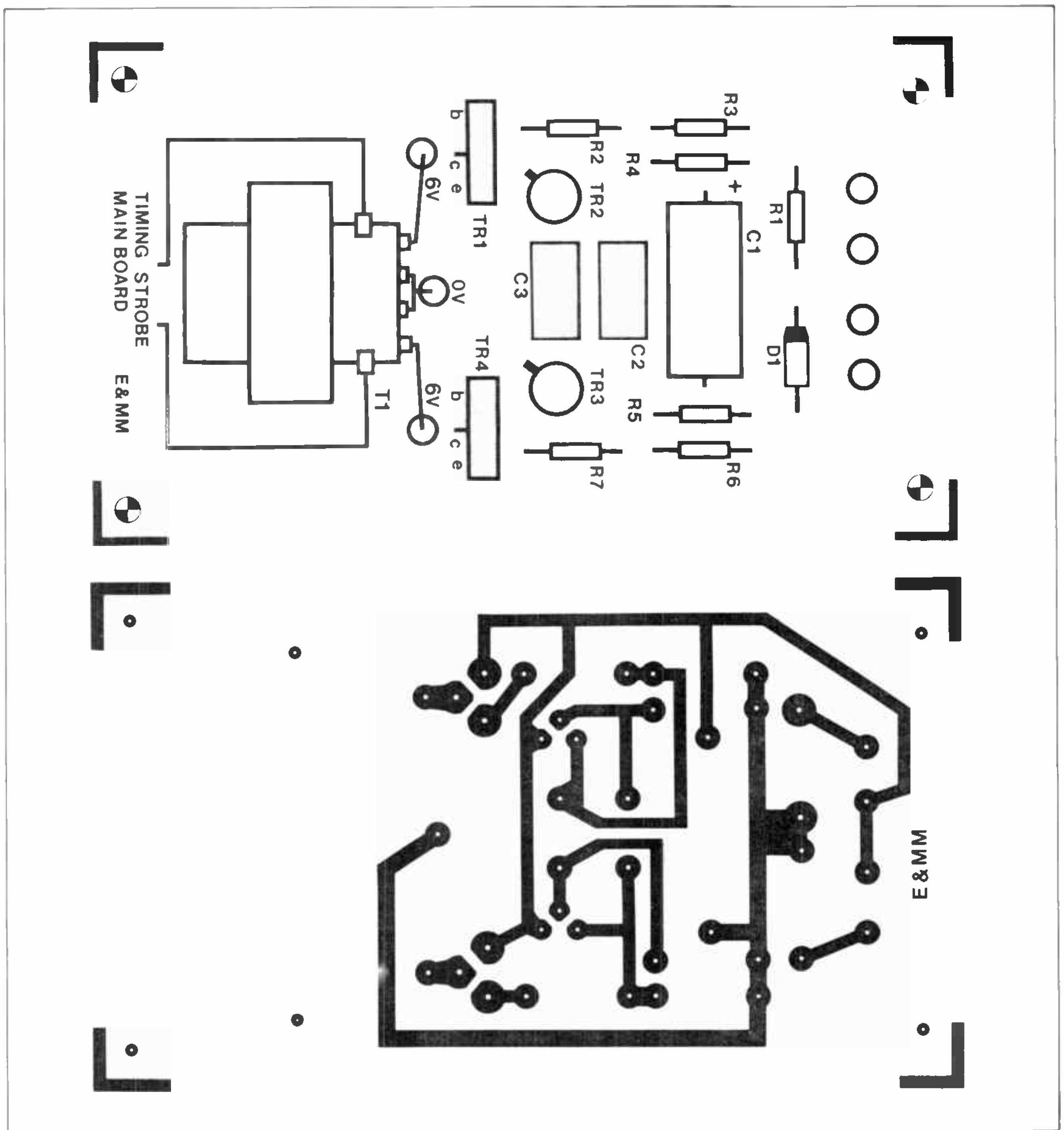
Construction

The Ignition Timing Strobe is built into two veroboxes bolted together. Each verobox has its own PCB. The first box, the bigger of the two, is where the cables enter the unit. The PCB holds the oscillator, power stage and transformer; the second PCB in the smaller box holds the bridge rectifier, the smoothing capacitor, R8, and the Xenon tube. The Xenon tube is mounted within the remains of a flashcube, this is to help in reflecting the light. You are strongly advised to follow the constructional details, in particular using the plastic boxes; the use of metal cases may lead to arcing to the case and electric shocks may occur. Begin by building up the PCB's as shown in Figure 2.

To mount the Xenon tube cut open a used flashcube and cut away one of the four sections of the reflective plastic after removing it from the flashcube base. Cut a hole through the top of the reflective section (where the top of the bulb would have been) to fit the tube. With the bulb removed, mix up some filler as in the manufacturer's instructions. Car body filler was used in the prototype to mount the reflective section securely on the PCB. Allow the filler to dry before proceeding to build up the board. The Xenon tube is mounted as shown in Figure 3; use an insulated wire on the trigger lead and remember to observe polarity.

Cut or drill the holes for the cables, LED, fuseholder and PCB's, and those for bolting the boxes together. Observe the fact that the mounting holes for the





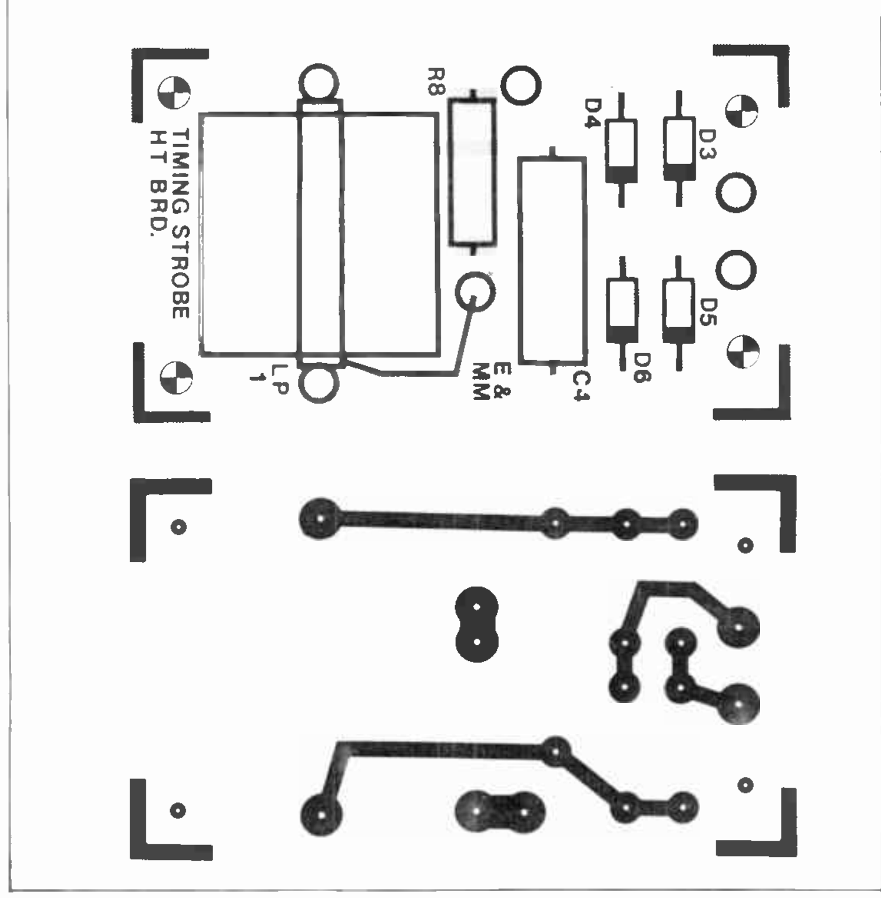


Figure 2. Timing strobe PCB's.

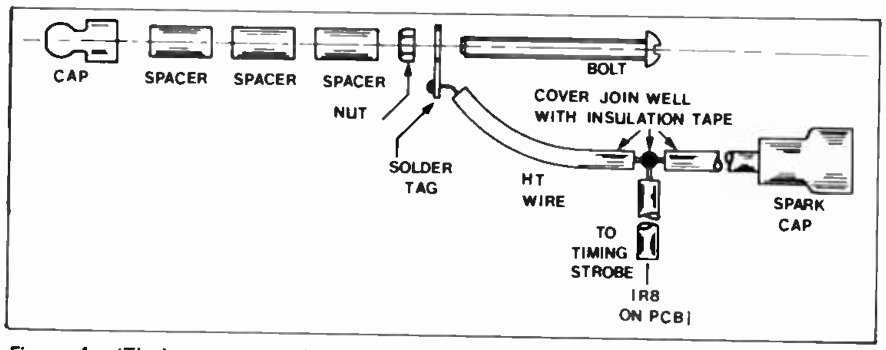


Figure 4. 'T'-piece construction.

PCB are spaced at different distances at each end of the box in both boxes and cut the holes in the end faces accordingly. Now bolt the two boxes together using ½ in. 4 BA bolts and nuts. Assemble the fuse holder and LED to the large case and then mount the PCB's in the boxes. Wire the unit as in Figure 2. If it is desired, the two units can be left separated; just extend the cable from the transformer. You will find it more convenient, however, to build the project up in one unit.

The next constructional step is to build the take-off point for the spark plug. This is essentially a 'T' piece of HT wire with one end made up of spacers, a 1½ in. screw and the cap of a spare spark plug. The other end is fixed on to a spark plug cap and the third end goes to the unit. Solder the wires at the T-junction and cover with insulation tape. One layer is not enough — if necessary use a whole reel. During operation there is 20,000 volts at this point. The assembly diagram is shown in Figure 4.

The final part of construction is to cut the hole in the top of the smaller box for the strobe. When cut cover with a piece of clear acetate or plastic.

Using the unit

Setting the ignition timing on most cars is quite straightforward, but the details vary from one make and model of a car to another. Referring to the manufacturer's handbook, locate the following:—

1. The cylinder plug which is used to determine the timing setting.
2. The appropriate ignition timing marks.
3. The adjustment on the distributor.

Before setting the timing it is recommended to take the distributor cap off the distributor and with a set of feeler gauges check the gap in the points as in the manufacturer's specification. If the points have not been changed for some while then it is always wise to replace them, cost approximately £1. Replace the cap of the distributor in the correct position and see that it is held down tight.

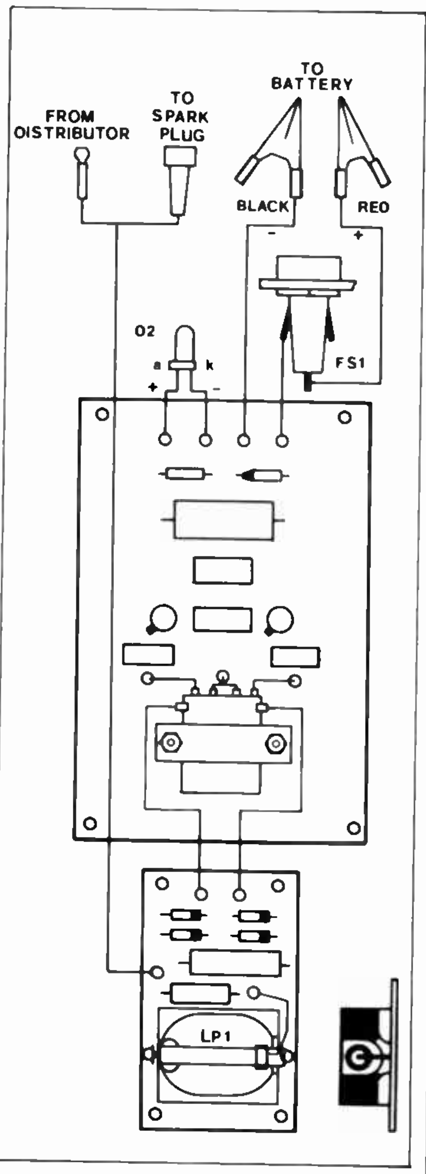


Figure 3. Timing strobe wiring diagram.

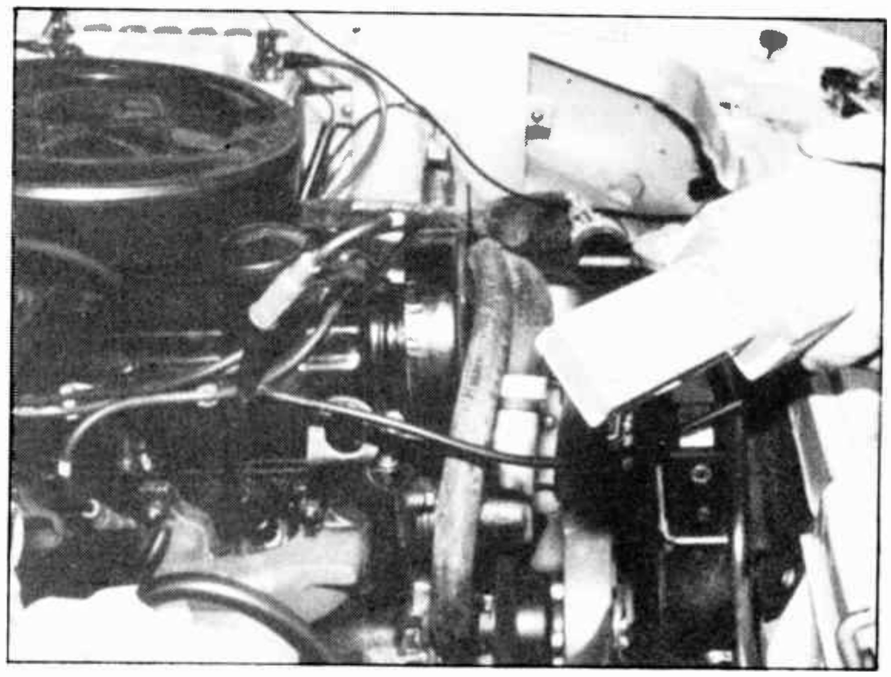
With the engine stopped and ignition off, remove the appropriate HT lead from the spark plug and connect the timing strobes HT lead in its place. Connect the unit to the battery. If all is well, the red LED will come on. A faint whistling noise may be heard from the unit.

Following the manufacturer's instruction start the engine and run it at the recommended speed. Also on some cars it may be necessary to remove the vacuum advance pipe, again this information should be given in the handbook.

Flashes of light should be given out by the strobe, if the engine is running at high revs, the light will appear to be continuous. Shine the light at the timing marks, which should appear to be aligned. Again the timing marks and their location will vary from car to car. On some cars there is a steel ball in the flywheel and this should line up with a recess in the timing aperture. If the timing marks are aligned, the timing is correct and no further adjustment is necessary. If, however, they are not aligned, slacken the nut on the distributor and turn the distributor in the required position until the timing marks are aligned. Switch off the engine, remove the HT leads and replace the HT lead from the distributor on to the spark plug. Disconnect the strobe supply from the battery. Do not remove these leads while the engine is running or you may receive an electric shock.

CAUTION:

The fan may appear to be stationary or revolving slowly when viewed in the light from the timing strobe — this, however, is not the case and injury may result if you put your hand in the vicinity of the fan or fan belt whilst the engine is running.



PARTS LIST

Resistors — all 5% ½W carbon

R1,2,3,6,7	1k0	5 off	(S1K0)
R4,5	10k	2 off	(S10K)
R8	4k7		(S4K7)
Capacitors			
C1	100uF 40V axial elect.		(FB50E)
C2,3	68nF polyester	2 off	(BX75S)
C4	100nF 250V		(FF56L)
Semiconductors			
TR1,4	MJE3055	2 off	(QH56L)
TR2,3	BC107	2 off	(QB31J)
D1	IN5400		(QL81C)
D2	0.2in. LED, red		(WL27E)
D3-6	IN4007	4 off	(QL79L)
LP1	Xenon tube		(YQ62S)
Miscellaneous			
	Verobox 102		(LH01B)
	Verobox 106		(LL03D)
T1	Transformer 240v Prim., 0-6, 0-6v Sec. 6VA		(WB06G)
	Panel fuse holder, 1¼in.		(RX97F)
FS1	Fuse, 1A, 1¼in.		(WR11M)
	0.2in. LED clip		(YY40T)
	4BA ½in. Bolts		(BF03D)
	4BA 1½in. Bolts		(LR52G)
	6BA ¼in. Bolt		(BF05F)
	4BA Nuts		(BF17T)
	4BA Solder tags		(BF28F)
	4BA ½in. spacers		(FW32K)
	Mains cable. 3A, 2-core		(XR47B)
	HT Wire		*
	Used flashcube		
	Spark plug cap		*
	Alligator clip, red		(HF24B)
	Alligator clip, black		(HF23A)
	Main PCB		(GA22Y)
	HT PCB		(GA23A)

**Obtainable from most motorist's shops*

CAR BATTERY MONITOR

by David Hough

Any number of things from a faulty alternator to left-on headlights can result in a flat car battery - and the first you are likely to know about it is when you turn the key one morning and the car won't start! This useful little unit is designed to warn you in advance by displaying the battery's state of charge with a row of ten LED's. The Monitor costs less than a fiver to build, and since it consumes a miserly 20mA, can be left connected directly to the battery.

The Car Battery Monitor will even reveal faults like a slipping fan-belt, which prevent the battery charging but leave the dashboard battery warning light off, and show how the battery is handling the strenuous work of starting the car (it takes 20 minutes of running to put back what a five-second start takes out).

Circuit

The National LM3914 bar-graph IC is used to drive a row of red, orange, and green LED's, indicating the battery charge voltage in ten steps of approx. ½V each from 9V to 14V. The IC contains an input buffer, a potential divider chain, comparators, and an accurate 1.2V reference source. Logic is also included which gives the choice of bar or dot-mode operation - the latter is used in this application. The comparators cause the LED's to light at 0.12V intervals of the input voltage.

TR1 acts as an amplified diode and raises the lower end of the divider chain and the negative terminal of the reference source (pins 4 & 8) to 1.9V. The upper end of the chain (pin 6) is connected to the reference source output (pin 7) and therefore is at about 3.1V. The potential divider formed by R1 and RV1 attenuates the supply voltage and uses it as the signal input to the comparators such that a supply range of 9-14V covers the span of the divider chain and is indicated over the whole of the ten-LED display. The LED brightness is held constant by an internal constant current source.

Construction

Bend the leads of each LED through 90° about 5mm from the body, then solder them in position on the PCB with all the other components. Drill a line of ⅛" holes in the front of the base for the LED's, two in the bottom for mounting the PCB, and one in the back for the supply leads. Mount the PCB using 6BA bolts with ¼" spacers, connect the supply, and after setting up screw the lid on.

Setting Up

Adjust RV2 until the voltage at its wiper is 1.9V, and then adjust RV1 until the end green LED lights with a fully charged battery (alternatively use a variable voltage bench PSU)

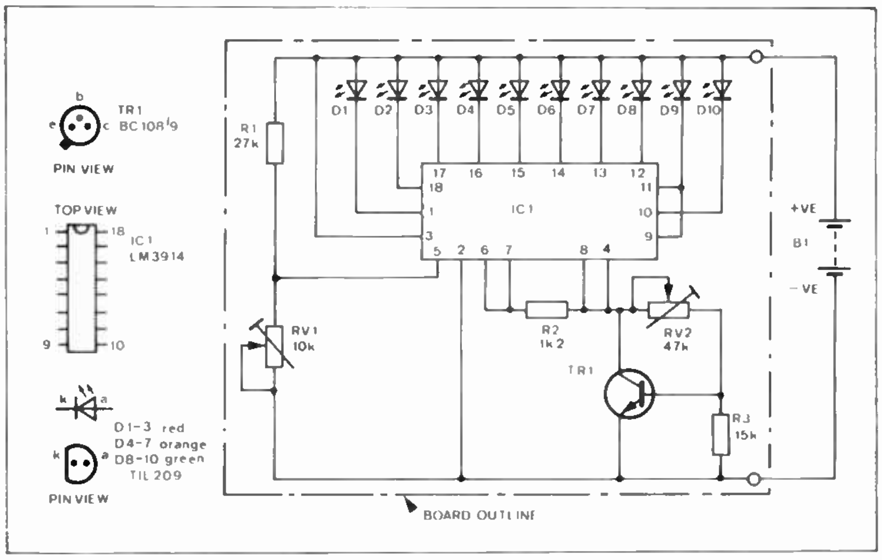


Figure 1. Circuit diagram of the Car Battery Monitor.

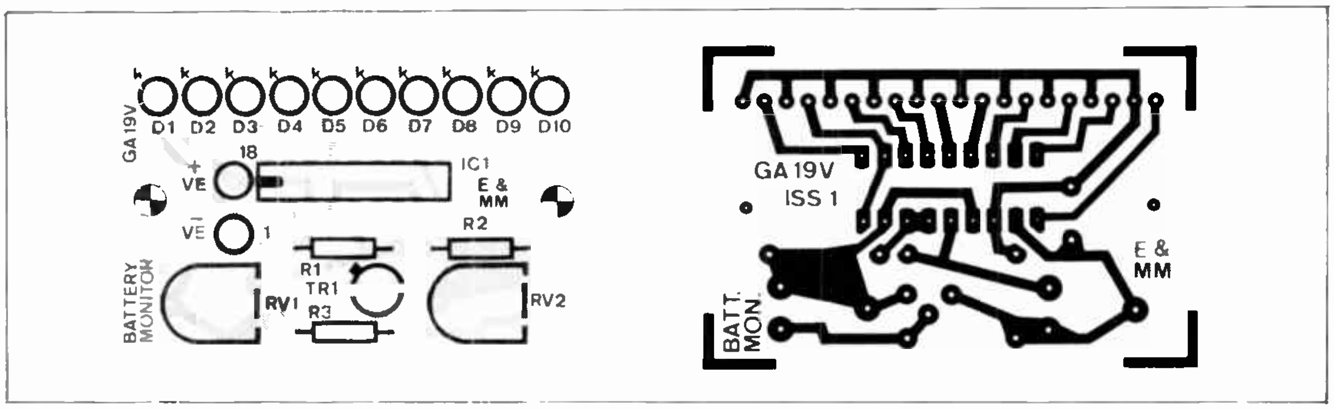


Figure 2. Car Battery Monitor PCB.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified.

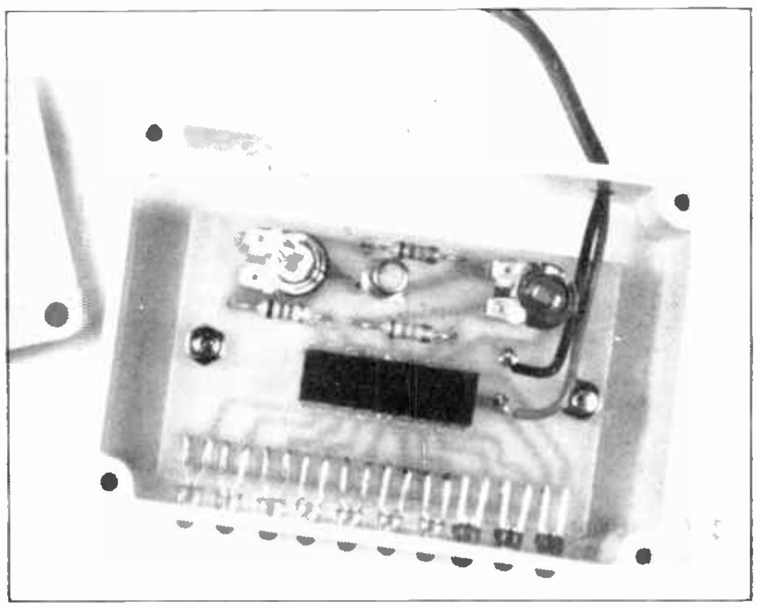
R1	27k		(M27K)
R2	1k2		(M1K2)
R3	15k		(M15K)
RV1	10k Hor. s. min preset		(WR58N)
RV2	47k Hor. s. min preset		(WR60Q)

Semiconductors

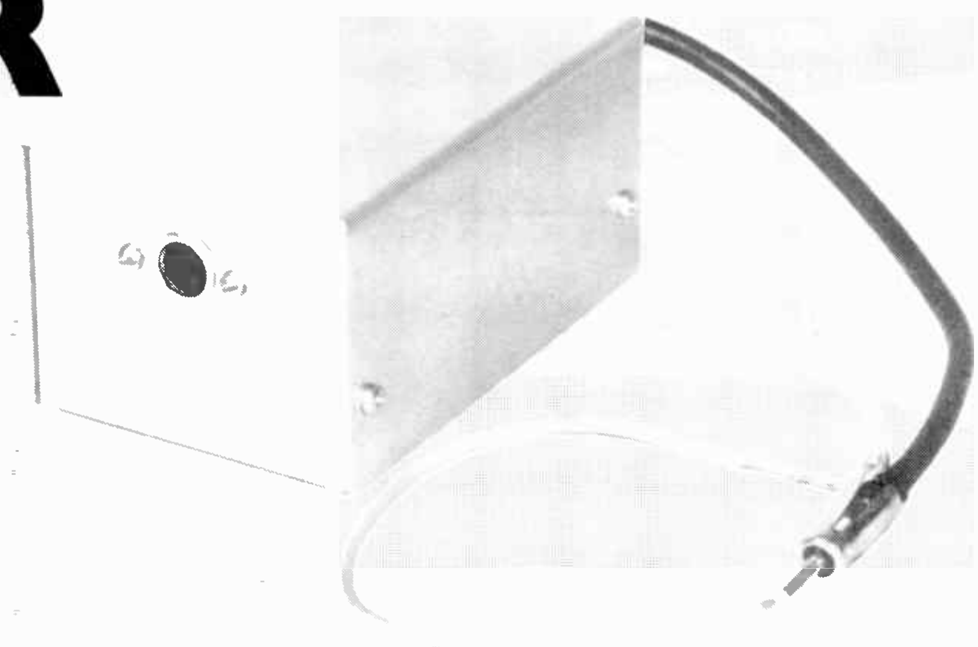
D1-3	TIL209 red	3 off	(WL32K)
D4-7	TIL209 orange	4 off	(WL34M)
D8-10	TIL209 green	3 off	(WL33L)
TR1	BC108		(QB32K)
IC1	LM3914		(WQ41U)

Miscellaneous

Printed circuit board	(GA19V)
Verobox 301	(LL12N)



CAR AERIAL BOOSTER



- ★ **Basic but functional design**
- ★ **AM and FM reception**
- ★ **Ideal for areas with poor signal reception**
- ★ **Low cost**
- ★ **Easy to install**

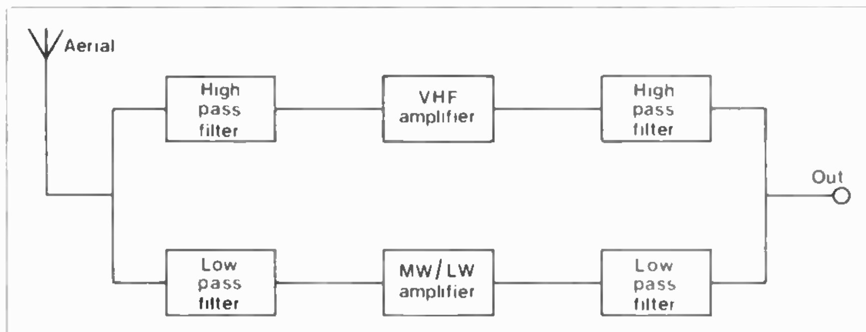


Figure 1. Block diagram of the car aerial booster.

by Robert Penfold

Although a normal car aerial has the useful feature of being omnidirectional, it is less than ideal in terms of signal pick-up. This often results in weak and noisy reception on both the AM and FM bands, especially in areas of relatively low signal strength.

This aerial booster is simply inserted between the aerial lead and the car radio aerial socket, the only other connection that is required is one to the positive side of the car battery. The booster is only suitable for 12 volt negative earth systems but this system is used in most vehicles today. The unit is effective over medium, long, low frequency short wave, and VHF broadcast bands. The degree of improvement obtained depends on a number of factors but in general there would be a substantial improvement in results if the booster was employed with an insensitive receiver in a poor reception area, and little or no improvement if it was used with a receiver having "state of the art" design in a strong reception area.

Basic Arrangement

The booster uses separate amplifiers for the AM and FM bands with a form of crossover arrangement to ensure that the two amplifiers hinder each other as little as possible. The block diagram in Figure 1 outlines the system used.

A high-pass filter blocks AM band signals from the VHF amplifier while a low-pass filter blocks VHF band signals from the AM amplifier. This ensures that no significant amounts of aerial signal are wasted by being fed to the

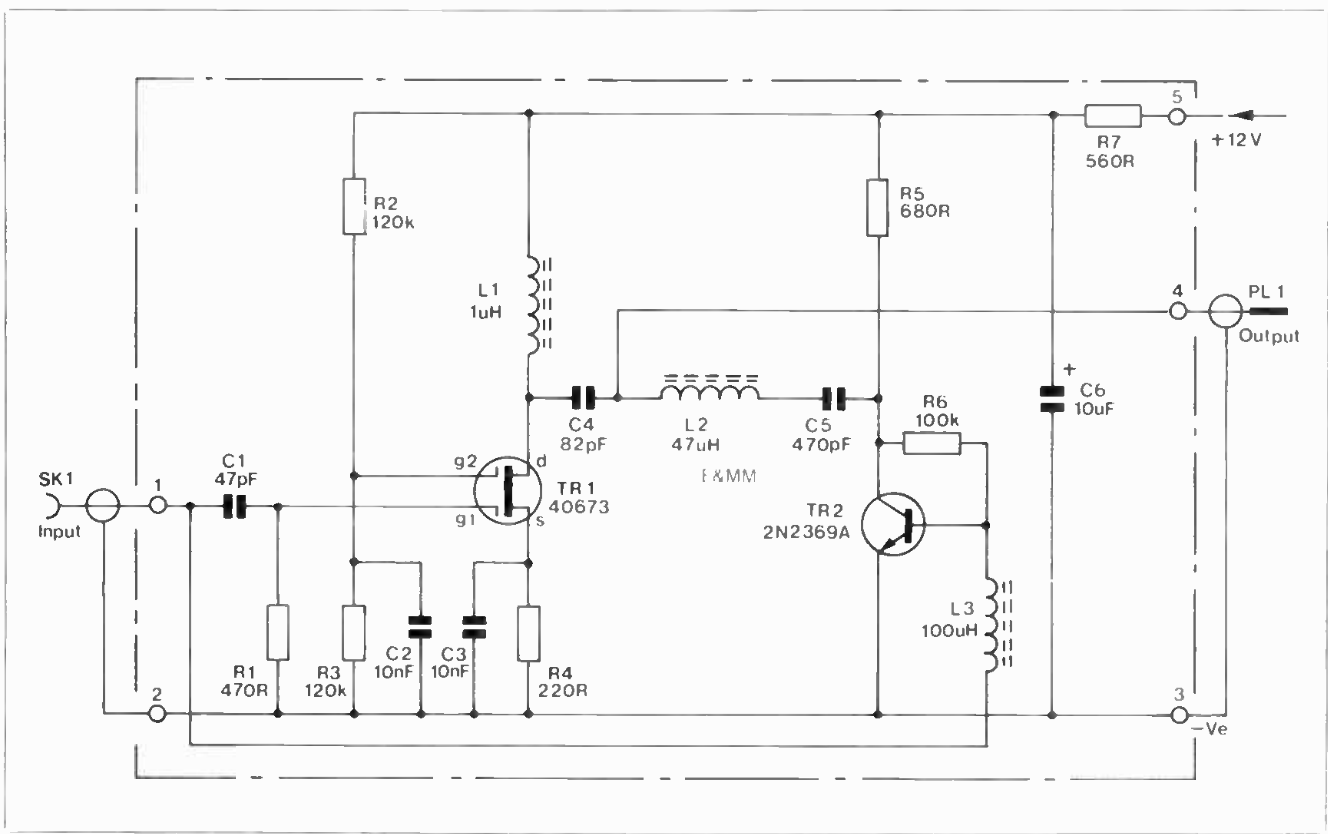


Figure 2. Car aerial booster: circuit diagram.

wrong amplifier stage. Both amplifiers are simple broadband, untuned types. The high-pass and low-pass filters at the outputs prevent either amplifier from feeding any significant amount of its output signal into the other amplifier stage, which would obviously give reduced gain. It also prevents VHF noise from the AM amplifier reaching the output and effectively reducing the noise performance of the FM amplifier stage. Noise from VHF over the AM bands is not really a problem, but the high-pass filter at the output will tend to block any noise of this type that is produced.

Circuit

Figure 2 shows the complete circuit diagram of the booster.

TR1 is used in the VHF amplifier and is a dual gate MOSFET used in the common source mode of operation. A dual gate MOSFET is used as it gives good gain and low noise at VHF. A choke load (L1) is used for TR1 but in other respects it is quite conventional.

The high-pass filter action at the input and output is provided by using low value capacitors (C1 and C4 respectively) at these points. C4 also provides DC blocking at the output. This very simple method of filtering seems to be more than adequate in practice.

TR2 is used in the common emitter mode and is the AM amplifier. It is run at a fairly high collector current (about 8mA) and has a fairly low collector load resistor value so that it maintains good voltage gain up to frequencies of several megahertz. This enables the unit to be used on the low frequency, short-wave broadcast bands (25, 31, 39 and 49 metre bands) if desired.

L2 and L3 have high impedances at VHF, and therefore give the low-pass filtering at the input and output of the AM amplifier. C5 merely provides DC blocking at the output of TR2 and is not a filter component.

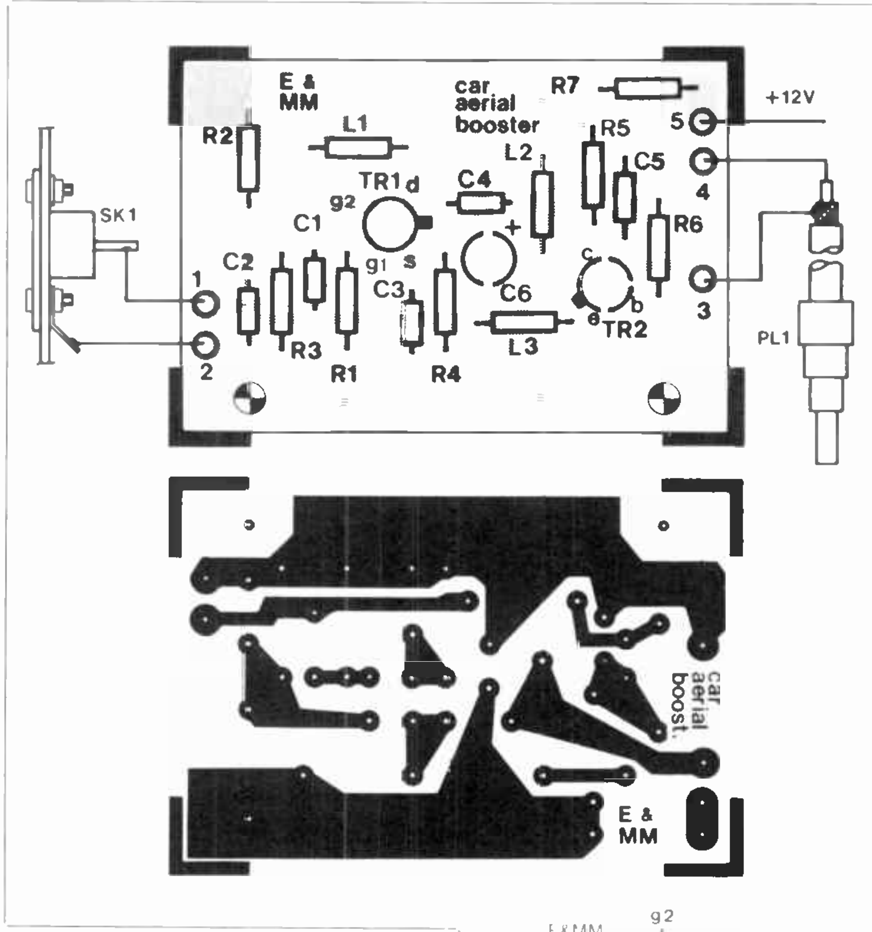
R7 and C6 form a supply decoupling network and these prevent the inevitable noise spikes on the vehicle's supply from degrading the noise performance of the booster. No on/off switch is needed since the current consumption of the circuit is only around 10mA and this is negligible to a car battery which has a very high capacity (in the order of 38Ah or more).

Construction

The aerial booster is built on a printed circuit board, the track and component layouts for which are shown in Figure 3. TR1 is a MOSFET device but as it has internal protection diodes, does not require any special handling precautions.

A small aluminium box makes an inexpensive and practical case for the project, but a non-metallic case could be used if preferred. SK1 is mounted at one end of the box and a solder tag is fitted onto one of its mounting bolts. The opposite end of the case is drilled to produce an exit hole for the output lead which is a coaxial type. This hole should be fitted with a grommet to protect the lead. It should be possible to use this hole as an exit point for the positive supply lead also, without any difficulty.

If Veropins are used on the printed circuit board to carry the off-board connections, the board can be mounted in the case prior to making the connections to SK1, the output lead, and the positive supply lead. If a metal case is used, fit spacers over the mounting bolts so that the connections on the underside of the board do not short-circuit to the case. It is then only necessary to fit a car aerial plug to the output lead and the booster is ready for installation and use.



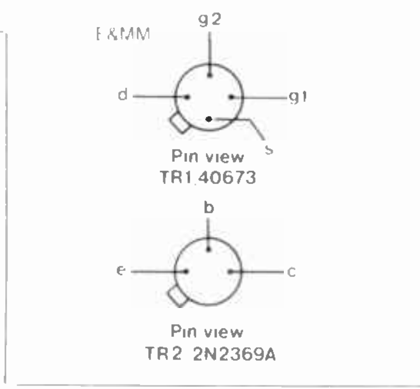
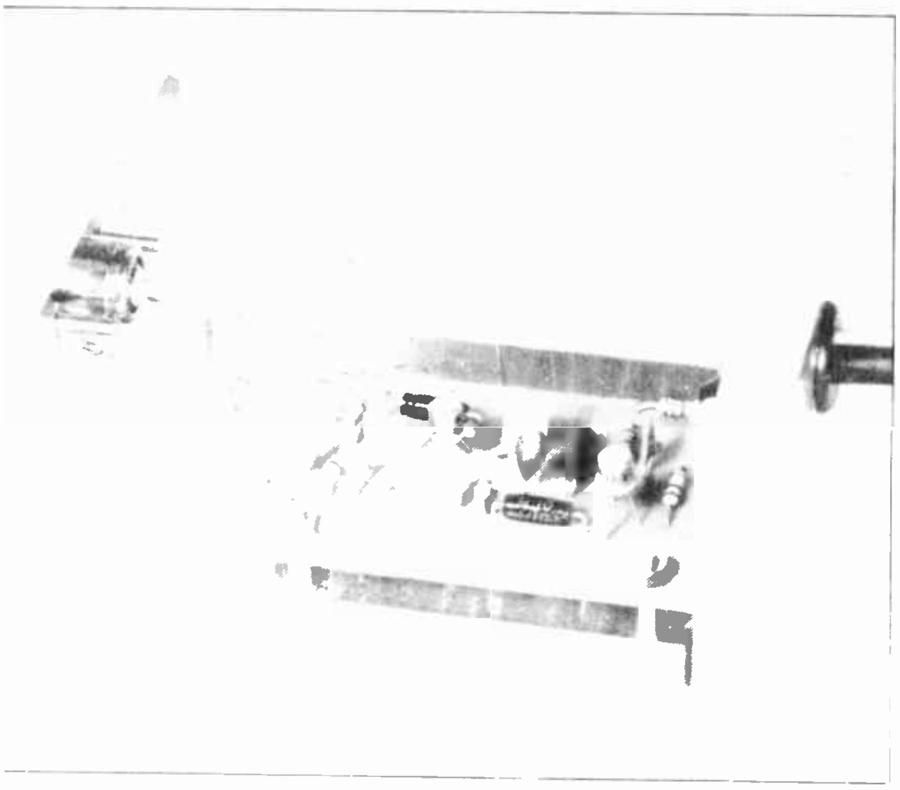


Figure 3. PCB track and component layout and SK1 fixing details.



Internal view of car aerial booster.

PARTS LIST

Resistors - all 1% 0.4W metal film.			
R1	470R		(M470R)
R2,3	120k	2 off	(M120k)
R4	220R		(M220R)
R5	680R		(M680R)
R6	100k		(M100k)
R7	560R		(M560R)
Capacitors			
C1	47p ceramic plate		(WX52G)
C2,3	10n ceramic plate	2 off	(WX77J)
C4	82p ceramic plate		(WX55K)
C5	470p ceramic plate		(WX64U)
C6	10u 16V tantalum bead		(WW68Y)
Semiconductors			
Tr1	40673		(QX34M)
Tr2	2N2369A		(QR12N)
Inductors			
L1	1uH		(WH29G)
L2	47uH		(WH39N)
L3	100uH		(WH41U)
Miscellaneous			
	Case type AB9		(LF10L)
SK1	Chassis Car Aerial Socket		(HH14Q)
PL1	Skeleton Car Plug		(HH13P)
	Coax cable 1 metre		(XR30H)
	Bolt 6BA ½in	4 off	(BF06G)
	Nut 6BA	4 off	(BF18U)
	Spacer 6BA ¼in	2 off	(FW34M)
	Grommet		(FW59P)
	Printed circuit board		(GA40T)
	Washer, shakeproof 6BA	2 off	(BF26D)
	6BA Tag		(BF29G)

CAR DIGITAL TACHOMETER

by Peter Marriott

In these days of ever-higher motoring costs the unit described here will help the driver to change gear at the most advantageous point to save fuel and extend engine life. Anyone using a car to tow a trailer or caravan will also benefit by being able to make the best use of the torque available from the engine.

Conventional tachometers give a display of engine speed on a milliameter, usually with a scale of about 270° arc. Pulses produced by the action of the contact breakers are integrated and fed to the meter to give an analogue display of engine revolutions. The disadvantages are that an average reading is displayed, which can easily lag behind rapid speed changes, and the meters tend to be somewhat fragile.

The tachometer described overcomes both of these disadvantages by counting pulses and displaying engine revolutions over a very short time, the digital display being continuously updated. Two digits display the number of revolutions x100. The unit is designed for negative earth cars. If you are not sure of the polarity on your car a glance at the owners manual or even at the battery connections will tell you.

As can be seen from the photographs, the case chosen gives an extremely professional looking unit, with only one hole needing to be drilled. Construction is very straightforward, using two printed circuit boards which fit directly in the case without the need for mounting bolts, so the project can be tackled by any but the most inexperienced constructor.

Circuit

The complete circuit is shown in Figure 1. Pulses produced by the make-and-break action of the engine contact breaker points are fed to IC1a which is a dual Schmitt trigger monostable, the other half being used elsewhere via a resistor/capacitor network composed of R1, R2 and C1. This network helps to smooth out any high voltage spikes which may be present on the contact breaker pulses. The zener diode D1 limits the input pulse at IC1a to 4.7 volts, to avoid any damage to the device. To prevent any false triggering due to contact points bounce (produced when the points do not open and close cleanly) the monostable period is set to 3 milliseconds by R3 and C2. This chosen time also means that the monostable is ready for retriggering by the next pulse and so the maximum count for a 4-stroke, 4 cylinder engine is limited to 10000 r.p.m. — a speed not often attained on normal road cars! The maximum count of 10000 r.p.m. corresponds to 20000 pulses/minute and the time for 1 pulse is 60/20000 seconds or 3 ms. A higher engine speed would not allow enough time between pulses for triggering of the monostable. This design is for 4 cylinder cars only and anyone using it on a 6 or 8 cylinder car would have to modify the count period accordingly, or use a compensating factor on the

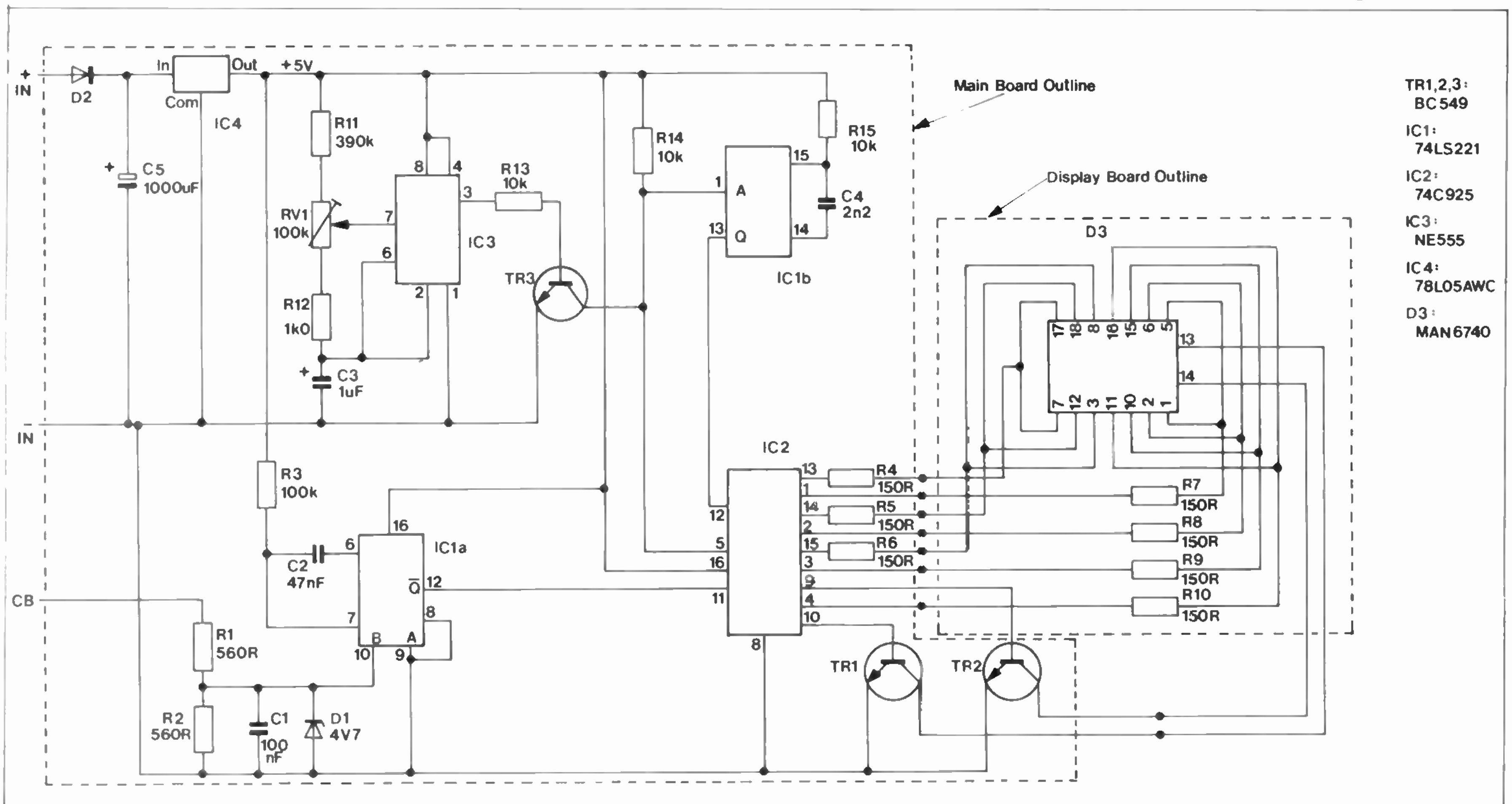


Figure 1. Circuit diagram of Digital Tachometer.

readings — not easy to do while driving!

The output pulses from IC1a, pin 12, are fed to the count input, pin 11, of IC2. This is a 4-digit counter with latch and reset. It drives the multiplexed 2-digit display directly, with transistors TR1 and TR2 selecting the digit and resistors R4—R10 limiting the segment current.

The counter requires latch pulses to give a sensible reading and these are provided by IC3, TR3 and their associated components. IC3 is the ever useful 555, used as an oscillator whose frequency is controlled by RV1. The oscillator output waveform, arranged so that there is a long high and a short low period, is inverted by TR3 so that a short high is achieved. This short pulse is used to control the latch on the counter integrated circuit IC2, so that when this input goes high the information in the counter is transferred to the internal latch and displayed. The short pulse is also used to trigger the monostable IC1b whose output pulse is used to reset the counter so that it starts counting from 00 again. Use of a separate monostable to reset the counter ensures that the reset pulse always occurs after the latch pulse so that a true reading is displayed.

Because the voltage (nominally 12 volts) on a car varies slightly with engine speed, integrated circuit IC4 is used to regulate this to 5 volts. This is used to supply IC1, IC2 and IC3 and is important for stability of the oscillator (IC3). Diode D2 and capacitor C5 remove noise on the supply.

Construction

The Digital Tachometer is constructed on two PCBs: the main board and the display board. The display board is mounted at 90° to the main board by veropins and holds the display so that it can be viewed through the filter at the end of the case.

Begin construction of the main board by fitting the resistors, capacitors, preset and three veropins, making sure C5 is the right way around. Then solder in the transistors, regulator (IC4) and diodes, paying attention to the orientation. The flats on the packages of TR1-3 and IC4 all face the furthest away long edge of the main PCB.

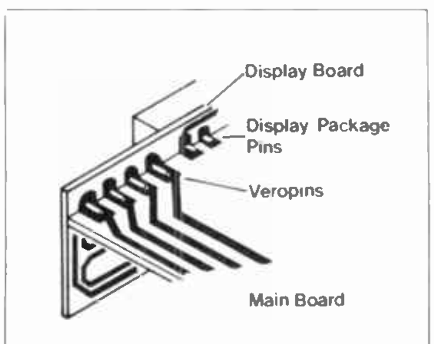


Figure 3. Board construction details.

Fit the sockets for the DIL ICs. IC2 is a CMOS device and costs over £5.00, so don't be tempted to economise on a socket for this one. Insert the other ICs but leave IC2 until the display board is completed and soldered to the main board.

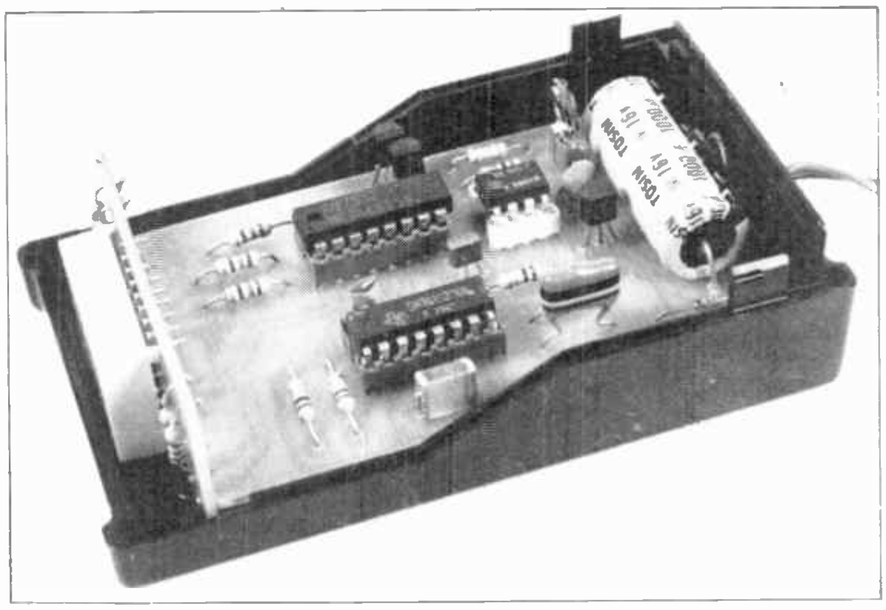
Fit the resistors and veropins to the display board, with the latter inserted from the component side. Be careful not to strip the pads off when doing this and push them home with the soldering iron before soldering them on the track side. Next fit the display. For some reason best known to the manufacturers, this has no formal package orientation mark, the device number being the only guide. This is on the same side as pin 1 and faces downwards on the completed unit. Solder the display in position.

The display board should now be fitted to the main board at right-angles by soldering the pins to the pads provided on the edge of the main board. This method of construction is shown in Figure 3. Fit IC2 to the main board, taking the normal precautions for CMOS devices, and solder long wires for power and input signal to the three pins, labelling the function of each wire at the end that will connect to the car electrics.

The metal front plate of the West Hyde case is replaced by a piece of red filter cut to 24 x 49 mm with a pair of scissors or craft knife. This slots neatly into the case, which is moulded in two sections. Drill a hole in the back for the wires and proceed to setting-up before fitting the boards in place and clipping the case sections together.

Setting Up

One advantage of a digital over an analogue tachometer is the ease of setting-up and calibration. Only one adjustment (RV1) needs be made and, barring accidents, will prevail for the life of the unit. This setting ensures that the oscillator runs at the correct frequency, and the method of calibration depends on the equipment available. Calibration against another tachometer is possible, setting RV1 to give a display of 30 when the standard tachometer reads 3000 rpm. If you have access to a signal generator, set the frequency to 100Hz and the output level to maximum (more than 4.7v). Connect this signal to the I/P pin on the PCB and adjust RV1 to give a reading of 30. Should the output of the signal generator be less than 4.7V, calibration may still be possible by feeding the output directly to pin 10 of IC1a.



Internal view of the Digital Tachometer.

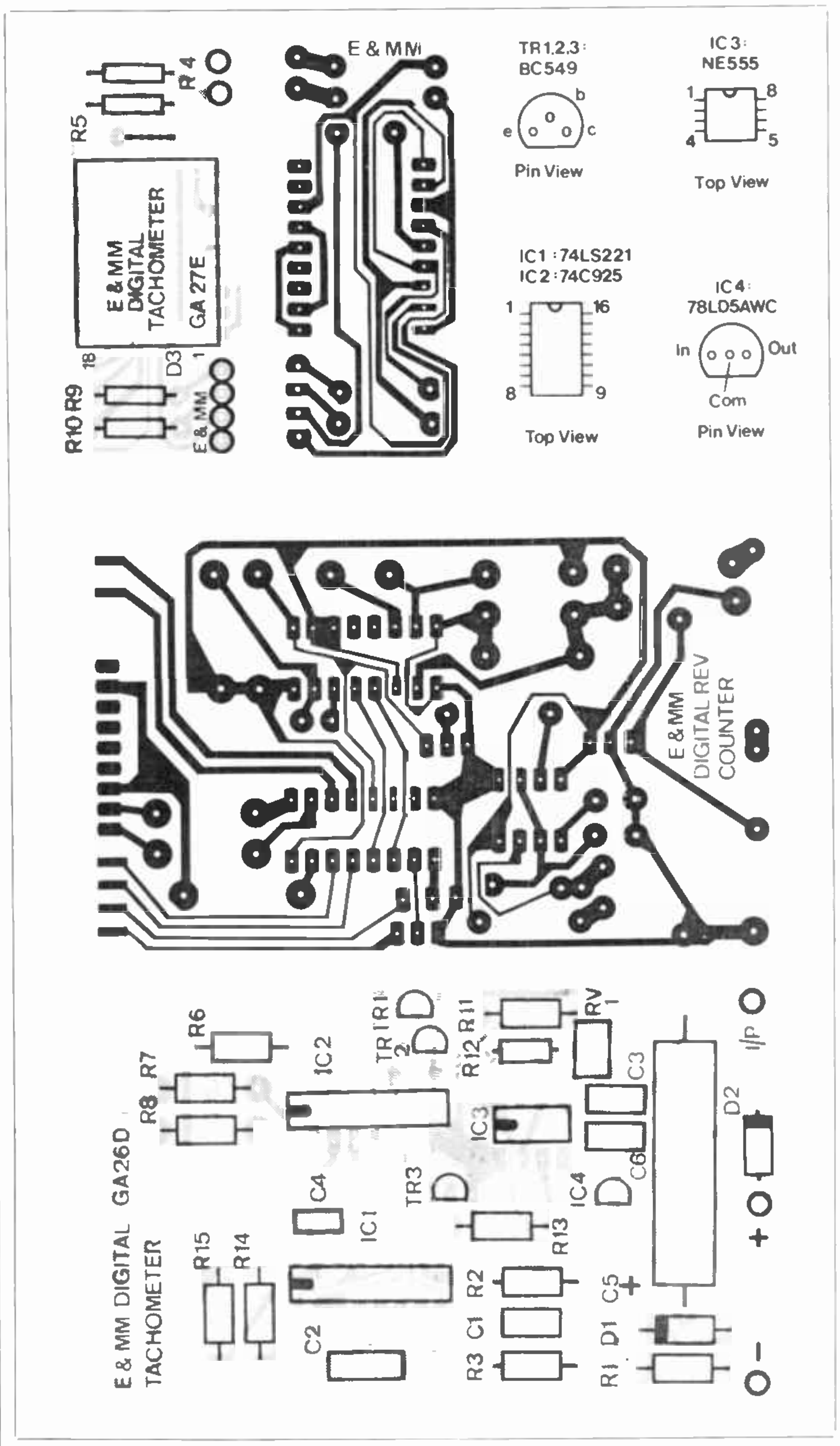


Figure 2. Digital Tachometer printed circuit boards.

Continued on page 64.

HI-FI SUB BASS WOOFER

by Jeff Macaulay

Add full deep Hi-Fi bass down to 27 Hertz to your stereo system

This woofer system is designed to be used in conjunction with existing speakers to extend the bass response, but at the same time is capable of being integrated into a full scale active speaker system.

For maximum flexibility the woofer is fed from the output terminals of the power amplifier used in the existing system. A variable cut-off 2nd order Butterworth filter is employed so that the woofer can be rolled on to complement the bass roll-off of the existing speaker system.

Before discussing the woofer in more detail it is instructive to consider the deficiencies of most speaker systems when it comes to reproducing deep bass, where an important constraint is the maximum permissable size of the enclosure. It is well known that the infinite baffle enclosure rolls off at 12dB/octave below the bass unit's resonant frequency. This situation can be improved to some extent by making the enclosure more inefficient, but then a high power amplifier is required to reproduce good bass. Another problem facing designers is that the bass unit is usually intended to reproduce the midrange as well. It is clearly advantageous to roll off the bass at some reasonable point to avoid intermodulation distortion which would occur due to the large cone excursions that are required. Interestingly the required output at 30Hz is some 8dB less than is required in the midrange. The peak power in music and speech signals occur at around 150 Hz.

Of all the possible forms that a woofer can take the most simple and effective method is to employ a bass reflex system. By suitable choice of drive unit a fairly compact, and hence domestically acceptable enclosure can be built that will respond down to 30Hz without problems. I write from experience of two such systems, one of which has been working in my own lounge for over a year, where the bass is often felt as well as heard. Even at high volume levels there is no apparent distortion and efficiency is also very high.

30Hz seems to be the optimum value at which to fix the lower -3dB point. Any lower and the cabinet begins to assume massive proportions. Moreover the dimensions of the average domestic listening room limits the lowest frequency that can be reproduced to around 30Hz. Going this far down will usually add another octave to the response in any case. The output power at this low frequency depends on the available cone excursion. At 30Hz the port is radiating sound as well as the speaker and this effectively doubles the area of the cone. When the relevant calculations have been made the output sound pressure level (SPL) is found to be 90dB at 1 metre. Put another way, if the main system is producing 96dB the bass unit will still have plenty of headroom. In fact these output figures are average - the bass speaker is capable of handling larger peaks.

The electronics required consist of an amplifier and a variable active filter system, designed to roll off between 50-100Hz adjustable by means of a potentiometer. The filter and power amp are installed inside the speaker cabinet, with the controls externally accessable.

The size of the cabinet is closely related to the characteristics of the drive unit employed. After some research the most suitable unit was found to be the Kef B200, requiring an amplifier power of only 20 watts RMS in this application. This unit has a free air resonance at 25Hz and when mounted in a sealed undamped enclosure of 2.4 cu.ft. this rises to 45Hz. From this information the acoustic volume (ie. the cabinet volume which enables the response to extend to 25Hz at -3dB when reflexed) can be calculated.

$$V_{as} = V\left[\left(\frac{f_c}{f_o}\right)^2 - 1\right]$$

Where f_o = free air resonant frequency, f_c = resonant frequency in cabinet, V = volume of cabinet. This gives a value for V_{as} of 5.38 cu.ft. By rearranging the formula for V the cabinet volume can be determined for any chosen value of f_c. In order to maintain a smooth response to 30Hz the resonant frequency must be 42Hz ($\sqrt{2} \times 30$Hz). For the B200 the volume is 3.4 cu.ft. This is a little large to be accommodated in the average lounge and so experiments were undertaken to lower the resonant frequency. The easiest way is to add a small amount of extra mass to the cone itself. This was conveniently done by adding two ¾" sq. pieces of bitumised felt panels to the cone, spaced equally on opposite sides of the centre. Suitable material is readily available as self adhesive car damping panels. This lowered f_o from 25Hz to 21Hz and, more importantly, allows a 2.4cu.ft. cabinet to be used for a f_c of 40Hz. By the time damping has been added to the enclosure it can be reflexed down to 27Hz.

Having actually determined the required enclosure size attention can now be turned to its mechanical details. Because the highest frequency to be handled is 100Hz the cabinet will be acoustically small. What this means is that air resonances inside the cabinet cannot occur because the dimensions are small compared with wavelength of the sound being emanated. The wavelength of a given frequency can be found simply by dividing the speed of sound, $344 ms^{-1}$, by the

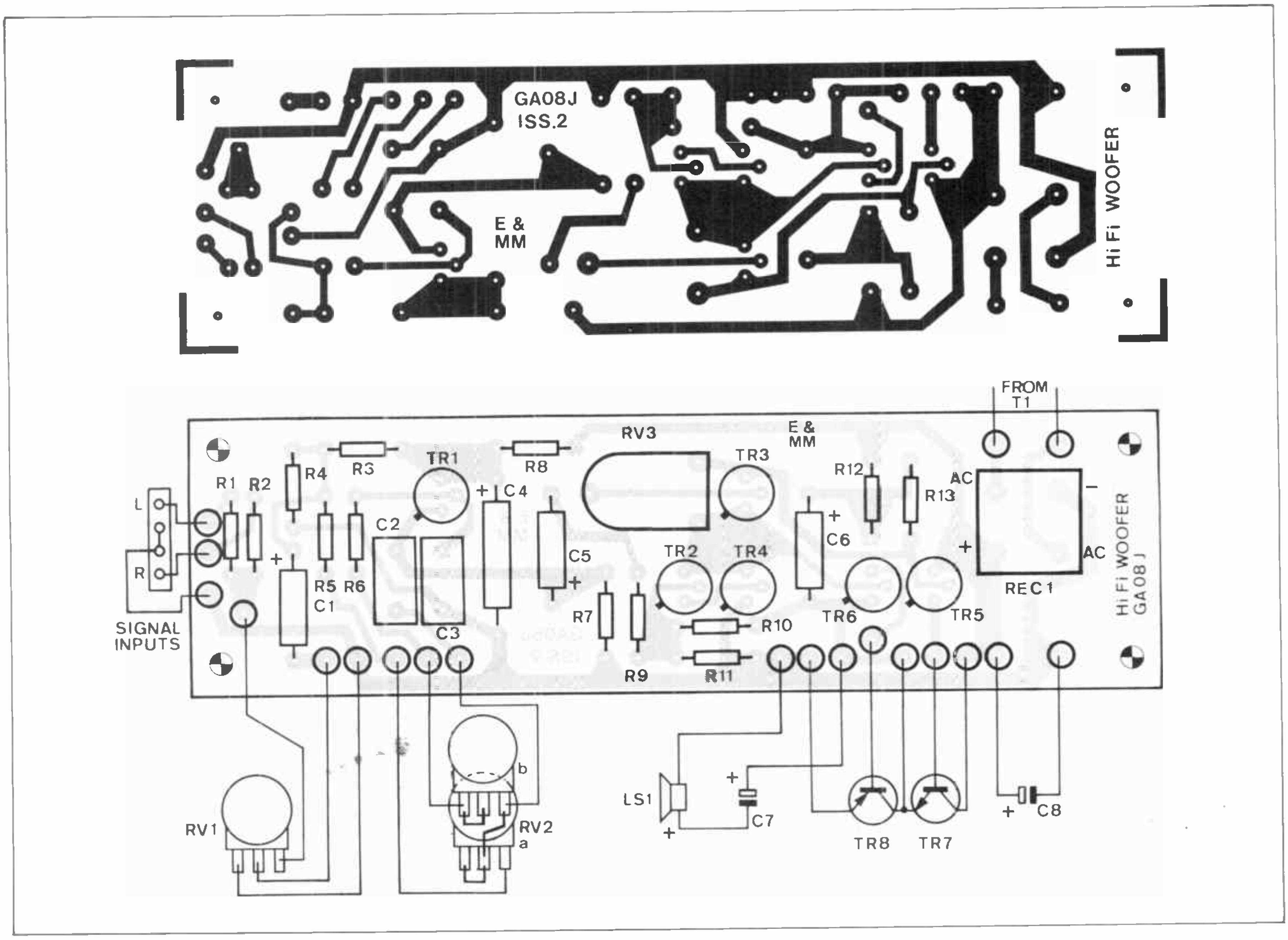


Figure 2. The Woofer PCB.

frequency. Thus the wavelength of a 100Hz tone is some 3.44m and a 30Hz tone has a wavelength of 344/30 = 11.45 metres. The resonances that plague normal speaker systems occur when one of the internal dimensions is equal to or a multiple of the wavelength of a reproduced signal. Since the largest internal dimension of our cabinet is 0.53m these resonances will be avoided. Another consequence of the smallness of the cabinet is that the polar diagram is totally symmetrical across the working range. In other words the speaker is omnidirectional. However, the wavelength of the sounds emanating from the enclosure have practical consequences for the positioning of the unit in the listening room.

The lowest note that can be sustained in a room is a function of its maximum dimension. The lowest note, in fact, that can be reproduced is found from the relationship, F = 344/2L where L is the longest room dimension in metres. Different dimensions will cause peaks and dips in the response, but of course this will happen whatever form the woofer may take and occurs naturally even in large halls. It does have a bearing on the siting of the enclosure which must be chosen for best results by empirical methods.

Circuit

Figure 1 shows the complete circuit of the sub-bass woofer electronics. For descriptive purposes it can be divided into three sections; mixer, filter and power amplifier.

R1, R2 and RV1 form a passive mixer and gain control. The signals from the speaker sockets of the amplifier are fed into the 'top' of R1 and R2. The signal from the wiper of VR1 is fed via the DC blocking capacitor C1 into the filter built around TR1. The values of the components are such that a 2nd order Butterworth response is obtained with maximum slope and minimum ripple in the pass band. RV2 allows the cut-off frequency of the filter to be varied from 50-100Hz to suit the bass roll-off of the existing pair of speakers.

The active element of the filter, TR1, is configured in the emitter follower mode. This produces a low impedance drive for the power amplifier, which is a little unusual in that the circuit is of the shunt feedback type. The reason for its adoption here is that it is easy to build and unconditionally stable. The 2N3055's on the output are more than capable of delivering the 4mV/us slew rate required for a bandwidth of 100Hz! The output power of the amplifier is 20W RMS and unlike the majority of current designs the output is capacitatively coupled to the speaker. This has the advantage that if a breakdown were to occur in the amp then the speaker will be protected from DC current. The value of C7 is such that the response is 3dB down at 10Hz.

Construction

The printed circuit board should be assembled following the component overlay, in usual order of resistors first, then capacitors, followed by the semiconductors. The pins for the off-board wiring can be soldered in at this stage, but the wiring should be left until after the cabinet is completed. Check the orientation of the electrolytic capacitors and semiconductors, and make sure there are no tracks shorted by solder bridges. Drill holes in the cabinet back for the push connector, volume and cut-off frequency pots, on-off switch, and mains chassis plug. Fit these components. Mount the transformer and fuseholder using woodscrews and then the two capacitors C7 and C8 using clips and screws. Drill the heatsink to take the TO3 power transistors, and mount them using mica washers etc. Mark fixing positions for the PCB and heatsink, and then wire up all the off-board components. This is best done by soldering wires of the right length to the veropins on the PCB before mounting it, then connecting the mains and output components. Finally mount the heatsink and PCB, and solder the wires from the latter in place.

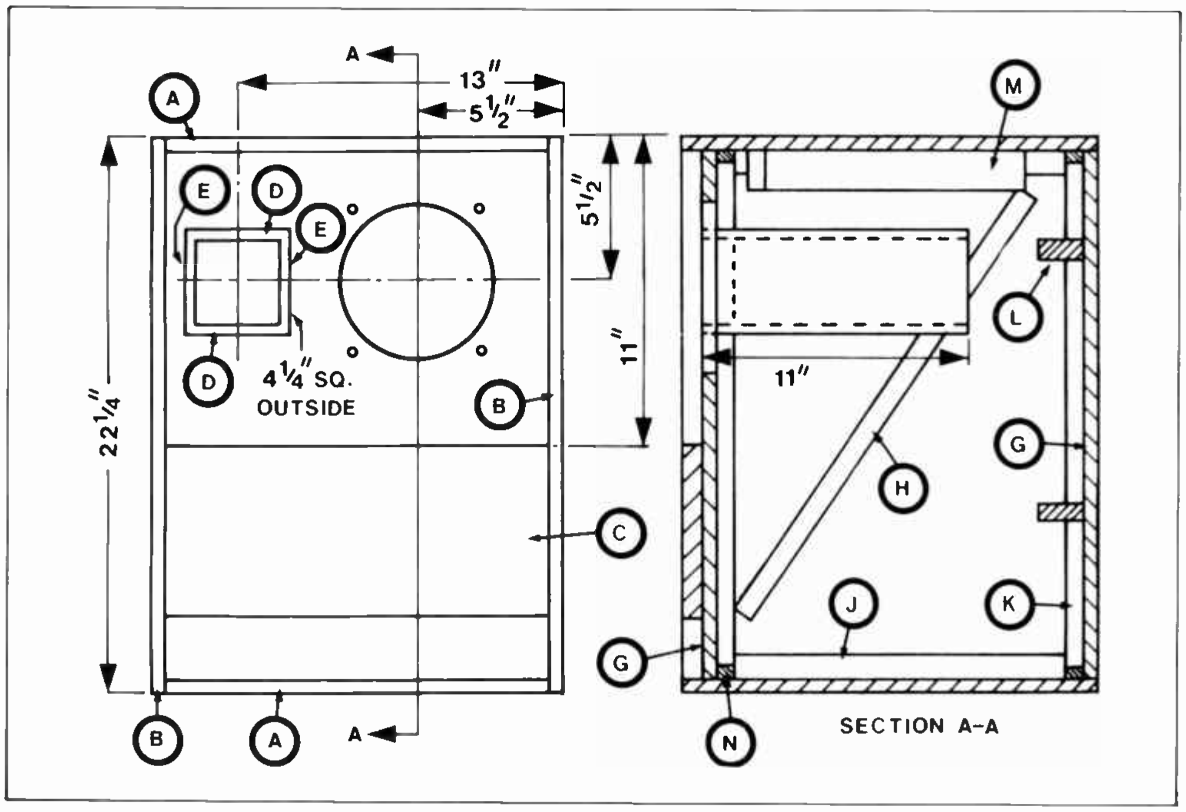


Figure 3. Woofer cabinet construction.

CUTTING LIST

Quantity	Dimensions (inches)	Material	Part
2	15 × 15 × ⅝	Veneered Chip	A
2	22¼ × 15 × ⅝	Veneered Chip	B
1	15 × 7 × ⅝	Veneered Chip	C
2	11 × 3 × ⅝	Veneered Chip	D
2	11 × 4¼ × ⅝	Veneered Chip	E
2	21 × 15 × ⅝	Veneered Chip	G
2	20 × 2 × 1	Soft Pine	H
4	10⅛ × ⅞ × ⅜	Hardwood	J
4	21 × ⅞ × ⅜	Hardwood	K
2	13 × 2 × 1	Soft Pine	L
1	13 × 2 × 1	Soft Pine	M
4	15 × ⅞ × ⅜	Hardwood	N

Cabinet Construction

There are several possible materials that could be employed for the cabinet. Domestic considerations, and the desire for a ready made finish dictated the use of veneered chipboard. Melamine teak board has an added advantage of being denser material than either the white or wood veneered board. These facts led to its adoption for this project. The internal volume of 2.4 cu.ft. and the desire to keep the woodworking simple determined the dimensions at 22½" × 15" × 15" external. This means that the cabinet, excluding battens, can be fabricated entirely from 15" boards. If the wood is cut to size at the timber yard the only tools required will be an electric drill and jigsaw attachment.

Assembly is straightforward and should proceed as follows:

1) Label each panel with its respective part letter on the worst face. This saves any possible confusion as work proceeds.
2) Cut the battens H, J, K, M and N to size and mark as in 1).
3) Mark out the positions of the battens on the panels with a felt tip pen. This is about the only thing that will mark the surface of the boards without smudging.
4) Glue the battens into their respective positions. The best glue to use with this material is 'Thixofix'. This adhesive is often employed for fixing table tops. It is a contact adhesive having the advantage that the glued surfaces can be moved relative to one another before they are permanently joined. A permanent joint is made by simply pressing the parts together.
5) Having fixed the battens with adhesive secure more permanently with ¾" panel pins. Use four for each batten.
6) Using the B200 template mark out the position of the four fixing screws on the front panel. Remove the central area of the template and mark the inner circle. Remove the template and drill out the mounting holes.
7) Mark out the 4¾" square cut-out for the port. At this point it is advisable to drill four ⅜" holes near the corners of the port cut-out to facilitate the use of the jigsaw.
8) Take part C and glue it into position on the baffle.
9) Cut out the front baffle apertures. The baffle can be painted matt black at this stage. Blackboard paint is suitable.
10) Take parts D and E and glue and pin together to form a square tube as shown in detail two.
11) Insert the port just constructed into the aperture on the front front baffle. If there are any gaps between the port and front panel it should be filled from the rear of the baffle. The port must be mounted so that the end is flush with the front of the baffle.
12) Take the side, bottom and top panels (A and B) and the front baffle, apply adhesive to all surfaces that will butt together, and leave for 15 minutes.
13) Assemble the cabinet except the back panel, using screws in addition to the glue to fix the front panel to its supporting battens. Check that the back panel will fit tightly in position.
14) Glue and screw the battens (L) onto the back panel (G).
15) Add the felt panels to the KEF B200's cone as previously described, then attach the drive unit to the front baffle using the bolts and T nuts provided.
16) Install the electronics. Stretch a 12" square piece of acoustic wadding across the back of the drive unit and fix it with a dab of glue in each corner.
17) After the setting-up stage, roll up 2 metres of wadding and place it in the cabinet (the position is not critical) then screw the back on.

Setting Up and Use

The quiescent current of the output pair must be set to eliminate crossover distortion. Before applying power turn RV3 to its most clockwise end and cover the mains terminals of the transformer switch, and fuse with insulating tape. Apply power and if all seems well measure the voltage at the positive terminal of the output capacitor C7. This should be 20V ± 2V. Switch off and disconnect the wire to the collector of TR7. Insert a multimeter to read current and switch on again. The current should be 10mA. Adjust RV3 until it reaches 30mA. Reconnect TR7. Remove the tape from the mains terminals and screw on the back of the cabinet.

Play some programme material with a good bass content. Experimentally adjust the volume control on the bass unit for what you judge to be the correct level. At this point it is as well to go and sit in the stereo seat and listen carefully. Often further adjustment will be required since the level of bass heard depends on one's listening position. The filter is best set with a voice signal. Radio 4 is a good source of assorted voice signals. Start adjusting from the 100Hz (clockwise) end downward. Speech will probably sound a trite boomy but as you adjust a point will be found where the voice sounds natural and well balanced. Play some music, preferably a piece that you know well with a reasonable bass content. In all probability it will take some time to find the optimum position for the controls. However, even before that you should find that your enjoyment of all signals will be enhanced. It should go without saying that these adjustments should be made with the tone controls in the flat position, or better switched out.

It is most important that the woofer be sited with care. Although no stereo information is broadcast below 100Hz it is important not to disturb the stereo image. If you sit too near the woofer the sound is likely to suffer because of the Haas effect (if two independent speakers are reproducing the same signal it will appear to come from the nearer source).

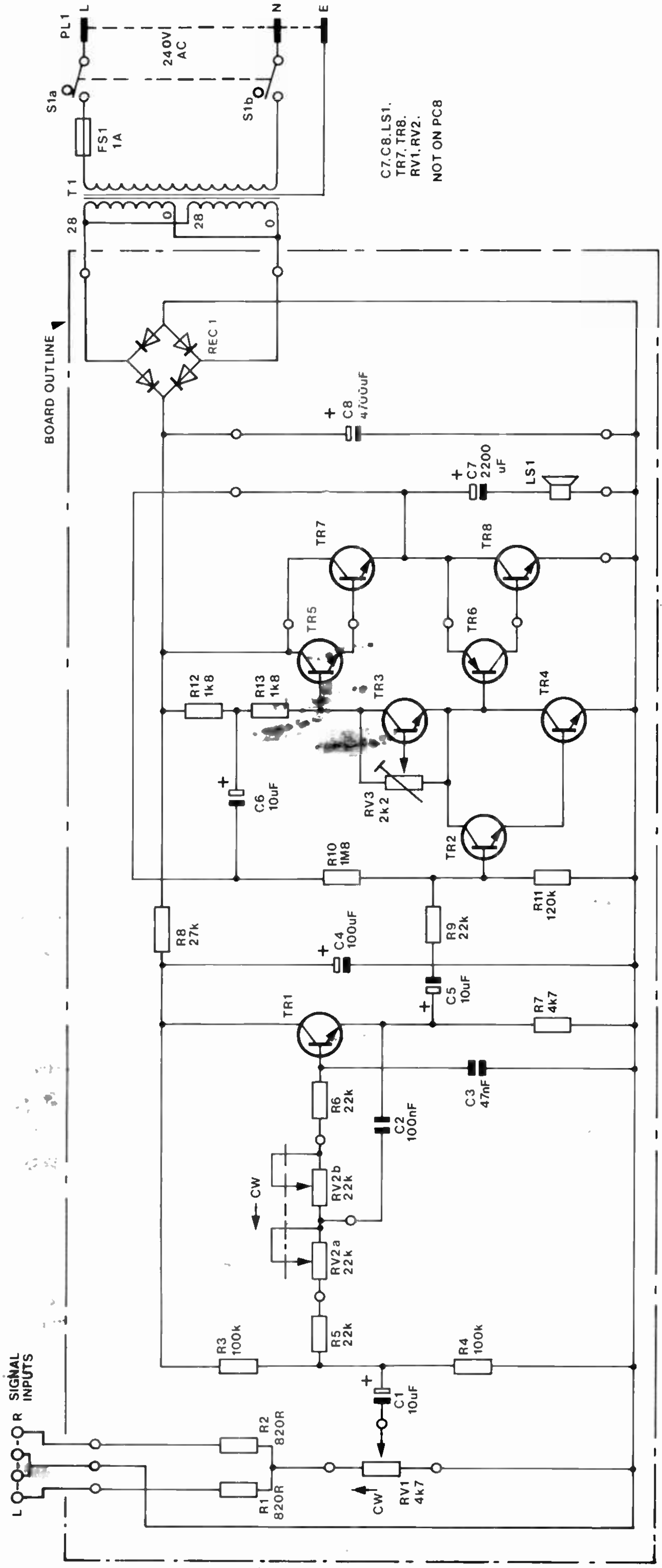
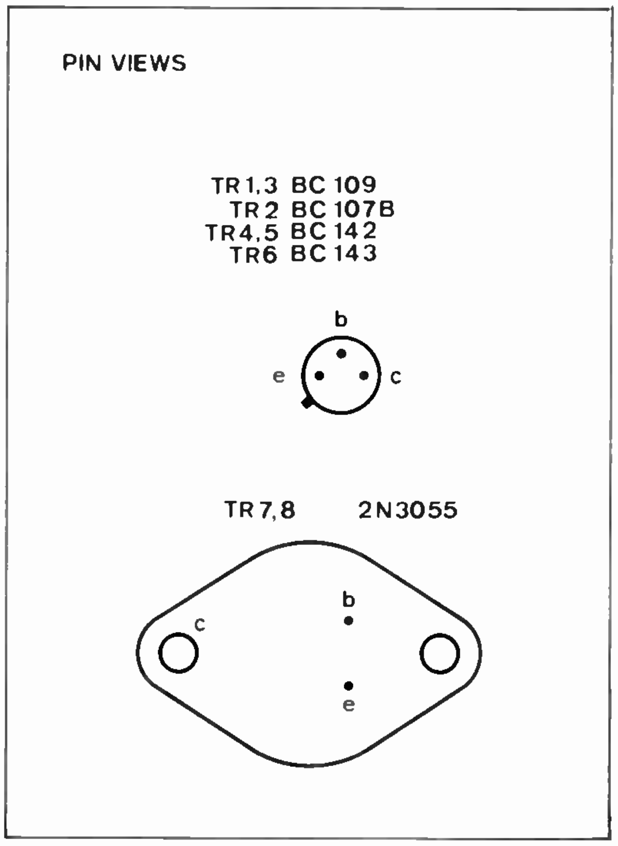


Figure 1. Circuit diagram of the Hi-Fi Sub-Bass Woofer.



PARTS LIST

Resistors - all 1% 0.4W metal film unless specified

R1,2	820R	2 off	(M820R)
R3,4	100k	2 off	(M100K)
R5,6,9	22k	3 off	(M22K)
R7	4k7		(M4k7)
R8	27k		(M27K)
R10	1M8		(M1M8)
R11	120k		(M120k)
R12,13	1k8	2 off	(M1k8)
RV1	4k7 log. pot.		(FW21X)
RV2	22k lin. dual gang pot.		(FW86T)
RV3	2k2 carbon preset		(WR82D)

Capacitors

C1,5,6	10uF 25V electrolytic	3 off	(FB22Y)
C2	100nF polyester		(BX76H)
C3	47nF polyester		(BX74R)
C4	100uF 25V electrolytic		(FB49D)
C7	2,200uF 63V electrolytic		(FF22Y)
C8	4,700uF 63V electrolytic		(FF28F)

Semiconductors

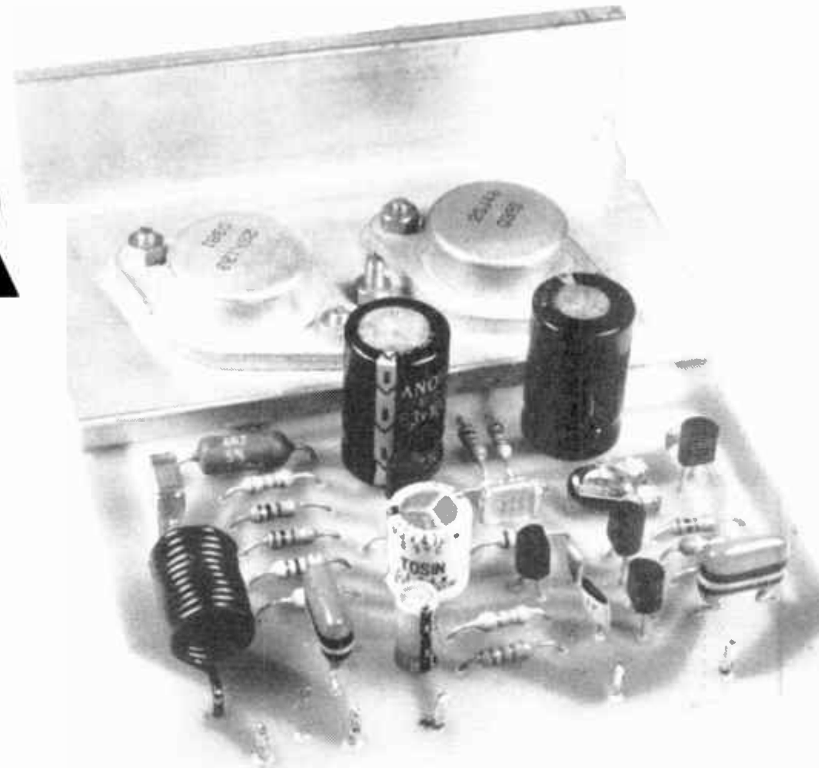
TR1,3	BC109	2 off	(QB33L)
TR2	BC107B		(QB31J)
TR4,5	BC142	2 off	(QB39N)
TR6	BC143		(QB40T)
TR7,8	2N3055	2 off	(BL45Y)
REC1	S005		(QL09K)

Miscellaneous

T1	Transformer 240V prim. 0-28, 0-28 sec. 84VA		(WB17T)
LS1	Kef B200 SP1014		
FS1	Fuse, 20mm 1A Quick Blow		(WR03D)
	Chassis fuseholder, 20mm		(RX49D)
S1	Toggle switch, DPDT		(FH39N)
PL1	P429 3-pin chassis plug		(HL20W)
	P646 3-pin line socket		(HL44X)
	4-way push-type connector		(BW71N)
	TO3 Mounting kits	2 off	(WR24B)
	Heatsink, 2.1° C/W		(FL54J)
	Control knobs	2 off	(RX08J)
	Acoustic wadding	1 m	(RY06[illegible])
	Mains cable, 3A	3 m	(XR00A)
	Connecting wire	3 m	(XR37S)
	Capacitor clip to suit C7		(FF33L)
	Capacitor clip to suit C8		(FF35Q)
	Printed Circuit Board		(GA08J)

MOSFET AMPLIFIER

An incredible Hi-Fi Amp that's virtually bomb-proof — like the best valve amps



by Dave Goodman

Specification

Power output:	**>75W RMS into 4Ω**
	>50W RMS into 8Ω
Sensitivity:	**650mV RMS for rated output**
Input impedance:	**47kΩ**
Power supply:	**44-0-44V DC, 2A**
Frequency response:	**20Hz to 20kHz virtually flat**
	10Hz to 40kHz ±1dB
Total harmonic distortion:	**20Hz to 20kHz <0.005%**
	1kHz <0.002%
Signal-to-noise ratio:	**120dB**

Power MOSFETs are a relatively new addition to the range of semiconductor devices available. Small signal MOSFETs have been around for some years, mostly finding uses in high frequency applications, but it was found difficult to make MOSFETs with gate-to-drain voltages greater than 30V (most are rated at 20V), and with high current capabilities, such as would be required in power amps. The high voltages possible with power MOSFETs are achieved by separating the gate and drain layers with a layer of ion implanted silicon dioxide. In addition, a field plate is provided on the source, near the gate. These two measures prevent electric field concentration, which at high voltages would otherwise destroy the gate. The high current capability is achieved by using a comb-shaped structure for the drain and source regions.

Figure 1 shows the connection configuration for the MOSFETs used in this amplifier. Although a T03 package is used to give excellent heat dissipation, MOSFETs are far superior to bipolar transistors in their response to high temperatures. As a bipolar transistor heats up for a given voltage, the current through it becomes greater; i.e. it has a positive temperature characteristic. If the temperature were allowed to continue to rise thermal runaway would ensue and the transistor would be destroyed. A MOSFET, however, has a negative thermal characteristic. As the transistor becomes hotter, the current tends to decrease, so power MOSFETs are most unlikely to be destroyed due to high temperatures.

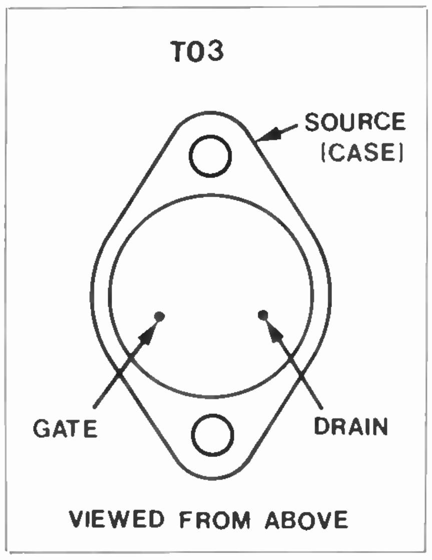


Figure 1. Power MOSFET package.

Power MOSFETs also have a far wider frequency response than bipolar power transistors, so that a very wide and extremely flat frequency response can be obtained, without any complicated circuitry. Figure 3(a) shows the typical output characteristic of a power MOSFET for gate-to-source voltages (V_{GS}) from 1V to 10V in 1V steps. Figure 3(b) shows the remarkably low total harmonic distortion generated by this amplifier. It is scarcely measurable, even with the best test equipment available and certainly far below the minimum audible level.

Circuit Description

TR1 and TR2 form a stable, differential input buffer amplifier, the bias current for each transistor being set to 0.5mA. The 2SA872 transistor is used because it has a very low noise

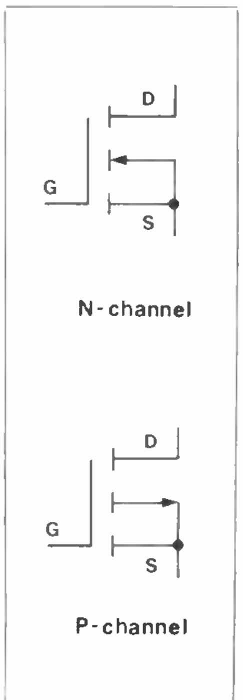


Figure 2. MOSFET symbols.

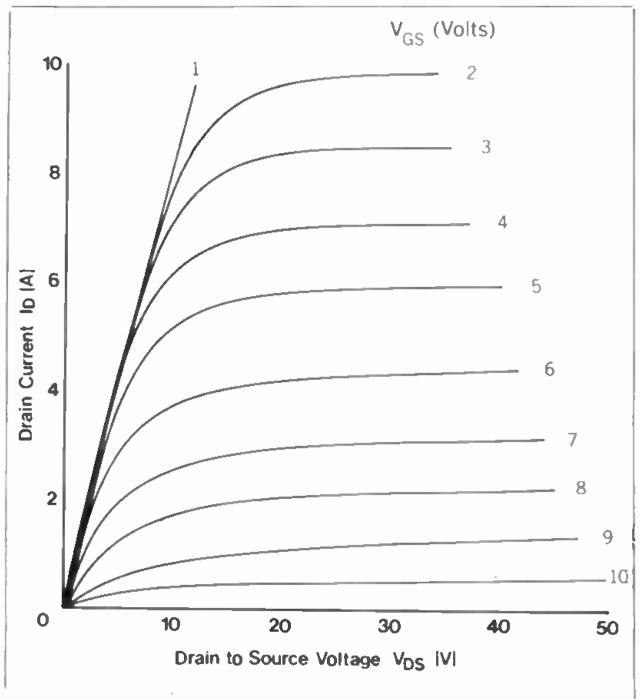


Figure 3(a). Typical output characteristic.

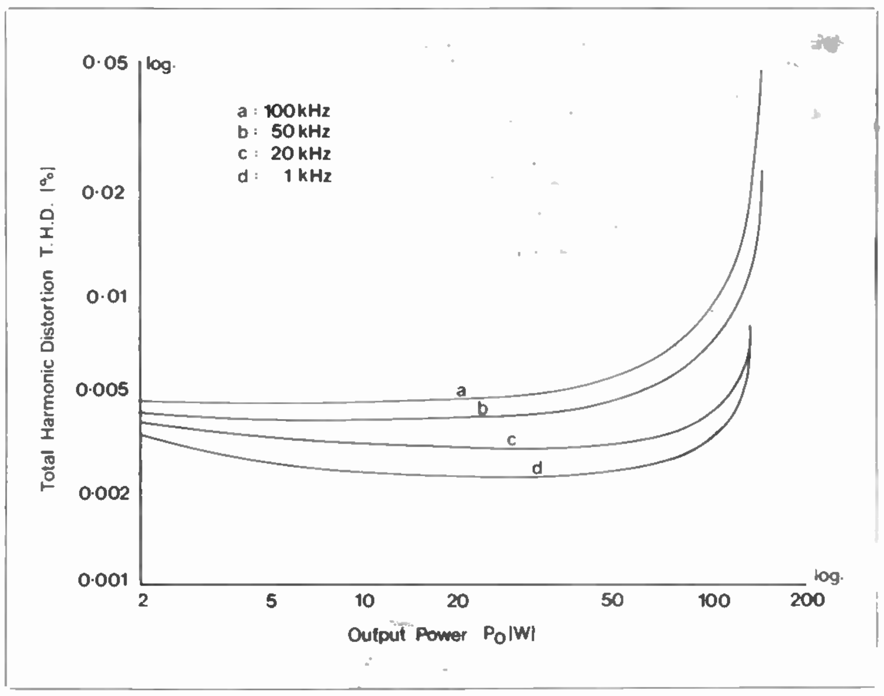


Figure 3(b). Harmonic distortion graph.

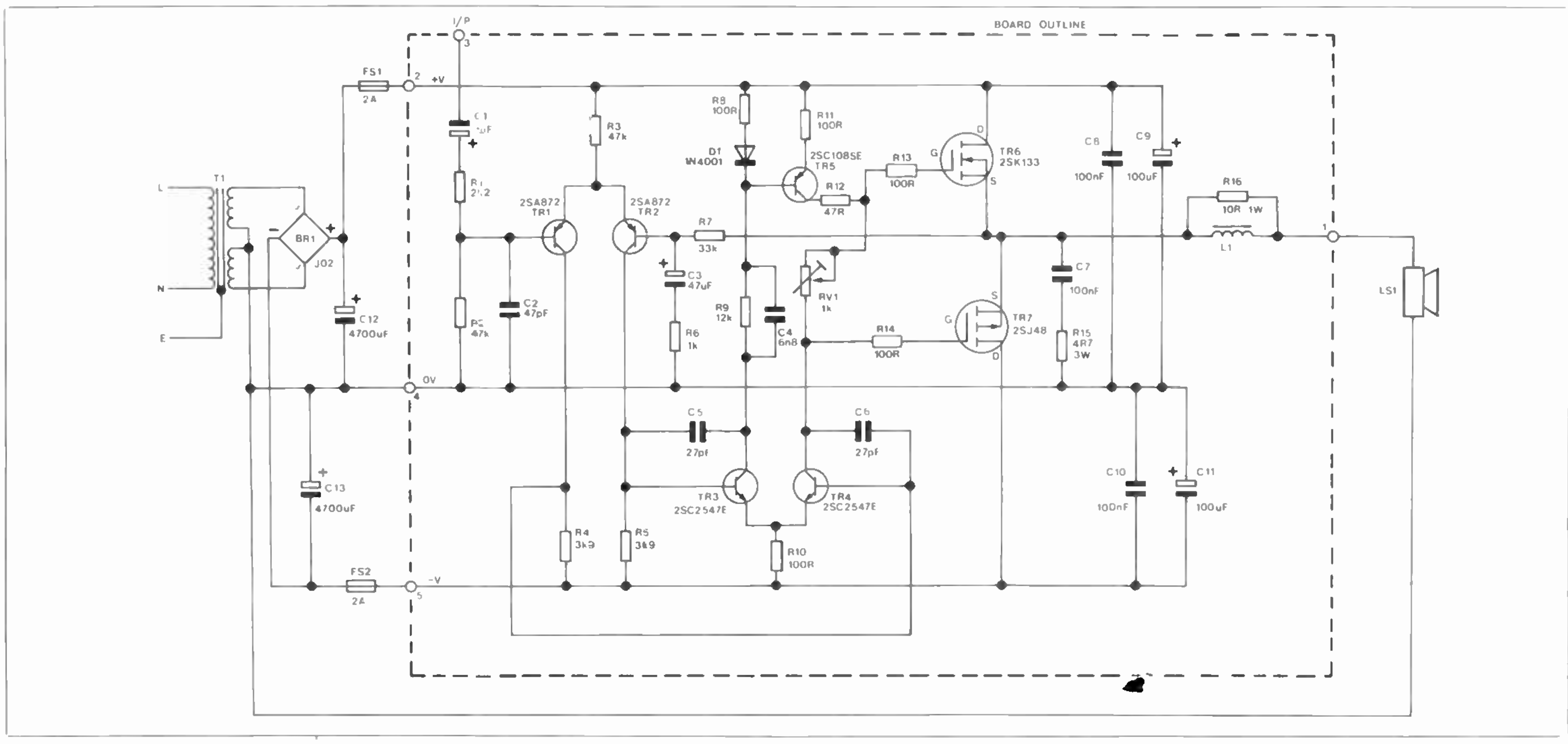


Figure 4. Circuit diagram for single channel amplifier.

output but can handle high voltages. TR3 and TR4 form a 'current mirror' to give a high open-loop voltage gain. TR5 acts as a constant-current load and this low-noise, high-gain, class A amplifier stage is all that is required to drive the power MOSFETs TR6 and TR7. The transistors in the driver stage need to have a high voltage durability, high F_T and low C_{ob}. They also have to supply sufficient power to charge and discharge the gate-to-source capacitance of the power MOSFETs. In this case a bias current of around 50mA is sufficient to ensure adequate power is available at all frequencies and power levels.

The input impedance of the amplifier is set, by R2, to 47k, and C2 bypasses any RF signals present at the input. The amplifier has a gain of 33, and this is set by R7 and R6 via decoupling capacitor C3. R13 and R14 improve the stability at high frequencies by reducing the effective gate load capacitance. C7 and R15 are a Zobel network which, in conjunction with R16 and L1, ensures excellent stability into reactive loads at high frequencies.

Construction

Fit the five Veropins, labelled 1 to 5, to the PCB and solder. Fit and solder diode D1 taking care that it is the right way round. Fit and solder all the resistors *except* R16, and all the capacitors, taking care with the polarity of the electrolytic ones, C1, C3, C9 and C11 (refer to Figure 5). Scrape or burn the enamel off one end of the piece of enamelled copper wire and solder it to one lead of R16, close to the body of the resistor. Now wind the wire tightly around the resistor ten times to form L1, as shown in Figure 6. Do not cut the wire, but hold it tightly and scrape off the enamel where it will touch the other lead-out wire of the resistor, then wrap it around the lead and solder. Fit this composite component to the PCB and solder. Fit and solder the preset to the PCB, then the transistors (TR1-5).

Make the heatsink bracket shown in Figure 7. (Note that this is available ready-made, and is included in the kit supplied by Maplin Electronic Supplies Ltd.) The mounting bracket fits to the component side of the PCB as shown in the photograph. Align it with the holes in the PCB and put one bolt through the centre hole from underneath using a 6BA nut, bolt and shakeproof washer. Referring to Figure 8, place a nylon bush in each of the four large holes in the bracket, smear both faces of both mica washers with Thermpath silicone grease and place these in position. Mount the two power MOSFETs, ensuring that TR6 (2SK133) is fitted closest to the coil L1. Put in the 6BA bolts to hold the transistors from underneath and secure them using nuts and shakeproof washers. Solder the bolt heads to the track on the PCB. Finally solder the drain and gate pins to the PCB and re-check all component positions, polarisations and solder joints.

Power Supply

The PSU (T1, BR1, C12 and 13

PARTS LIST

Amplifier

Resistors — all 1% 0.4W metal film unless specified

R1	2k2		(M2K2)
R2,3	47k	2 off	(M47K)
R4,5	3k9	2 off	(M3K9)
R6	1k		(M1K)
R7	33k		(M33K)
R8,10,11,13,14	100R	5 off	(M100R)
R9	12k ½W STD		(S12K)
R12	47R		(M47R)
R15	4R7 3W,W/W		(W4R7)
R16	10R 1W, carbon		(C10R)
RV1	Hor. S-Min. Preset, 1k		(WR55K)

Capacitors

C1	1uF, 100V, PC. elect		(FF01B)
C2	47pF, ceramic		(WX52G)
C3	47uF 63V, PC, elect.		(FF09K)
C4	6n8, polycarbonate		(WW27E)
C5,6	27pF, ceramic	2 off	(WX49D)
C7	100nF, polycarbonate		(WW41U)
C8,10	100nF, polyester	2 off	(BX76H)
C9,11	100uF, 63V, PC elect.	2 off	(FF12N)

Semiconductors

D1	1N4001		(QL73Q)
TR1,2	2SA872	2 off	(QQ30H)
TR3,4	2SC2547E	2 off	(QY11M)
TR5	2SA1085E		(QY12N)
TR6	2SK133		(QQ36P)
TR7	2SJ48		(QQ34M)

Miscellaneous

L1	Enamelled copper wire, 18swg ¼m		(BL25C)
	T03 mounting kit	2 off	(WR24B)
	Thermpath		(HQ00A)
	Printed circuit board		(GA28F)
	Mounting bracket		(GA29G)
	Pin 2141	5 off	(FL21X)
	Bolt 6BA, ½in	5 off	(BF06G)
	Nut 6BA	5 off	(BF18U)
	Shakeproof washer, 6BA	5 off	(BF26D)

Power supply

C12,13	4700uF, 63V, Can elect.	2 off	(FF28F)
BR1	Bridge J02		(BL36P)
T1	Transformer, 32-0-32V, 2A.		(YK02C)
FS1,2	Fuse, 2A 20mm	2 off	(WR05F)
	Fuseholder	2 off	(RX49D)

Test components

	100R, 5W, W/W	2 off	(L100R)
	Fuse 250mA 20mm	2 off	(WR01B)

Note A complete kit (LW51F) of all the parts listed under Amplifier is available for just £11.49 inc. VAT and P&P from Maplin Electronic Supplies Ltd. The kit does not include the power supply or test components.

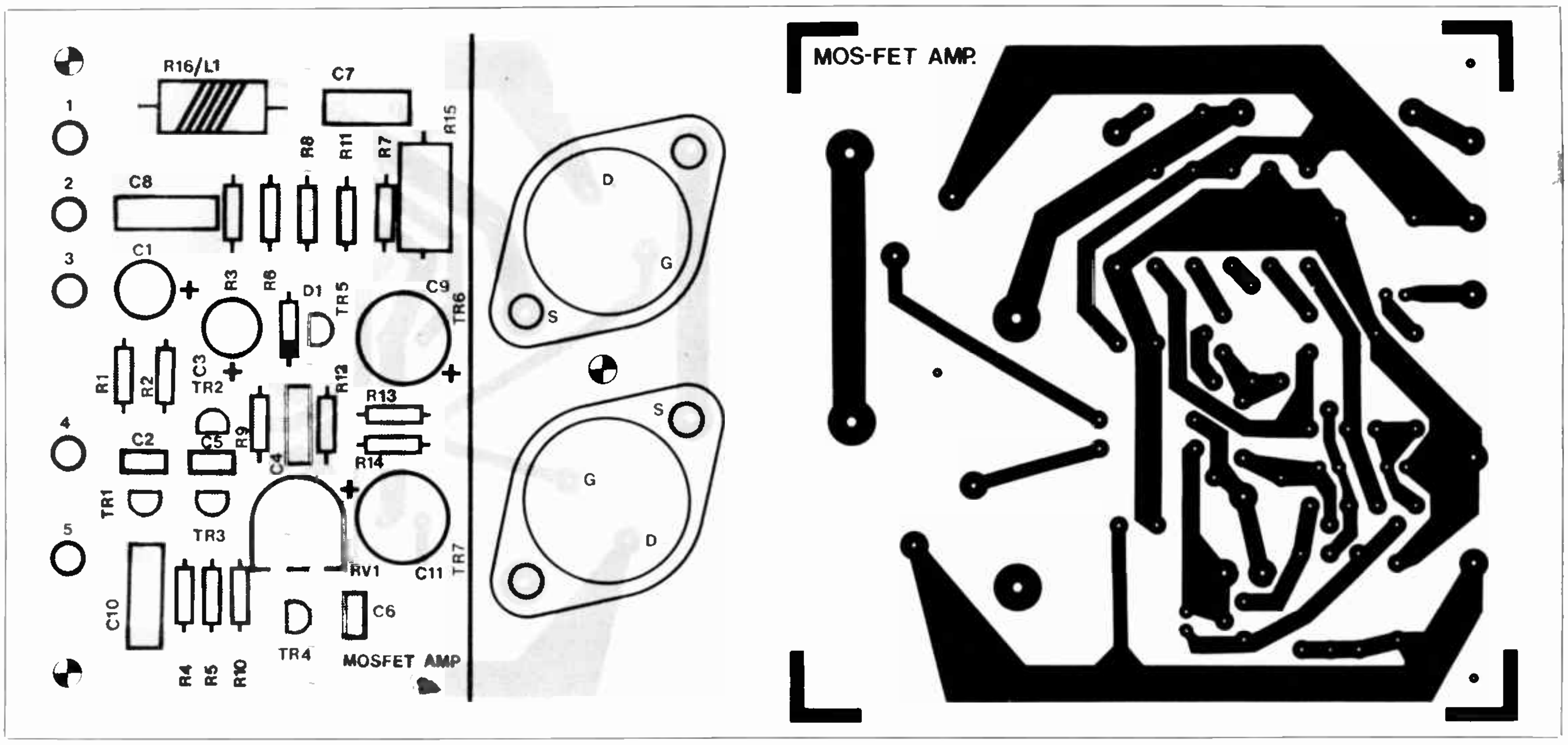


Figure 5. PCB component overlay and track layout

and FS1 and 2) will produce approximately 44-0-44V DC. For a stereo pair use a 4A transformer at 32-0-32V instead of a 2A type. Alternatively, toroidal transformers rated at 35-0-35V could be used, 160VA for a single amp and 300VA for a stereo pair. If the transformer voltage is increased to 40-0-40V and TR6 and 7 replaced by a 2SK134 and 2SJ48 respectively, output powers in excess of 75W RMS into 8 ohms are possible.

Figure 9 shows how simple it is to parallel the output transistors to achieve even higher powers. Using the higher voltage and transistor types just mentioned power levels in excess of 125W RMS into 4 ohms are possible with a 1V RMS input signal if this circuit is used.

Setting Up

With no speaker connected and fuses not inserted, check that the voltage across C12 is approximately 45V (± 5V) and that the voltage across C13 is the same. Switch off and short C12 and C13 in turn with a resistor (e.g. one of the test resistors). Now connect FS1 and FS2, via 100R 5W resistors, to pins 2 and 5 respectively. Connect 0V to pin 4. Check with a multimeter set to the highest resistance range, that there is no connection between the MOSFET cases and the mounting bracket. Turn RV1 fully clockwise.

Insert 250mA fuses for test purposes as FS1 and FS2 and switch on again. If either fuse blows or any component gets excessively hot switch off immediately. If all is well, connect a

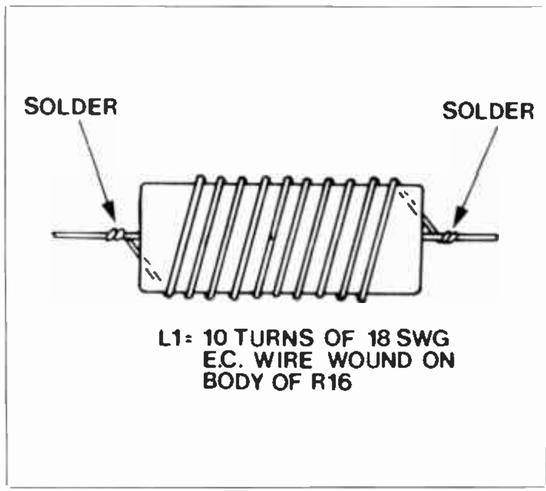


Figure 6. Making the inductor L1.

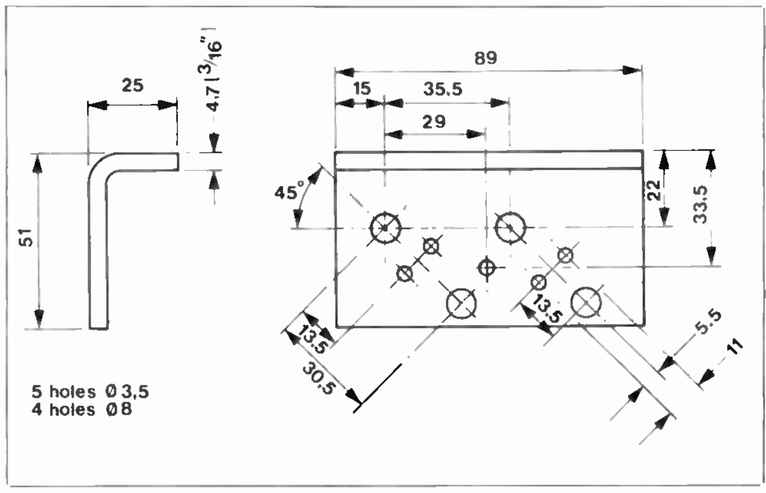


Figure 7. Mounting bracket.

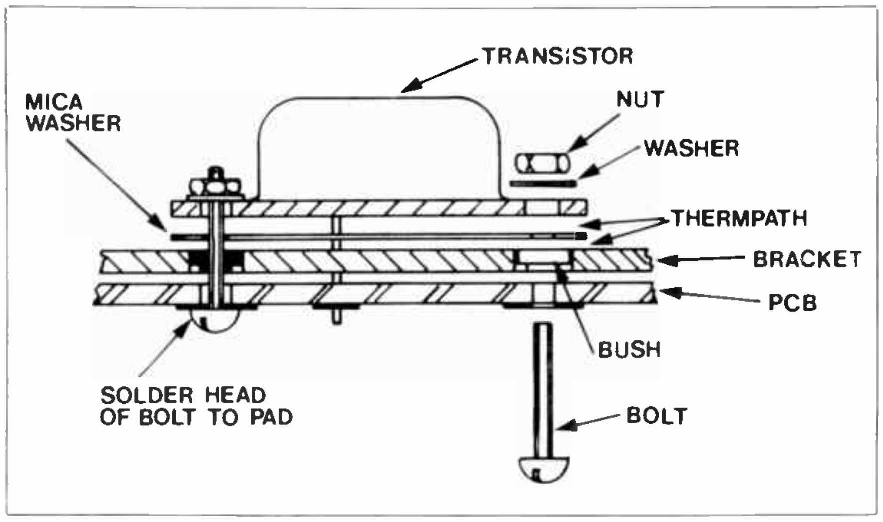


Figure 8. Method of mounting MOSFETs.

DC voltmeter between pin 1 and pin 4. The meter should read about 0V (not more than ± 100mV). Switch off and remove the two 100R resistors. Connect FS2 directly to pin 5 and connect a multimeter switched to about 100mA DC between FS1 and pin 2 (+ve lead to fuse and -ve lead to pin 2). Switch on again and rotate RV1 slowly until the meter reads 50mA. Leave for 10 minutes and re-adjust.

Switch off, disconnect the meter and connect FS1 direct to pin 2. The mounting bracket must now be bolted to a good sized heatsink or a substantial chassis. Finally, connect a loudspeaker to pin 1. Note that the speaker negative terminal must be returned to the 0V in the power supply and *not* to pin 4 of the amplifier. We recommend making the negative

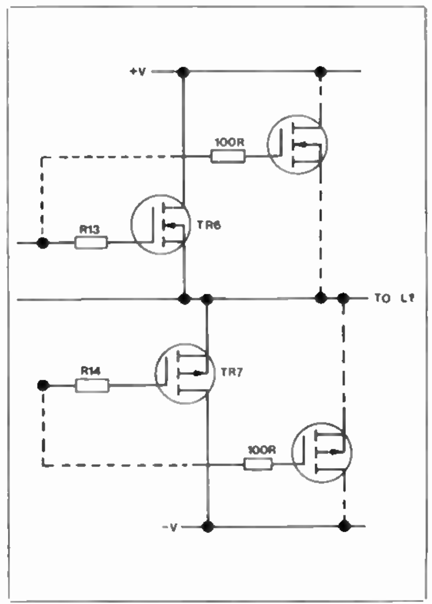


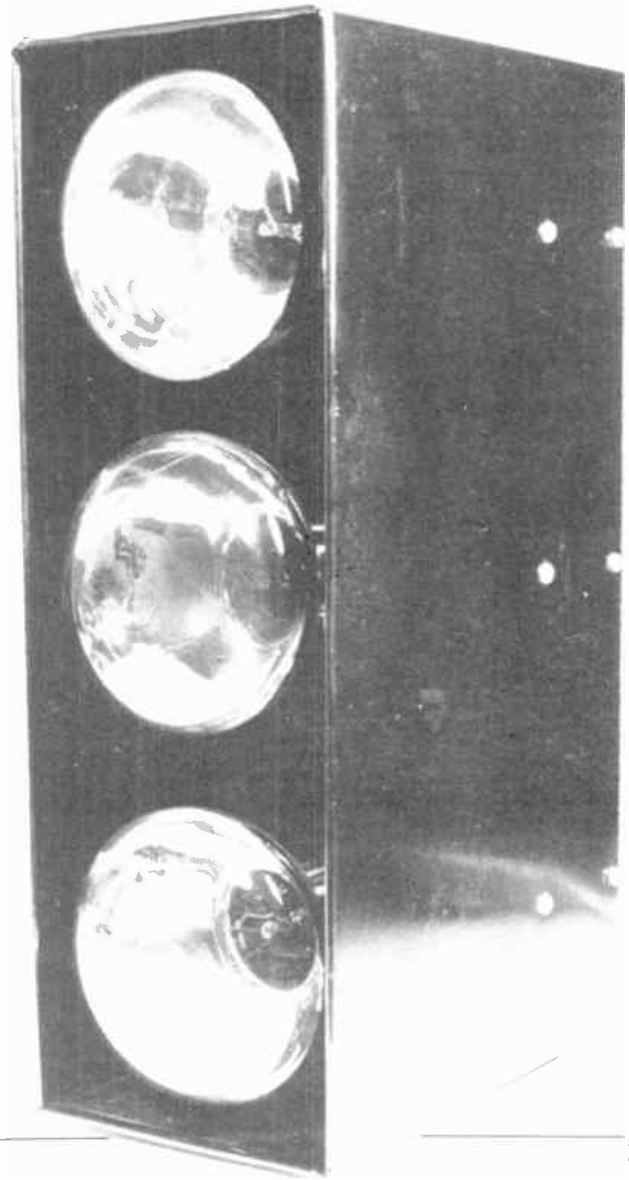
Figure 9. Parallel output connections.

tag of C12 the only 0V point with more than one wire attached. Replace the 250mA fuses with 2A types and connect an input between pins 3 and 4. Note that if you use a toroidal transformer you will have to use antisurge fuses and the test fuses used should be 500mA rating.

PARTYLITE

A 3-Channel sound-to-light modulator

by Clive Button



- ★ No connection to your sound system required
- ★ Automatic level adjustment
- ★ 3-channel operation
- ★ Operates from any sound level

KIT COST ONLY £8.45 inc. VAT

The idea of a three-channel sound-to-light modulator is obviously not a new one, there being a multitude of units of this type already available, ranging from professional products to the types available at supermarkets for domestic use. Nevertheless, the Partylite is a worthy addition to the range because of its simplicity. It is fully automatic - no knobs to re-adjust every time the level or tonal content of your music alters. The Partylite also has its own built-in microphone eliminating the need for an audio connecting lead, making a completely free-standing unit and also avoiding the possibility of damage to your hi-fi or power amp. The Partylite employs zero voltage triggering of the thyristors. Consequently no interference is generated to produce

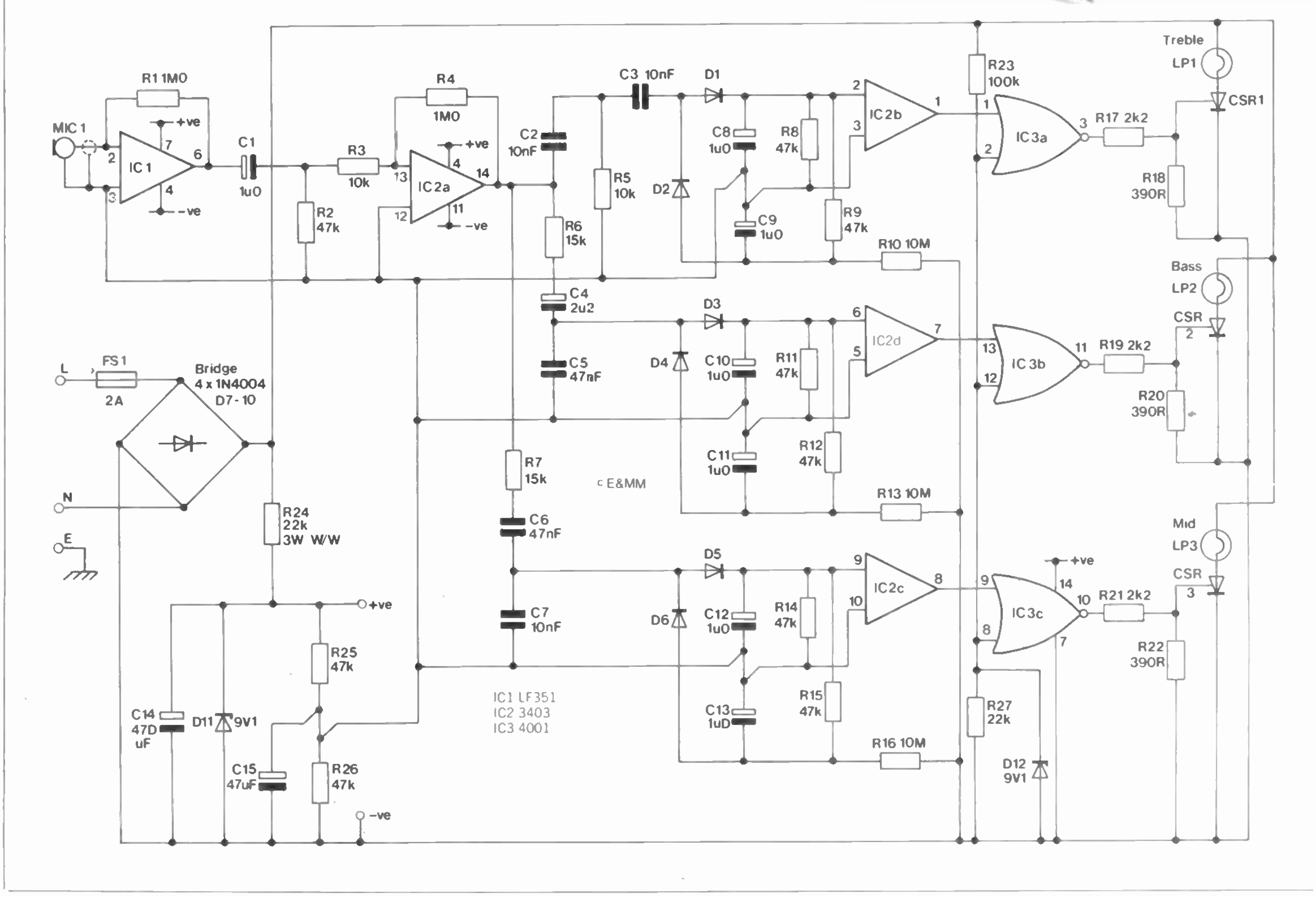


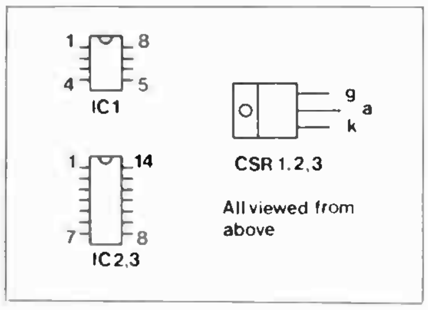
Figure 1. Partylite circuit diagram.

those annoying clicks through the speakers, so common with cheaper sound-to-light units. It will work effectively on all three channels with music at normal domestic listening levels or in a disco environment with say 100W pumping out. This is achieved by having independent, automatic level control circuits for treble, middle and bass frequencies.

Circuit Description

The circuit diagram of the Partylite is shown in Figure 1. IC1 acts as the input amplifier for the crystal mic; this signal is then fed to IC2a which provides amplification to a level sufficient to feed into the three filter networks. C2, C3 and R5 pass treble frequencies; R6, C4 and C5 pass bass frequencies, and R7, C6 and C7 pass mid band frequencies. The outputs of these filters feed individual diode pump/comparator/automatic level control circuits (ALC). We only need to look at one of these in detail because all three are identical in operation. Consider the circuit around IC2b, the treble ALC. Its input comes from C3 with D1 passing the positive-going parts of the waveform and 'pumping-up' capacitor C8 with a positive charge. This level change is fed to the inverting input of the comparator IC2b and every time the level exceeds the non-inverting input, the comparator output will change state from positive to negative. Across C8, however, is a 47k resistor leaking its charge away, giving only short term level changes in response to the treble input. Working against this positive charge is D2, passing the negative part of the waveform and pumping up C9 in a negative direction but with no resistor across it, thus giving a slow response to the treble input. This negative charge is summed with the positive charge on C8 via resistor R9 giving ALC action as follows. If the input level increases (i.e. the music loudness increases), C8 will charge to a higher positive voltage, C9 will also charge, but negatively, consequently they will cancel each other out and the fast response of C8 to the input peaks will be the only signal to flip over the comparator output. This ensures that no matter what the level of the music is or what its tonal content may be, all three channels will always function and never stay permanently on or off. As C8 also has a delay time constant, even the fastest of treble spikes will give a finite 'on' time for the comparator not just an instant flip over, eliminating the annoying flicker associated with many sound-to-light units.

The 10M resistor R10 ensures that all lamps are properly extinguished when no signal input is present by providing a small negative bias to the comparator input. IC3 provides zero voltage triggering making sure no interference is generated by the circuit (another common fault of cheap units). IC3 is a quad NOR gate of which three gates are used. Looking again at the treble channel IC3a has two inputs both of which must be at logical '0' for it to give an output. One input is fed from



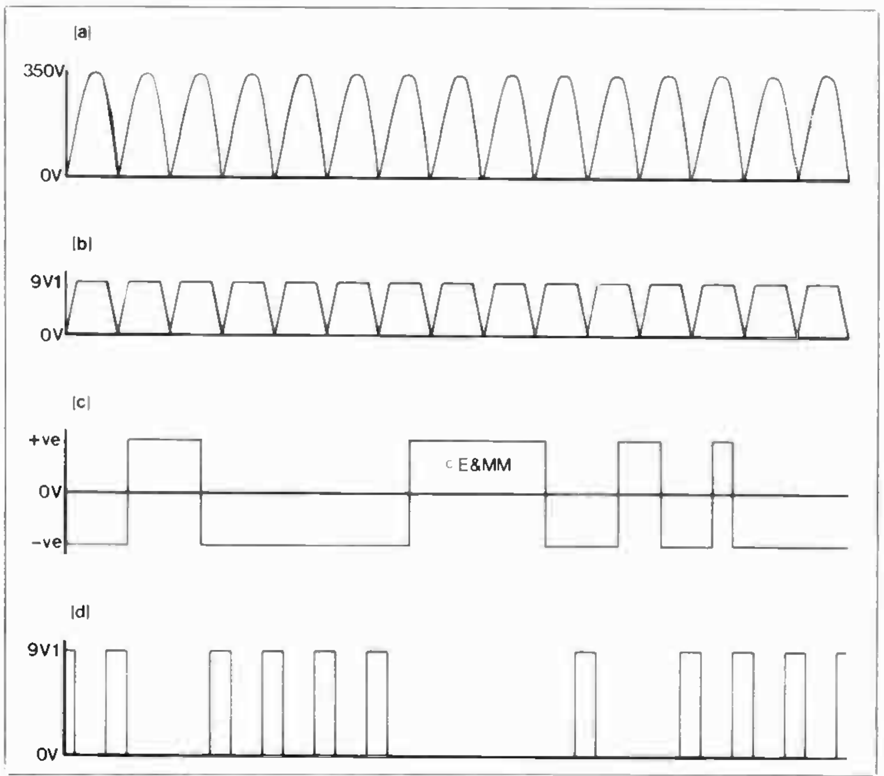


Figure 2. Triggering waveforms.

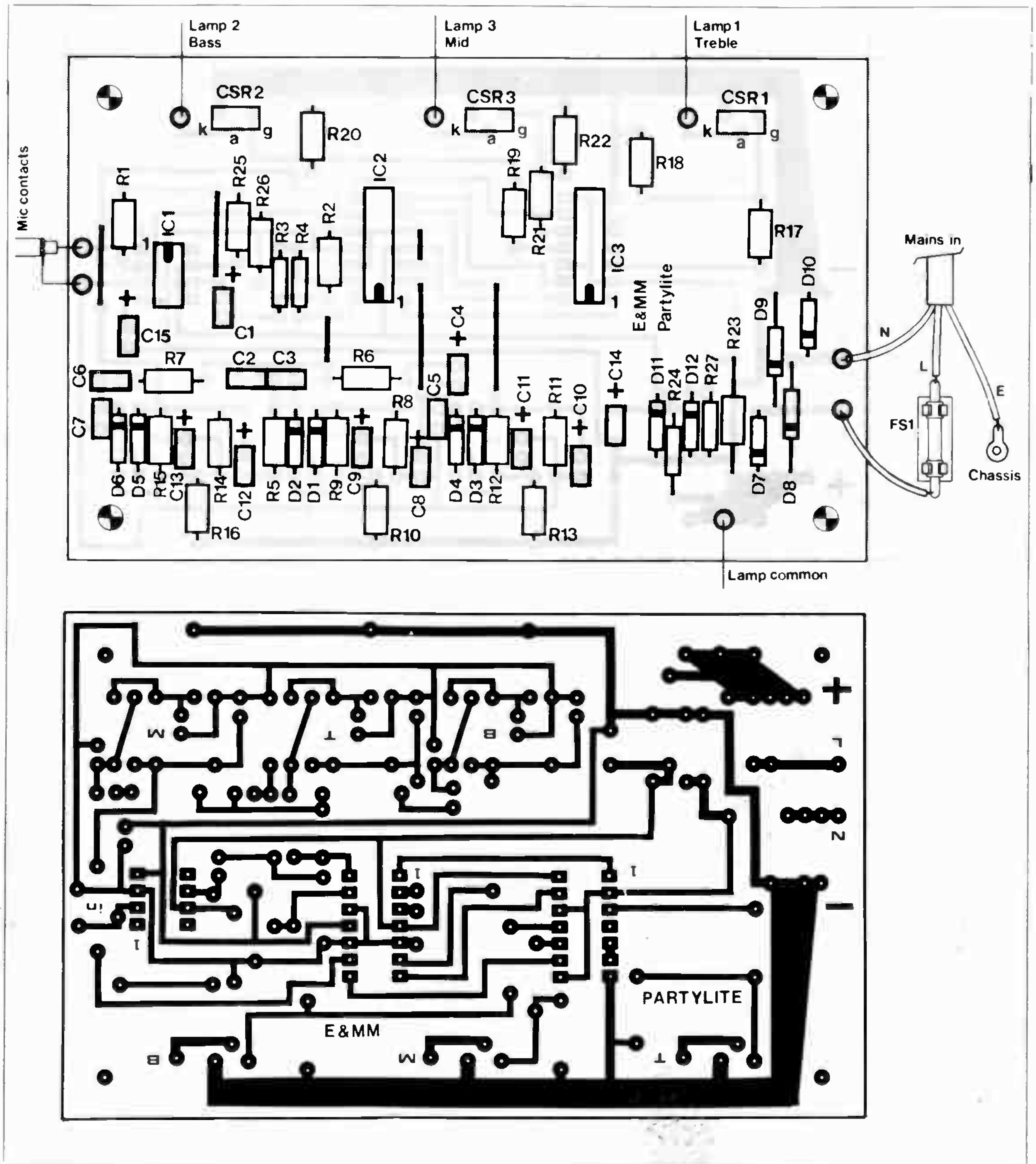


Figure 3. PCB track and legend details.

the comparator tnat goes to '0' on detecting treble pulses; the other input is fed directly from the bridge rectifier with 100Hz positive pulses; these are suitably attenuated and limited to 9.1V to suit the inputs (see Figure 2). Thus an output can only be obtained when the mains voltage is zero, which happens at a rate of 100Hz, providing suitable gate pulses for switching the thyristors.

Warning:

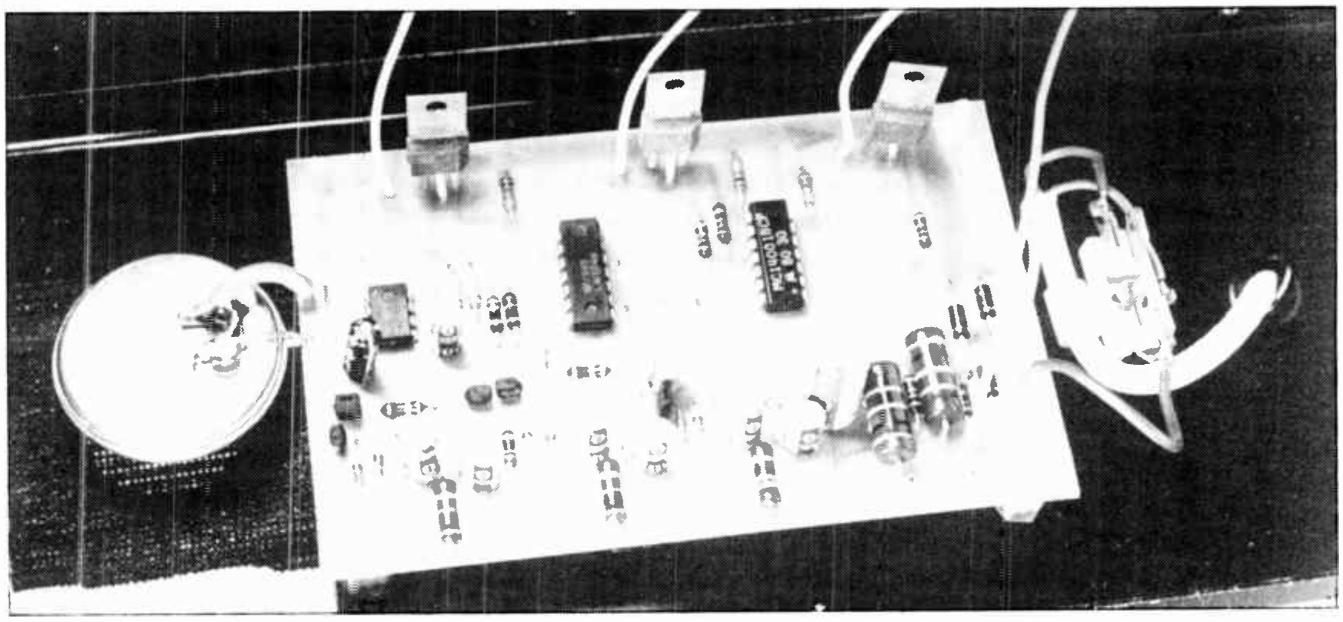
The entire circuit is at live mains potential so be sure that it is properly mounted and avoid tampering with it when the unit is plugged into the mains supply. This includes the crystal mic which must be of the metal body type for input screening purposes and MUST be mounted to the PCB and not the unit case. The case must also be securely earthed.

Construction

The unit can be built either as a complete set-up containing modulator and lights as shown in the photographs, or as a separate modulator having output sockets to which the lights can be connected. In either situation the maximum power handling with the circuit as shown is 150W per channel. This limitation is imposed by the rating of the diodes in the bridge circuit and can easily be up-graded to 500W per channel by using 1N5404 diodes instead of 1N4004's and increasing the mains fuse to 10A (1¼" type).

Pay special attention to the warning given at the end of the circuit description and when wiring up the lamps/lamp outlets avoid running leads too close to the microphone end of the PCB as pick-up from these may be enough to operate the bass channel.

Mount all components and wire links to the PCB leaving the ICs until last (handle the 4001 carefully as it is a CMOS device). Mount the PCB on plastic mounts to keep it well isolated from surrounding casework etc. If necessary, insulate it from close proximity metal by covering the metal with insulating material. Fit a fuse external to the board in the live side of the mains input; this fuse should have a maximum rating of 2.5A for the circuit shown. Mount the microphone by twisting a length of heavy gauge tinned copper wire around its circumference, point solder it in position at, say, three points, and use this as the connection to the non-inverting input of IC1, thus providing a rigid mounting for the microphone.



Fully assembled PCB.

The microphone must be mounted to the PCB and not to any casework, for the unit to operate successfully. Use a short length of screened lead to connect the microphone input to the circuit board; the sensitivity of the unit is sufficient to cause premature turn-on if a plain lead is used.

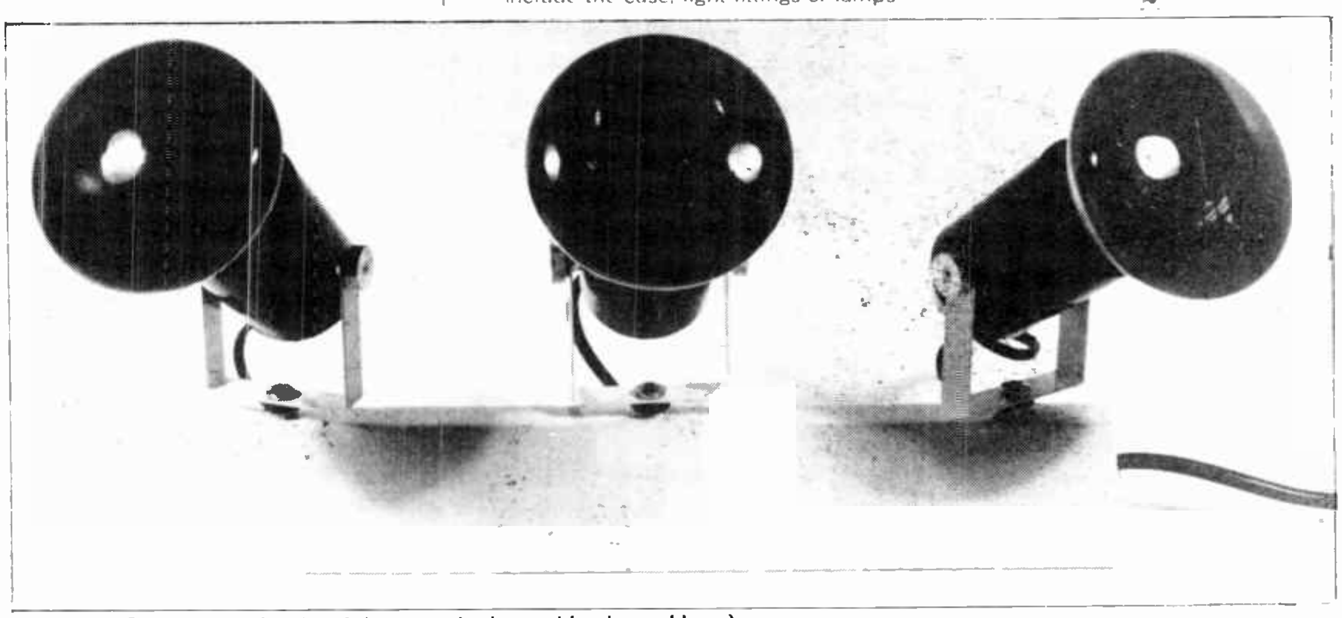
Testing

After the unit has been completed and the lamps connected, check that there is no continuity between the board and the case. The unit can then be plugged into the mains for testing. Should it not function correctly, disconnect the unit from the supply before making any checks. If the bass channel lamp stays on it is probably due to pick-up either from lamp outputs or main leads; re-route these and try again. If the unit is not sensitive enough the 10k resistor from the output of the first op-amp to the input of the second may be reduced in value or alternatively, a 10k pre-set may be fitted in its place to allow adjustment of the sensitivity. *(Use an insulated screwdriver for adjusting this pre-set if it is fitted).*

PARTYLITE PARTS LIST

Resistors - all 1% 0.4W metal film unless specified			
R1,4	1M	2 off	(M1M)
R2,8,9,11,12,14, 15,25,26	47k	9 off	(M47K)
R3,5	10k	2 off	(M10K)
R6,7	15k	2 off	(M15K)
R10,13,16	10M, 10%	3 off	(M10M)
R17,19,21	2k2	3 off	(M2K2)
R18,20,22	390R	3 off	(M390R)
R23	100k, ½W, 5% carbon		(S100K)
R24	22k, 3W, wire wound Min		(W22K)
R27	22k		(M22K)
Capacitors			
C1,8,9,10,11,12,13	1u, 100V PC electrolytic	7 off	(FF01B)
C2,3,7	10n minidisc ceramic	3 off	(YR73Q)
C4	2u2, 63V PC electrolytic		(FF02C)
C5,6	47n minidisc ceramic	2 off	(YR74R)
C14	470u, 16V PC electrolytic		(FF15R)
C15	47u, 25V PC electrolytic		(FF08J)
Semiconductors			
IC1	LF351		(WQ30H)
IC2	3403		(QH51F)
IC3	CD4001		(QX01B)
D1-6	1N4148	6 off	(QL80B)
D7-10	1N4004	4 off	(QL76H)
D11,12	BZY88C9V1	2 off	(QH13P)
CSR1-3	C106D	3 off	(QH30H)
Miscellaneous			
	Case } *		
	Light fittings } *		
	Lamps		
	Mains cable	2m	(XR01)
	Mains plug		(RW67)
	PCB mounting pillars		(FW16)
	Fuse holder		(RX49)
	Fuse, 2 Amp		(WR5)
	Crystal mic		(HY33L)
	PCB		(GA42V)
	DIL socket, 14 pin	2 off	(BL18U)
	DIL socket, 8 pin		(BL17T)

Note: A complete kit (LW93B) of all the parts listed under Partylite is available for just £8.45 inc VAT & P&P from Maplin Electronic Supplies Ltd. The kit does not include the case, light fittings or lamps.



Alternative 3-light mounting (modulator can be housed in channel base).

The Syntom DRUM SYNTHESISER

Join Warren Cann in the drum revolution with this unique touch sensitive instrument costing only £10.90 in kit form

by Clive Button

The Syntom is a very effective drum synthesiser that can produce a variety of fixed and falling pitch effects, triggered either by tapping the unit itself, or by striking an existing drum to which the device is attached.

Four potentiometers give control over different characteristics of the sound, the Volume control being used to switch off the internal battery as well as determining the level of the signal sent to the external amplifier. The Decay pot. governs the time taken for the sound to die away after each strike, from less than 1/10 sec. to several seconds, giving a wide range of envelopes. The frequency of the note is variable over the entire audio range by means of the Pitch control, and the Sweep control introduces a voltage causing the pitch to fall as the amplitude decreases. These controls, when used in combination with each other enable the most popular drum synthesiser effects heard on commercial recordings to be obtained.

Circuit

The Circuit is in three main parts: the envelope generator, the Voltage Controlled Oscillator (VCO), and the Voltage Controlled Amplifier (VCA). IC1 forms the first stage of the envelope generator, detecting the signal produced by the crystal earpiece when the unit or the drum to which it is fitted is struck. The trigger signal charges C1 via D1, and the capacitor is then discharged slowly by RV1 and R3. This envelope voltage is buffered by IC2c and sent to the VCA. It is also fed (via RV2 — the Sweep potentiometer) to IC2d, the VCO control voltage summing amplifier where it is mixed with a voltage from the Pitch control, RV3.

The VCO consists of an integrator formed around IC2a, and a Schmitt trigger (IC2b) driving TR1. When the integrator voltage approaches the upper threshold of IC2b, TR1 is turned on shorting the non-inverting input of the integrator to earth, causing it to act in inverting mode. Hence the output voltage falls until the lower threshold is reached, IC2b changes state, turning off TR1, and the output of IC2a starts to rise, as it is once more in non-inverting mode. The resultant triangle wave is fed to the VCA section, which consists of a CA3080 transconductance amplifier, IC3. The gain of this amplifier is controlled by the output of the envelope generator, such that as the envelope voltage decays,

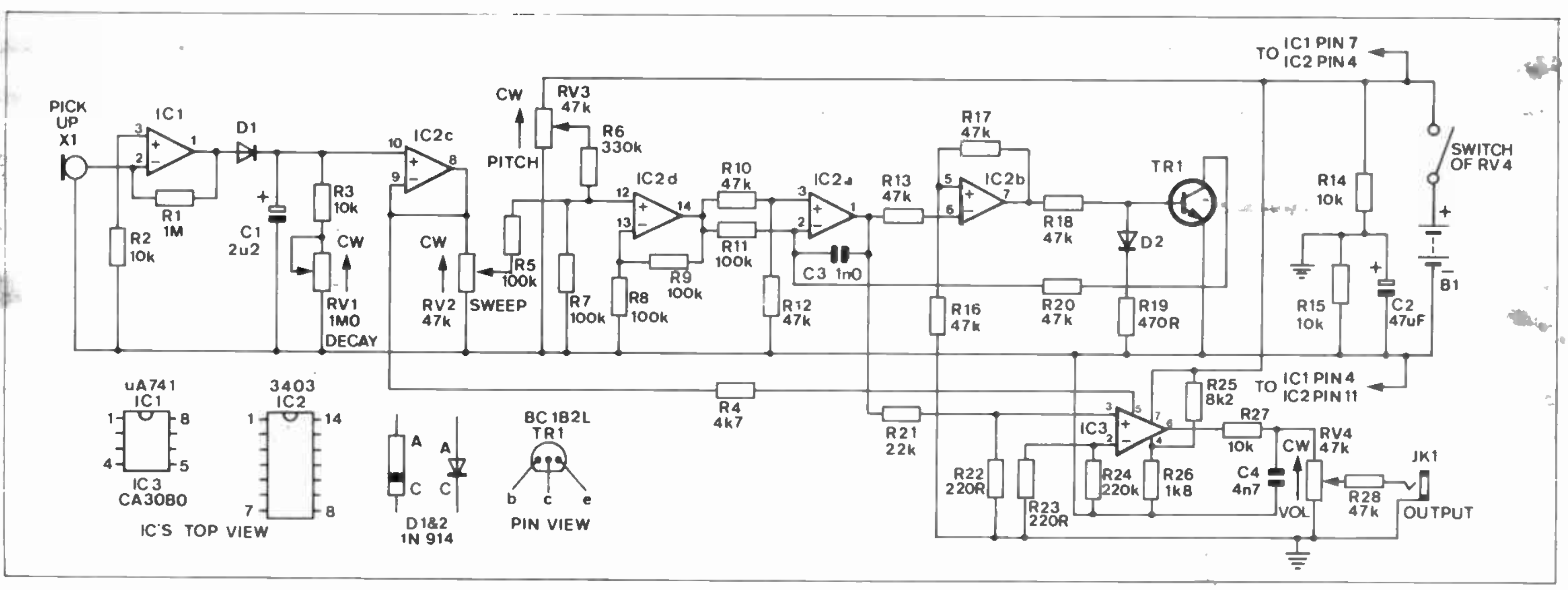


Figure 1. The circuit diagram of the Syntom.

the triangle wave is increasingly attenuated until it is reduced to a very low, inaudible level. The output of the CA3080 is fed to RV4, the Volume pot, and then on to the jack socket.

A dual supply is derived from the single 9V battery by a potential divider formed by R14 and R15, providing a 0V supply which is stabilised by C2.

Constructional Details

All resistors, capacitors and semiconductors except R28 are mounted on the printed circuit board in that order, taking care as always with the orientation of electrolytic capacitors, IC's, diodes, and the transistor. If the suggested case is used, veropins for connection of the pots, jack, battery and earpiece must be mounted from the component side since this side faces away from them and there is no room for the wires to pass around the edge of the board. Otherwise they fit from the track side, or can be left out altogether, the wires being soldered directly to the tracks.

The potentiometers are mounted on the front side (which is the side opposite the removable side if using the case suggested in the parts list), after their spindles have been sawn to a length suitable for the knobs. The jack socket is best mounted on the back, where the lead to the external amplifier will be out of the way during use, but take care here since the board, battery and earpiece all fit near the back of the case. The connections to the off-board components can now be made, and the PCB fitted in the special slots on the inside of the case (with the track side facing towards the pots). Note that R28 is connected directly from the wiper of RV4 to the signal terminal of the jack socket.

For use with an existing drum, the Syntom is attached to the drum by a securing bolt and a bracket made from 25mm aluminium channel section which is fixed to the case by two bolts with washers. A simple hexagonal-head bolt could be used, but the handwheel bolt specified in the parts list is much easier to use, and lends a professional appearance to the finished unit. One side of the bracket must be drilled and threaded to accommodate the bolt, and it is a good idea to stick a small piece of rubber on the inner face of the opposite side to prevent scratching of the drum rim. The final constructional stage is to fit the knobs, connect the battery using a PP3 connector, and screw on the back of the case. A piece of foam glued to the inside of the back will hold the battery against the potentio-

PARTS LIST

Resistors - all 1% 0.4W metal film unless specified

R1	1MΩ			(M1M0)
R2,3,14,15,27	10k		5 off	(M10K)
R4	4k7			(M4K7)
R5,7,8,9,11	100k		5 off	(M100K)
R6	330k			(M330K)
R10,12,13,16-18,20,28	47k		8 off	(M47K)
R19	470R			(M470R)
R21	22k			(M22K)
R22,23	220R		2 off	(M220R)
R24	220k			(M220K)
R25	8k2			(M8K2)
R26	1k8			(M1K8)
RV1	1M0 log. pot.			(FW28F)
RV2,3	47k log. pot.		2 off	(FW24B)
RV4	47k log. pot. with switch			(FW65V)

Capacitors

C1	2u2 63V axial electrolytic	(FB15R)
C2	47u 10V axial electrolytic	(FB38R)
C3	1n0 Mylar Film	(WW15R)
C4	4n7 Mylar Film	(WW17T)

Semiconductors

IC1	uA741, 8-pin DIL		(QL22Y)
IC2	3403		(QH51F)
IC3	CA3080, 8-pin DIL		(YH58N)
TR1	BC182L		(QB55K)
D1,2	1N914	2 off	(QL71N)

Miscellaneous

X1	Crystal earpiece		(LB25C)
SK1	Mono-jack socket (open type)		(HF91Y)
	Case MB2		(LH21X)
	Handwheel bolt		(YL23A)
	M4 6mm bolts		(BF33L)
	Printed circuit board		(GA05F)
	PP3 connector		(HF28F)
B1	PP3 battery		
	Ribbon cable (10 way)	1m	(XR06G)
	1mm Veropins		(FL23A)
	Knobs	4 off	(YG40T)
	Blue knob cap		(QY01B)
	Green knob cap		(QY02C)
	Grey knob cap		(QY03D)
	Red knob cap		(QY04E)
	Front Panel		(BH60Q)

Note: a complete kit LW86T of parts listed is available from Maplin Electronic Supplies Ltd. price just £10.90 inc. VAT & P&P. The kit does not include batteries.

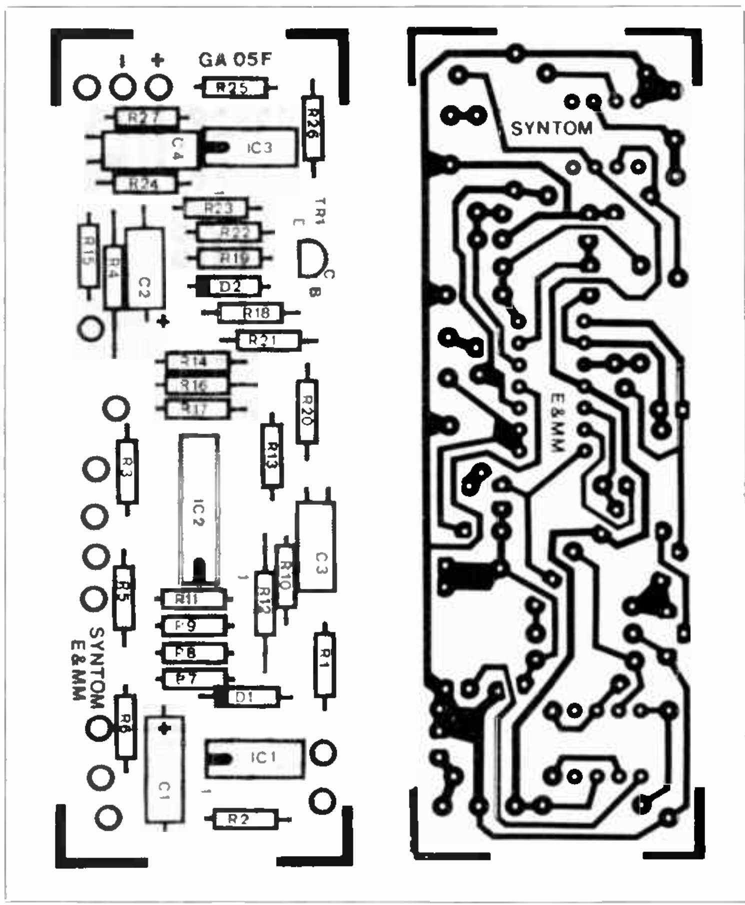
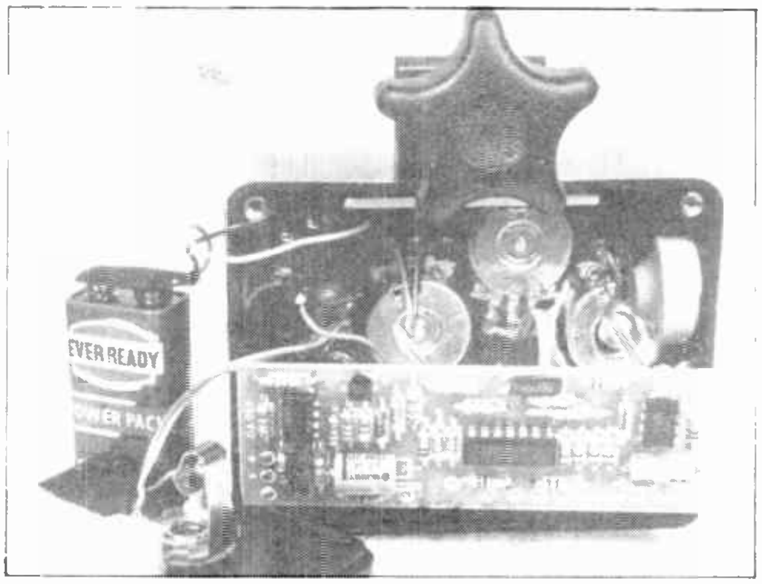


Figure 2. The Syntom PCB.



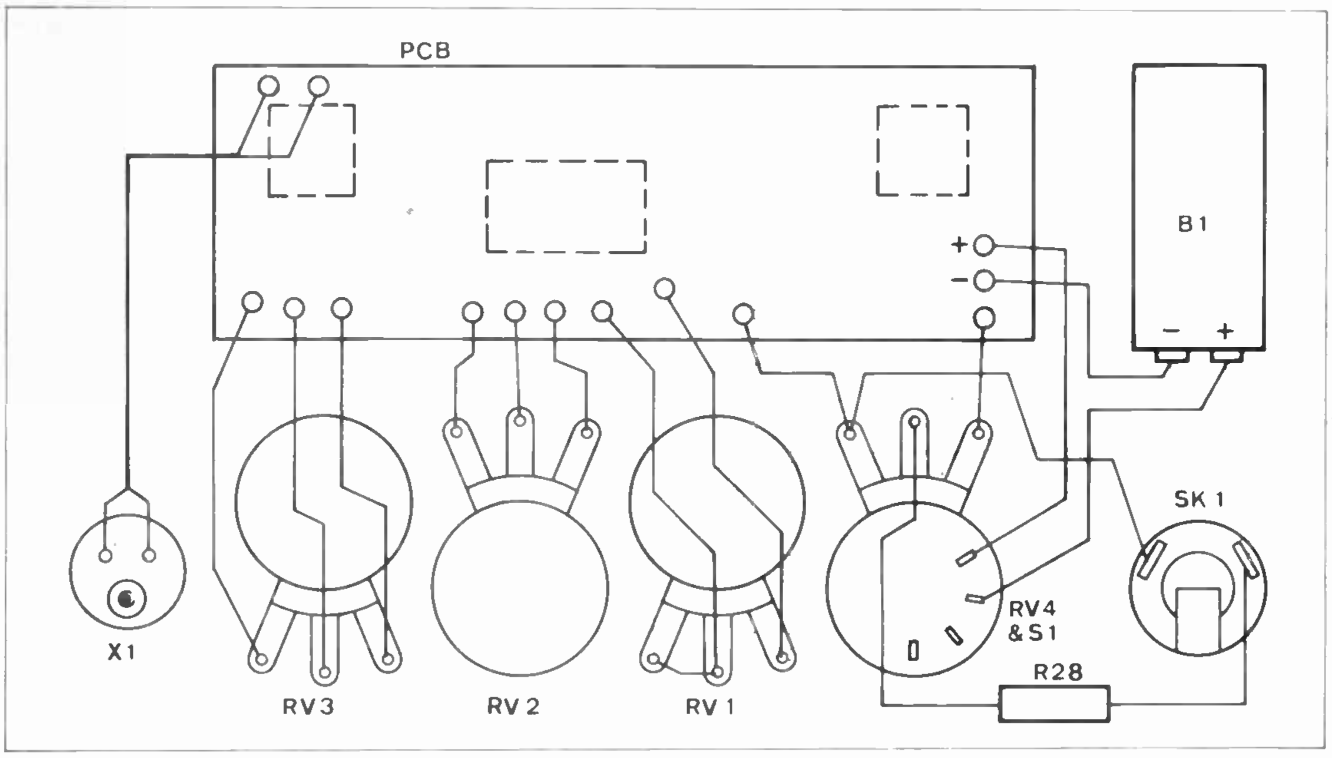


Figure 3. PCB wiring, with the board viewed from the track side.

Continued on page 29

SYNWAVE

* Sounds from swirling sea to cymbal and wood block
* Easy to build and play
* Dual trigger operation - percussive/external trigger
* Touch sensitive

KIT COST ONLY £10.25 inc. VAT

Design - Mike Beecher
Development - Robert Penfold

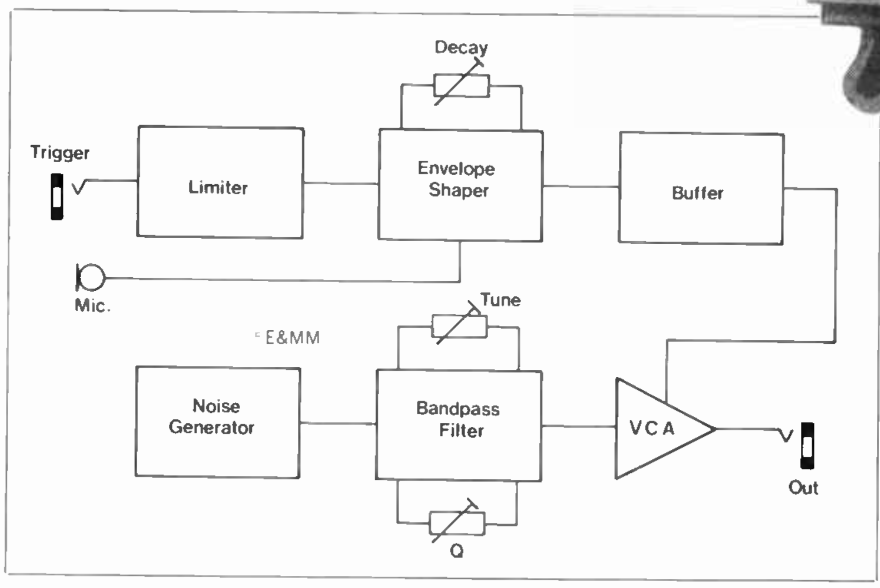
RECORDED ON CASSETTE NO. 3

The Synwave continues our series of small projects that produce sounds for use in electro-music by percussive control. The minimum number of controls have been selected to give a wide range of 'sea-wave' sounds. In addition, different settings of the controls will produce wind, cymbal and woodblock sounds.

Like the 'Syntom' project shown elsewhere in this issue, the unit can be triggered by tapping the case or by striking a drum (on which the Synwave is mounted). These projects are also ideal for triggering from an external source (e.g. a sequencer synthesiser or micro) and thus a second mode of operation can be from an electronic trigger using a positive-going edge of about 7 to 15 volts in amplitude. Interaction of the two modes of use is possible so that complex rhythms can be made from a steady 'external triggered' beat mixed with hand or drum taps providing syncopation.

The four controls are Volume (with on/off switch), for setting output level; Decay - adjusts the time it takes for the sound to die away; Pitch - sets the frequency range of noise from low to high; 'Q' - a resonance control that narrows and highlights the pitch range selected.

Circuit



The block diagram of Figure 1 shows the general arrangement used in the Synwave. An envelope shaper can be operated by either an internal microphone or an external trigger signal, or both. The envelope shaper has a fast attack and slow decay, with the latter being adjustable from less than 100ms to about 5 seconds. The output of the envelope shaper is fed to the control input of a voltage controlled amplifier (VCA) via buffer stage.

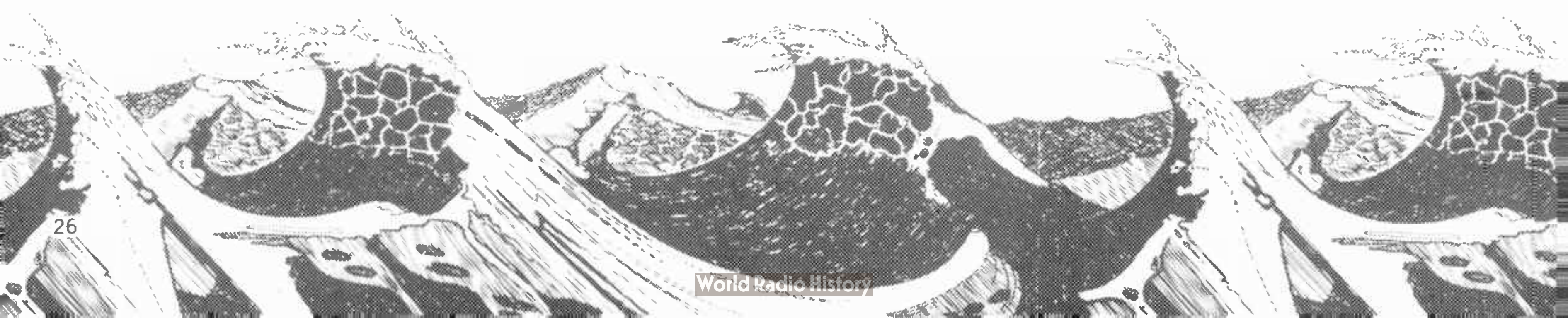
A simple noise generator feeds a bandpass filter which in turn feeds the input of the VCA. The bandpass filter is tunable from a few hundred hertz to more than 10kHz, and therefore gives considerable control over the sound produced by the unit. The bandwidth of the filter can be varied from a very broad response to a very sharp peaky response by means of the Q control, and again, this permits the output sound of the unit to be varied greatly.

If we now consider the full circuit diagram of the Synwave (Figure 2); C1, R1, D1, D2, and R2 process the trigger signal so that on its rising (positive) edge a brief positive pulse of about 7 volts in amplitude is supplied to the base of TR1. TR1 and TR2 form a Darlington pair emitter follower ing the brief input pulse to TR1. R3 and RV1 provide a discharge path for C2, and the setting of RV1 determines the discharge time of C2 (and therefore the length of the output signal).

If crystal microphone X1 is subjected to strong vibrations it will give an output level of several volts peak to peak, and positive going output half cycles will result in C2 being rapidly charged. The output signal level and hence the charge produced on C2 depends on how hard the unit or the drum to which it is attached is struck, and this gives a degree of touch sensitivity.

The signal across C2 must be only lightly loaded or the decay time will be greatly shortened by the charge current leaked away through the loading circuit. TR3 is therefore used as an emitter follower buffer stage which couples the output from C2 to the control input of the VCA. The VCA uses a CA3080 transconductance amplifier, and it gives a level of gain that is proportional to the control voltage. The output signal thus rises rapidly to its peak level, and then decays relatively slowly, in sympathy with the envelope voltage across C2. The output of the VCA is coupled to the output socket by way of volume control RV2. Dual balanced supplies are required by the VCA circuitry and a central 0V rail is effectively produced by R10, R11 and C5.

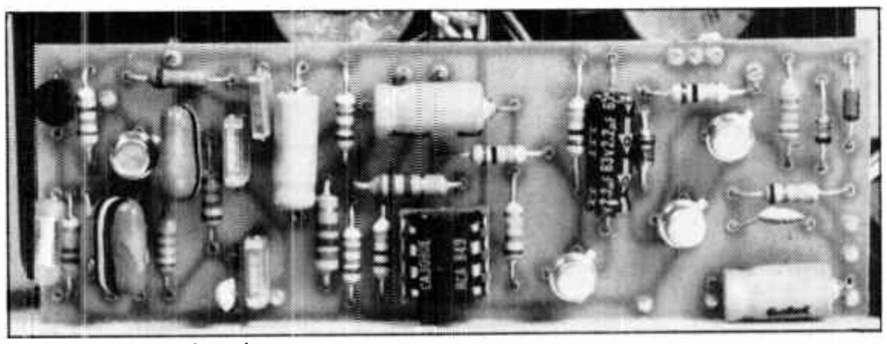
TR5 is used as the noise generator and R18 applies a reverse bias to its base-emitter junction. This junction behaves rather like a Zener diode and like a Zener diode produces noise spikes. This arrangement is preferable to using a Zener diode though, as it gives a higher output at audio frequencies. The high frequency output of the noise generator is



excessive, and so C11 is used to give high frequency attenuation to correct this.

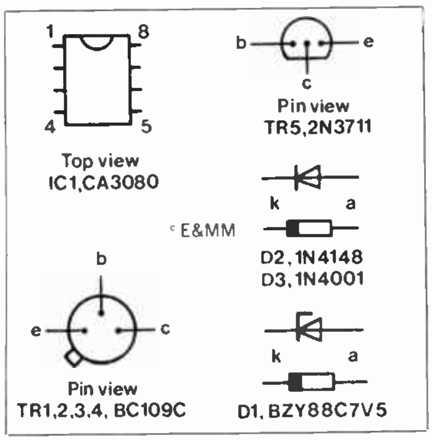
A twin T filter is used as the basis of the bandpass filter, but as a twin T network gives a notch at its centre frequency rather than a peak, the filter network is connected to give negative feedback over a common emitter amplifier. This amplifier features TR4 in a conventional configuration. TR4's emitter circuitry enables a certain amount of negative feedback to be applied to the amplifier and the amount of feedback is controlled by RV4. With the slider of RV4 at or near the lower track connection there is little or no feedback; giving the circuit a high Q value and a narrow, peaky response. Moving the wiper of RV4 towards the upper end of its track gives increased feedback and a consequent reduction in Q together with a broader, flatter response.

RV3 is part of the twin T network, and varying the setting of this component alters the centre frequency of the filter. Ideally all three resistive elements in the filter should be varied when tuning the filter, but this is not really practical. This simple system works quite well though, and the only minor drawback is that the Q of the filter varies somewhat with changes in the setting of the pitch control. At some settings of RV3 it may be found that setting RV4 for a very high Q causes the filter to oscillate at its centre frequency. If desired, this can be avoided by adding a resistor of about 150 ohms in value between the positive terminal of C9 and the upper track connection of RV4. However, as this would limit maximum Q available, especially at the highest and lowest pitch control settings, it would reduce the effectiveness of the unit and is not really worth while.



Completed circuit board.

C13 is needed to prevent the filter becoming unstable due to stray high frequency feedback. C4 couples the output of the filter to the input of the VCA. The current consumption of the circuit is only about 1.5mA., or a little higher than this when it is triggered.

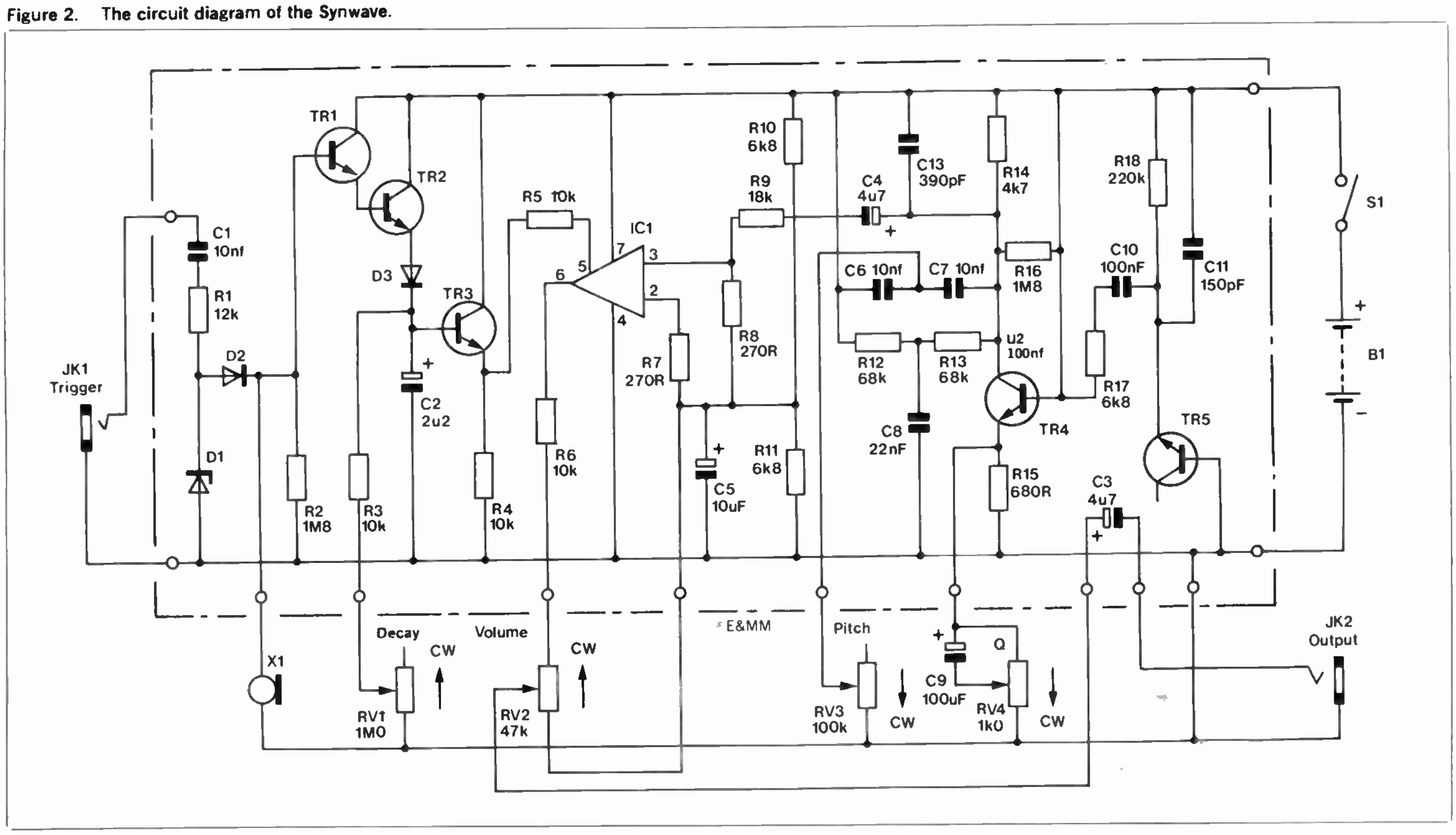


Construction Details

Except for C9 all the resistors, capacitors, and semiconductors are fitted onto a printed circuit board. Fit the semiconductors last, and make quite sure that the electrolytic capacitors and semiconductors are connected the right way round. The printed circuit board fits into the mounting rails of the specified case, but this leaves insufficient room for leads to be taken around the board to the controls, battery clip, microphone, and sockets. This makes it necessary to fit Veropins to the board at the points where it connects to these components, or if preferred, these leads can simply be soldered direct to the copper tracks. Details of the printed circuit board are provided in Figure 3.

The removable lid of the case is used as the rear panel in this application, and the two sockets are mounted on this panel. The front panel is drilled to take the four potentiometers and the microphone is mounted on the right side of the case (as viewed from the front). The microphone is

Figure 2. The circuit diagram of the Synwave.



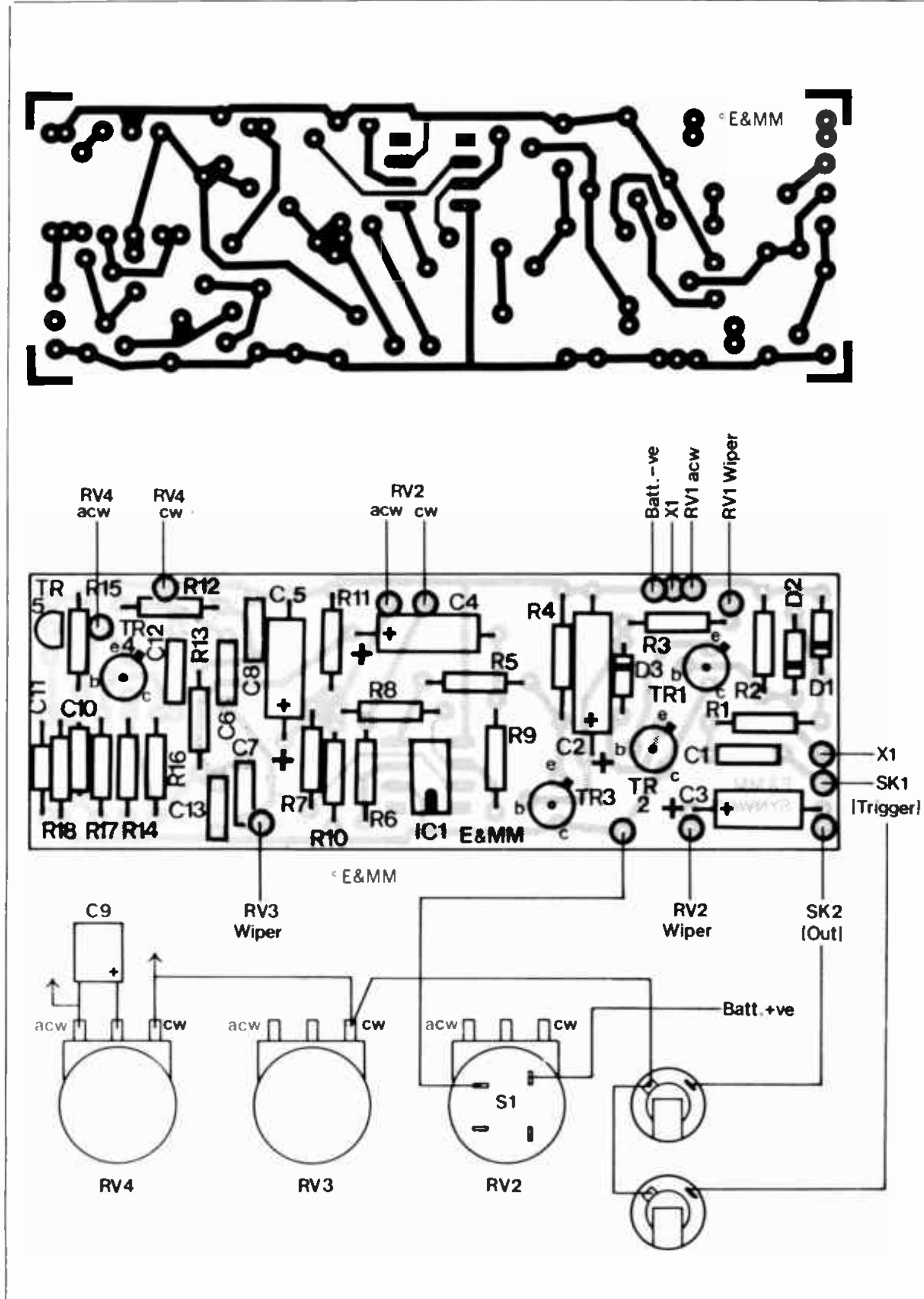
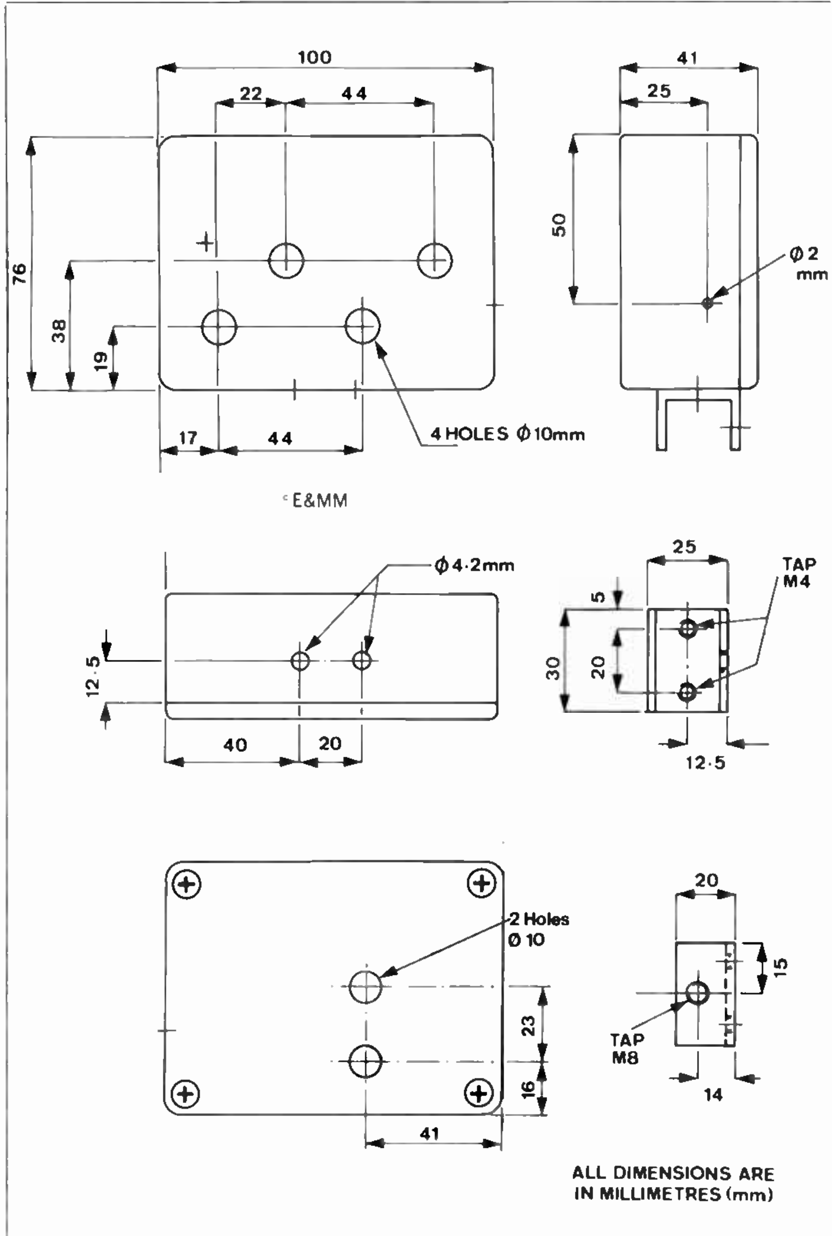
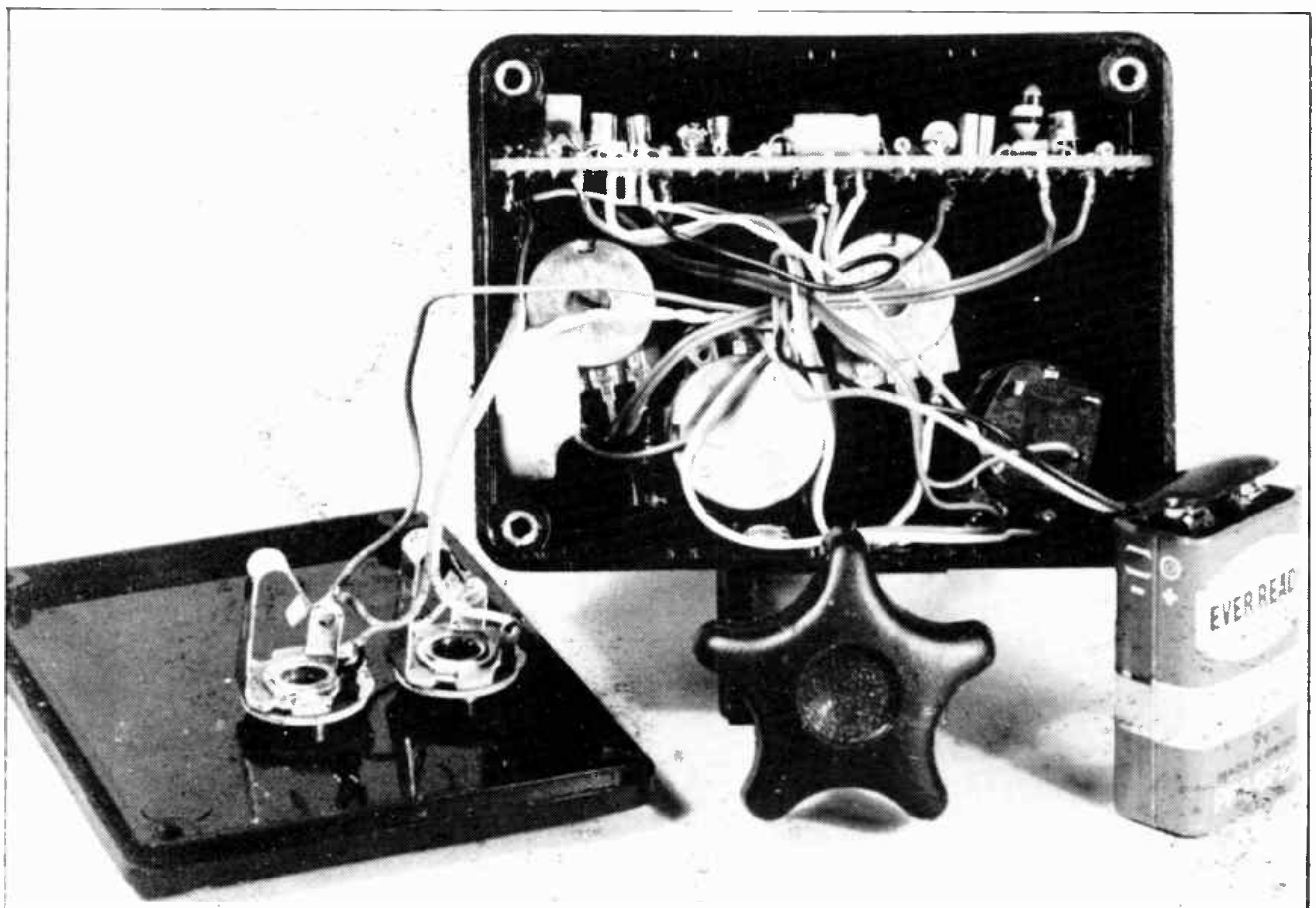


Figure 3. PCB track, component layout and wiring details.



Note: connections to PCB must be made from the track side

Figure 4. Case and bracket construction.



Synwave with wiring completed and PCB board inserted in case.

actually a crystal earphone having the earpip unscrewed. The small screw at the rear of the earphone is removed, and this is then used to fix the earphone to the case. Figure 4, shows the drilling of the case, and it is advisable to follow this as accurately as possible since there is not a great deal of excess space inside the case and it might otherwise be found that all the parts cannot be fitted into the case.

Next C9 is connected to RV4 and the other wiring to the off-board components is completed. Figure 3 gives details of all this wiring. The printed circuit board fits into the upper set of mounting rails in the case with the component side uppermost. The battery fits vertically into the case in the space between the two sockets and the microphone. A piece of foam material can be glued to the rear panel of the case to keep the battery firmly in place.

If the Synwave is to be fitted onto a drum it is necessary to fit the unit with a mounting bracket. This can consist of a piece of

25mm aluminium channel section which is fixed to the case using a couple of 6mm M4 bolts. A large bolt is used to clamp the Synwave onto the drum, and the bracket is drilled and threaded on one side to take this bolt. A handwheel bolt is ideal for use in this application, but an ordinary type can be used of course. It is advisable to fit a small pad of rubber on the part of the bracket opposite the mounting bolt as this will help to prevent the rim of the drum from becoming scratched when the Synwave is fitted in place.

To complete your project use our smart brushed aluminium panel with electric blue legend and sticky back as shown.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified.

R1	12k		(M12K)
R2,16	1M8 (10%)	2 off	(M1M8)
R3,4,5,6	10k	4 off	(M10K)
R7,8	270R	2 off	(M270R)
R9	18k		(M18K)
R10, 11,17	6k8	3 off	(M6K8)
R12,13	68k	2 off	(M68K)
R14	4k7		(M4K7)
R15	680R		(M680R)
R18	220k		(M220K)
RV1	1M lin. pot.		(FW08J)
RV2	47k log. pot. with switch		(FW65V)
RV3	100k lin. pot.		(FW05F)
RV4	1k lin. pot.		(FW00A)
Capacitors			
C1	10n ceramic plate		(WX77J)
C2	2u2 63V axial electrolytic		(FB15R)
C3,4	4u7 63V axial electrolytic	2 off	(FB18U)
C5	10u 25V axial electrolytic		(FB22Y)
C6,7	4n7 polycarbonate	2 off	(WW26D)
C8	10n polycarbonate		(WW29G)
C9	100u 10V PC electrolytic		(FF10L)
C10,12	100n polyester	2 off	(BX76H)
C11	150p polystyrene		(BX29G)
C13	390p ceramic plate		(WX63T)
Semiconductors			
IC1	CA3080, 8-pin, DIL		(YH58N)
TR1,2,3,4	BC109C	4 off	(QB33L)
TR5	2N3711		(QR34M)
D1	BZY88C7V5		(QH11M)
D2	1N4148		(QL80B)
D3	1N4001		(QL73Q)
Miscellaneous			
X1	Crystal earpiece		(LB25C)
SK1,2	Mono jack socket (open type)	2 off	(HF91Y)
	Case MB2		(LH21X)
	Handwheel bolt		(YL23A)
	M4 6mm bolts		(BF33L)
	Printed circuit board		(GA35Q)
	PP3 connector		(HF28F)
B1	PP3 battery		
	1mm Veropins		(FL23A)
	Knobs	4 off	(YG40T)
	Knob cap blue		(QY01B)
	Knob cap grey		(QY03D)
	Knob cap red		(QY04E)
	Knob cap yellow		(QY06G)
	Front panel		(BX99H)

Note. A complete kit (LW87U) of all parts listed is available from Maplin Electronic Supplies at a price of just £10.25 inc VAT and P&P. The kit does not include batteries.

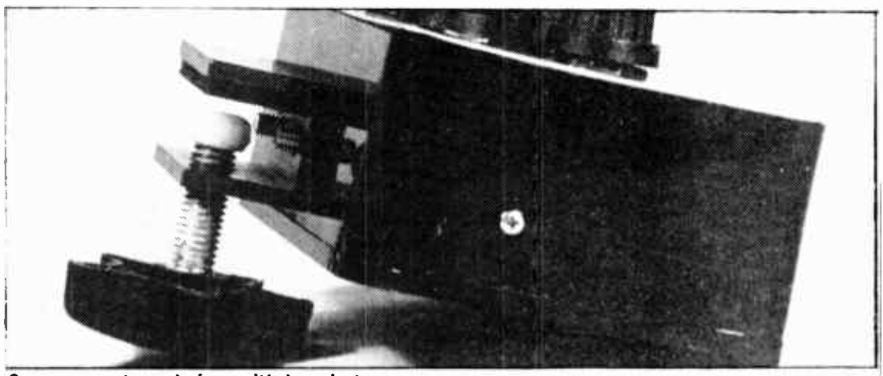
Testing and Use

Connect the Synwave to an amplifier via SK2 and switch on with the Volume control set to midway. Set Decay to maximum, Pitch to minimum and Q to midway. Give the unit a sharp tap or use a suitable trigger signal (applied to SK1) and a 'seawave' should be heard.

Using short Decay and high Pitch and Q, the woodblock sound can be obtained. Cymbal effects require high Pitch and Q with slightly longer Decay.

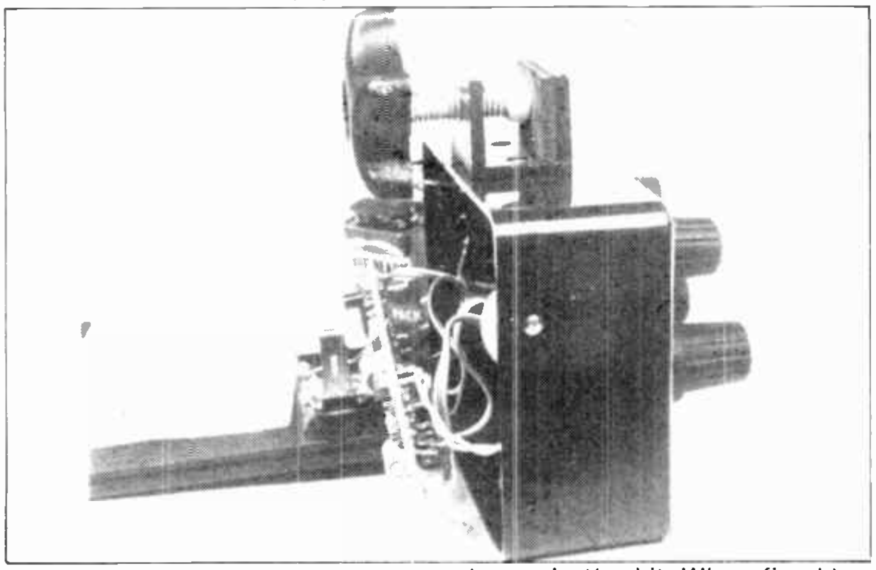
In addition, the Q control can put the filter into oscillation and Pitch will then vary the frequency. Of course, this effect may not be desirable, especially as the volume increases substantially, and is simply removed by reducing the Q control or inserting a resistor as mentioned earlier.

A little experimentation with the controls will soon give an idea of the wide range of useful effects that can be produced.



Synwave external view with bracket.

SYNTOM continued from page 25.



meters and prevent rattling, which could cause unwanted triggering of the unit.

Testing & Use

Connect the drum synthesiser to an external amplifier, and with all controls at midway position, firmly tap the case. A medium duration falling pitch effect should be heard, and experimentation with the controls will soon reveal the whole range of sounds available. The sensitivity of the unit has been fixed to respond to a direct hit or a hit on the drum to which it is fixed but not to external sounds and vibrations, including those from other drums in the kit. When fixed to a drum, the Syntom can be set off by just hitting the drum rim with the stick, or caused to sound along with the drum if the skin is hit. Since the sound varies with stick impact, particularly interesting effects can be produced by, for example, using a sharply falling pitch with an envelope of similar length to the natural drum sound, and playing single hits and rolls of differing impact force on the drum skin.

Since the drum synthesiser is battery powered, it should be turned off when not in use to conserve power, though a single PP3 will still provide for up to 60 hours of continuous playing.

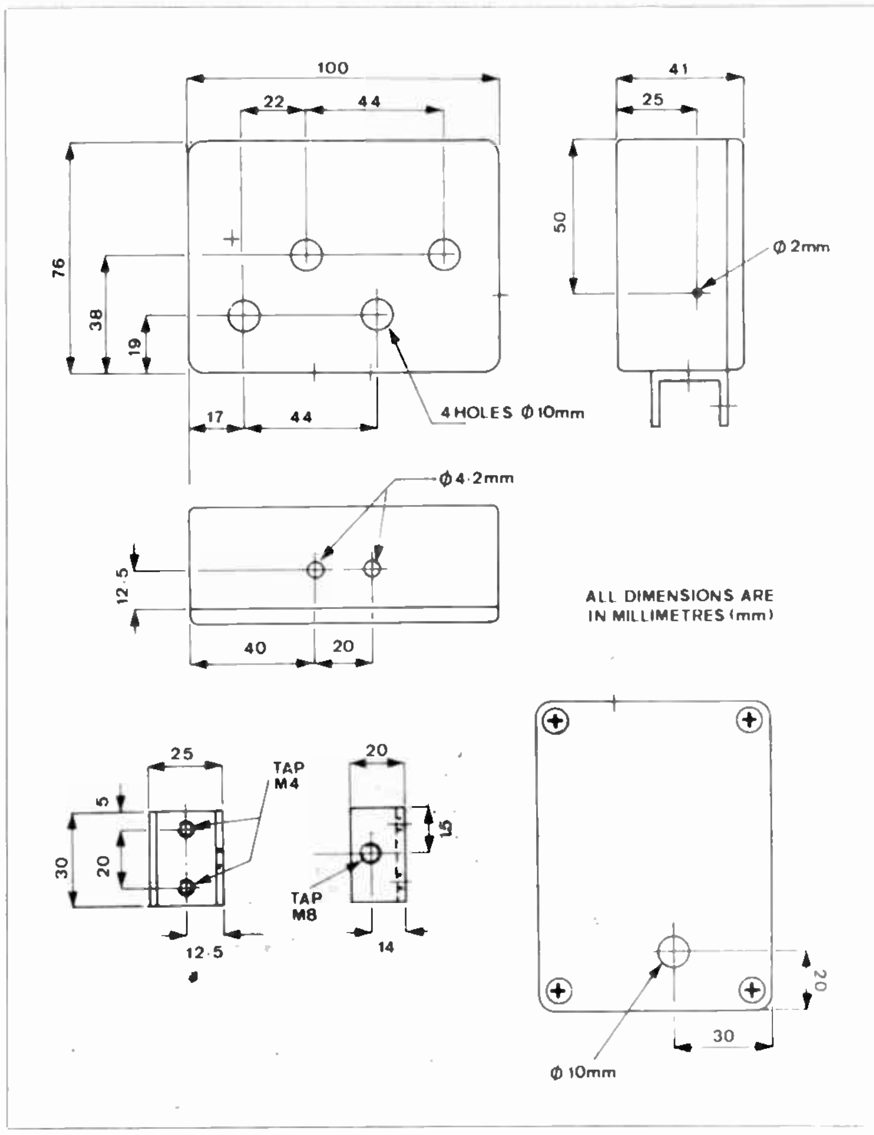


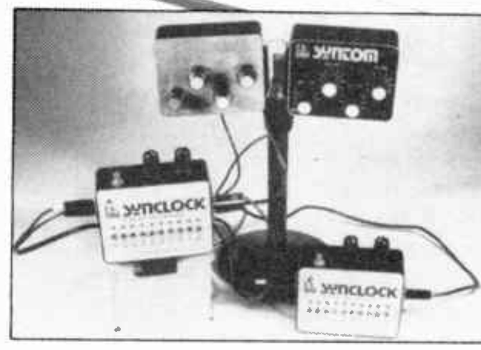
Figure 4. Case and bracket construction.

SYNCLOCK

by Glenn Rogers

KIT COST ONLY £19.75

- ★ **Rhythmic triggering for sequencers and synthesisers**
- ★ **Complement E&MM's Syntom and Synwave with this universal unit**
- ★ **Many rhythmic combinations possible with different pulse lengths**
- ★ **Any number of Synclocks can be connected together in series or parallel**
- ★ **Powered from a single 9V battery**



Units for generating drum rhythms have been around for many years. They first started to become popular when home electronic organs incorporated them in their accompaniment sections. The early rhythm unit produced typical dance rhythms such as Rock, Bossa Nova, Swing and Waltz, with sounds that could not really be called typical of a drum kit. Over the years the sounds have been improved considerably as knowledge of synthesis has widened and now the rhythms themselves have come into a period of change. The musician is no longer satisfied with a choice of preset rhythms, he wants to plan out his own rhythms to try and break the monotony of the drum machine. Enter the programmable rhythm units; now it is possible to create your own drum rhythms and set up your own drum sounds.

There are many different analogue and digital sequencers now available which are used to set up a pattern of electronic information for controlling one or more parameters on a synthesiser. Because they generally supply and control voltages as well as a trigger output, the sequencer is more obviously applied to synthesiser pitch and EG control to produce repeated note patterns, bass lines and special rhythmic effects. The sequencer can control a synthesiser without the use of a keyboard and thus enables complex patterns to be sounding whilst the performer plays his instruments.

Many well known soloists and groups have based their compositions on the use of sequencers, including Tangerine Dream, Kraftwerk, Klaus Schulze, Jean Michel-Jarre and Logic System. Even though sophisticated micro-memory sequencers, such as the Roland CSQ-600 computer controlled digital sequencer can be regarded as state-of-the-art systems, often the most convenient sequencer is simply one that can count just a few bars of the chosen time signature to provide live performance control of each pulse within the bar.

So using the Synclock as the music proceeds, the player manipulates the switches and makes subtle changes to the rhythmic control of percussive and melodic sounds. No bar need sound the same and because speed can be set over a very wide range, the sequence can be running faster than it's physically possible to play.

Since the Synclock PCB board with its components measures only a few inches and is very cheap to construct, it should enable any electro-musician to experiment with this fascinating and important part of music-making.

The E&MM Synclock is a compact and uniquely expandable control sequencer which can be used to trigger the Syntom and Synwave, as well as most synthesisers and other sound generators, to give new and exciting rhythms and sounds. When used with the Syntom or Synwave, the internal triggers of these devices still operate allowing an even larger scope for filling

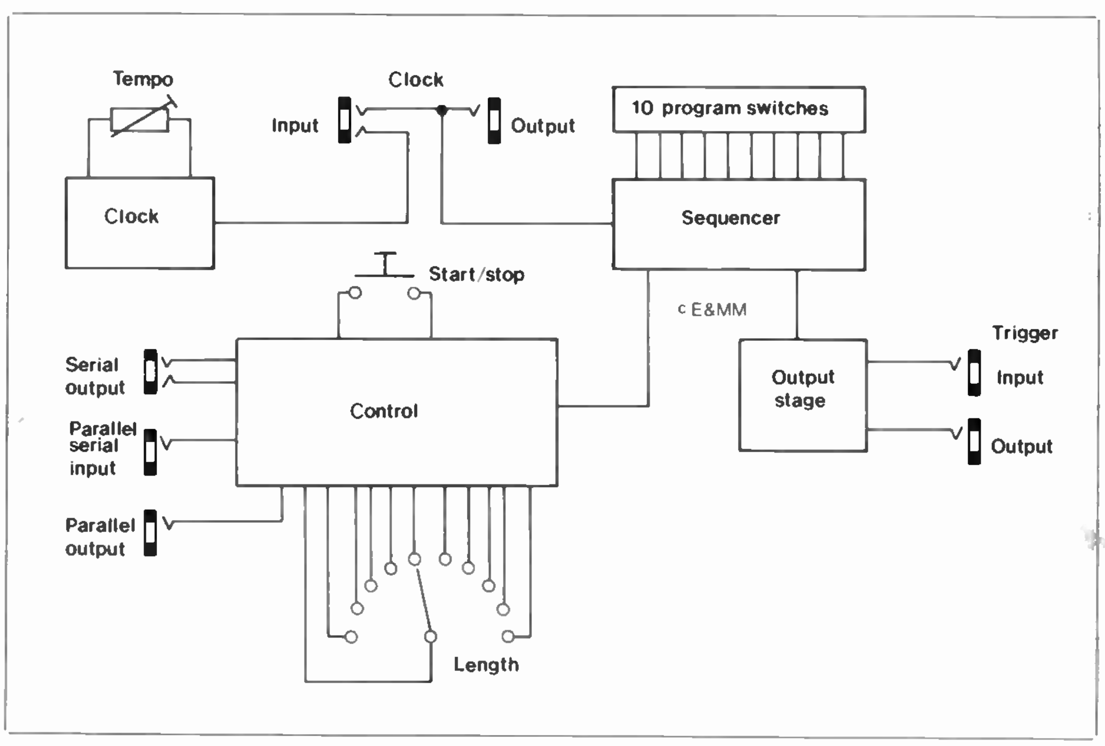
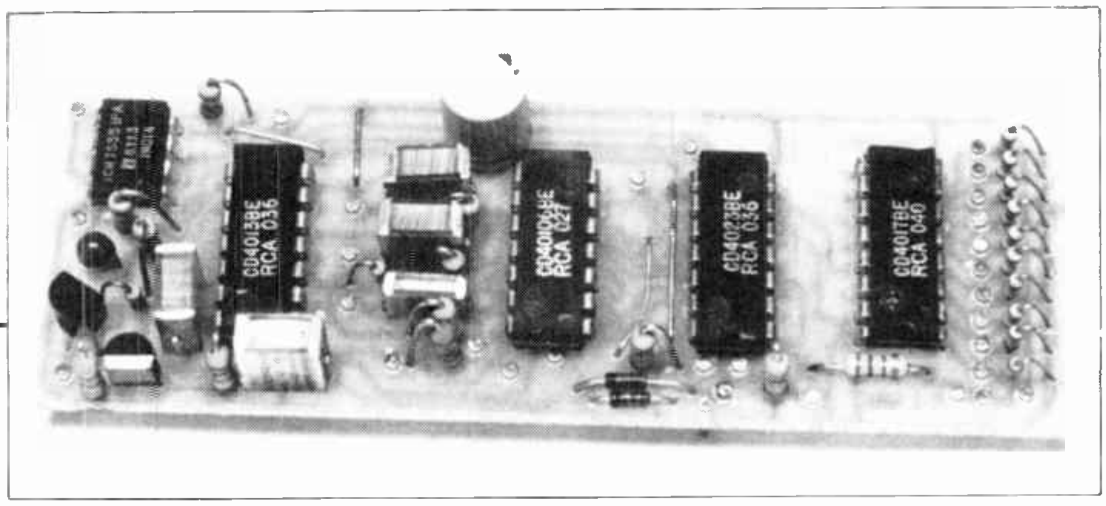


Figure 1. Block diagram of the Synclock.

in further rhythms by hitting the box.

The Synclock has a sequence length variable between 1 and 10 beats and this is of course expandable with further Synclocks. This system allows any number of beats in any time signature to be programmed. There are only a minimum number of controls for ease of use and setting up. The three controls on the top of the unit are: Stop/Start, Sequence Length, and On-Off/Tempo with the programming switches and indicators on the front panel. The sockets for interfacing and control are mounted on the side of the box and finally the clamp is mounted on the bottom if required. All the components except the LEDs and controls are mounted on a PCB and everything fits into the same size box as the Syntom and Synwave.

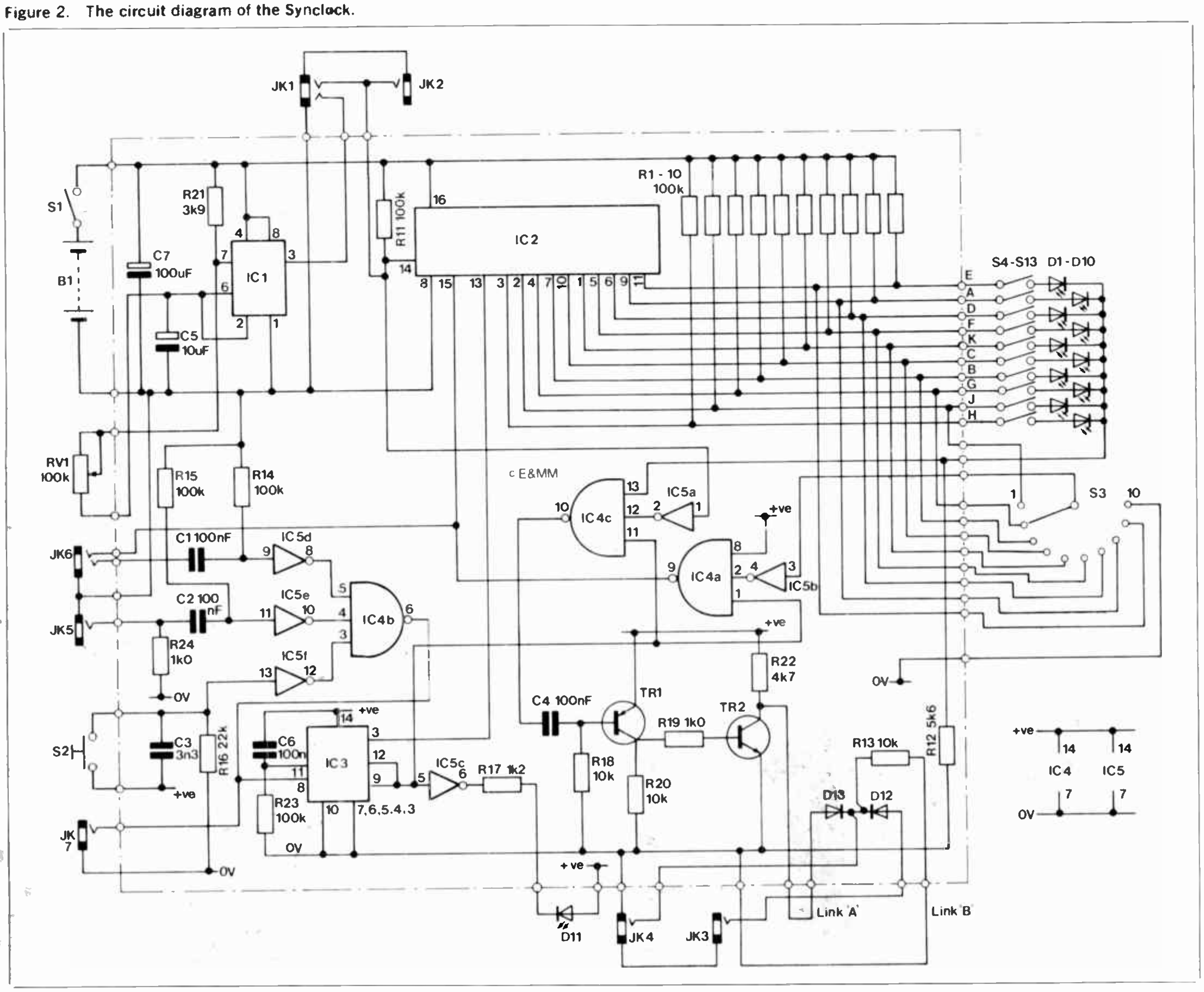


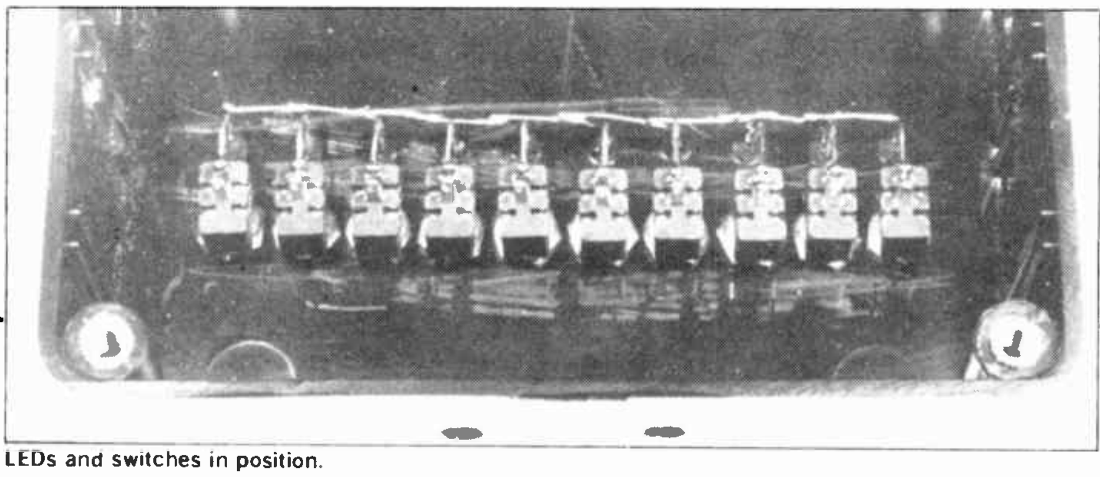
Completed circuit board.

The Circuitry

The block diagram of the Synclock is shown in Figure 1. As can be seen, there are four blocks: the Clock, Sequencer, Control and Output stage. The Synclock design is based around the popular 4017 (Decade counter divider with ten decoded outputs) and 555 (Timer). The CMOS version of the 555 has been used to keep current consumption as low as possible. The circuit diagram of the Synclock is shown in Figure 2. The timer (IC1) is used in its astable mode to generate the clock signal for driving the 4017 (IC2) the rate being controlled by RV1. The 'clock in' and 'clock out' jack sockets have been included to enable another unit to drive or be driven by the Synclock. The clock signal is also used to gate the output through IC4c. The outputs of IC2 are pulled high by resistors R1-R10 and fed to the ten programming switches (S4-S13) and the sequence length switch (S3), the other side of the programming switches being connected to the anodes of the ten miniature LEDs. The cathodes of the LEDs are all commoned together

Figure 2. The circuit diagram of the Synclock.



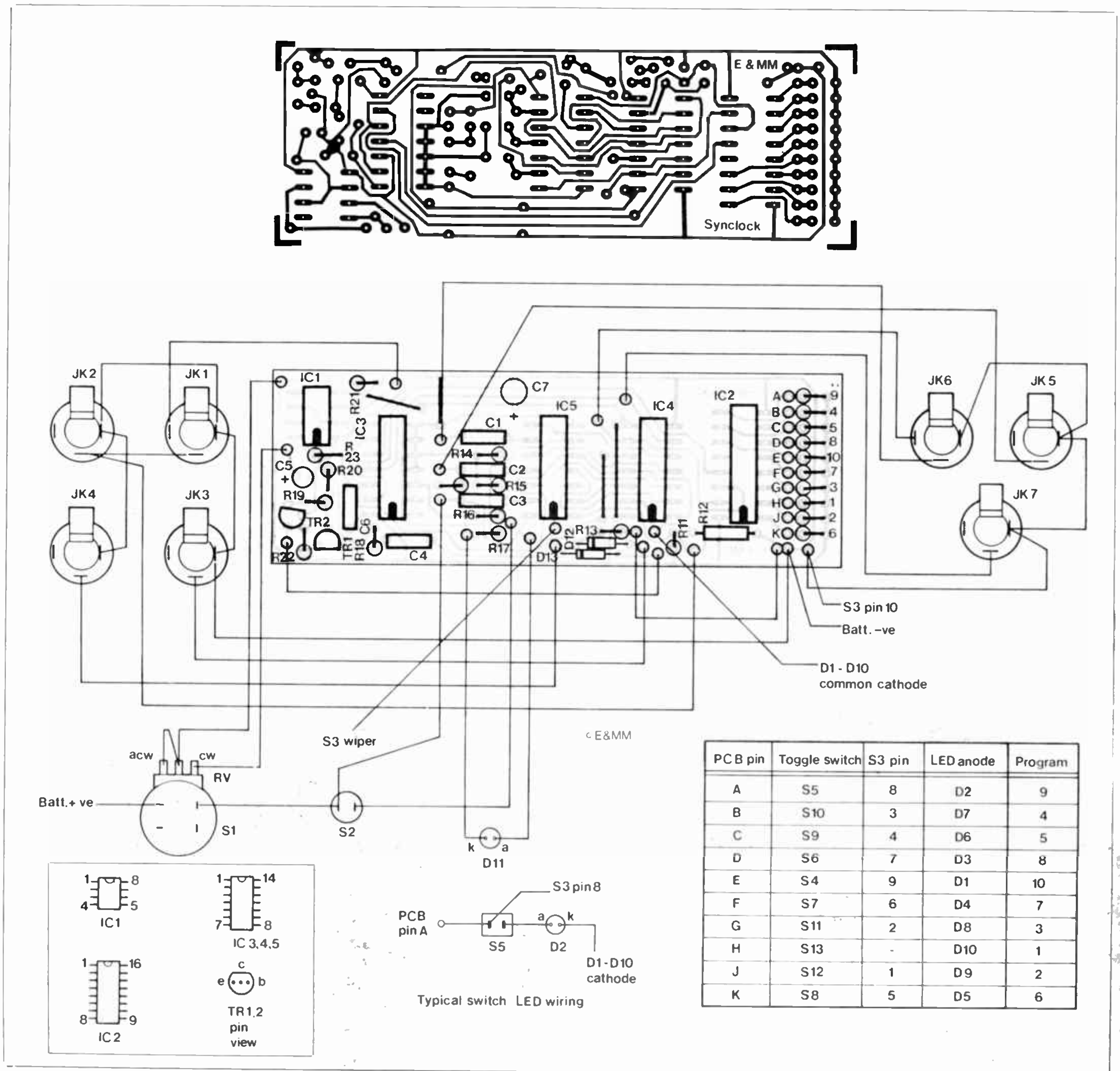


LEDs and switches in position.

and taken to ground via R12 to form a ten input diode 'OR' gate. The intensity of the LEDs can be adjusted if required by changing the value of R12, the lower the value the brighter the LEDs, but remember that battery life will be shortened. The output of the 'OR' gate is then fed to IC4c. The other input to IC4c (pin 11) comes from the control circuitry to ensure no output occurs when the sequencer is stopped. IC4c then feeds into the output stage consisting of TR1 and TR2, and then to another diode 'OR' gate to combine the signal with any other trigger information being used.

The most complex part of the Synclock is the control circuitry. The complexity arises due to the need for expansion of the system both serially and in parallel. ICs 3, 4 and 5 are all used in this section and the heart of the circuit is IC3a, a D type flip flop which is connected so that each clock pulse on pin 11 causes the outputs (pins 12 and 13) to change. R23 and C6 are used to provide a power-up reset to ensure the unit is not running when you switch on. The stop/start switch is debounced and fed into IC5f, a Schmitt inverting buffer, and then to IC4b. The control input is differentiated, then inverted by

Figure 3. PCB track, component layout and wiring details.

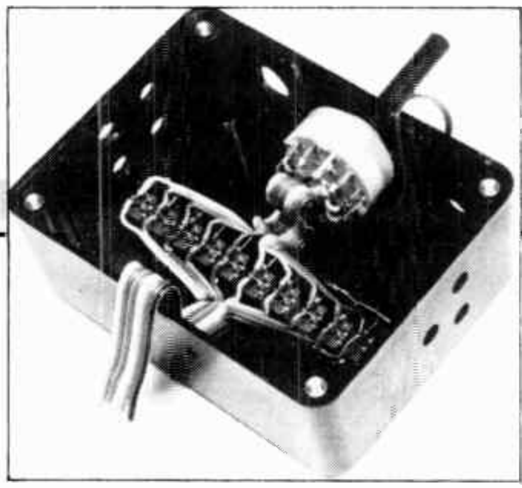


PCB pin	Toggle switch	S3 pin	LED anode	Program
A	S5	8	D2	9
B	S10	3	D7	4
C	S9	4	D6	5
D	S6	7	D3	8
E	S4	9	D1	10
F	S7	6	D4	7
G	S11	2	D8	3
H	S13	-	D10	1
J	S12	1	D9	2
K	S8	5	D5	6

IC5e and fed into IC4b. The other input to IC4b comes from the serial output via the jack socket switch through a differentiator and an inverter (IC5d). The stop/start LED is driven by IC5a from the flip flop $\overline{Q}$ output. S3, IC5b and IC4a are used to control the sequence length by resetting IC2 at the appropriate point.

Construction

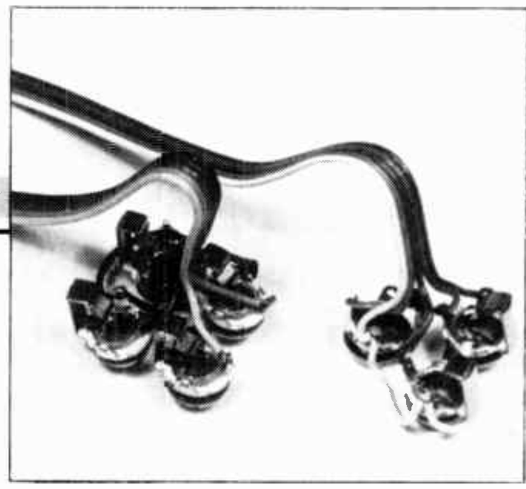
The Synclock is housed in a plastic box (Maplin type MB2) which is the same as that used for the Syntom and Synwave, and a reasonable amount of care is required to fit all the components into the box. If you have only a limited experience in the construction of compact projects you will probably find it easier to use a larger box. Construction can begin with the assembly of the PCB (see Figure 3). The veropins should be fitted first and these are pushed in from the component side of the board. The four wire links can then be soldered in place.



LEDs, switches and S3 wired up.

Next fit the resistors and capacitors in place, remembering to get the micro resistors in the correct positions and the polarities of C5 and C7 correct. The diodes and transistors can then be soldered in place on the PCB. The PCB assembly can then be completed with the insertion of the ICs. The prototype unit was not fitted with IC sockets as the space on the PCB is very limited, but sockets can be used if there is sufficient room in the case used.

All the ICs are CMOS and require a certain amount of care when being handled - use a low leakage soldering iron and avoid physical contact with the IC pins if possible. The ICs should be left in their protective packing until you are ready to insert them.

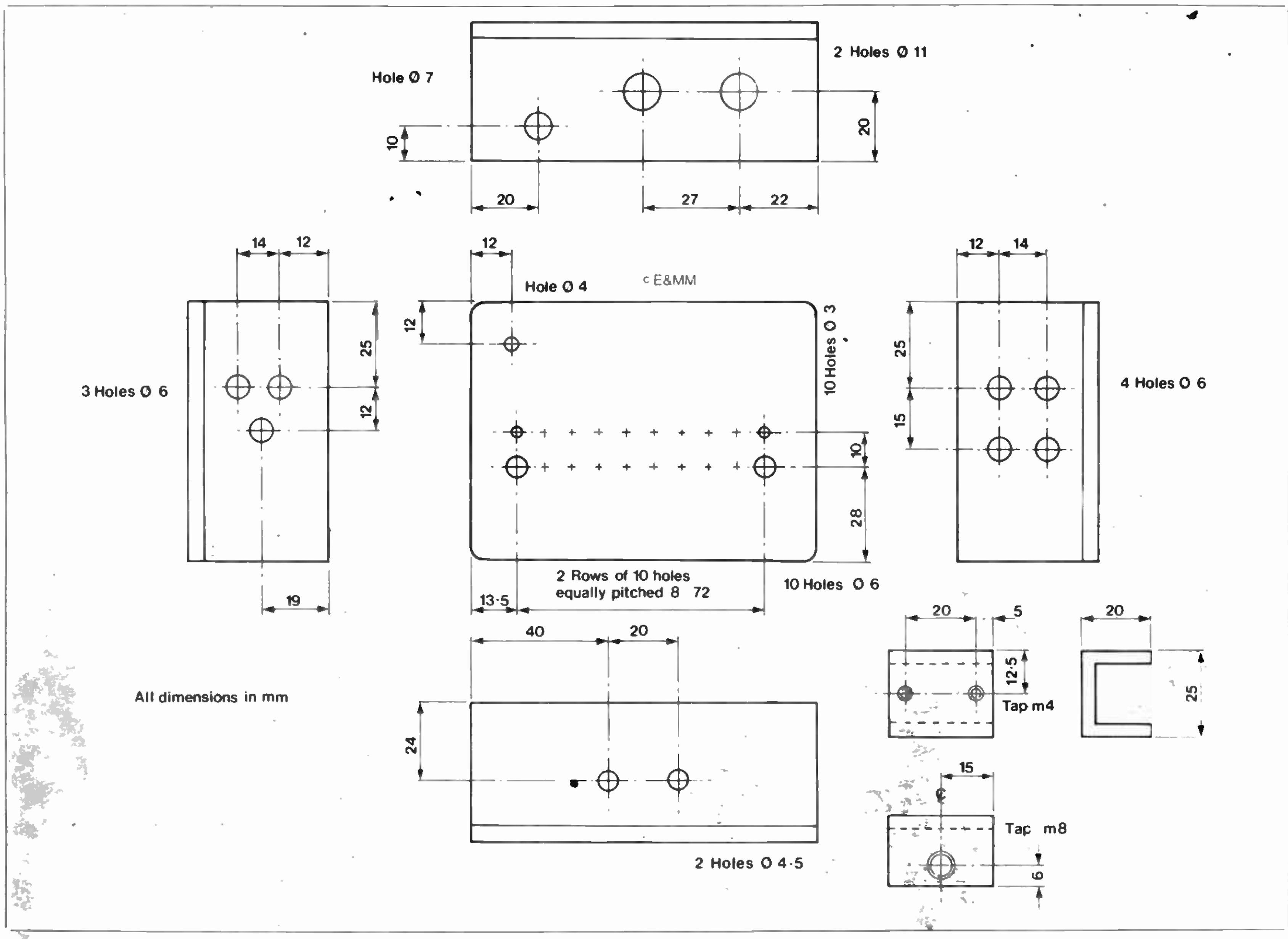


Jack socket wiring before fitting into case.

The board is now complete and can be put to one side while the box is assembled. Figure 4 shows details of the box and the drilling should be as accurate as possible to ensure all the components will fit into the box. When the drilling has been completed you will need to cut away some of the ribs of the box to ensure that all the components will seat properly - this can be done with a sharp knife.

The ten miniature LEDs can then be pushed into place with the anodes towards the switches (see Photograph) and should not be glued in place until the unit has been tested. Next fit the ten ultra-min. toggle switches in place (due

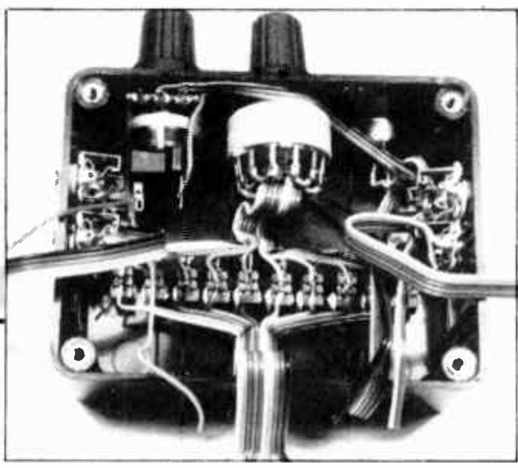
Figure 4. Case and bracket construction.



to the closeness of the switches the washers supplied with the switches cannot be used). Now connect the anodes of LEDs 1 to 10 to the nearest contact of switches 1 to 10 respectively and connect the cathodes of the LEDs together. Cut a 4 inch and a 5 inch length of 10 way ribbon cable and then cut and wire one end of each wire to the programming switches (Photo 3). Make use of the wire colour coding, with programming switch 1 connected to the black wire and switch 10 to the white wire. Next take the other end of the 4 inch length of ribbon cable, cut the black wire off an inch from the end and then strip and tin the ends of the remaining 9 wires. Then solder wire 1 (brown) to S3 pin 1, wire 2 (red) to S3 pin 2 etc. Cut a 5 inch length of two way ribbon cable, strip and tin one end of each wire and connect one wire to S3 pin 10 and the other to the wiper of the switch. Next fit and wire up S2, D11, RV1 and S1 with suitable lengths of ribbon cable leaving the end that connects to the PCB floating for the moment.

The jack sockets should now be wired up - this has to be done outside the box. The first thing to do is to change JK6 from a 'break' contact type to a 'make'-contact type. This can quite easily be achieved by bending the outer contact to the other side of the spring contact with a pair of pliers (see Figure 5). Cut one 5 and one 6 inch length of 5 way ribbon cable and use the 6 inch length to wire JK1, 2, 3 and 4 as shown in the wiring diagram and the other piece to wire JK5, 6 and 7 (Photo 4). The two groups of sockets can then be mounted in the box and the positive of the battery connector soldered to S1.

All the components in the box should now be connected up (Photo 5) and the connections to the board can be made. If you wish to fit a clamp to the unit (for a drum rim) this should be fitted prior to connecting the PCB. There are two wire links (A and B) to be made on the back of the PCB first, then



Completed off board component wiring.

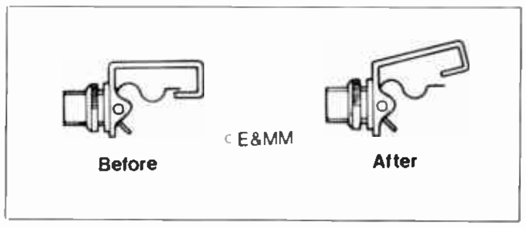


Figure 5. Jack socket modification.

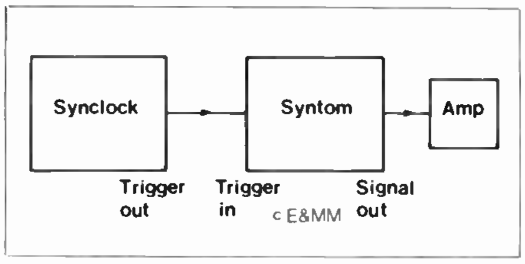


Figure 6. The Synclock driving a Syntom.

the board can be wired as indicated in the wiring diagram (Figure 3). The PCB can then be slotted into the box and the battery connected (Photo 6).

Testing

Set the sequence length to 10 and the ten programming switches down, then switch the unit on. LED number 1 should be on and all the others should be off. If all is correct then press the

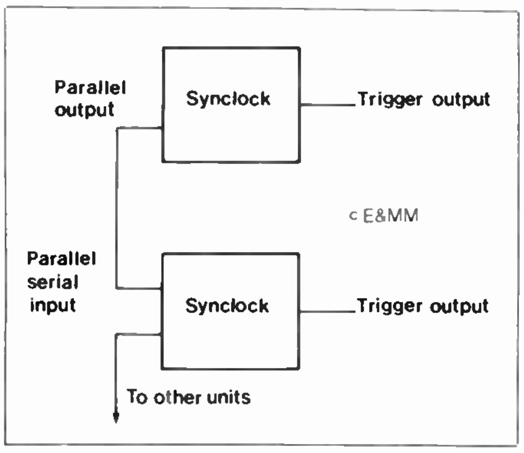


Figure 7. Parallel connection of Synclocks.

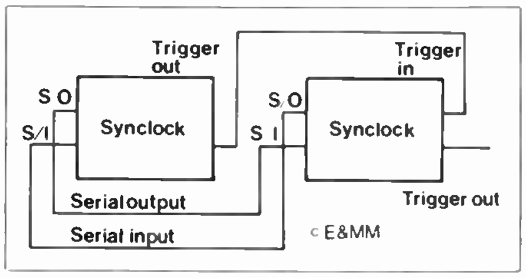


Figure 8. Serial connection of Synclocks.

stop/start button. The start LED should now be lit and the program LEDs should light sequentially. Next, check the sequence length switch operates, the tempo control alters the clocking rate and that the stop/start button will also stop the sequencer and reset the unit so that LED number 1 is on.

If the Synclock has functioned correctly up to now, you can connect the trigger to your Syntom or Synwave trigger input (see Figure 6). Start the Synclock and the Syntom or Synwave should now trigger once on every step. Check all the programming switches are operating by changing the rhythm pattern. The other input and output sockets are best tested with another Synclock unit.

Using your Synclock

It is very easy to learn the best ways of controlling the Synclock and together with the Syntom and/or Synwave the variations on rhythms and sounds give you endless possibilities. There are only a few points to note when using one Synclock: the clock input required is a 9V square wave, the clock output is a 9V square wave the frequency of which is set by the tempo control and the trigger output is an 8V pulse.

The potential of the system really comes to light when two or more Synclocks are available. The wide combination of interconnections and control settings give unlimited scope. Two of the many possibilities are shown in Figures 7 and 8. The parallel connection allows one Synclock to start and stop all the other Synclocks simultaneously and each Synclock can be used to trigger a different sound with its programmed rhythm. The units can all

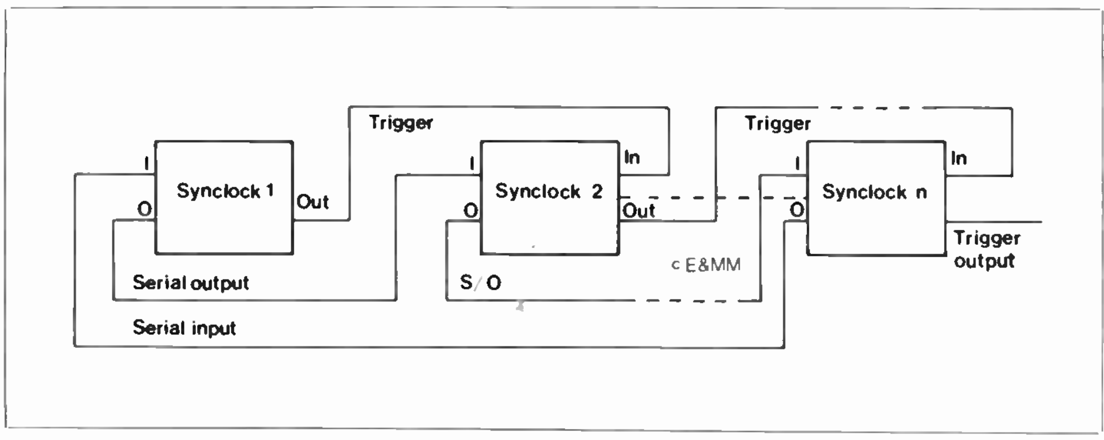
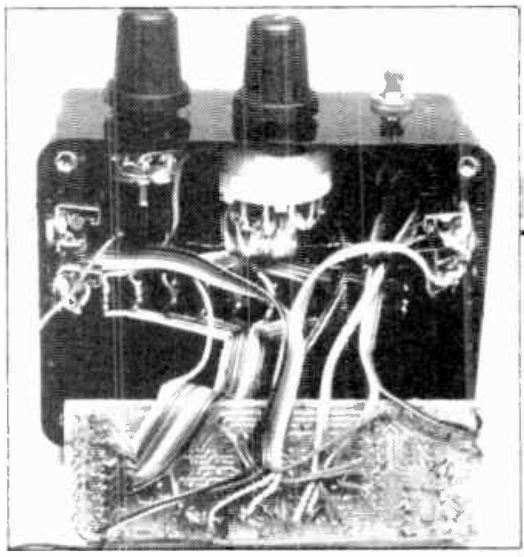


Figure 9. Synclocks connected for long serial sequence.



Synclock with wiring completed and PCB inserted in case

Edgar Froese of Tangerine Dream looks with interest at the Synclock.

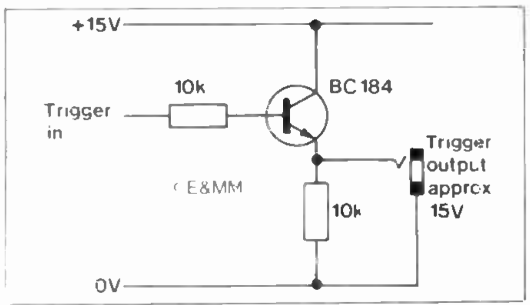


Figure 10. Output stage modification.

be run from one clock control using the clock in and out jack sockets or they can be controlled with their own tempo pots.

Figure 8 shows how to connect two Synclocks so that a sequence of up to 18 beats can be used. Note the maximum of 18 (and not 20) is due to the fact that the tenth beats have a different reset mechanism. The first Synclock can be used to control the system and the clock signals can again be commoned if required. If a long sequence (more than 10) is being used for one sound generator then the trigger signals need to be combined and this is simply a matter of connecting the output of one to the input of the next. In the serial mode the Stop/Start button will only stop the sequence if it is operated on the unit in which the sequence is running.

Figure 9 shows how a long serial chain may be set up and the legend for the front panel is shown full size in Figure 10 to enable you to give a professional finish to your project.

Extending the use of the Synclock

The ratio of R22 to R13 determines the amplitude of the output pulse. The amplitude can be increased by reducing the value of R22 and increasing R13 - remember, however, that the effective resistance of R13 is the parallel combination of R13 and the input stage being driven.

The Synclock can be used to drive commercial equipment but a few modifications may be needed. The industry standard for trigger signals is +15V and for this the Synclock really needs an additional output stage. This can consist of a single transistor buffer as shown in Figure 11. The +15V supply is needed as it is impossible to obtain a +15V trigger pulse from a 9V supply.

The length of the trigger pulse in the Synclock is set to approximately 1ms by R18 and C4. The pulse length can be increased by increasing the value of R18 (or C4). Some interesting effects at higher clock rates have been found when using the Synclock with the Syntom, by lengthening the pulse to about 5ms - due to the Syntom trigger circuitry the pitch will vary automatically!

PARTS LIST FOR SYNCLOCK

Resistors — all 1% 0.4W metal film unless specified			
R1-R10	100k ⅛W	10 off	(U100K)
R11,13,15,16,23	100k	5 off	(M100K)
R12	5k6		(M5K6)
R18,20	10k	2 off	(M10K)
R14	22k		(M22K)
R17	1k2		(M1K2)
R19,24	1k0	2 off	(M1K0)
R21	3k9		(M3K9)
R22	4k7		(M4K7)
RV1	100k lin pot with switch		(FW45Y)
Capacitors			
C1	3n3 carbonate		(WW25C)
C2,3,4,6	100n carbonate	4 off	(WW41U)
C5	10u 16V tantalum		(WW68Y)
C7	100u 10V PC electrolytic		(FF10L)
Semiconductors			
IC1	7555		(YH63T)
IC2	4017		(QX09K)
IC3	4013		(QX07H)
IC4	4023		(QX12N)
IC5	40106		(QW64U)
TR1	BC214		(QB62S)
TR2	BC184		(QB57M)
D1-D10	LED min red	10 off	(WL32K)
D11	LED red		(WL27E)
D12,13	1N4148	2 off	(QL80B)
Miscellaneous			
S1	see RV1		
S2	Push button switch HQ		(YR67X)
S3	Rotary switch 1 pole 12 way		(FF73Q)
S4-S13	SPST ultra min toggle	10 off	(FH97F)
JK1-JK7	3.5mm open jack socket	7 off	(HF82D)
	Case plastic box type MB2		(LH21X)
	PCB		(GA54J)
	Battery clip PP3		(HF28F)
	LC collet knob	2 off	(YG40T)
	Collet knob cap black	2 off	(QY00A)
	10 way ribbon cable	1m	(XR06G)
	Front panel		(XX44X)

All the parts for the Synclock are available in a kit from Maplin Electronic Supplies Ltd, PO Box 3, Rayleigh, Essex SS6 8LR, order number LW55K, price £19.75. Also available separately are the lettered, brushed aluminium stick-on front panel: order number XX44X price £1.50; and the PCB: order number GA54J, price £1.65. Prices are inclusive of VAT & p&p.

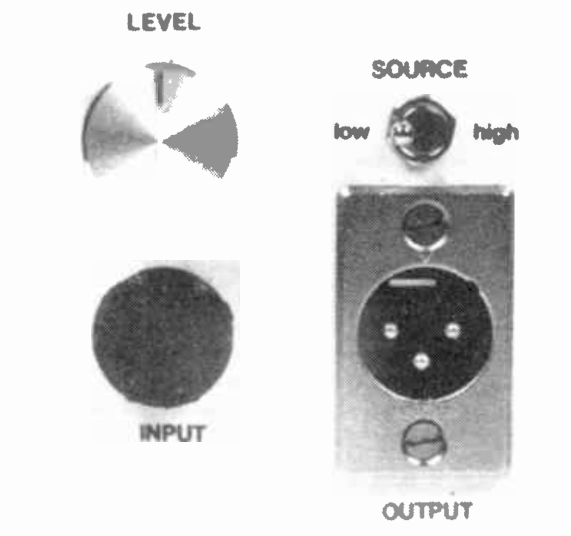
DIRECT INJECT BOX

DI BOX

E&MM

by Chris Lare

The Direct Inject Box (D.I. Box) allows the signal from an amplified instrument to be fed directly into a balanced line mixing desk, and as such is invaluable on stage, and in the home or professional recording studio avoiding many of the disadvantages of using a microphone. It is much cheaper to build the D.I. Box than to buy a good microphone, and it eliminates acoustic feedback and 'spill-over' of other sounds into the instrument channel.



The Circuit

The D.I. Box takes its input from the instrument amplifier and converts it to a balanced line output at microphone level. Figure 1 shows the circuit employed. The input signal is fed via a switchable attenuator to the 47k potentiometer. This means that the box has an input impedance of about 47k when used as a low level (line) input and over 700k in the speaker level mode. As shown the input is dc coupled and if a dc offset appears on the input the potentiometer will be noisy as it is moved. If this occurs a 470nF polyster capacitor should be connected in series with the input.

Two J-FET op-amps, chosen mainly for their very low power consumption, form the phase and antiphase generator. The first op-amp inverts the signal and divides its level by 4, whereas the second op-amp merely re-inverts the output from the first. The two outputs are thus of the same level but exactly out of phase and can be used directly. A 100R resistor is included in each output as a protection against short circuits, and a capacitor is obviously required to block the dc level. The op-amps are biased to half rail by R9 and R10 which hold the non inverting inputs at 4.5 volts and R3 which provides a dc offset for the input signal. Diodes D1 and D2 protect the op-amp in the event of severe overload, and play no part in the normal operation of the circuit.

A single 9 volt battery is used to power the circuit. This is switched in the usual way by using a stereo jack socket on the input.

Construction

A printed circuit board holds all the resistors, capacitors and semiconductors except R1 and RV1. Mount and solder the components and Veropins on the PCB, with IC1 left to last. Bolt the PCB to the lid of the box and the connectors, pot, and switch to the base. If a box other than the one recommended is used, check that it is deep enough to take the chassis Cannon plug. Solder R1 and the battery connector in position, and wire up the connections to the PCB using screened cable.

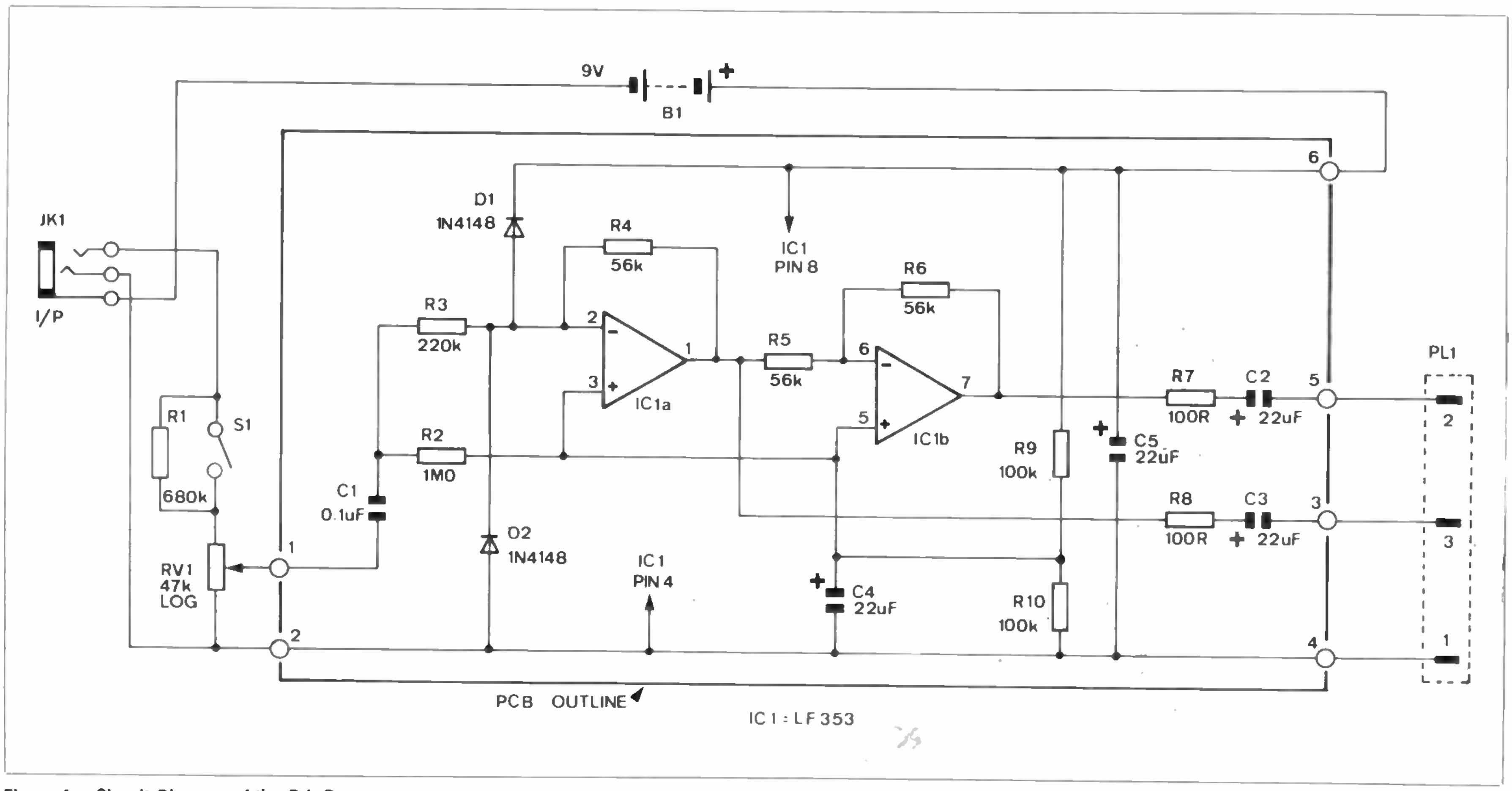


Figure 1. Circuit Diagram of the D.I. Box.

The prototype used a small piece of polystyrene foam glued to the lid above the battery position to hold the battery in place. Finishing consisted of lettering and varnishing the front, and sticking four rubber feet on the bottom.

Operation

Use a jack-to-jack to connect the D.I. Box to the extension speaker socket on the amplifier or speaker cabinet, or the amplifier slave out jack, remembering to set the high/low switch accordingly. If an additional speaker output jack is not available, a jack-to-two-jack splitter lead can be used to connect the D.I. Box input in parallel with the speaker cabinet. RV1 should be adjusted for a convenient signal level to the desk.

Unlike direct injection of the instrument output or pre-amplified signal, the D.I. Box passes the full sound of the amplified instrument, including the effects of tone controls, signal processors, and amplifier distortion (the latter is often an important part of guitar sound) from the speaker outputs to the mixing desk for recording or amplification by the group P.A. Alternatively, the unit can be connected to the 'Slave Out' or 'Link' jack socket of the amplifier avoiding the distortion of the output stage. This is particularly useful for amplifiers which are also used as sub-mixers e.g. with keyboards, since output stage distortion is especially noticeable on a mix of different signals.

It is important to note that the D.I. Box design that follows is intended for mixers with balanced line inputs. If the mixer in question does not have such inputs it is debatable if the D.I. Box is worth using; a simple wire connection being the easiest. If hum problems do occur, or particularly long connections are required, better results will be obtained with the E&MM Line Driver/Receiver presented in last month's issue. The Balanced Line System, although designed for microphone use, will handle signal levels of up to 400mV without any trouble.

Obviously some instruments cannot be D.I.'ed — the most notable example being an organ with a Leslie cabinet. The D.I. is also a matter for personal opinion, indeed many claim that the sound produced is too dry. Additionally, problems will occur if tonal adjustments are made before the D.I. connection to compensate for a poor speaker cabinet. It is, however, a much under-rated technique, offering several advantages — cheaply.

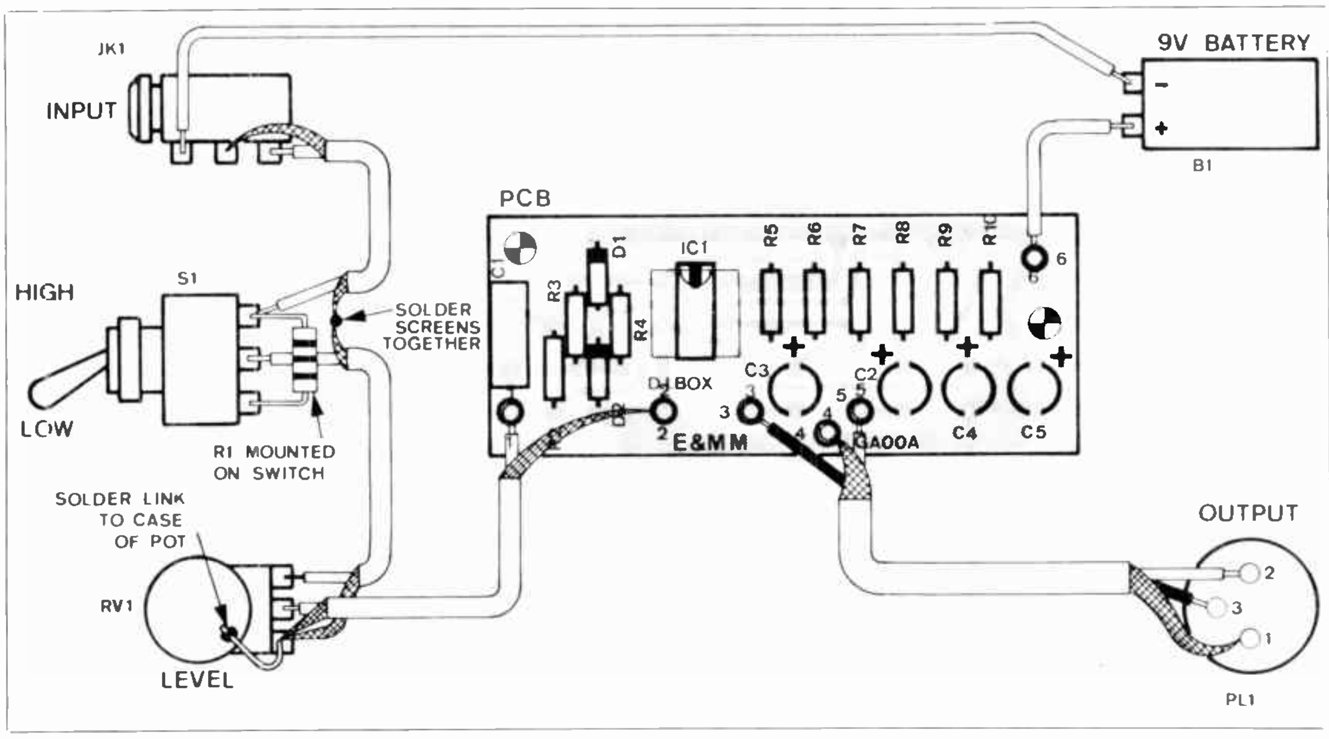


Figure 2. Internal Wiring of the D.I. Box.

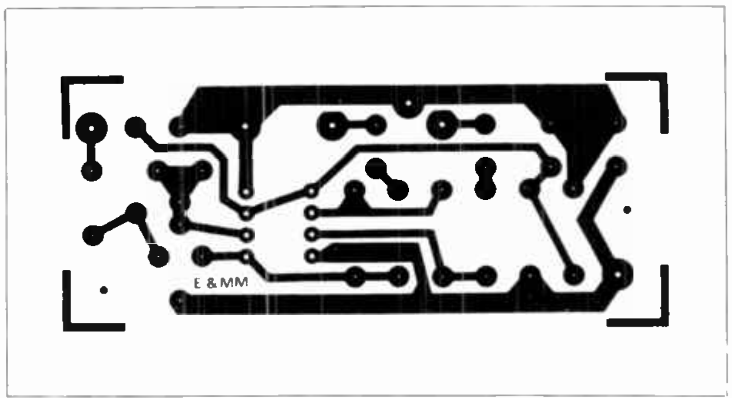


Figure 3. The D.I. Box PCB.

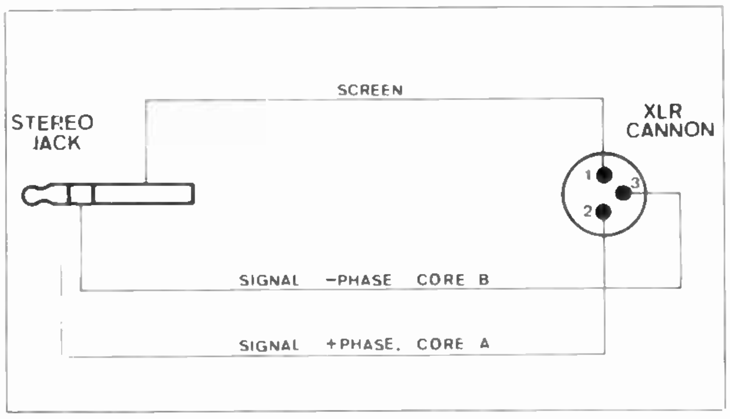
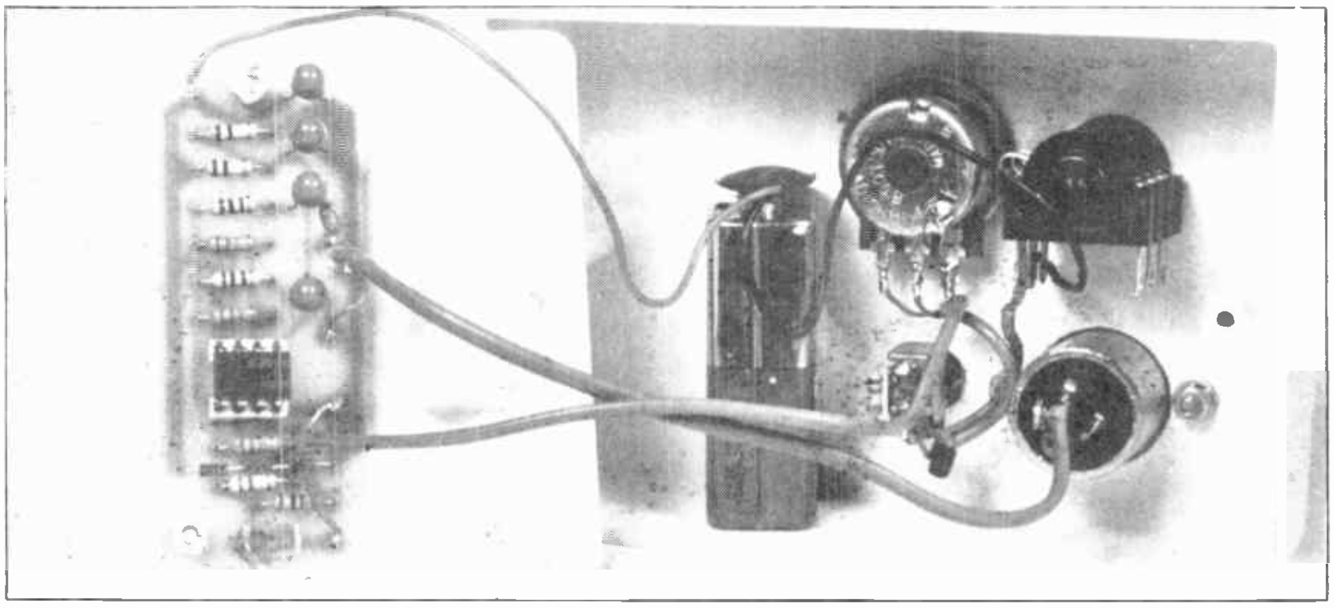


Figure 4. XLR Cannon-to-jack lead connections.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified

R1	680k		(M680K)
R2	1M0		(M1M0)
R3	220k		(M220K)
R4,5,6	56k	3 off	(M56K)
R7,8	100R	2 off	(M100R)
R9,10	100k	2 off	(M100K)
RV1	47k log. pot.		(FW24B)
Capacitors			
C1	100n carbonate		(WW41U)
C2,3,4,5	22u 16V tantalum	4 off	(WW72P)
Semiconductors			
IC1	LF353		(WQ31J)
D1,2	1N4148	2 off	(QL80B)
Miscellaneous			
PL1	XLR chassis plug, 3-pin		(BW92A)
S1	Sub-min toggle 'A'		(FH00A)
JK1	Jack socket, stereo		(HF92A)
B1	PP3 battery		
	Printed circuit board		(GA00A)
	Case PB1		(LF01B)
	(or alternative)		
	Veropins, 1mm		(FL24B)
	PP3 connector		(HF28F)
	Knob R52		(HB29G)
	(or alternative)		
	Twin screened cable	1m	(XR21X)

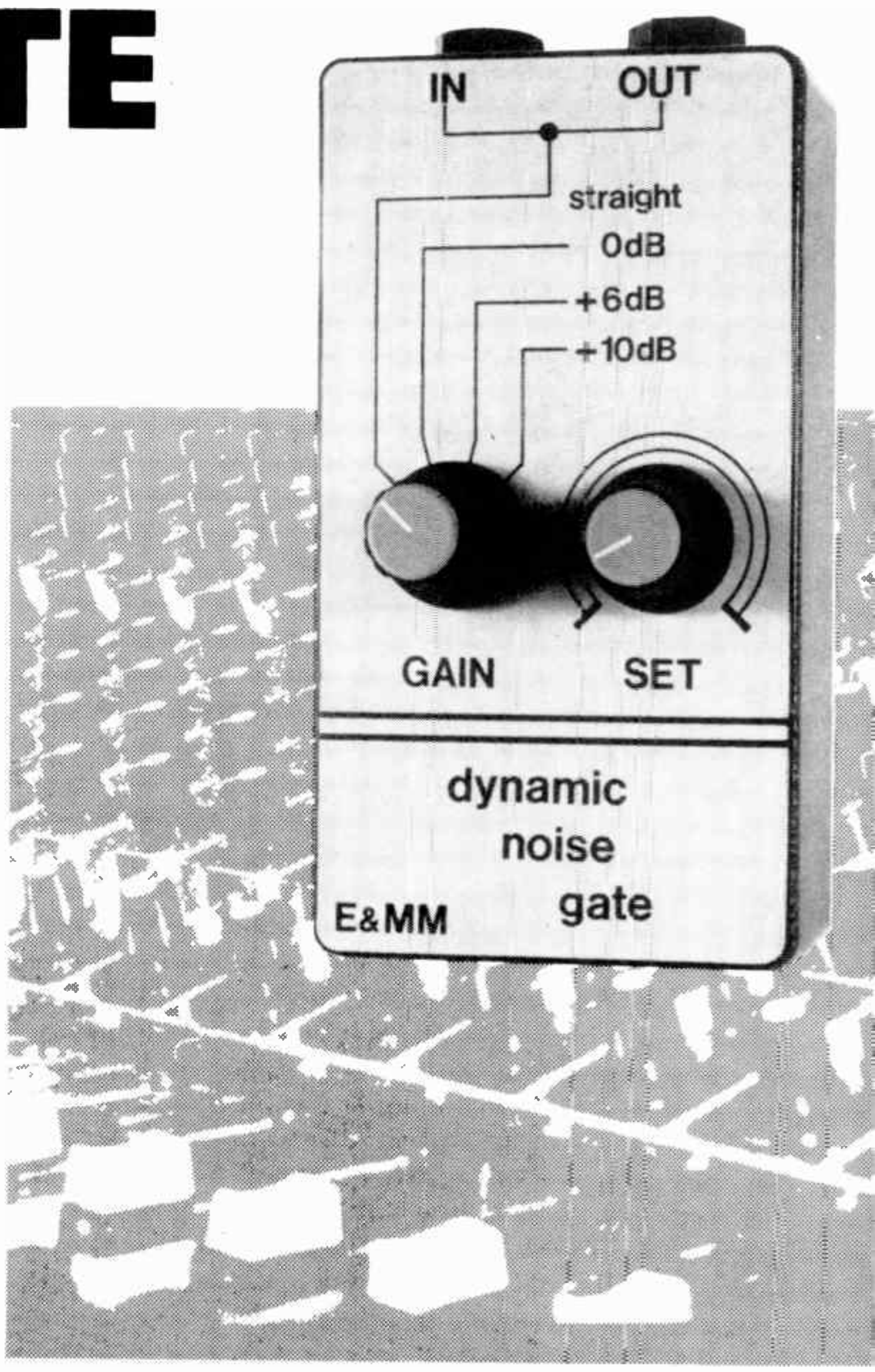


Internal view of the D.I. box.

NOISE GATE

by Dave Roffey

★ Provides automatic shutdown of unwanted noise during 'pause' conditions
★ Compander technique eliminates 'signal snapping'
★ User adjustable characteristics for high or low level network insertion
★ Allows the use of otherwise 'too noisy to use' effects units
★ Can effectively cancel crosstalk in multi-microphone set-ups
★ Can be used in multi-instrument layouts for instant unit shutdown on changeovers
★ Will eliminate 'beehiving' in older type 'spaghetti' wired organs
★ No circuit trimming required or tight specification devices used
★ Can be used in its own right as an effect to create soft attack bowing characteristic
★ Uses only two low cost and readily available ICs
★ Self contained, jack-in-jack-out unit allows instant in-line connection



In these modern times of multi-device usage, noise or cross-talk can often become such a nuisance that desired sounds often have to be forfeited. Although this unit can do nothing for the possibly distracting noises emanating from audiences, producers etc., it can considerably reduce, or even eliminate, pause noise from devices in which it is otherwise impossible to improve on signal to noise ratios.

Several types of noise gate are available for this kind of noise elimination - 'snap-off' units - programmable types - externally controlled units - low level expansion devices etc. In order that the unit may have as wide a range of applications as possible, the low level expansion or dynamic gate has been selected. This type tends to be less critical in set-up

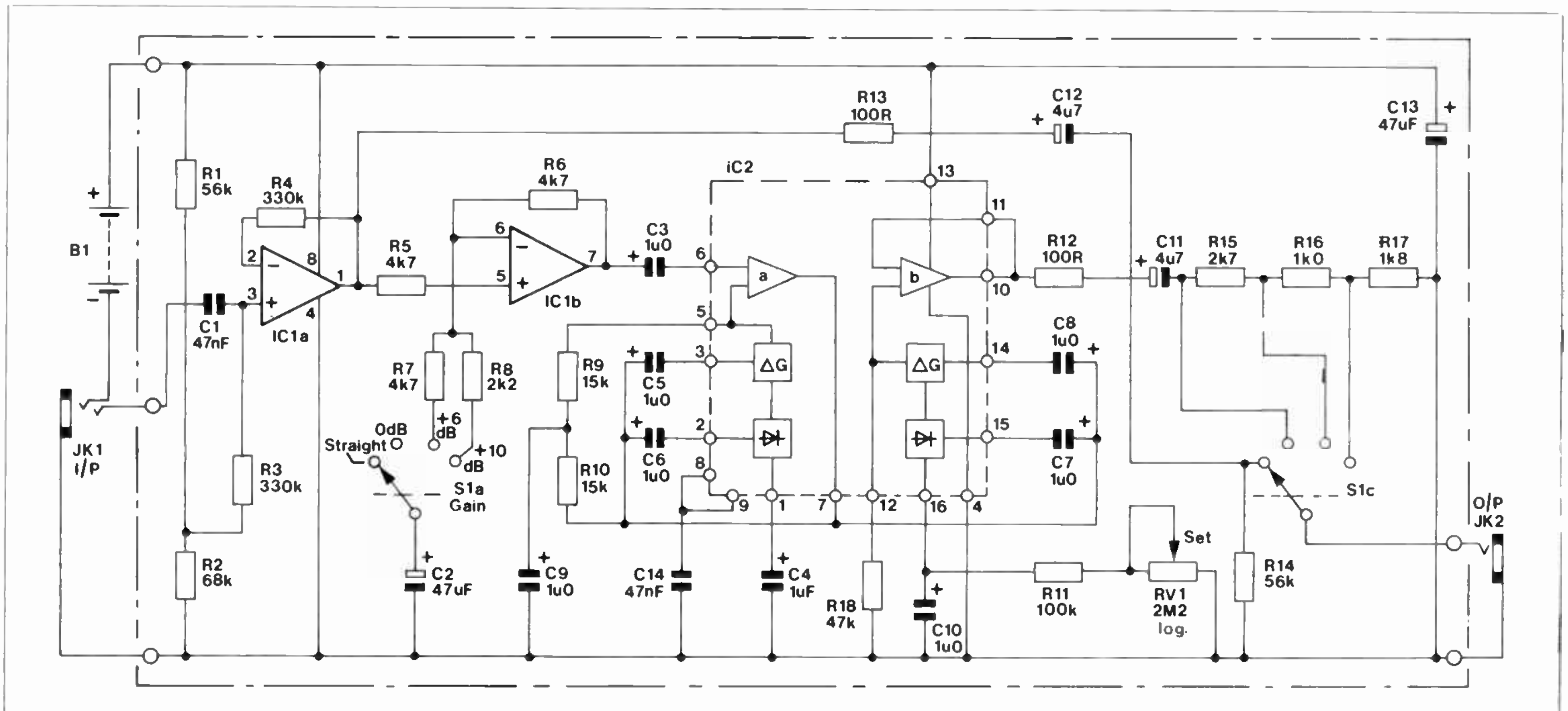
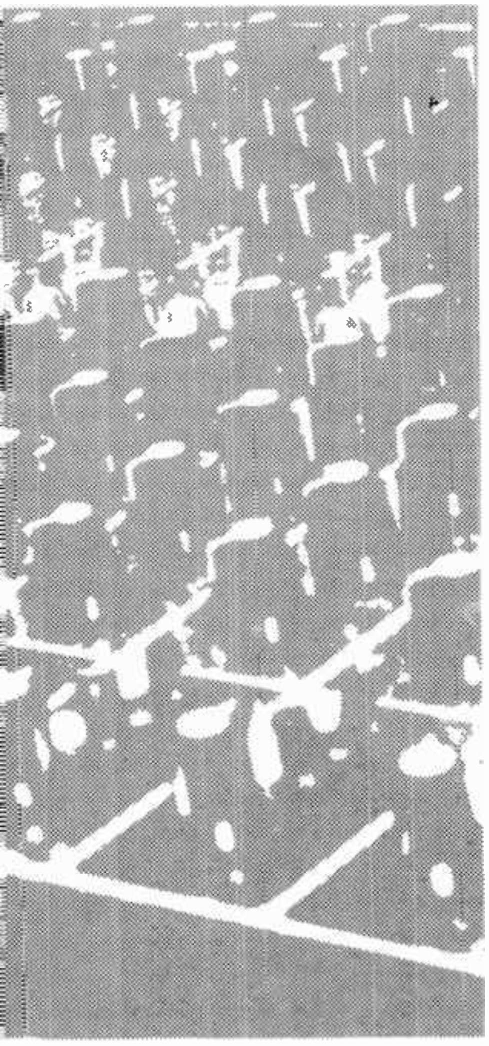


Figure 1. Circuit diagram for noise gate.

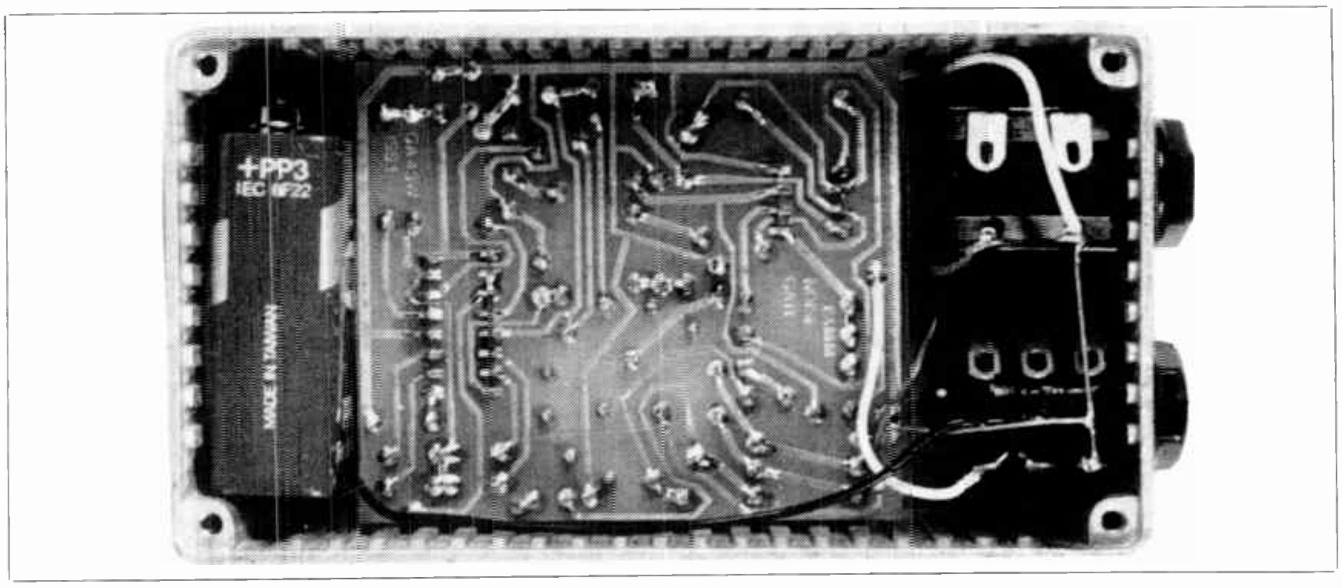


and general use, giving a more musically acceptable sound entrance and exit than the 'sudden shutdown' units.

Having a wide range of user adjustable characteristics, it should find many a useful working place with, for example, guitar / organ / keyboards / mic levels to mixer desk/P.A./recording levels. Not only can it be used for its main purpose, that of closing down noise or unwanted signals below a selected level, but as an effect in its own right, creating soft attack, bowing type characteristics.

Noise gates have been in use for considerably longer than most people would imagine. In fact, the first application of these devices was in the 1930's, when they were used to reduce unwanted crackles and pops from soundtrack film that had become dusty or scratched.

The most popular use for the dynamic noise gate is that of removing undesirable noise during pauses in live or recorded performance. To achieve this, the threshold would normally be set just under the required signal level



Internal view of noise gate.

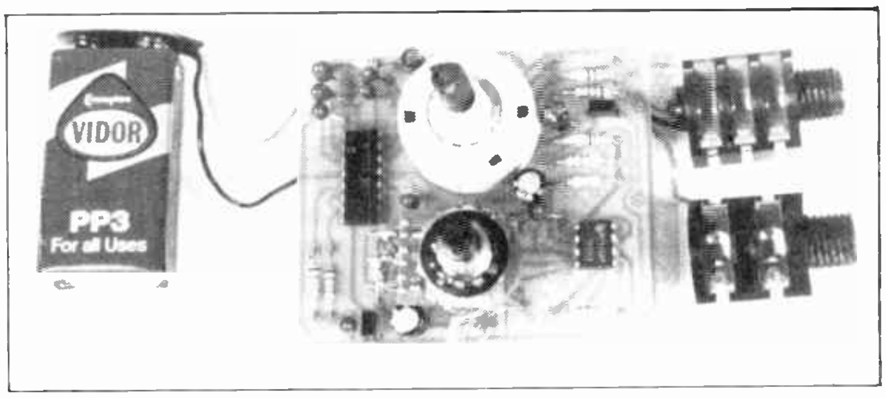
The unit is easily constructed and does not require any 'setting up'. As a self contained unit, it can be used in different applications at will, although being very compact it may be just as easily panel mounted for 'one-off' noise conditioning.

Circuit

Block Format

1. Jacking input connects the -ve battery line to the common earth line by the use of a stereo socket. (A mono jack will short ring and earth contacts on the socket to effect power up).
2. A high input impedance stage is used to prevent input device loading. The output of this stage serves also as the straight driving unit for pre-post comparisons.
3. A switched gain stage provides adjustable sensitivity and allows flexible device usage.
4. The first stage of the dynamic noise gate's active circuitry consists of a fixed 2:1 compression network.
5. The second stage of this active circuitry consists of a 1:2 expansion network with a simple resistive control element (potentiometer) for adjustable low level mistracking. This action is responsible for the total characteristics of the unit.
6. A passive attenuator is switched in conjunction with (3) such that an overall 1:1 input/output level is maintained. Included in the switch position is a straight-through route which allows comparison tests.

For those interested in such details, phase inversion between input and output does not occur.



Internal parts removed from case.

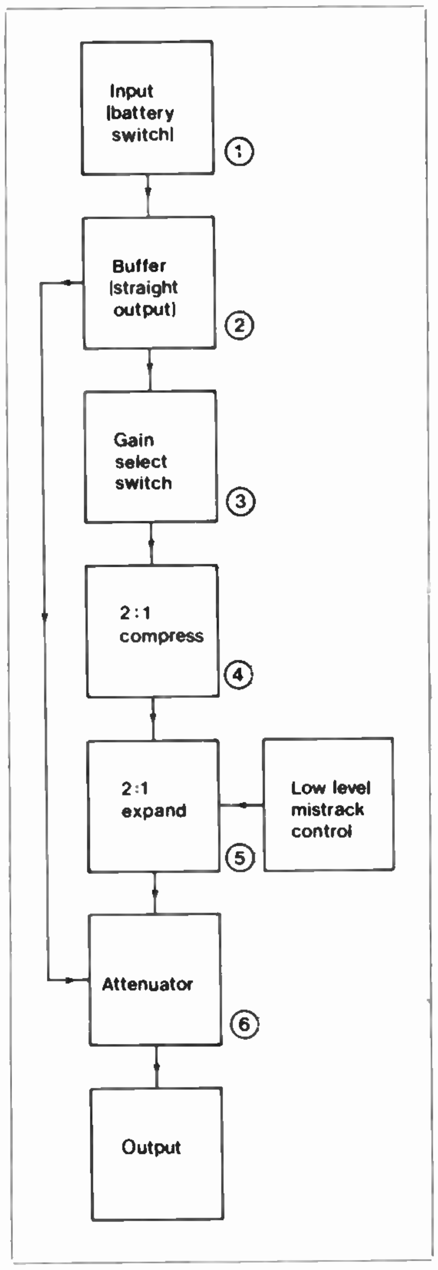


Figure 2. Noise gate block diagram.

Electronic Circuitry

Signals enter the +ve input of IC1a via the decoupling capacitor C1. IC1a is biased by R1 and R2 via R3 (R3 sets the input impedance level plus R1/R2). The voltage set by R1 and R2 ensures that outputs from IC1a and b will, at maximum levels, be evenly clipped. (At lower than normal V_{cc} levels maximum +ve excursions will be greater than available -ve excursions due to internal circuitry). R4, connected in the negative feedback line, equalises input biasing (R3=R4) for minimum output offset in IC1a.

IC1b is used in the non-inverting mode and has gain levels of 0dB, +6dB and +10dB, selectable via S1a, which introduces feedback reducing resistors R7 and R8.

Entry to the Compander section is now effected via C3, with compression achieved through IC2a and expansion by IC2b. All components in this section have been selected for the best overall performance in terms of frequency, speed of operation and distortion when bearing its 'musical' application in mind.

In the expansion section, RV1 is the resistive control element and operates by giving an increasingly false representation of lower level. Rectified signal voltages appear on C10, as RV1 resistance is decreased, causing an overall increase in attenuation rate.

This increasing attenuation characteristic creates low level expansion and is used to reduce the dynamic range of any signal within its domain with consequent drop in noise level. At higher signal levels RV1 becomes less effective in its role and allows a return towards original signal dynamics.

Returning to normality, S1c selects the resistors R15-R17 that are required to achieve an overall device gain of unity. S1a in conjunction with S1c also selects the output of IC1a, enabling pre and post 'gate' comparisons to be made.

Construction

The construction details relate to a cased unit, for panel mounting all that is necessary is to 'fly-lead' the input, output and RV1 wiring. Circuit board fixing is effected by the rotary switch on which the board is mounted. PCB assembly should begin with the two link wires.

Resistors, capacitors and ICs may now be fitted. Double-check polarised capacitor and IC orientation. Pins, or tinned copper wire should be soldered on to the board to hold RV1 in position, remembering that since RV1 is mounted above the board, components beneath it should lay flush to the board.

Solder the battery connector with the positive lead connected to the correct point on the board, and the negative lead to the centre tag of the stereo socket.

Fit the rotary switch next. This switch is normally obtained with solder type eyelet tags which need simple modifications for PCB fitting. Cut these eyelets off, as close to the solder hole as possible, then using snips, taper the ends to assist in aligning and fitting to the board. If the use of fixed pointer, push-on knobs is envisaged, remember to position the switch correctly bearing in mind the shaft 'flat' orientation.

Using tinned copper wire, earth, input and output leads can be connected to the board (using sleeving on the leads or slightly bending the wiring to avoid board shorts). After making sure all components have been fitted correctly, a few quick checks with a meter will verify correct basic operation. Switch to 100mA DC current range and connect the negative lead to battery and positive lead to earth. By touching the battery on to the connector a reading will be registered. A quick 'blip' and a reading under 10mA shows all is well. Voltage checks on outputs of ICs will confirm this - pins 1 and 7 on IC2 should be approximately 5V and pins 7 and 12 of IC2 about 3.6V. (These voltages are related to internal references that remain constant, and therefore allow for battery voltage reduction.)

All that remains now is to mount the unit in a suitable case. The metal case suggested in the parts list is ideal since it is small, easily workable, and provides good screening. Two appropriate holes in both the top and the side are all that are needed so that the whole assembly can just be fitted straight into the main case body and secured with the switch and jack sockets. An earth wire soldered to the edge of RV1 will ensure case earthing. If a plastic case is used, use adhesive backed metal foil tape (as used in pipe cladding) or glue household metal foil for internal screening.

Operation and Application

To obtain maximum usage of the unit, its functional characteristics should be fully understood. This can be achieved mainly by studying the response curves. The 1:1 gain slope reference allows visualisation of the deviation from normal input/output characteristics. The curve closest to the 1:1 gain slope shows input/output of the device when set in any gain position with RV1 set for minimum effect (clockwise). Note that just below -60dBm input levels, the output deviates more rapidly towards -85dBm. This is the operating region of the unit and any signal or noise below -60dBm will be attenuated, reducing its effective level. Any signal above -60dBm will have virtually normal dynamic range. Studying the curves of maximum effect characteristics shows that the noise gate can expand signals from even -15dBm down and completely shut down below -38dBm. (This setting will shut off most extraneous noises.)

Since the compander section uses rectified signal levels in its operation, speed of recognition of these levels becomes a compromise between several factors, one of which is the loading of the circuitry by RV1. It should, therefore, be remembered that the unit will take a finite time to attack and decay and that these times bear a direct relationship to the threshold level (that level selected at which deviation from normal characteristics occurs), and change in amplitude of the input signal. For example, with the noise gate set at 0dB gain with input levels gated from infinity to 0dBm, attack times will vary from typically 2ms to 0ms for minimum to maximum threshold settings.

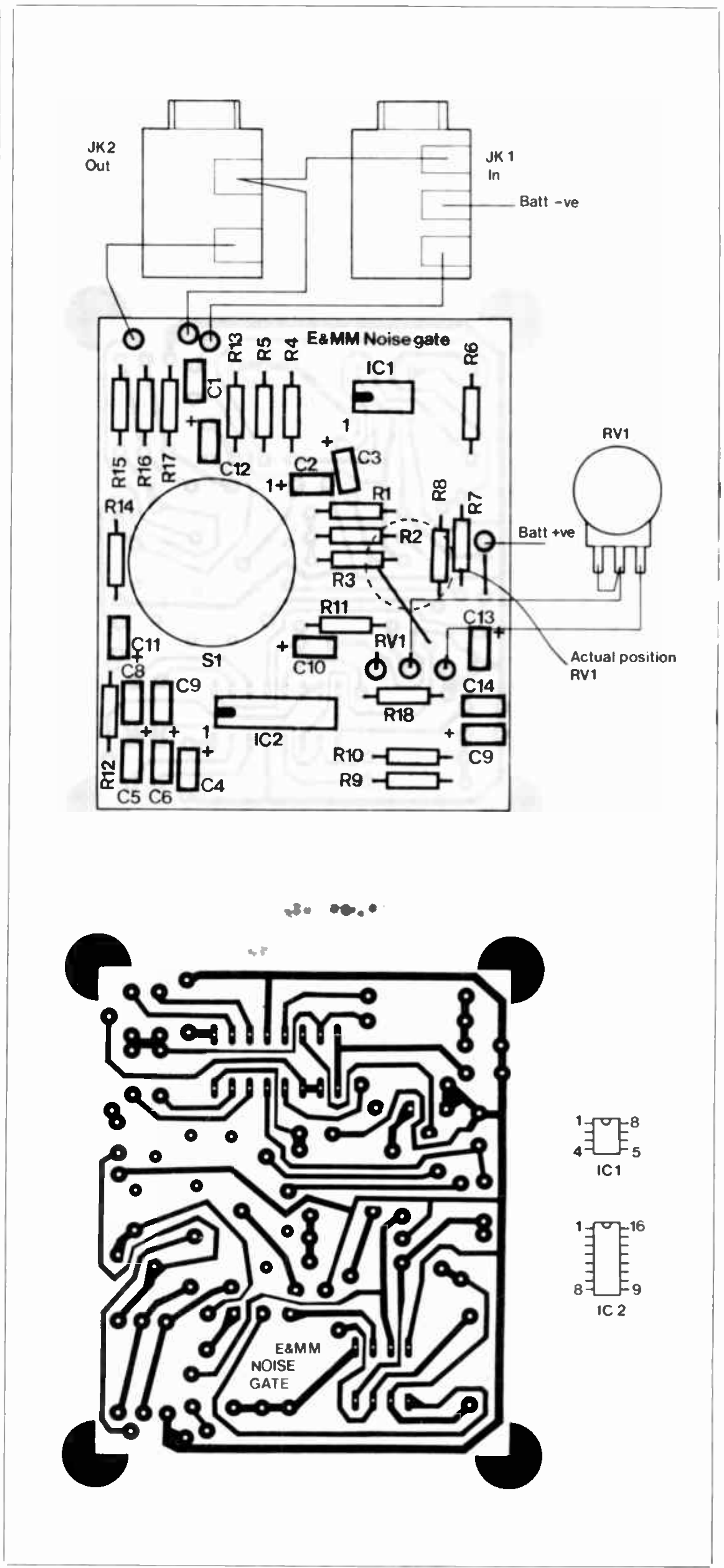


Figure 3. PCB track, legend and wiring details.

Before continuing with typical applications, it should be remembered that this unit is designed for 'pause' noise reduction, and has no magic ingredient for reducing any noise present in actual signals (see E&MM May 1981 for a noise reduction project).

Wherever noise exists, in-line connection of the noise gate should discriminate and further separate it from the required signal. Probably the lowest levels encountered will be from microphones and low output guitars (low Z mics will have to be transformed or the unit modified to suit). In the case of microphones, crosstalk (pick-up from sources other than that intended)

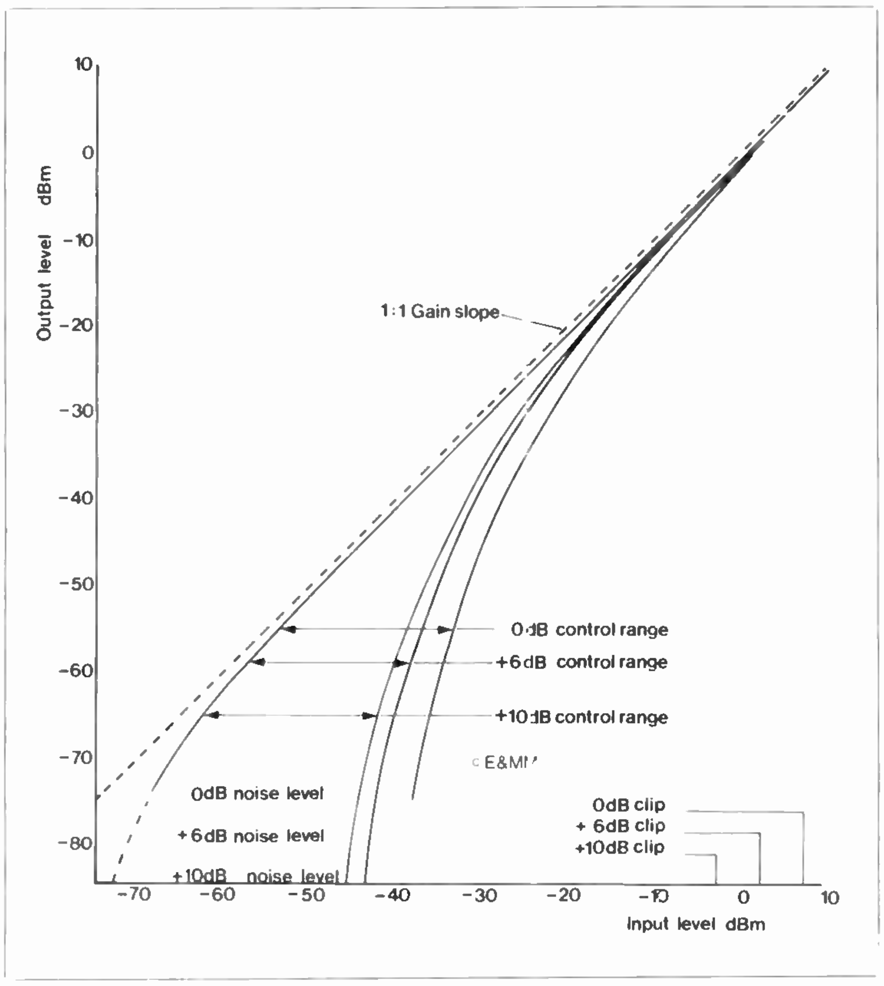
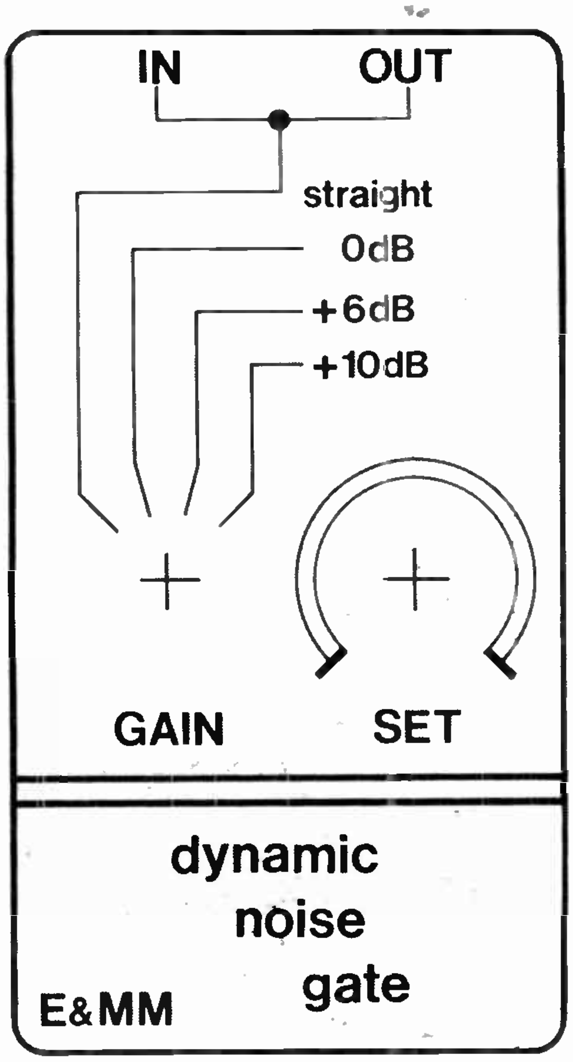


Figure 4. Response curves for noise gate.

NOISE GATE PARTS LIST

Resistors - all 1% 0.4W metal film unless specified			
R1,14	56k	2 off	(M56K)
R2	68k		(M68K)
R3,4	330k	2 off	(M330K)
R5,6,7	4k7	3 off	(M4K7)
R8	2k2		(M2K2)
R9,10	15k	2 off	(M15K)
R11	100k		(M100K)
R12,13	100R	2 off	(M100R)
R15	2k7		(M2K7)
R16	1k0		(M1K)
R17	1k8		(M1K8)
R18	47k		(M47K)
Capacitors			
C1,14	47nF min. ceramic	2 off	(YY10L)
C2,13	47uF 16V PC electrolytic	2 off	(YY37S)
C3-10	1uF 35V tantalum	8 off	(WW60Q)
C11,12	4u7 25V PC electrolytic	2 off	(YY33L)
Semiconductors			
IC1	MC1458		(QH46A)
IC2	NE571		(YY87U)
Miscellaneous			
S1	Switch 3-pole 4-way rotary		(FH44X)
RV1	2M2 log. pot.		(FW29G)
SK1	Stereo jack socket		(HF92A)
SK2	Mono jack socket		(HF90X)
	Battery connector		(HF28F)
	Case - M5004		(LH71N)
	Knobs, low cost collet	2 off	(YG40T)
	Cap, low cost, grey	2 off	(QY03D)
	PCB		(GA43W)



will be the problem to cure. The +10dBm setting will allow maximum dynamic range with fast attack to be achieved when dealing with low level crosstalk (drum kit miking, awkward placement instrument miking, outside recording in windy conditions excluded).

Guitarists working in cramped conditions may encounter induced hum from closely situated amps, especially via single coil pick-ups. This often embarrassing situation can be alleviated by using the noise gate between guitar and amp., adjusting the threshold control for the best overall effect. Another interesting use of the noise gate is for changing the attack characteristics of an instrument. If the threshold level is taken to extremes, soft attack bowing type sounds can be created.

Similar treatment can be applied to special effects units (Echo, Chorus, Phasing, Distortion etc.), by using the noise gate between the last unit and its main amplification. The +6dB mode will be the norm in this application, since typical peak levels may introduce clipping.

Multi-instrument set-ups can often make the background noise unacceptable, considering that all units probably remain set at the required output level when only one or two instruments are actually being used at any time. Fitting individual noise gates to the noticeably 'noisy' instruments will automatically shut off their outputs when not in use.

Extreme levels of noise, or higher input levels, can be coped with in the 0dB mode, making this setting suitable for most line and mixer desk levels.

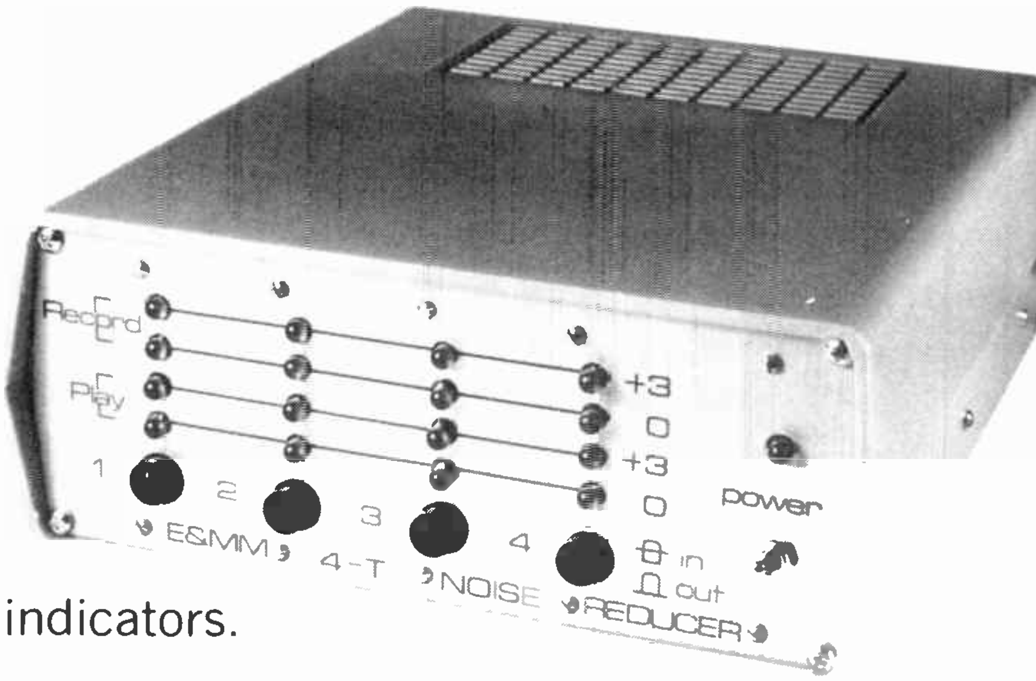
In the studio, the noise gate is often used with drum kit multi-miking, where as many as 14 microphones may be allocated to the kit in an isolation booth. A rather woolly sounding bass drum can be tightened up effectively by using a fast attack setting (further improved by coupling with a slow attacking limiter) that recovers some of the basic square wave response. It's even worth trying this method using an LFO signal source to create a synthesised drum sound.

'Dynamic reversal' effects can be obtained using the gate followed by a fast attack limiter. With slow recovery times the signal applied is thus strongly over-limited and during the limiter's slow recovery is gradually attenuated by the gate on a slow release setting. The resulting effect sounds like a 'backward' snare drum, tom-tom or cymbal.

It is also possible to reduce fixed delay times of instruments or reverberation treatments by suitable setting of the noise gate control parameters. One further application useful as a treatment in electronic music, is the 'keyed' or programmable gate where an external control voltage switches the signal being processed (used by Irmin Schmidt on his 'Last Train to Eternity' featured last month). But that's another project . . .

NOISE REDUCTION UNIT

by Dr David Ellis



Ideal for the home or semi-professional recording studio, a four-channel compander offering 30dB improvement of signal-to-noise ratio, simultaneous encoding/decoding and LED peak indicators.

It's on the cards that within the next five years there'll be a sixteen-track digital cassette recorder, complete with a touch control mixer, in a box the size of Teac's Portastudio. Even now, at the top end of the synthesizer spectrum, there's a new Crumar programmable polyphonic synthesiser with built-in digital recorder (but yet to arrive in this country) and the amazing Synclavier II complete with sixteen-track digital memory and every editing facility under the sun, a snip at a mere £16,000!

The great advantage of digital recording is the absence of noise, and, if one's working with digital instruments, as in the case of the above polysynths, then there's no A/D conversion before the sounds are committed to 'tape'. However, at what might be described as the tail end of the analogue era, most of us are stuck with trying to get the best possible sound out of our trusty Revoxes, Teacs, or whatever. The main problem with such machines is their annoying habit of burying your latest creation under a blanket of tape noise as soon as you depart from the first tape generation. And, with the new Teac 3440, the basic quality is so fine that some way of preventing the build-up of tape and machine noise seems a pretty logical step to take.

Noise reduction systems reduce the irritating noise of tape hiss and so on by an encode-decode process. Quiet sounds, especially those at the top end of the spectrum, are easily swamped by tape hiss, so an encoder is used to artificially boost these signals before they are recorded. During playback the reverse process decodes the recorded sound back to its original state and rids the music of tape-generated noise. Up until recently, noise reduction systems have fallen into three distinct types: Dolby B (domestic), Dolby A (professional) and DBX (professional). However, the near future is likely to see a confusing proliferation of other systems offering various degrees of noise suppression, including: Toshiba's Adres system, Telefunken's High-Com and Telcom, Sanyo's Super D, Dolby's C and HX systems and Tandberg's Dyneq. If there's any sense in this race to the pinnacle of perfect music reproduction, then hopefully there'll be some common standards of operation agreed upon! Table 1 gives the S/N ratios obtainable from various recording mediums with and without different types of noise reduction.

The various systems of noise reduction available at present basically work on the principle of complementary compression of the on-tape signal and expansion of the off-tape signal. Compression involves reducing the dynamic range of the material that is being recorded, so that, with a 2:1 compression ratio, if the input to the compressor increases by 12dB, then the output of the compressor (on-tape signal) will increase by only 6dB. Conversely, expansion involves increasing the dynamic range, so that an increase of 6dB in off-tape level will result in a 12dB increase fed to a subsequent mixer, thereby restoring the original dynamic level of the music. At the same time, the noise introduced in the recording chain, in particular tape hiss, will be rendered inaudible on expansion since this unwanted signal is not subject to the initial compression treatment and is therefore expanded downwards way below the lowest dynamics of the music signal. This process is illustrated in Figure 1.

Another feature of the compression/expansion process is that it allows the recording of signals with a dynamic range

Recording medium	Noise reduction	S/N ratio	Comments
Cassette (Sony TCK55 II)	- Dolby B	57dB	
	+ Dolby B	67dB	Above 4 KHz
	+ Dolby C	75dB	Above 1 KHz
	+ HighCom	75dB	Above 1 KHz
Four-track tape (Teac 3440)	No noise reduction	55dB	
	+ E&MM unit	85dB	Above 30 Hz
Two-track tape (Studer)	No noise reduction	70dB	
	+ Dolby A	80dB	Above 20 Hz

Table 1. Comparison of Noise Reduction Systems.

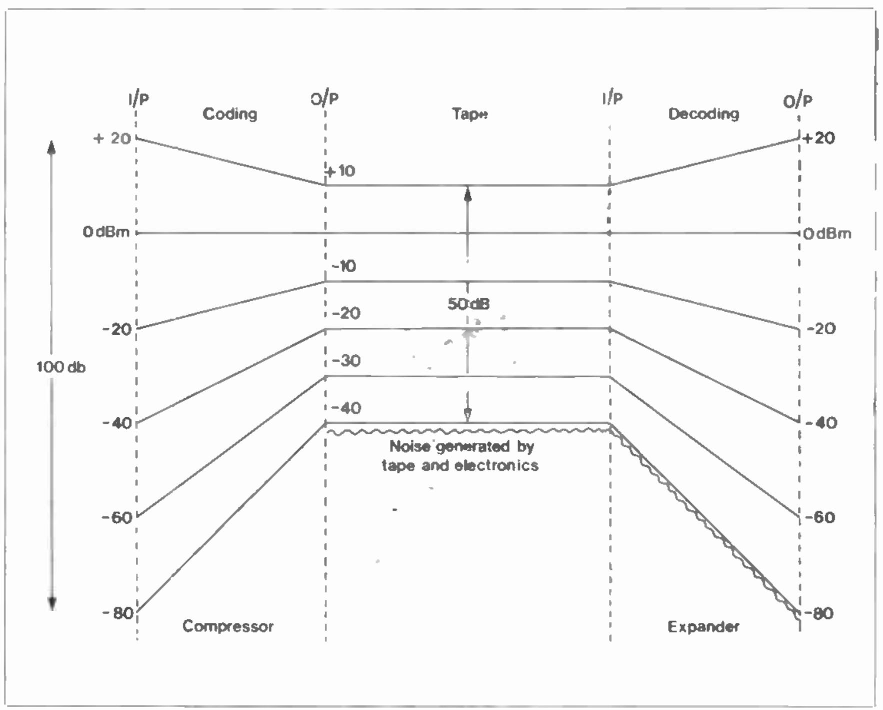
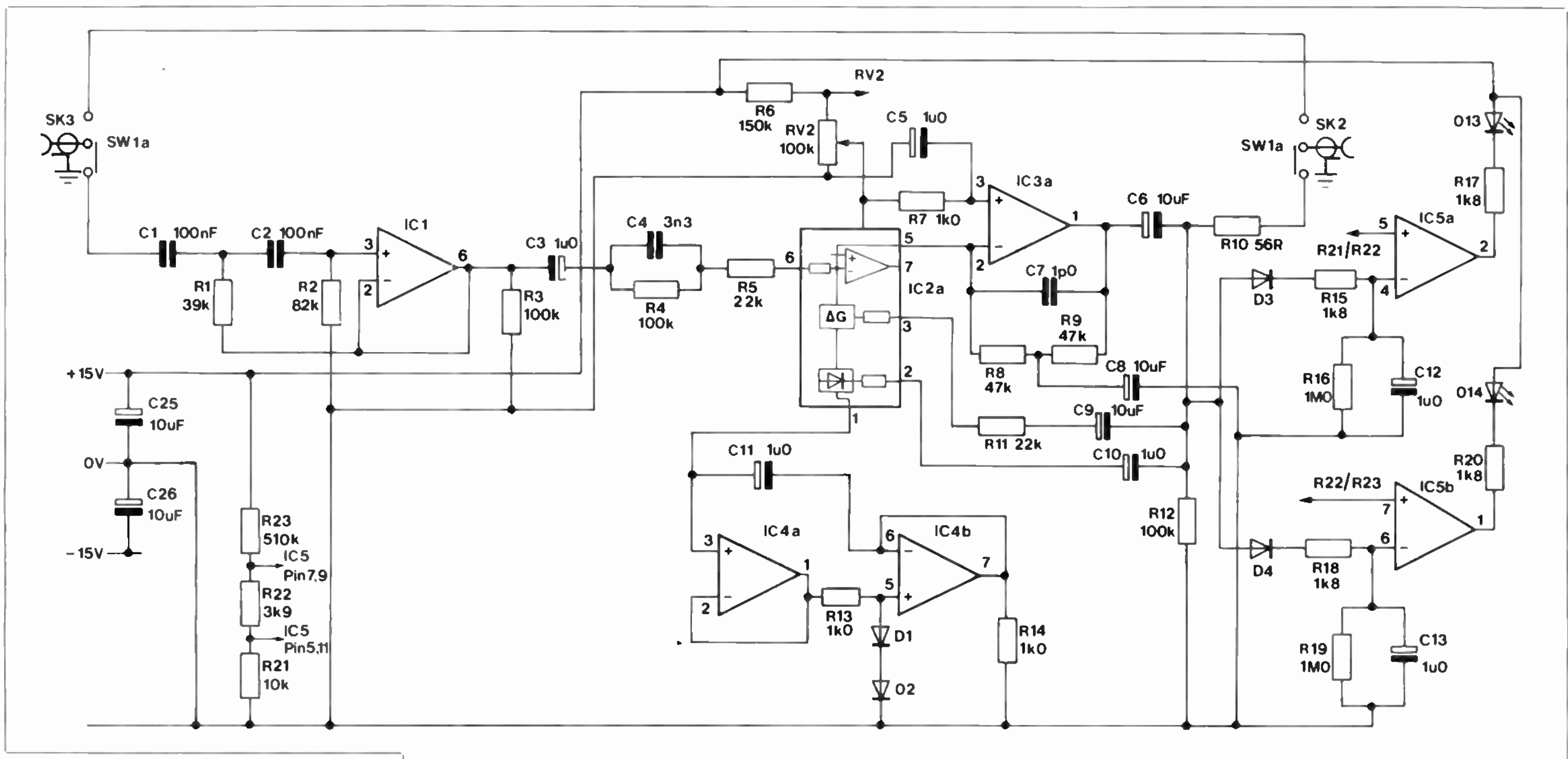


Figure 1. Operation of a compression/expansion system.



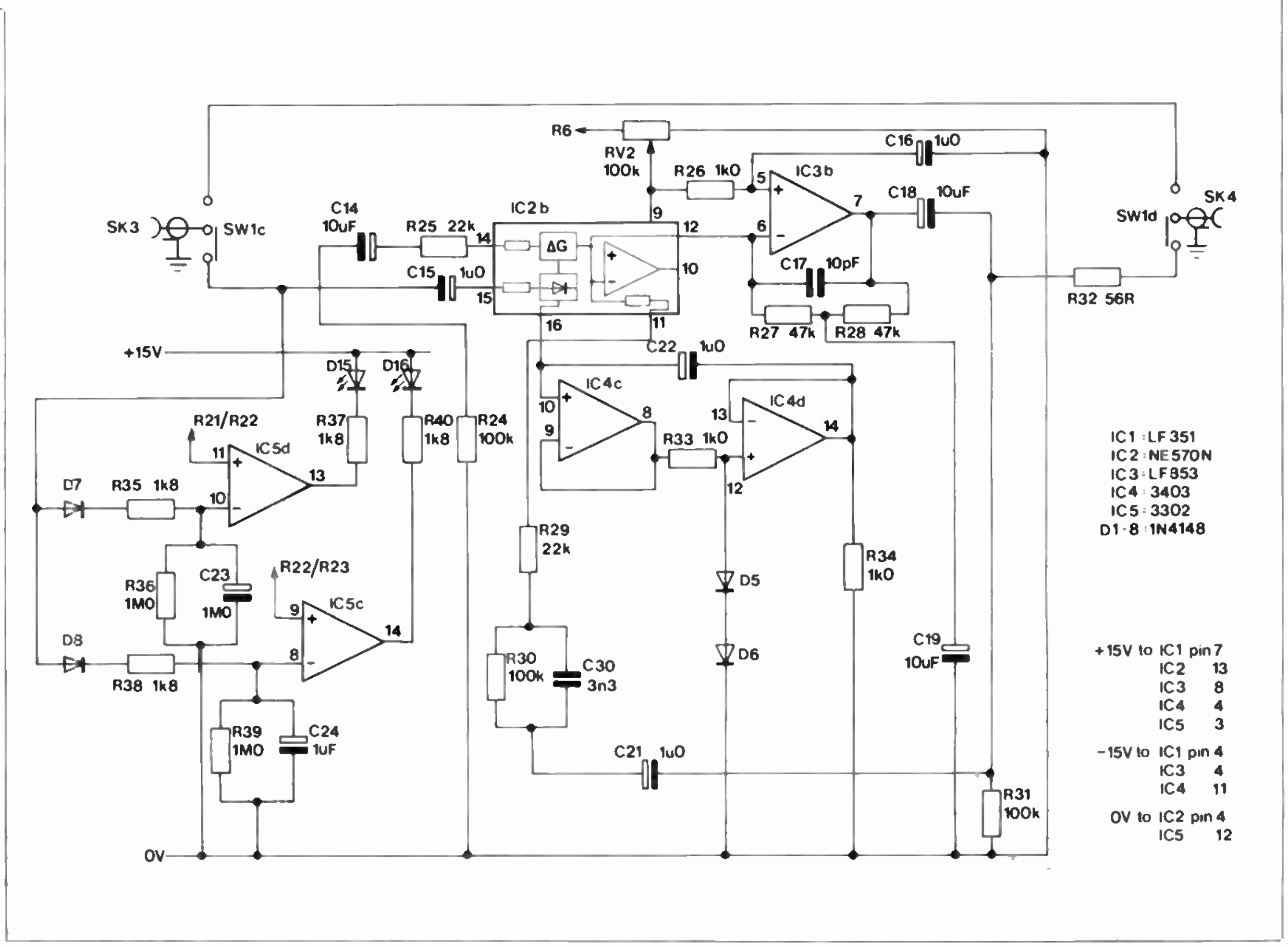


Figure 2. The circuit of the compressor and expander.

approaching the limits of audibility, i.e., 100 to 120 dB. However, since modern-day musical experiences tend either to be restricted to the bottom end of the dynamic range (muzak) or stuck at the top end (rock and heavy metal), this facility may be more theoretical than practical!

With careful shopping it should be possible to make the complete four-track unit for around £55.

Circuit

The circuit diagram for the compressor and expander is shown in Figure 2. The power supply circuit is given in Figure 3.

The compressor input is routed via SW1a, either directly to the output in the 'out' position, or to C1 in the 'in' position. IC1 and associated components form a second-order high pass filter with a 12 dB/octave roll-off below 30Hz. This removes sub-audible signals (infra-sonics) that might be generated from record warps or sub-octave tracking VCO's. The reason for this filtering is that once audio frequencies descend towards DC, the response of tape recorders drops-off dramatically, and on playback a signal compressed in response to high level low frequency signals will be expanded, resulting in phantom modulation by the missing low frequency component lost during recording. The output of the filter is AC-coupled to a simple RC network (C4, R4) which forms a high frequency pre-emphasis circuit providing a 12 dB treble boost. Without this pre-emphasis, and corresponding de-emphasis in the expander, a low level signal may be swamped by high level bass frequencies and typically results in a 'heavy breathing' or pumping effect as the expander attempts to adjust the gain accordingly. The signal is then applied to one half of the NE570 (IC2a) configured as a compressor using an internal variable gain cell and full-wave rectifier as well as an external output op-amp (IC3a) The variable gain cell is similar to a standard operational transconductance amplifier (OTA), except that, unlike OTA's, it is 'linearized' and therefore insensitive to temperature changes as well as offering low noise and low distortion performance. The signal at the output of IC3a is rectified and the resultant control voltage used to adjust the variable gain cell. By placing the gain cell in a feedback loop with the op-amp, a variable current generated in proportion to the input signal is used to adjust the overall gain of the op-amp. A 6 dB increase in output level produces a 6 dB increase in the gain of the variable gain cell, and, since this is effectively an expander inserted in the feedback loop, results in a 12 dB increase in feedback current to the input of the op-amp. Consequently, an increase in input level of 12 dB results in only a 6 dB increase at the output of the op-amp, thereby yielding the desired 2:1 dynamic range compression.

The current from the full-wave rectifier is averaged by an external filter capacitor (C11) with the result that the gain control is made proportional to the average value of the input signal. The speed with which this gain adjustment is made determines the transient response of the compressor and is a product of the value of the filter capacitor and an internal 10k resistor. The value of 1uF for C11 yields good transient response at average signal levels. However, at low signal levels, the gain of the op-amp increases and any mistracking that occurs between the compressor and expander will be magnified by the high gain levels. To improve tracking at low dynamic levels it is necessary to provide a level-adaptive circuit that speeds-up

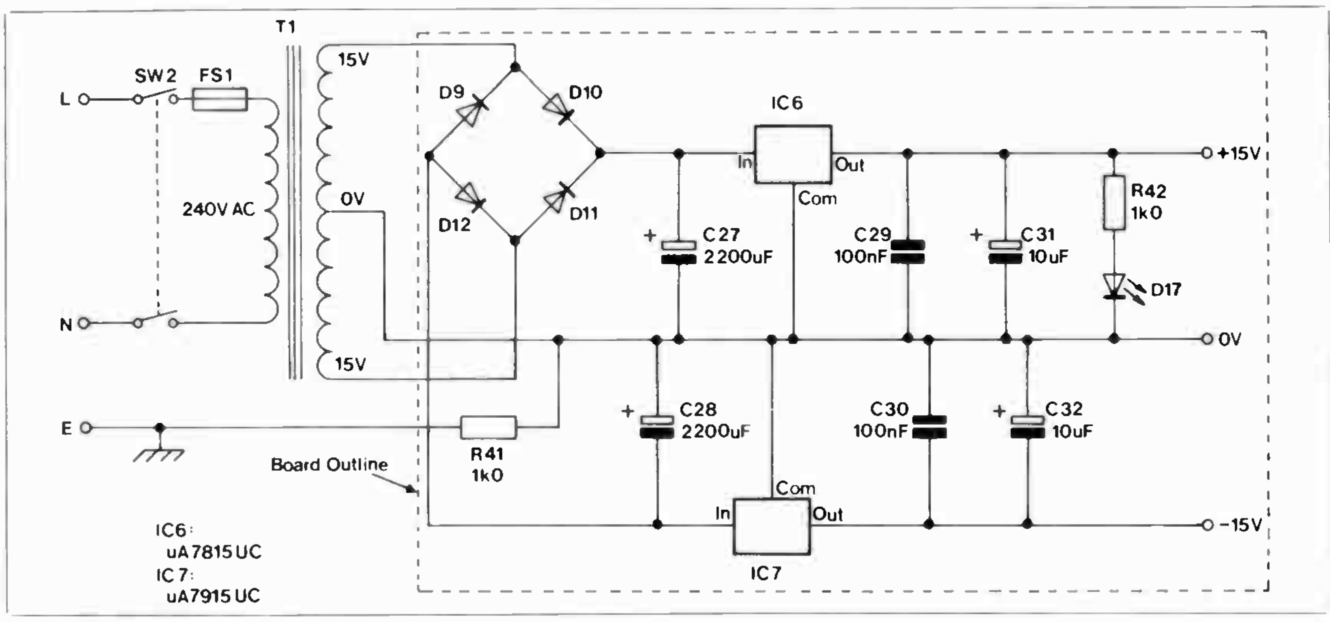


Figure 3. The circuit of the power supply.

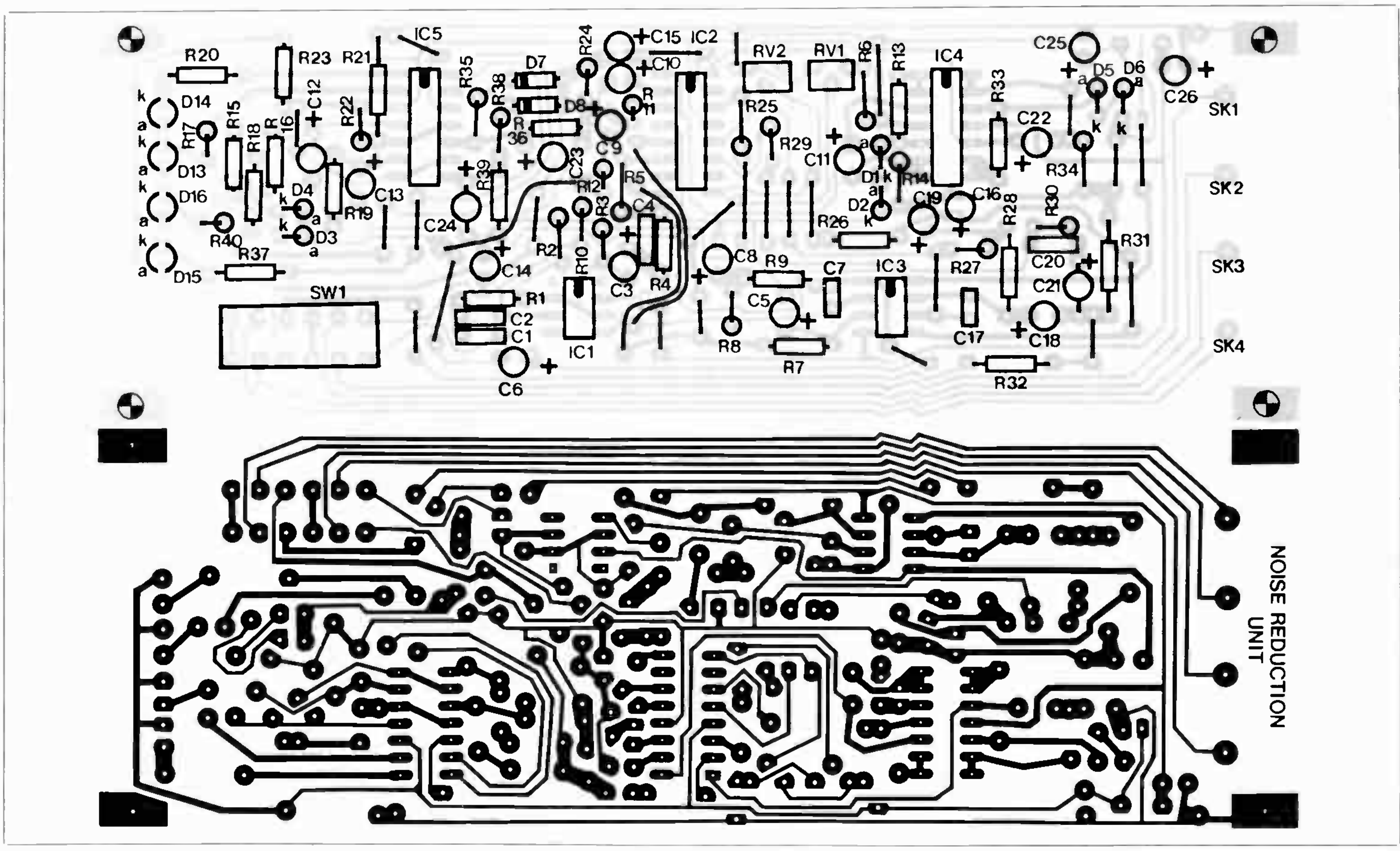


Figure 4. The compander PCB.

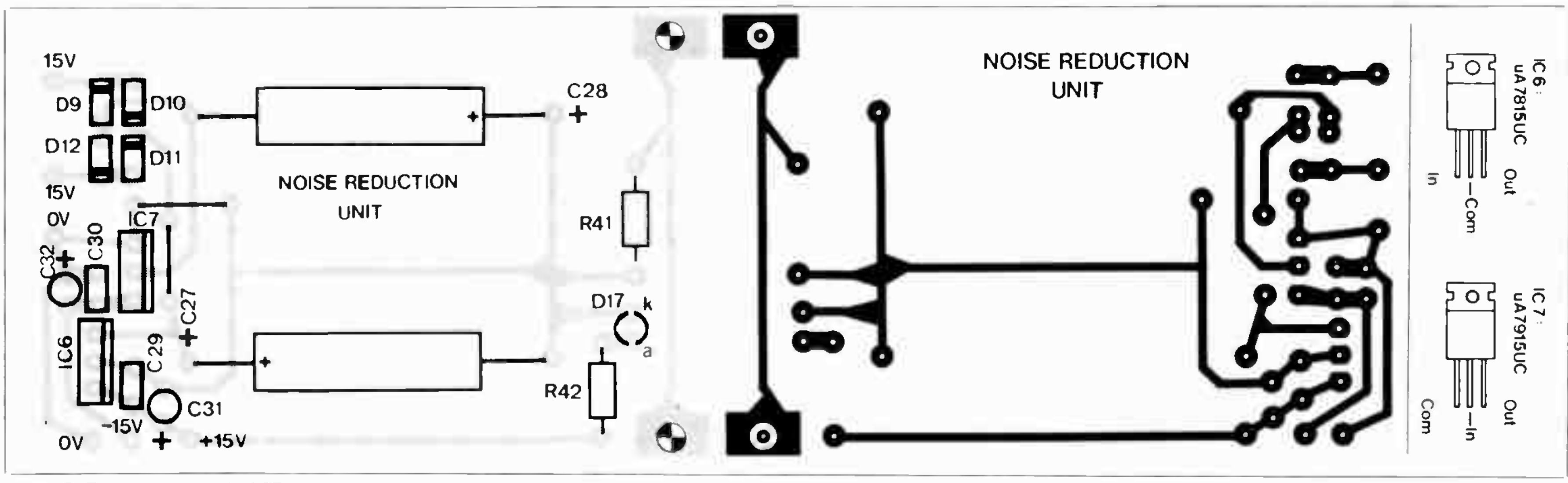


Figure 5. The power supply PCB.

the response time. This feature is derived from the circuit built around IC4a and b with series diodes (D1, D2) shunting the output of IC4a to ground.

Op-Amp slew rates

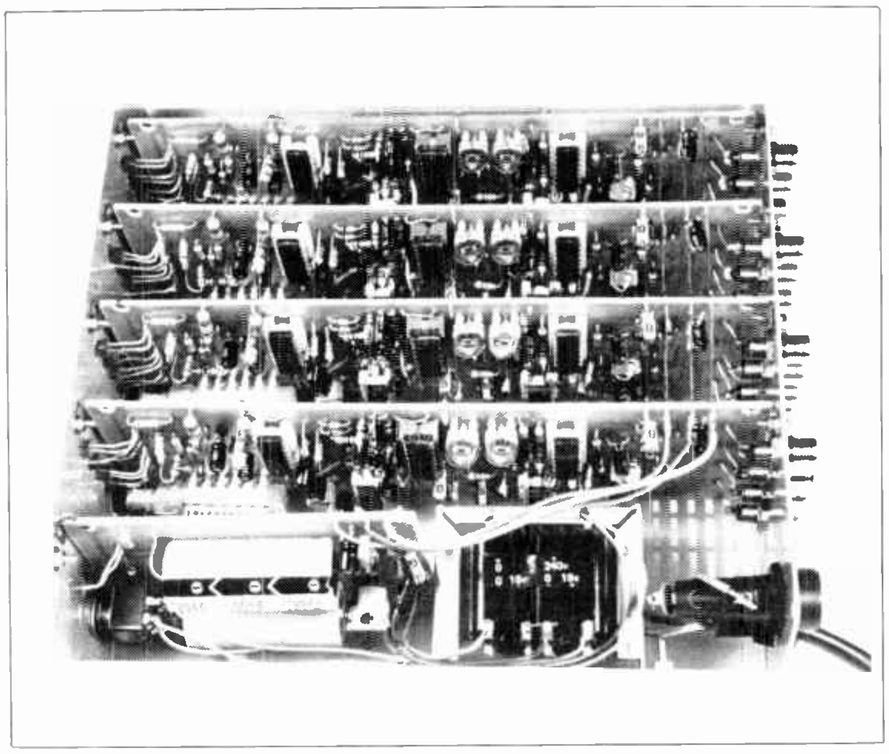
The RCR network (R8, C8, R9) around the op-amp, IC3a, provides DC feedback to bias the output at DC. C7 is an external compensation capacitor to provide stable operation over the audio bandwidth. It may seem curious to use an external op-amp when the circuit diagrams indicate that the NE570 has its own. This is because the op-amps in this IC are equivalent to 741-types with slew rate, noise, bandwidth, and output drive capability that aren't really adequate for demanding audio situations. With weak signals, the compressor circuit operates at high gain and the NE570 op-amp runs out of loop gain. Furthermore, a slew rate of 600mV/us means that high frequencies will suffer. By using a J-FET op-amp, such as the LF351 with a slew rate of 13V/us, these problems are eliminated. Additionally, the output swing can be larger since IC3a is powered by a dual supply rather than from the single-rail supply required by the NE570.

The non-inverting input of the NE570 op-amp is biased by an internal reference voltage of 1.8V. In the case of the external op-amp, IC3a, this is accomplished by tying it to pin 8 via an RC decoupling network (R7, C5) which filters out noise from the NE570 reference voltage. Pin 8 also serves another important function, that of providing the means for trimming distortion generated by IC2a. Even harmonic distortion is produced by voltage offsets in the variable gain cell, and RV1 enables adjustment of the offsets for minimum distortion.

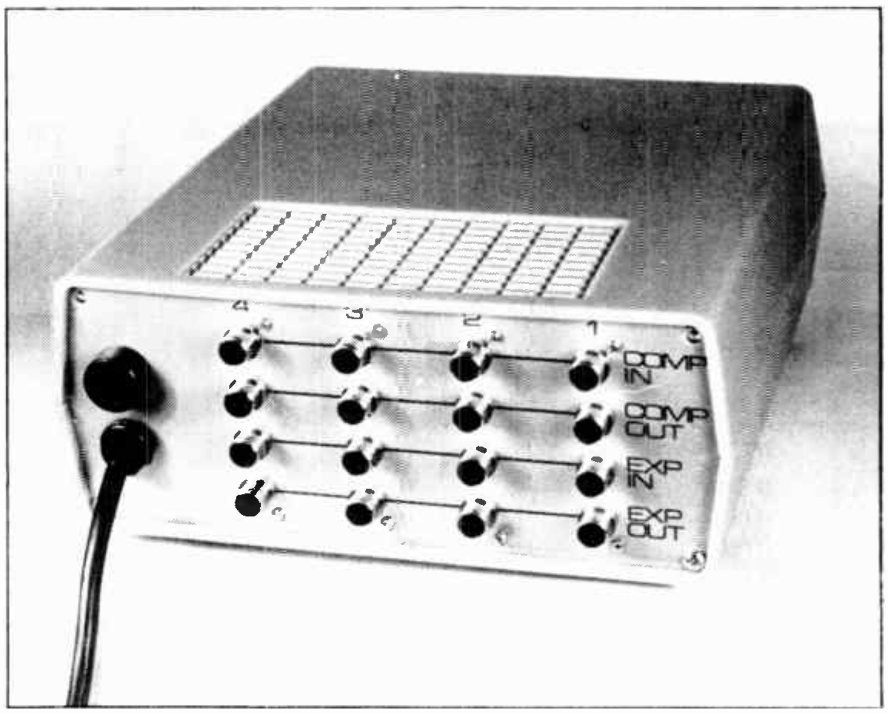
Comparator functions

The function of R10 is to isolate the output of IC3a from the potential capacitive load of a long length of screened cable connected to the compressor output which could lead to oscillation. SW1b selects the 'in' or 'out' mode of operation.

Comparators IC5a and b provide an indication of the signal level at the output of the compressor. The inverting inputs receive the half-wave rectified output signal which is compared with reference voltages derived from the potential divider network, R21, R22 and R23. C13 and R19 determine the fast attack/slow



Internal layout of the Noise Reduction Unit.



Rear view of the case showing connections.

decay operation of the comparators IC5a and b respond to signal levels of, respectively, -3 dBm and 0 dBm.

The expander configures the other half of the NE570, IC2b, with a different arrangement of the various blocks. Once the off-tape signal has been routed via SW1c to C14, the signal is applied to comparators, IC5c and d, to provide an indication of off-tape levels, and simultaneously to the full-wave rectifier and variable gain cell. The rectifier produces a control voltage that is used to adjust the gain cell, with a response time determined by the level-adaptive circuit of IC4c and d tied to the rectifier filter capacitor (C22). An RC network (R30, C20) is connected in parallel with the op-amp, IC3b, to provide a treble cut of 12 dB, therefore de-emphasizing the pre-emphasized signal emerging from the compressor via the tape recorder. When the input signal increases by 6 dB, the gain cell control current is raised by a factor of 2, resulting in an increase in gain of 6 dB. Since the input of the external op-amp, IC3b, is derived from the gain cell, the output level increases by 12 dB, giving the required 1:2 dynamic range expansion. RV2 enables adjustment of gain cell offsets for minimum distortion, as in the compressor. Finally, R32 isolates the output of IC3b from subsequent screened cable, and SW1d selects the mode of use.

Construction

The unit is designed on a modular basis so that each PCB provides simultaneous compression and expansion for one channel. Single sided PCB's have been used to keep the cost down, though double sided PCB's could easily be made from the layouts provided, eliminating 29 links from each one.

In order that decoding should be the exact inverse of coding, it is important that components are well-matched. This is obviously no problem with resistors, but the notorious variability of electrolytic capacitors necessitates the use of closer tolerance components such as the minielectrolytics available from Maplin. These capacitors offer ±20% tolerance at half the price of conventional tantalum types.

PCB designs and component overlays for the main board and PSU are given, respectively, in Figures 4 and 5. The threaded phono sockets suggested for the unit have the dual advantage of small physical size, enabling them to be fitted as rows of four on the back panel, and compatability with the connectors normally encountered in using Teacs and Revoxes. These sockets are mounted on the rear panel and connections to the signal pins made by short lengths of unscreened wire from the relevant holes on the PCB's.

The PSU is utterly standard, though it's important to note that mains earth is connected directly only to the front panel and indirectly via a 1k0 resistor (R41) to the 0V line. This should prevent the build-up of any hum loop when using the noise reduction unit with earthed equipment. Power line busses can be connected from the PSU to all four main PCB's.

Setting-up and use

The unit requires very little setting-up apart from adjustment of RV1 and RV2. The output level of a mixer is adjusted so that the compressor 0dBm LED's fire with louder dynamics. The record level is set to match the optimum requirement of the tape being used. Playback levels are then adjusted so that the expander 0 dBm LED's fire at the same level as the compressor 0 dBm LED's. This level matching isn't critical since the level-adaptive response time circuits take care of possible mistracking, but it does ensure really accurate decoding of the encoded signal.

A couple of points to note: the unit will not reduce the noise present in a noisy signal presented to the compressor input (this is territory best served by dynamic noise limiters) and any difference in the signal between compressor output and expander input introduced by the recording process will be exaggerated by expansion, including such horrors as common-or-garden drop-outs. Therefore to get the best out of the unit scrupulous attention should be paid of alignment and cleaning of tape heads!

To optimise the distortion levels of the unit, apply a 1V RMS (equivalent to +3 dBm) 10 kHz sinewave to the input of the compressor and expander in turn and adjust RV1 and RV2, respectively, for minimum distortion of the waveform viewed on an oscilloscope. This adjustment can also be carried out without testing equipment, but does require a good ear to get the audibly cleanest waveform.

Using the unit should be simplicity itself and basically you should be able to plug it in and forget all about it. Judging by the comments of two recording studios using the unit, it appears to be gentle and uncomplaining by nature, just like the ideal wife (or husband)!

Modifications

Apart from noise reduction, these circuits can also be adapted for use as a general purpose compressor-expander. The first modification to be made is to add a pre-emphasis/de-emphasis 'defeat' switch, bypassing C4 and R4 in the compressor and taking C20 and R30 out of circuit in the expander, so that the frequency response of the compressor or expander remains substantially flat. With this modified circuit, 2:1 compression can be performed on excessively wide dynamic range signals (guitar, piano, etc.) to get a more punchy sound, and 1:2 expansion can be carried out on previously compressed material (much rock music) to increase the dynamic range. However, this degree of compression or expansion is excessive for all but special effects and some means of altering the gain adjustment ratio is necessary. This is easily accomplished by introducing some variable feedback into the circuits using a couple of dual-ganged 4k7 potentiometers and some 4k7 resistors (see Figure 6). It should be possible to make these adaptations without too much destruction, but it will be necessary to break two tracks on the PCB, ie. that joining the track from C9, C10 and R12 with R10, and that from C15 and C14 to SW1a. It is important to note that the ganged potentiometers are connected so that the two halves operate in opposite directions.

The resistors and pots required to adapt a single compander board appear in the separate Modification Parts List.

With these adaptations, the compressor and expander sections will offer, respectively, a compression ratio adjustable from 1:1 to 2:1 and an expansion ratio ranging from 1:1 to 1:2.

Such a unit is similar to that marketed by dbx in the USA, and can be very useful in adding a bit of guts to that reluctant electric piano, or whatever. The dbx expander is claimed to restore the original dynamic range to recordings compressed during transfer to vinyl. I'd make no such claims but it can be effective on some internally compressed keyboards and also yields interesting results if used with some of the over-compressed heavy rock and heavy metal discs around today!

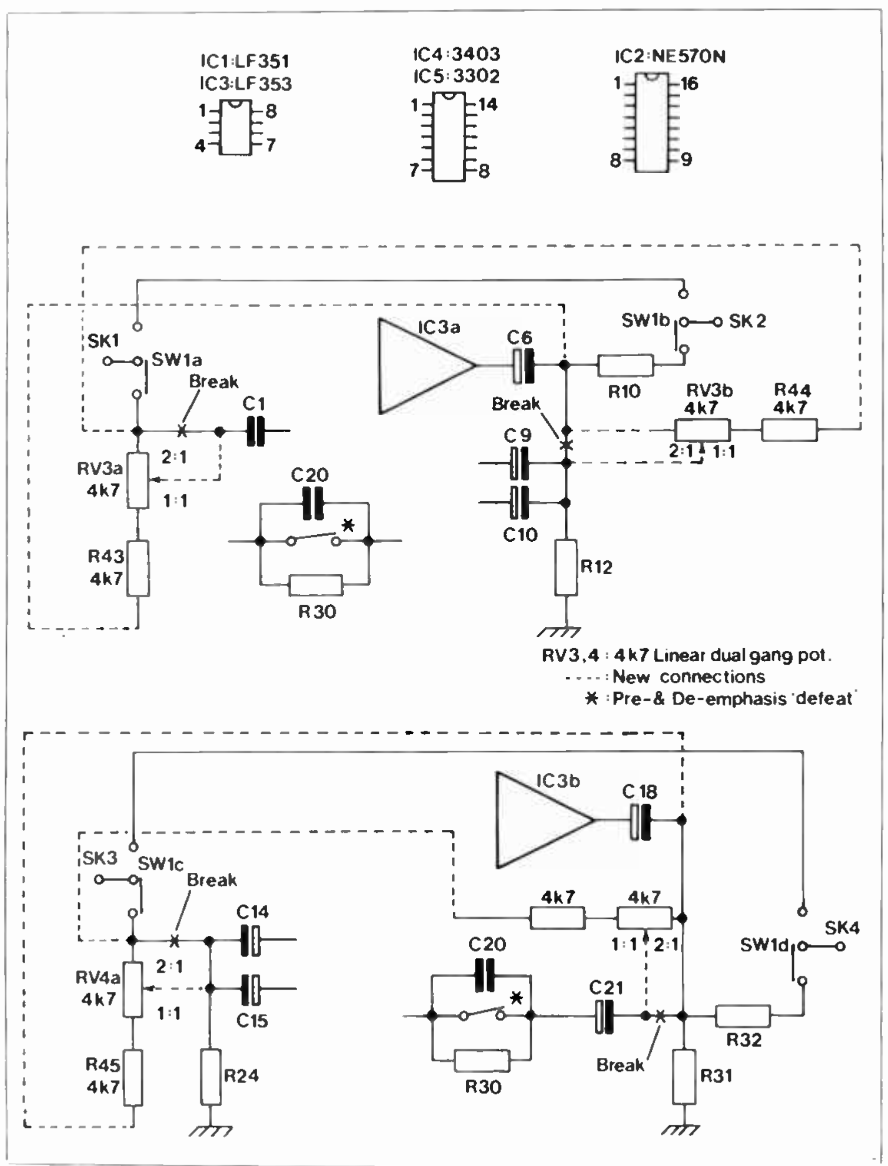


Figure 6. Modifications.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified.

R1	39k	4 off	(M39K)
R2	82k	4 off	(M82K)
R3,4,12,24,30,31	100k	24 off	(M100K)
R5,11,25,29	22k	16 off	(M22K)
R6	150k	4 off	(M150K)
R7,13,14,26, 33,34,41,42	1k0	32 off	(M1K0)
R8,9,27,28	47k	16 off	(M47K)
R10,32	56R	8 off	(M56R)
R15,17,18,20 35,37,38,40	1k8	32 off	(M18)
R16,19,36,39	1M0	16 off	(M1M0)
R21	10k	4 off	(M10K)
R22	4k7	4 off	(M4K7)
R23	510k	4 off	(M510K)
RV1,2	100k vert. s-min. preset	8 off	(WR74R)

Capacitors

C1,2	100nF carbonate	8 off	(WW41U)
C3,5,10,11, 12,13,15,16 21,22,23,24	1u0 50V mini-electrolytic	48 off	(YY31J)
C4,20	3n3 carbonate	8 off	(WW25C)
C6,8,9,14,18, 19,25,26,31,32	10uF 25V mini-electrolytic	34 off	(YY35O)
C7,17	10pF ceramic	8 off	(WX44X)
C27,28	2200uF 25V axial electrolytic	2 off	(FB90X)
C29,30	100nF mini disc ceramic	2 off	(YR75S)

Semiconductors

IC1	LF351	4 off	(WQ30H)
IC2	NE570N	4 off	(QY10L)
IC3	LF353	4 off	(WQ31J)
IC4	3403	4 off	(QH51F)
IC5	3302	4 off	(QH48C)
IC6	uA7815UC		(QL33L)
IC7	uA7915UC		(QL36P)
D1-8	1N4148	32 off	(QL80B)
D9-12	1N4002	4 off	(QL74R)
D13,15	0.2 in. LED, green	8 off	(WL28F)
D14,16,17	0.2 in. LED, red	9 off	(WL27E)

Miscellaneous

	Compander PCB	4 off	(GA30H)
	Power supply PCB		(GA31J)
	DIL socket, 8-pin	8 off	(BL17T)
	DIL socket, 14-pin	8 off	(BL18U)
	DIL socket, 16-pin	4 off	(BL19V)
S1	Push switch, 4-pole	4 off	(FH68Y)
	Switch button	4 off	(BW13P)
S2	DPDT toggle, sub-miniature		(FH04E)
SK1-4	Phono sockets	16 off	(YW06G)
	Chassis fuseholder, 20mm		(RX96E)
FS1	250mA fuse, 20mm		(WR01B)
	8BA ¼in. bolts		(BF08J)
	8BA nuts		(BF19V)
	8BA solder tags		(LR02C)
	Connection wire		(BL09K)
	Stick-on feet	set of four	(FW38R)
	Mains cable 3 amp	3 m	(XR01B)
	Cable grommet		(LR48C)
T1	Transformer 0-240V prim., 0-15V, 0-15V sec., 10VA		(LY03D)

MODIFICATION PARTS LIST

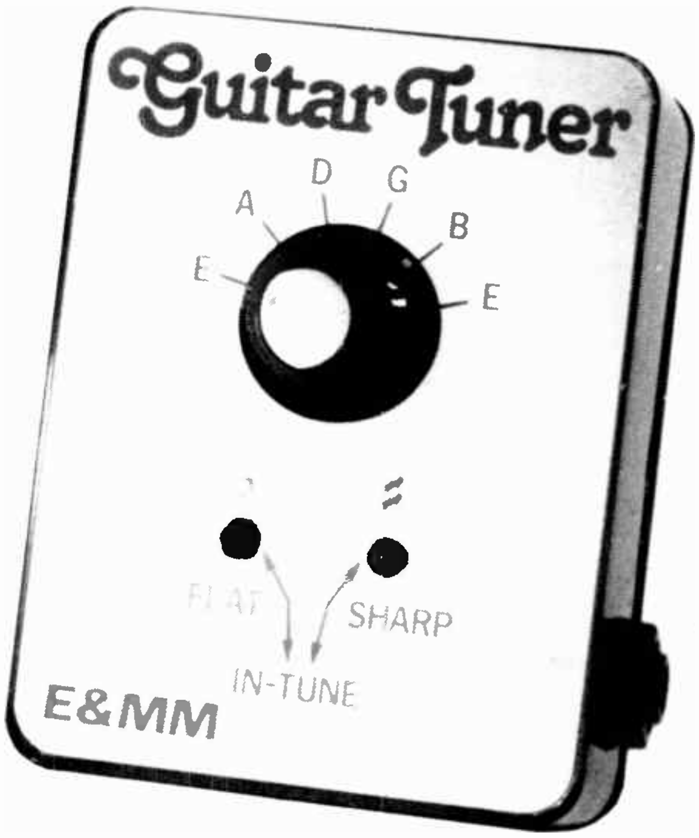
Resistors — all 1% 0.4W metal film unless specified

R43-46	4k7	4 off	(M4K7)
RV3,4	4k7 lin. pot dual gang	2 off	(FW84F)

GUITAR TUNER

by Clive Button

A low cost 'commercial' tuner that gives rapid accurate tuning for electric and acoustic guitars in a new simple way.



Many guitarists, especially beginners, experience difficulty in tuning their instruments. Even when using traditional 'pitch pipes' this can still be a problem if the string is out of tune by more than a semitone or two. I know from teaching basic guitar at evening classes that, with ten or twelve people trying to 'tune up' at the same time, the task is made almost impossible. There are of course electronic guitar tuners available and many are extremely good. Unfortunately, they also carry a rather high price tag. For this reason the Guitar Tuner described here was designed, the objectives being to produce a guitar tuner as quick and simple as possible to use, with a price closer to the old faithful 'pitch pipes' than to its commercially produced, electronic counterpart.

The system is extremely simple, using just two LED's, one for 'sharp' and one for 'flat'. To tune the selected string simply adjust both LED's to the same brightness and you're in tune. The accuracy of the device is quite sufficient to tune your guitar as well as can be done by ear alone, and remains at the set pitch with the battery voltage down as far as 8 volts (the unit being powered by only one

Circuit

Before looking at the circuit operation it is best to consider the requirements of the unit i.e. the frequencies we have to deal with. These are shown, for the six strings of the guitar, in Table 1. By the use of a frequency to voltage converting stage these are changed to DC voltage levels, which can be compared with reference voltages produced from a chain of high stability, close tolerance resistors, which are fed from a stabilised voltage supply. The frequency-to-voltage conversion is achieved by the single LM2917 chip (IC2).

The output of this device follows the expression: $V_{out} = f_{in} \times V_{cc} \times R_{11} \times C_5$ which is a linear relationship between the input frequency (f_{in}) and the output voltage (V_{out}), see Figure 1. Vcc is the internally stabilised voltage, between 7.3 and 7.5V, produced by the internal Zener of IC2, see circuit diagram, Figure 2. From the above expression, values of R11 and C5 are chosen to provide suitable output voltage changes for the guitar frequencies. The calculated voltages are also shown in Table 1.

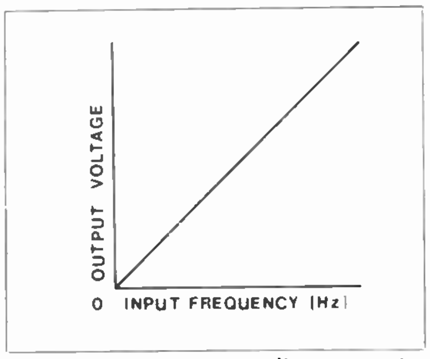


Figure 1. **Frequency-to-voltage converter characteristic.**

Now to the circuit itself. As we only need to deal with frequencies between 82.4 and 329.6Hz, the input is filtered by the high-pass combination of C5, R11, which attenuates frequencies below 72Hz and thus helps eliminate any mains hum pick-up. The low pass combination of C2, R2 attenuates frequencies above 338Hz, reducing the effect of harmonics and spurious noise etc. above this

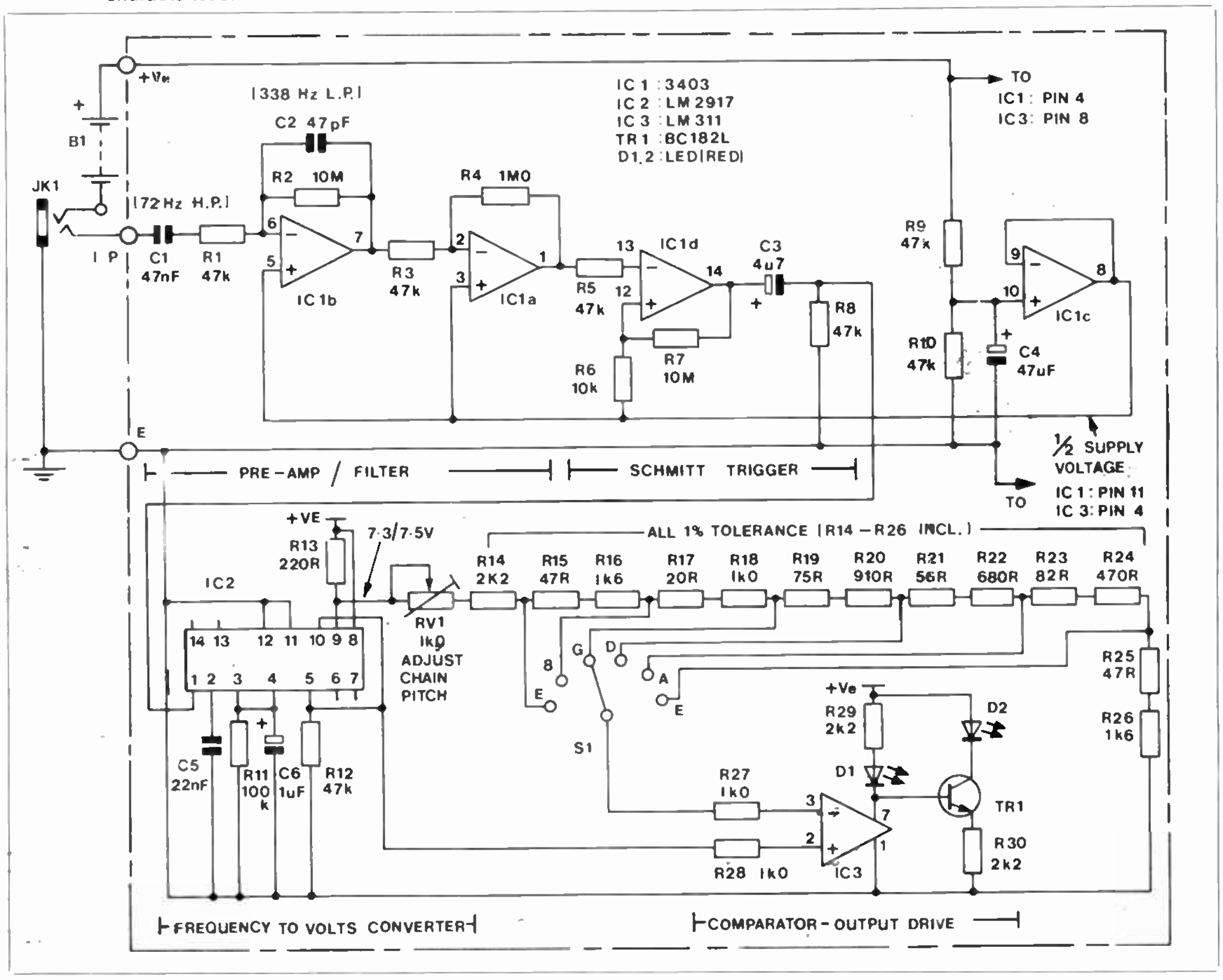


Figure 2. **Circuit diagram for guitar tuner.**

frequency. The chosen values of R1,R2 set the gain of the first stage at 200. This high gain is tolerable because the effects of clipping of the signal are relatively unimportant in this circuit.

The next stage produces a gain of 20, such high gain being required to produce a strong signal even though the decay of the string is fairly rapid. This stage is followed by a Schmitt trigger to give a clean square wave at the input frequency; a square wave that will remain virtually constant until the guitar string has ceased to vibrate. This square wave is coupled to the input of IC2, the frequency-to-voltage converting stage, the fourth op-amp in IC1 being used solely to supply a mid-point voltage for biasing the rest of the circuit.

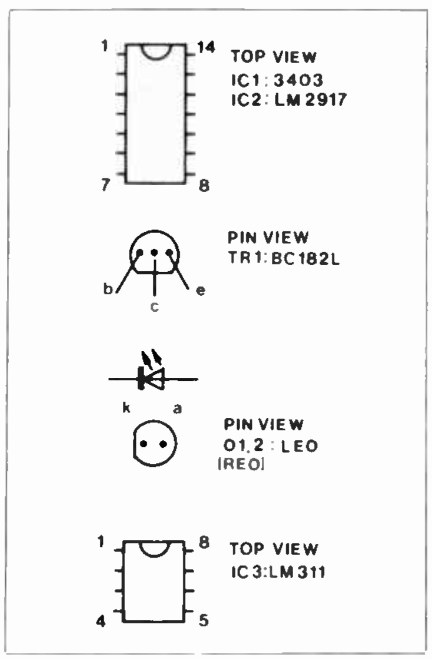
IC2 generates the linear conversion from frequency to voltage which remains linear over a fairly wide temperature range. Use is also made of its internally stabilised voltage supply (pin 9). Any variation in the stabilised voltage, whether caused by supply variation or temperature change, will

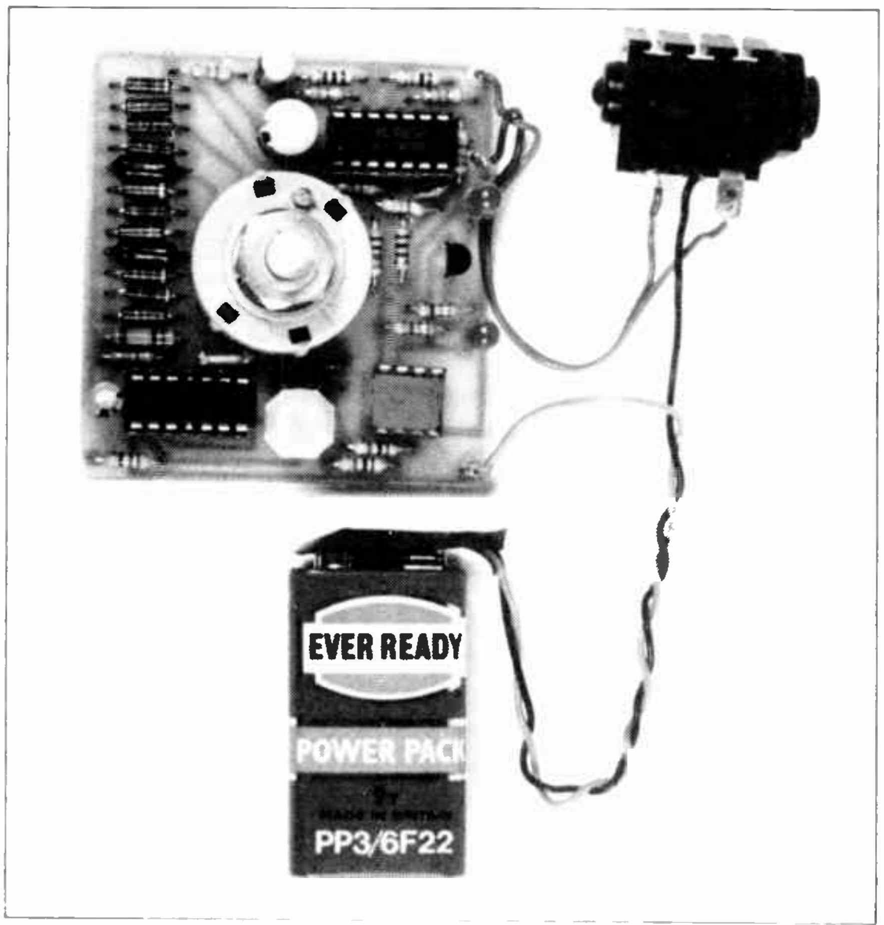
Note	Frequency (Hz)	F-V Output (volts)
E	82.4	1.360
A	110.0	1.815
D	146.8	2.422
G	196.0	3.234
B	246.9	4.074
E	329.6	5.438

Table 1. Calculated values of frequency-to-voltage converter output.

effect the IC's output slope and the reference slope equally, and thus produces no noticeable effect on the unit's performance.

The reference chain comprises a series of resistors, the values of which have been chosen to give a voltage division equal to the intervals between the guitar strings. The resistors must be high stability, close tolerance types, as the accuracy of the





Internal view of guitar tuner.

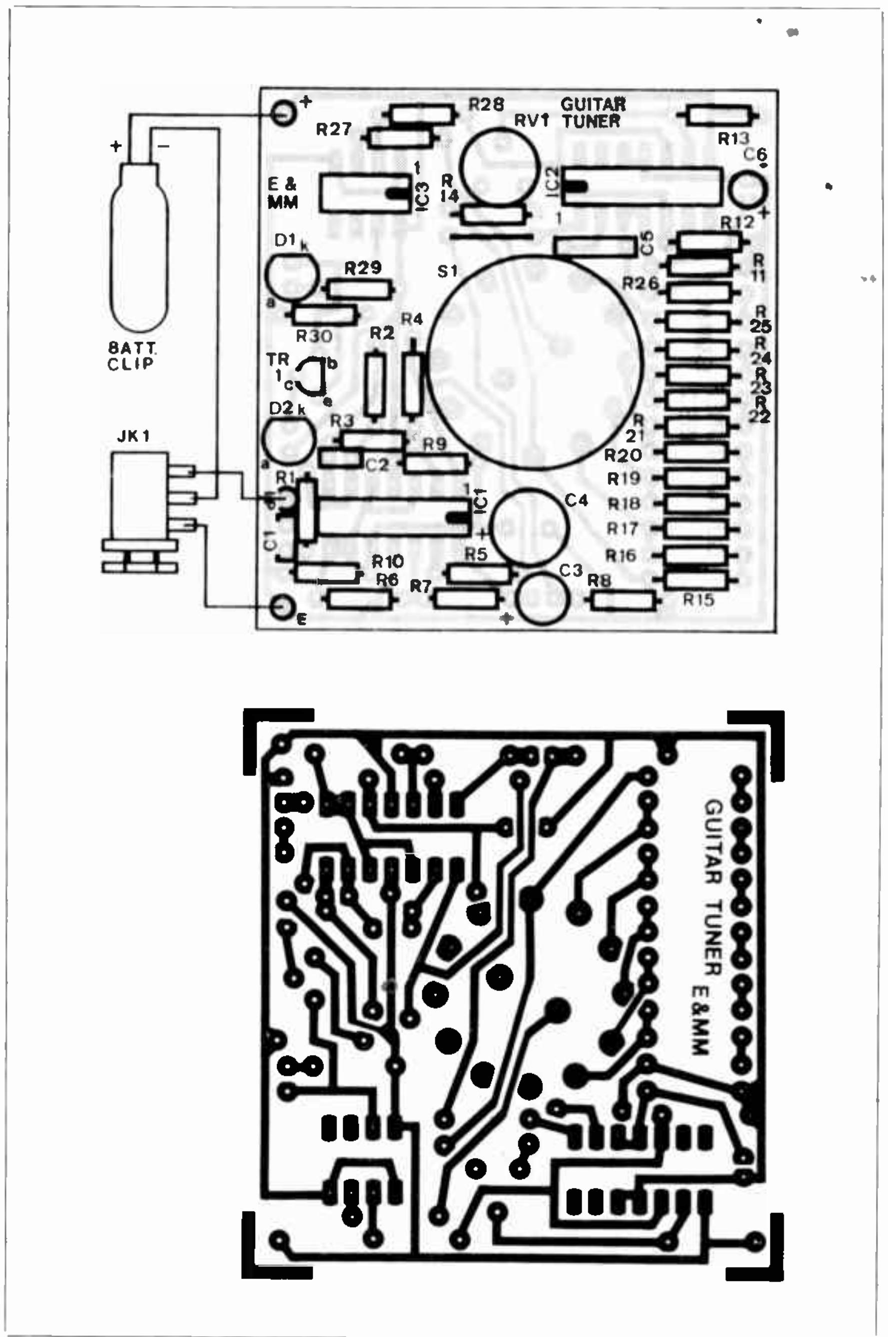


Figure 3. PCB Track layout and component overlay.

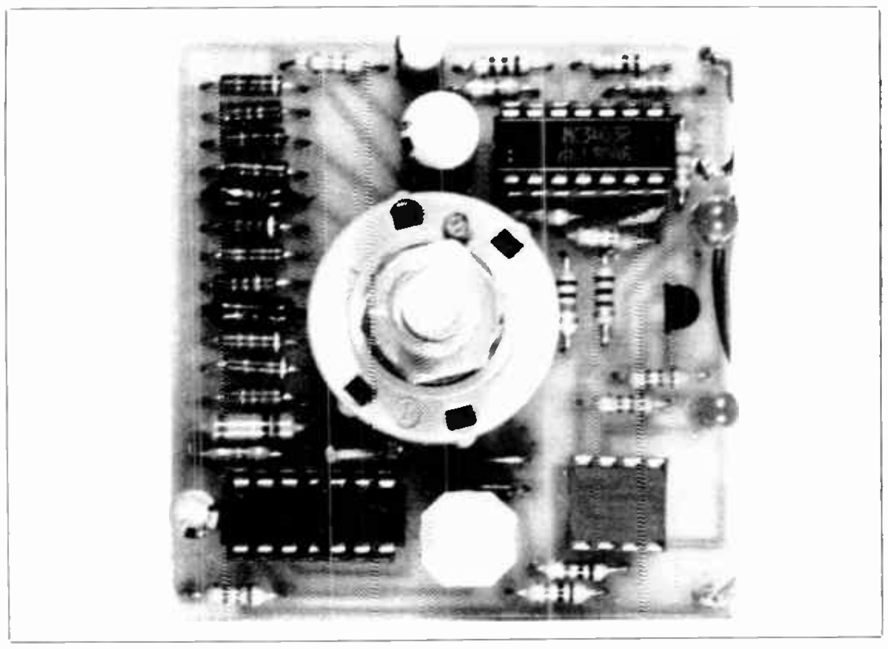
Tuner is entirely dependent upon the accuracy of the reference chain. Most of the resistance values in the chain are fairly small and any slight variations from their actual values will not significantly affect the overall accuracy.

There are however two 1k6 resistors (R16, R26) in the chain; 1% resistors could vary by as much as ±16 ohms from their required values (though they probably won't vary by more than 2 or 3 ohms) and this would adversely affect the accuracy, so it is preferable to select these, or at least check their actual values before connecting them in circuit. The voltages derived from this chain are selected by the six-way switch (S1) and compared with the voltage produced by IC2. This comparison is performed by IC3, with its own output directly driving D1, the LED which indicates that the note is flat. The 'sharp' LED (D2) is driven by a separate transistor (TR1).

Construction

Figure 3 shows the PCB and component overlay. Fit all the resistors to the PCB. Next, fit the capacitors, observing the polarity of the electrolytic types. The preset (RV1) and the three IC's can now be fitted, noting the position of pin 1 in each case. The LEDs stand up above the PCB, the base of each LED body being ½in. (13mm) above the board. It may help here to cut two pieces of insulation from some suitable gauge wire, each half an inch in length, and slide these over the legs of the LEDs to give the correct lead lengths. LED polarity is indicated by a flat on the body, next to the cathode connection, and this must be carefully observed. The last thing to fit to the PCB is the switch and for this the connecting lugs have to be cut off the switch contact tags, just below the circular end-pieces to allow them to be inserted through the board and soldered into position. The switch body then serves as the PCB support when it is finally fitted into the box.

Mark out and drill the holes in the box section, as shown in Figure 4. If this is done accurately, the switch bush and LEDs should fit neatly through the box front. Fit leads, about two inches in length, from the jack-socket to the board input connections and bend the tags of the jack-socket flat against its body. Connect the battery negative lead to the centre contact of the jack-socket and the positive lead to the board connection. With the battery connected,



and before fitting the board into the box, the unit can be set-up as detailed below. Finally the unit can be completed by fitting the board, jack-socket and battery into the box, as shown in Figure 5.

Even though the unit will remain at the set pitch for battery voltages as low as 8V, it is advisable to use the Duracell type battery, as its full voltage is maintained for 90% of its useful life, thus ensuring good results for some considerable time. An added advantage is, should you run it flat, it will not leak.

Setting-up

After construction the unit will have to be adjusted to concert pitch i.e. A = 440Hz. This has to be done using a pitch source of known accuracy, be it a keyboard, audio-oscillator or whatever. It is best to set the chain from the top 'E' by selecting this on the switch, injecting the note of E (329.6Hz) at the input and adjusting the preset until both LEDs are at the same brightness, which will automatically ensure that all six notes are set at the correct pitch.

An added feature can be incorporated by bringing the preset control out to the front panel. The unit could then be used for tuning to pitches other than A = 440Hz, which would be useful if you are playing on a keyboard that is slightly sharp or flat. This would be done by feeding in a note from the keyboard, adjusting the tuner to this and in turn using it to tune your guitar.

Operation

As explained earlier, for correct tuning of the selected string both LEDs will appear to be at the same brightness level. We now want to find the best way of doing this. First, when the guitar lead is plugged into the unit, and before a string is struck, the LED indicating string 'flat' will be illuminated. Now select the appropriate string note (e.g. bottom E) on the switch and, with the guitar's volume control at maximum, strike the string; then either LED may light, showing 'flat' or 'sharp', (this will be true however far the string is cut of tune). If the 'flat' LED stays lit, then wind the string up as quickly as you like until just the 'sharp' LED is illuminated indicating that the required note has just been passed.

Now detune the string (i.e. 'flat' LED lights) and gradually bring the string back up until the two LEDs come to the same brightness; the string is now in tune. It is always better to come up to the required note as this makes sure that any slack in the gearing of the machine-head is taken up and avoids any chance of the string slipping down out of tune again.

No doubt while carrying out the above operation the string will probably have to be picked more than once to maintain an input to the Tuner. This is more applicable to the higher pitched strings, as their vibrations die away more quickly, although once used to using the unit, you will find that all strings can be brought into tune with just one pick. On the initial pick of the string you will probably notice that the sharp LED will first light momentarily and then go out. This is due to the rich harmonic content of a plucked string and can be minimised by picking the string on or around the twelth fret, thus ensuring a strong fundamental vibration along the string's length, giving a quicker and longer-lasting reading on the LEDs. Incidentally, this is also true when using any other guitar tuning device.

It is not necessarily idea' to have a strong signal output from

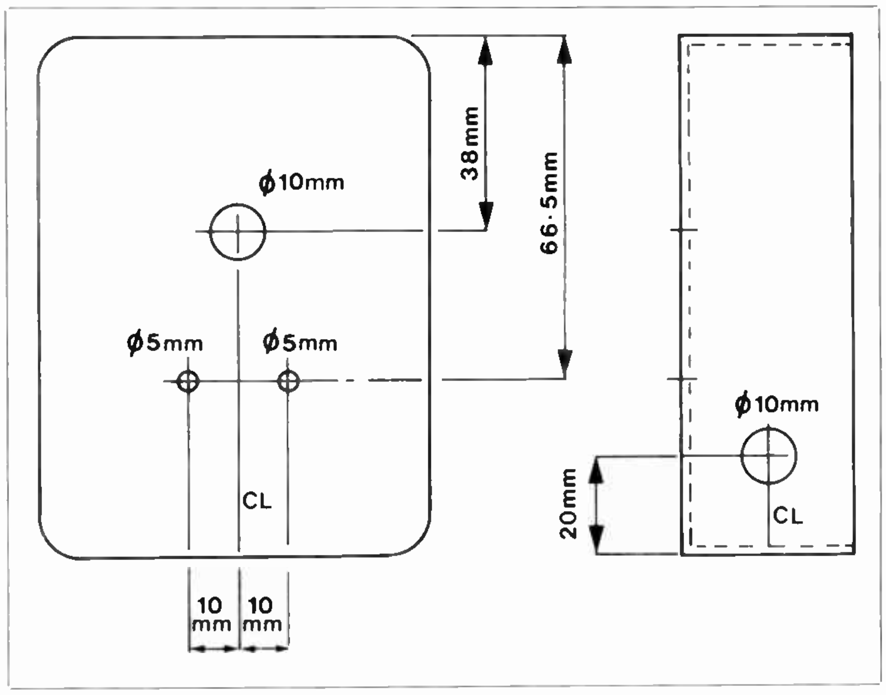


Figure 4. Drilling details.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified

R1,R3,R5,R8 R9,R10,R12	47k	7 off	(M47K)
R27,R28	1k0	2 off	(M1K)
R29,R30	2k2	2 off	(M2K2)
R2,R7	10M	2 off	(M10M)
R6	10k		(M10K)
R4	1M0		(M1M)
R13	220R		(M220R)
R11	100k		(M100K)
R16,R26	1k6	2 off	(M1K6)
R15,R25	47R	2 off	(M47R)
R14	2k2		(M2K2)
R17	20R		(M20R)
R18	1k0		(M1K0)
R19	75R		(M75R)
R20	910R		(M910R)
R21	56R		(M56R)
R22	680R		(M680R)
R23	82R		(M82R)
R24	470R		(M470R)
RV1	1k Cermet, preset		(WR40T)
Capacitors			
C1	47nF Minidisc		(YR74R)
C2	47pF ceramic		(WX52G)
C3	4u7F 63V PC elect.		(FF03D)
C4	47uF 25V PC elect.		(FF08J)
C5	22nF Carbonate		(WW33L)
C6	1uF 100V PC elect.		(FF01B)
Semiconductors			
TR1	BC182L		(QB55K)
D1,D2	LED, red	2 off	(WL27E)
IC1	3403		(QH51F)
IC2	LM2917		(WQ38R)
IC3	LM311		(QY09K)
Miscellaneous			
JK1	Jack socket stereo, plastic type		(HF92A)
S1	Rotary switch, SW6B 2 pole 6-way		(FF74R)
	Knob M3		(RW90X)
	PP3 Clip		(HF28F)
	ABS Box MB2		(LH21X)
	Printed circuit board		(GA24B)
	Front Panel		(YL26D)
	14-Pin DIL socket	2 off	(BL18U)
	8-Pin DIL socket		(BL17T)

Note. A complete kit (LW90L) is available from Maplin Electronic Supplies Ltd at a price of only £10.75 inc. VAT and P&P.

your guitar pick-up. Some guitars will give a clearer indication on the LEDs for several seconds when the string is hardly audible. An added advantage is that you will quickly detect an old string, that will be prone to slipping out of tune, by the Tuner's inability to hold the brightness of both LEDs without wavering.

It is also possible to use the Tuner for correct pitching of acoustic guitar strings, by plugging a microphone into its input. However, a small pre-amp may be needed to boost the microphone output in order to obtain a sufficiently good reading.

HARMONY GENERATOR

★ **Fixed accurate harmonies, unison, 3rd and 5th**
★ **Pitch shift up or down 3 octaves**
★ **Ideal accessory for the synthesist**
★ **Powered by a single PP3 battery**

by Paul Williams

KIT COST ONLY £15.45

Harmonisers are beginning to attract much attention from musicians, particularly for use in live performances where they can 'thicken up' the sound tremendously. Most musicians, however, cannot savour the delights of the harmoniser due to its very high cost. The only pitch change device within the price range of the average musician is the octave divider type of accessory used by guitarists. Between these two devices there appears to be a void.

The E&MM Harmony Generator is intended to fill this void, being a compromise between the simplicity of the octave divider and the versatility of the harmoniser. The Harmony Generator can give up to three octaves of pitch shift, up or down, including individually selectable intervals of '3rd' and '5th' harmonics. The pitch shifts are digitally derived and are thus very stable, obviating the need for precise setting-up and pitch shift adjustments during a performance. The Harmony Generator can, however, only accept monophonic signals from a source such as a mono synthesiser. Indeed this is an ideal device for use with a single VCO synthesiser, greatly extending its versatility.

The Harmony Generator will not only follow the pitch of the instrument, but also the amplitude, applying the same amplitude envelope to the harmony signal as that of the instrument. A mixer is provided so that the contrast between the instrument and harmony signals can be optimised.

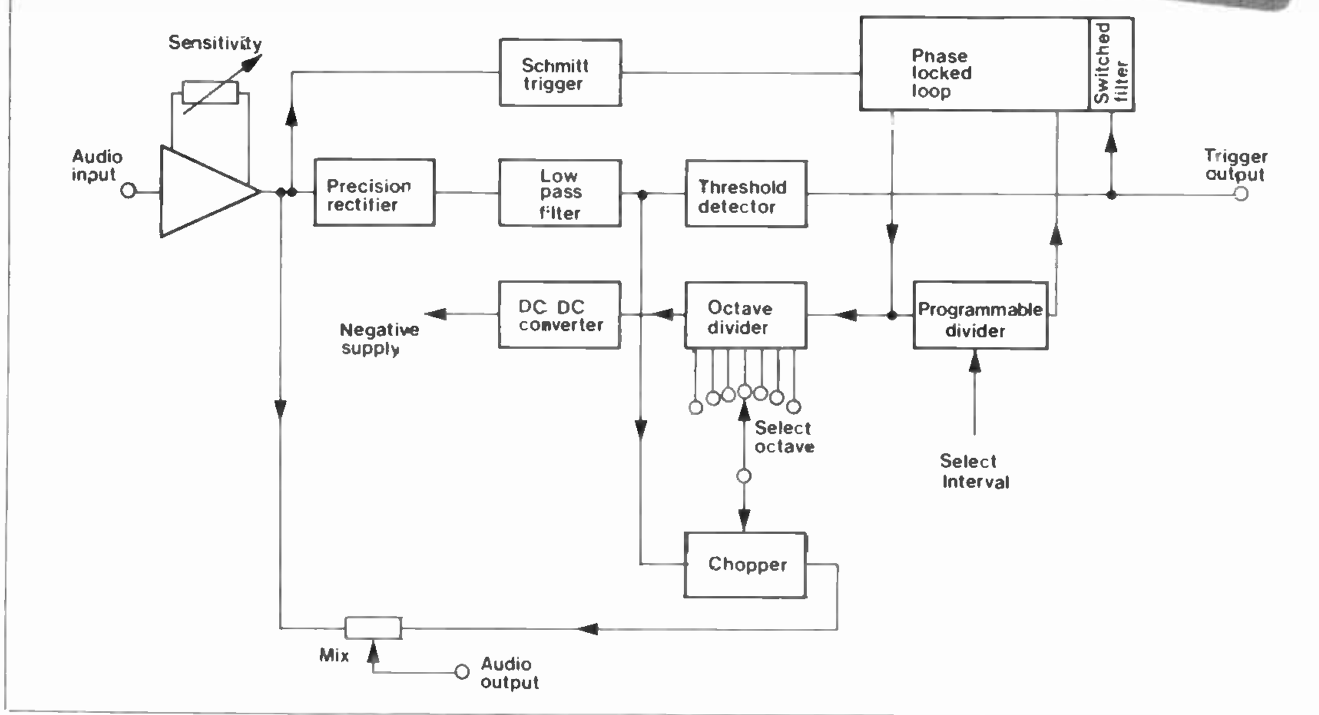


Figure 1. Harmony Generator block diagram.

Design Principles

The block diagram, Figure 1 shows the basic functional circuit elements. The heart of the unit is the phase-locked loop (PLL), which allows frequency multiplication to be performed. The PLL effectively compares the input frequency with that fed back from the output of its internal voltage controlled oscillator (VCO). It will adjust the voltage applied to the control input of the VCO, and hence alter its frequency until the two frequencies are the same. Since the programmable divider is in the feedback path of the PLL, the VCO output frequency will be a multiple of the input frequency. The ratio between them being the same as the division ratio of the programmable divider, which is selectable. The octave divider successively divides the frequency from the VCO output providing several outputs at one octave intervals below the VCO output frequency.

The nominal division ratio of both the programmable divider and octave divider is 64, thus for unison harmony, the VCO will operate at 64 times the input frequency. To achieve the interval '3rd', the pitch must be shifted up by 4 semitones, which is an increase in frequency of $(\sqrt[12]{2})^4=1.26$, close to $^{81}/_{64}$. Thus for the interval '3rd' the programmable divider must be set to $\div$ 81. Similarly for the interval '5th', a 7 semitone shift is needed, or $(\sqrt[12]{2})^7=1.5$, or $^{96}/_{64}$. Thus the programmable divider must be set to $\div$ 96

The precision rectifier provides a DC voltage which follows the amplitude envelope of the input signal. The threshold detector senses when the signal is of sufficient amplitude to reliably operate the Schmitt trigger in the PLL input line. Also at this signal level the trigger output is activated for the operation of external envelope generators, synthesisers etc. Below this threshold level, the PLL filter is switched in such a way as to 'freeze' the operating frequency, enabling the note to decay normally, even when the input signal level falls below the Schmitt trigger threshold.

Another DC output is taken from the precision rectifier which

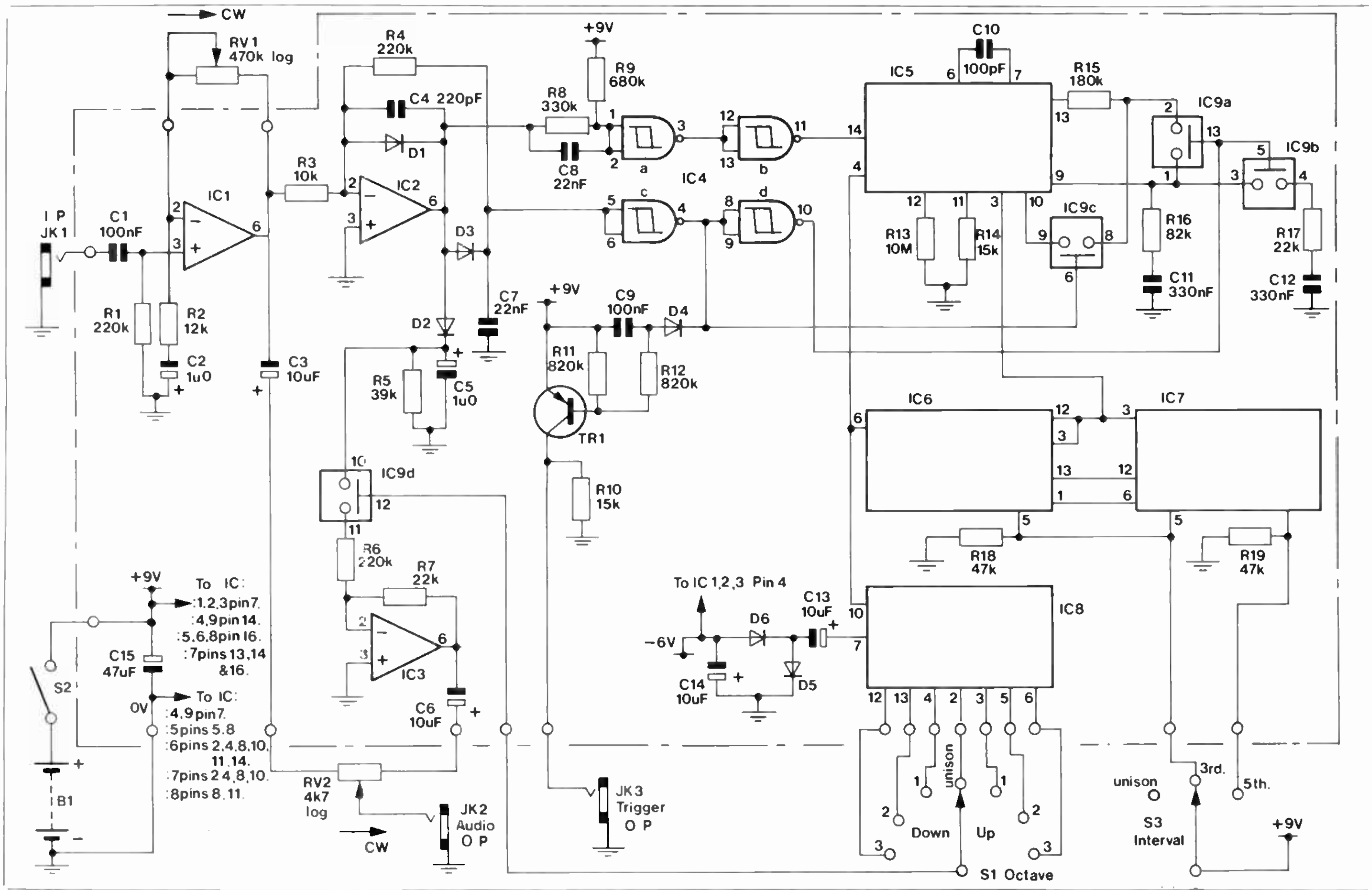


Figure 2. Harmony Generator circuit diagram

is chopped at the frequency selected from the octave divider. This AC signal which has the amplitude of the input signal and the frequency of the octave divider output selected, is the harmony signal. It takes the form of a square wave and is mixed with the input signal to become the combined output.

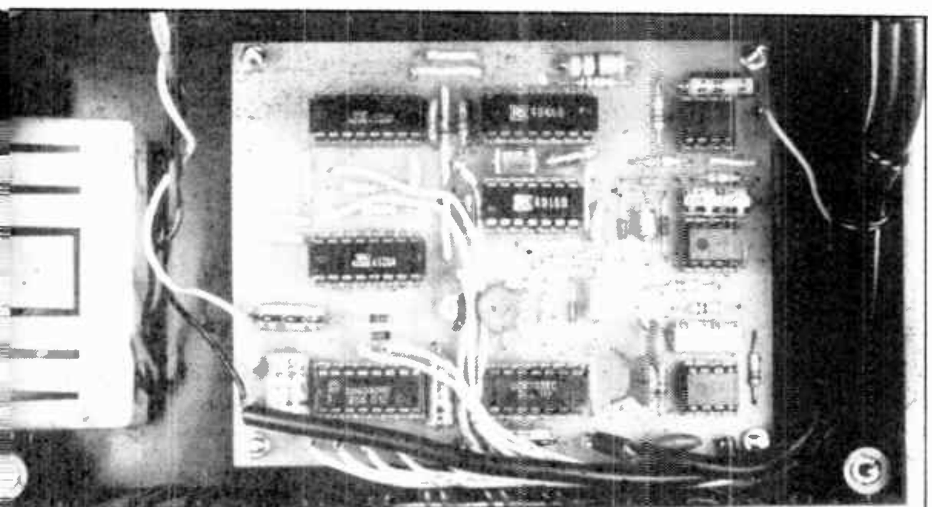
Circuit

IC1 forms a conventional non-inverting AC amplifier, its gain being adjusted by RV1, allowing an input signal range of 5mV to 200mV RMS to be accepted. IC4a, b form the Schmitt trigger which 'cleans up' the signal from the precision rectifier, IC2, to the PLL, IC5. C4 provides some attenuation of the higher frequency harmonics of the input signal.

The threshold detector IC4c, d controls the switches IC9a, c, which cause the PLL to freeze by disconnecting the voltage control source, pin 13, from the VCO input, pin 9. IC9b helps to keep the VCO frequency constant during a decaying note by disconnecting the short time constant filter R17, C12, leaving the longer time constant R16, C11 in circuit when the input signal amplitude becomes low. At higher signal levels, IC4 pin 4 will go low, causing TR1 to switch on to provide a positive-going trigger output.

IC6 & 7 comprise the programmable divider. Its division ratio is determined by the conditions on the programming inputs, selected by S3. A binary divider IC8 forms the octave divider, which again provides a division ratio of 64, with other outputs of divide by 8, 16, 32, 128, 256 or 512 selectable via S1. The selected signal controls the chopper IC9d, IC3. RV2 then mixes the chopped DC voltage from the precision rectifier with the untreated instrument signal from the output of IC1. The combined output signal at the wiper of RV2 has a typical RMS amplitude of 200mV.

An output from IC8 is used in conjunction with C13, 14, D5 and 6 to form a DC/DC converter, providing a negative supply voltage for the operational amplifiers. R13 ensures that the PLL VCO free-running frequency, and hence the frequency at pin 7 of IC8, is sufficiently high to allow the DC/DC converter to operate correctly. Since the circuit is CMOS based, the complete unit takes very little power; approximately 4mA from a PP3 battery.



Internal view of Harmony Generator.

Construction

All the components are contained on a single PCB, the track and component layout of which is shown in Figure 3. Assembly should commence with the resistors, diodes and capacitors, taking care with the polarisation of diodes and electrolytic capacitors, according to the component layout shown. Next solder in the IC sockets for all the CMOS ICs (IC4-9), but do not insert the ICs at this stage. The operational amplifiers IC1, 2 and 3 can be soldered directly into the PCB, along with the transistor TR1, again taking care with orientation. When assembly of the PCB is complete, the CMOS ICs may be inserted into their appropriate sockets, exercising the usual anti-static precautions.

Before mounting the jack-sockets, pots and switches on the front panel, adjust the rotary switch end-stops by inserting the tab of the washer into the appropriate end-stop hole; 7 for S1 and 3 for S3. Wire the sockets, pots, switches and the battery holder to the PCB according to the wiring diagram Figure 3. Use screened cable for the connections to the input socket, JK1, and sensitivity control, RV1. The screen of the cable to RV1 does not go to 0V but is used as the through connection from IC1 pin 6 to RV1.

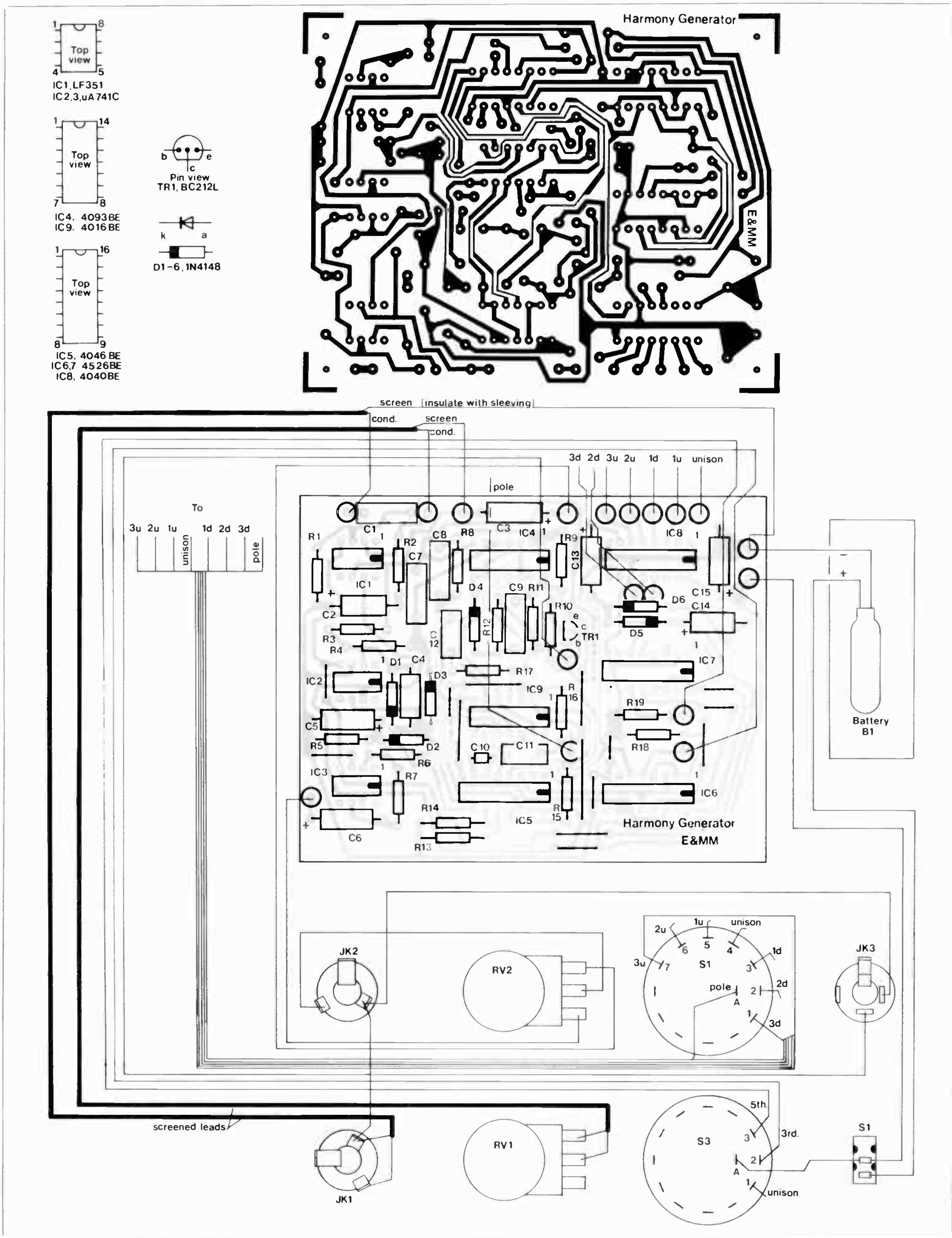


Figure 3. PCB track, component layout and wiring diagram.

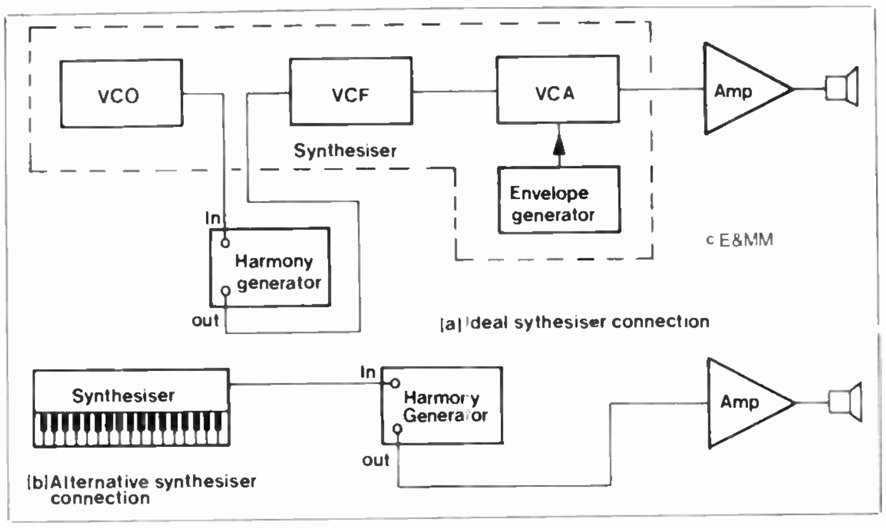


Figure 4. Harmony Generator/synthesiser connections.

Operation

There are no presets to adjust in this project; just connect the Harmony Generator's input to your synthesiser and the output to your monitor amplifier. Switch the unit on and set the controls initially as follows: sensitivity anticlockwise; mix clockwise; interval and octave as required. Play some notes on the synthesiser and advance the sensitivity control until the Harmony Generator just starts to lock on to and follow the frequency of the synthesiser. The mix control can then be adjusted to give the desired contrast between instrument and harmony signals. The Harmony Generator is now ready for use.

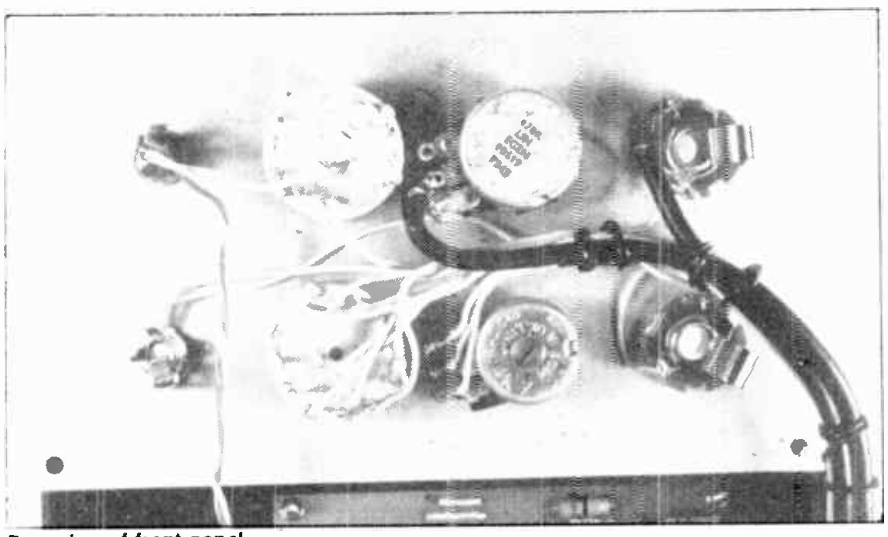
It is as well to understand the few limitations of this device so that the best use can be made of it. Only a single note at a time can be handled; it is not capable of dealing with polyphonic signals. Since high amplitude harmonics can also cause instability, high VCF Q values should be avoided. This applies also to the use of audio frequency FM, although the unit can cope with vibrato type FM. AM does not present quite such a serious problem, as long as the modulation depth is not too great. If a slow attack is used, then a pitch jump might be noticed as the amplitude passes through the threshold level. This effect can be reduced by advancing the sensitivity control.

The fact that the harmony signal is a square wave does not seem to be too much of a disadvantage since the mixed instrument signal gives the overall sound sufficient character. This can be further improved by putting all treatments such as phaser, flanger, reverb, echo etc, after the Harmony Generator.

Most of the above problems can be avoided altogether by using the ideal synthesiser connection scheme shown in Figure 4a. Here, the Harmony Generator is fed directly from the synthesiser VCO. The VCF Q setting will then make no difference to the stability. Also, the VCF will filter the Harmony Generator square wave, giving it extra character. The slow attack pitch jump problem also disappears since the amplitude envelope is applied after the Harmony Generator. If, however, you do not have access to these connections on your synthesiser, then you will have to settle for the connection scheme shown in Figure 5b.

Although this project was designed primarily for use with a synthesiser, there is no reason why the circuit should not be used, or adapted for use with other instruments such as guitars (Figure 5), brass and reed instruments or even vocals. The main criterion to be satisfied is to attenuate any high amplitude harmonics. This might be achieved by altering the value of C4, or more ideally by preceding the Harmony Generator by a sharp cut-off low-pass filter. With the circuit in its present form, bass instruments cannot be used since the response to pitch change becomes very slow at low frequencies. This could be improved somewhat by increasing the value of C7.

Once the synthesist has become familiar with this unit, he will find it an invaluable addition to his accessory collection.



Rear view of front panel.

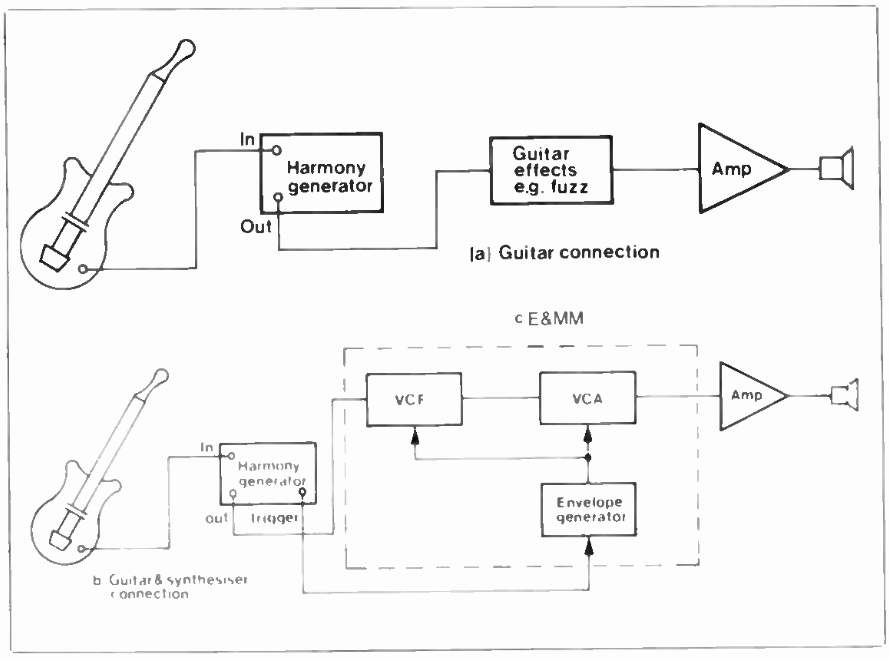


Figure 5. Harmony Generator/guitar connections.

PARTS LIST FOR HARMONY GENERATOR

Resistors — all 1% 0.4W metal film unless specified			
R1,4,6	220k	3 off	(M220K)
R2	12k		(M12K)
R3	10k		(M10K)
R5	39k		(M39K)
R7,17	22k	2 off	(M22K)
R8	330k		(M330K)
R9	680k		(M680K)
R10,14	15k	2 off	(M15K)
R11,12	820k	2 off	(M820K)
R13	10M		(M10M)
R15	180k		(M180K)
R16	82k		(M82K)
R18,19	47k	2 off	(M47K)
RV1	470k log pot		(FW27E)
RV2	4k7 lin pot		(FW01B)
Capacitors			
C1,9	100n ceramic	2 off	(BX03D)
C2,5	1u 63V axial electrolytic	2 off	(FB12N)
C3,6,13,14	10u 25V axial electrolytic	4 off	(FB22Y)
C4	220p polystyrene		(BX30H)
C7,8	22n ceramic	2 off	(BX01B)
C10	100p metalised ceramic		(WX56L)
C11,12	330n polycarbonate	2 off	(WW47B)
C15	47u 10V axial electrolytic		(FB38R)
Semiconductors			
D1-6	1N4148	6 off	(QL80B)
TR1	BC212L		(QB60Q)
IC1	LF351		(WQ30H)
IC2,3	uA741C	2 off	(QL22Y)
IC4	4093BE		(QW53H)
IC5	4046BE		(QW32K)
IC6,7	4526BE	2 off	(QQ44X)
IC8	4040BE		(QW27E)
IC9	4016BE		(QX08J)
Miscellaneous			
JK1,2	Jack socket	2 off	(HF91Y)
JK3	3.5mm socket		(HF82D)
S1,3	12-way rotary switch	2 off	(FF73Q)
S2	Ultra min toggle switch		(FH97F)
	14-pin DIL socket	2 off	(BL18U)
	16-pin DIL socket	4 off	(BL19V)
	Box M4005		(WY02C)
	PCB		(GA48C)
	Knob KB4	4 off	(RW87U)
	PP3 battery holder		(XX33L)
	PP3 battery		

Note A complete kit (LW91Y) of all the parts listed is available from Maplin ELectronic Supplies Ltd for a price of just £15.45 inc. VAT and P&P.

HEXADRUM

Sounds of the future are as simple to make as drumming your fingers on this novel instrument!

- ★ **Six electronic bongo tones**
- ★ **Touch sensitive**
- ★ **Battery powered**
- ★ **Ideal for home, education and studio**

KIT COST ONLY £18.95 inc. VAT & P&P

RECORDED ON CASSETTE NO 3

Until recently, many electronic percussion instruments have been little more than impact-triggered synthesisers of varying complexity, all subject to the disadvantage that the nature of the shock causing their sound in no way affects the character of the sound produced. To vary the loudness or timbre of the sound, some control must be adjusted, an action not exactly compatible with the fluent playing of percussion.

Hexadrum is a different kind of electronic percussion instrument: it is touch sensitive. The six sensors are arranged to be beneath the fingertips of a comfortably placed hand and are played by simply tapping with the fingertips. A harder tap produces a louder sound; a tap with an object harder than a fingertip produces a sharper sound. Any number of sensors may be struck at any time to produce a composite sound. The only electronic control is to set the overall signal level output, in other words, a volume control.

When played through an amplifier and speaker system designed to give faithful reproduction of audio, the sounds of Hexadrum are best described as similar to bongoes, though the lower range drums are of a lower range than normally encountered in bongoes and more like a bass drum. Like all other electronic instruments, Hexadrum may be played through any special effects unit, such as reverberation, echo, phaser, flanger or synthesiser external input, to obtain a different sound.

Its use is not restricted to trained percussionists, for the 'hand' layout virtually gives all 'finger-tappers' opportunity to experiment with rhythms. The potential of this low cost instrument makes it ideal for the music room - be it in school, home or studio. The touch sensitive pads give the Hexadrum some of that creative dynamic feel of the skin drum. Being battery powered, the unit can be connected via guitar coiled cable to group amplifier for on-stage performance. In the home or classroom, the output plugs direct into tape, mic or line inputs of your stereo (or mono) unit.

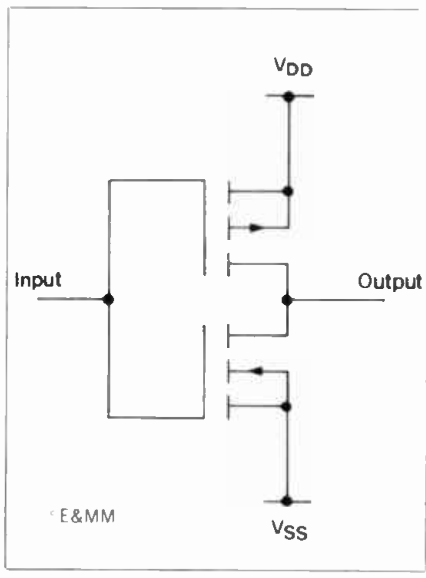


Figure 1. Output stage of 4069 inverter.

Piezo-ceramic Pick-ups

The touch sensors employed in Hexadrum are piezo-ceramic wafers deposited on thin brass plates about the size of twopenny pieces. Striking such a brass plate causes it to vibrate, distorting the ceramic layer attached to it and creating a small piezoelectric potential across the ceramic. This voltage, of the order of millivolts, is picked off from the brass backplate and another very thin electrode deposited on top of the ceramic.

As the voltage across the ceramic is proportional to its distortion, the signal produced by striking the brass plate contains all the frequencies of vibration of the plate. Amplified and reproduced in its raw form, this signal sounds (not surprisingly) like the sound actually made by the brass plate when struck: a short burst of almost white noise with a sharp attack and relatively rather longer decay. Though the plates do resonate, this is at several kilohertz, as they are far too small and rigid to mimic exactly the vibrations of the skin of a drum. However, the envelope of the sound is similar to that of a drum, and contained within the noise are the frequencies required to simulate the timbral qualities of a drum

Filters, logically!

The way the sound of Hexadrum is extracted from the noise signal produced by the piezo transducers is, of course, by filtering. To implement six large-gain filter preamplifiers as cheaply as possible, a single CMOS hex-inverter pack is used. Placing DC feedback around each inverter converts the digital gates to simple large-gain, analogue, inverting amplifiers. Since the output from the amplifiers is not intended to be a faithful, amplified reproduction of their input signal, the limitations of the gates as amplifiers are acceptable in this application.

A problem associated with the novel use of CMOS logic to perform analogue functions occurs as a result of the transfer characteristics and the biasing, or rather the lack of biasing, of the FETs making up the device used. The output section of each inverter is shown in Figure 1. The gates of the two FETs are also DC coupled to the input of the inverter. When an input voltage within the noise margin of the inverter, a band between the two supplies, is applied to the gate both FETs will be partially on together. Such a condition exists when the inverter is included in a feedback loop. The result is that a quiescent current that is substantially larger than quoted for the pack flows through the FETs. This mode of operation is analagous to amplifier class A operation.

To minimise the FET bias current, the inverter pack is run from a 5.6V supply provided by a Zener. This reduces its quiescent current from about 20mA at 9V to 5mA compared to a manufacturers quoted value of 10nA!

The circuit diagrams of the Hexadrum and the component values for each drum are shown in Figure 2.

The frequency response of the filters is bandpass, with a 12dB per octave cutoff except in the case of the lowest range drum in which the roll-off is not so steep. RV11-61 provide some control of

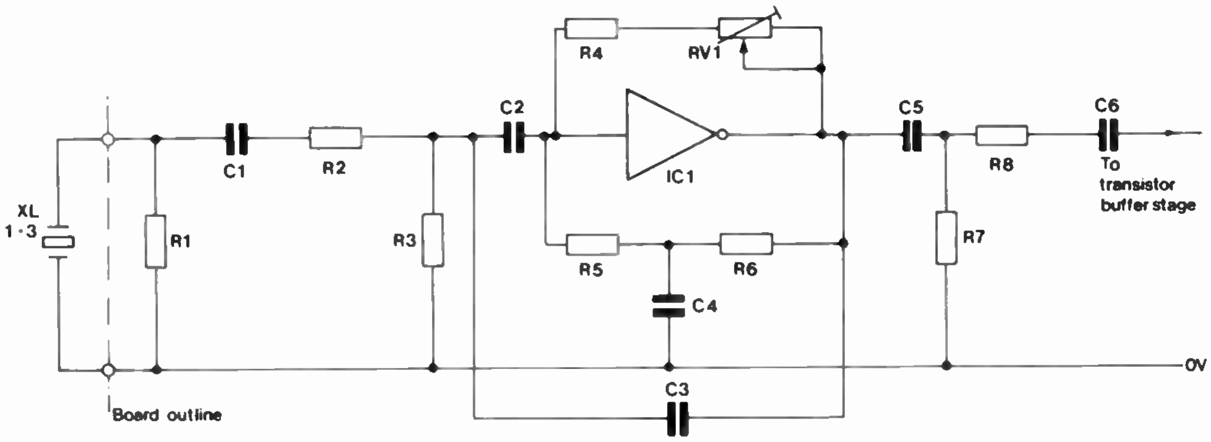


Figure 2(a). Circuit of upper range drums.

Component ref.	R1	R2	R3	R4	R5	R6	R7	R8	C1	C2	C3	C4	C5	C6
Drum 1	180k	68k	10k	180k	68k	68k	180k	150k	15nF	4n7	4n7	12nF	12nF	1n5
Drum 2	Values as for drum 1								27nF	10nF	10nF	27nF	27nF	1n5
Drum 3								470k	39nF	15nF	15nF	39nF	39nF	4n7

Figure 2(b). Resistor and capacitor values for Figure 2(a).

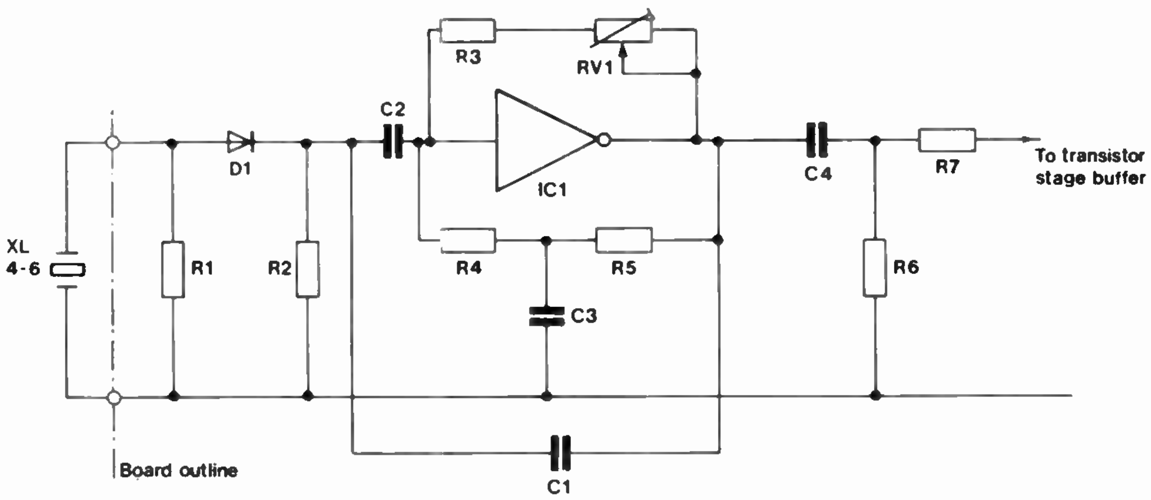


Figure 2(c). Circuit for lower-range drums.

Component ref	R1	R2	R3	R4	R5	R6	R7	C1	C2	C3	C4
Drum 4	180k	10k	180k	68k	68k	180k	680k	22nF	22nF	68nF	68nF
Drum 5	Values as for drum 4						470k	33nF	100nF	100nF	150nF
Drum 6				Not used			470k	47nF	47nF	Not used	150nF

Figure 2(d). Resistor and capacitor values for Figure 2(c).

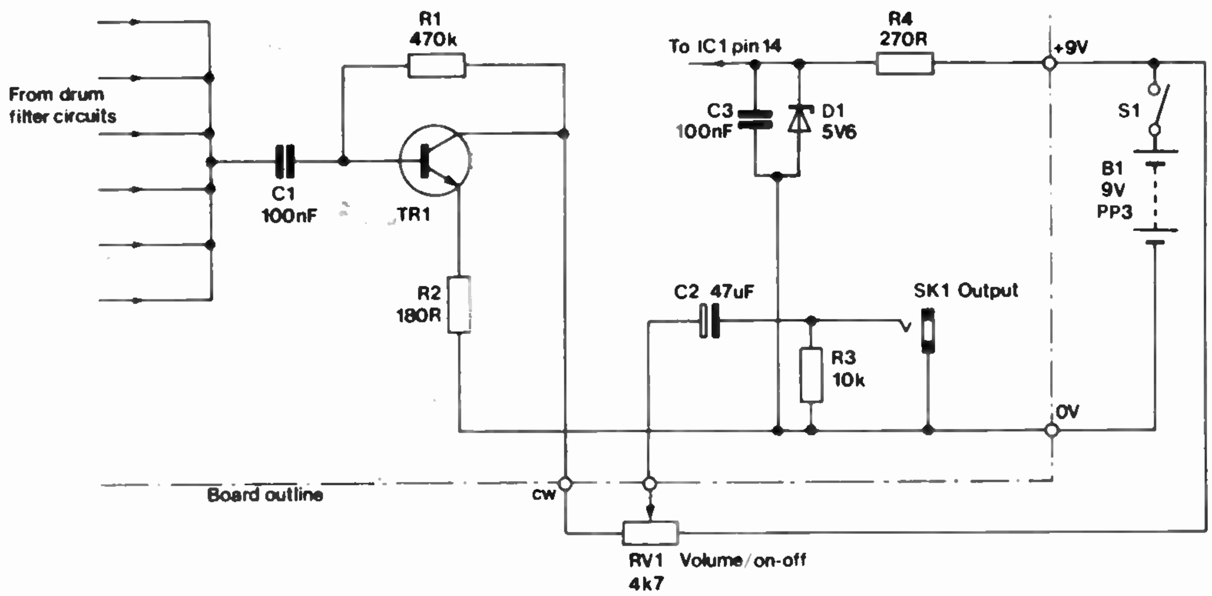


Figure 2(e). Circuit of transistor buffer stage and supply wiring.

the resonance of each filter. At their resonant frequencies, the filters have a gain of about 40dB. The frequencies covered by the drums encompass several octaves, from 40Hz up to 5kHz.

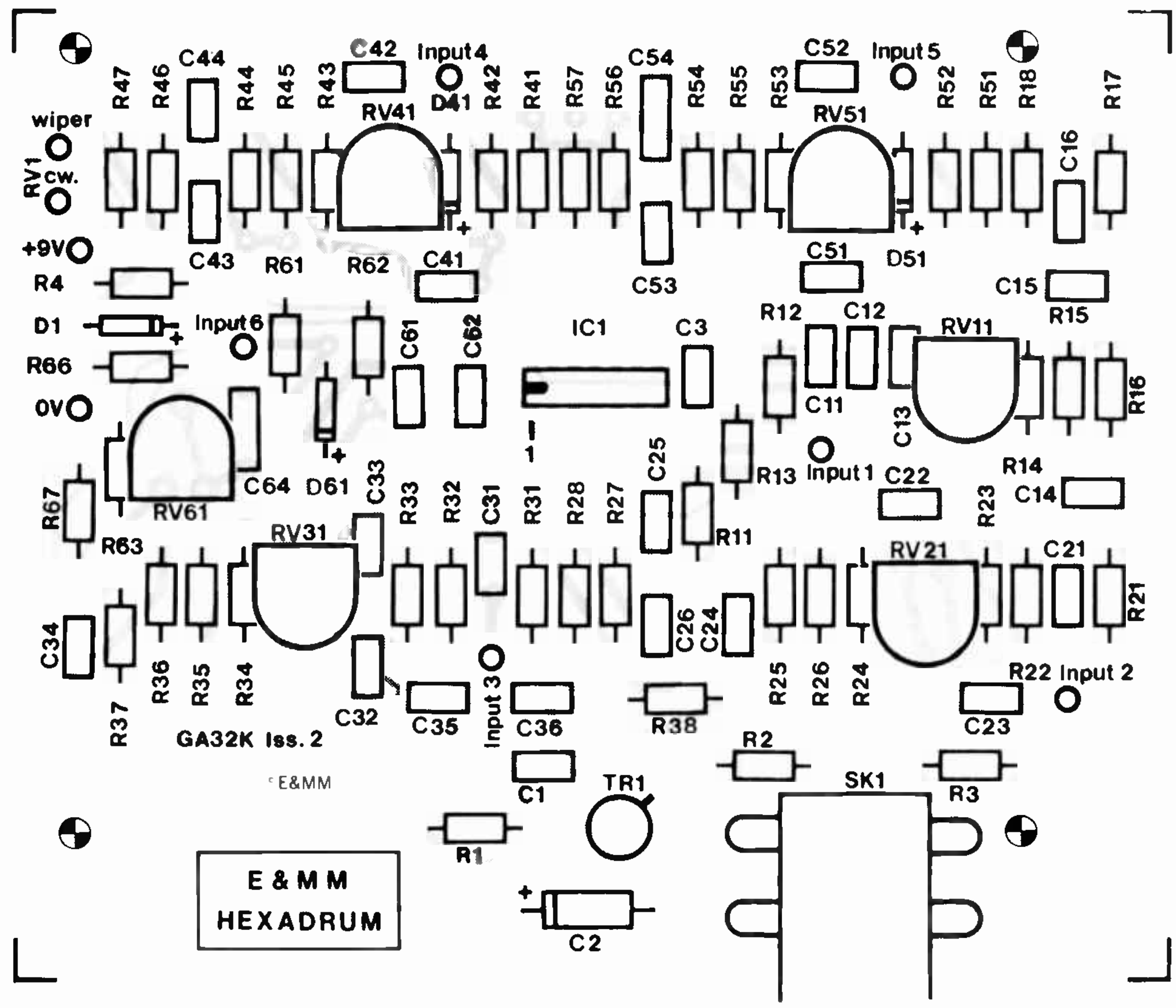
Although the filters may look a little unorthodox, they can be regarded as modified versions of the well known multiple feedback/bandpass type with adjustable resonance.

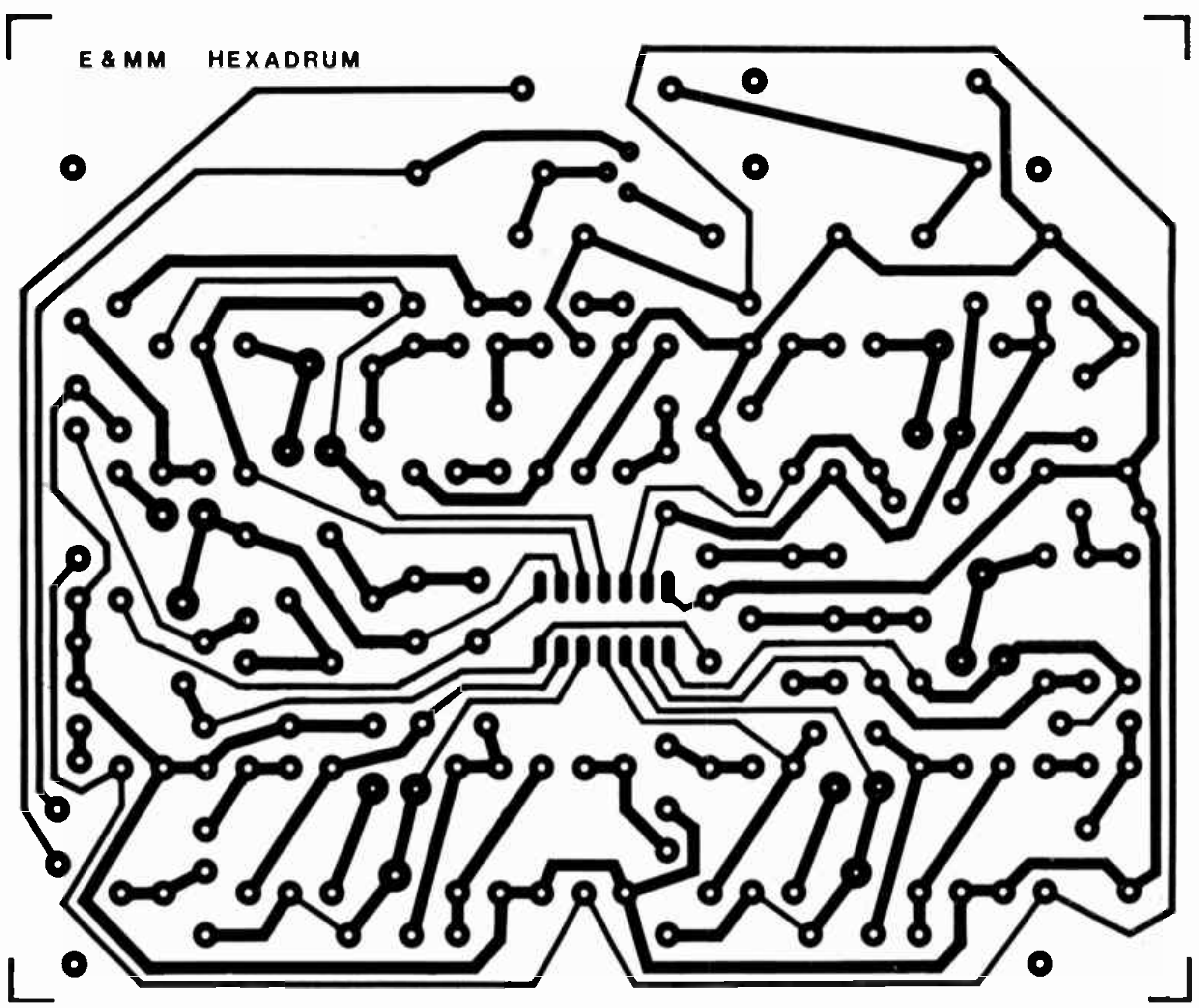
Construction

Assembly of the printed circuit board is straightforward, following the normal practice of installing pins, passive components and finally the integrated circuit which should be mounted in a socket. The output jack socket is best fitted first to the side panel, the tags then bent to lay flat against the circuit board and only soldered in place when the board is screwed down.

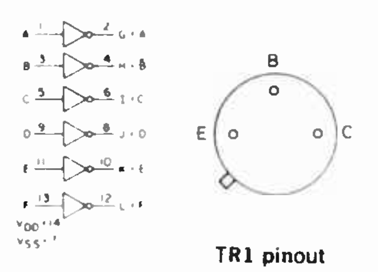
Before undertaking any work on the case, double check the layout of all parts to be mounted on it. The reasons for this are that the circuit board and its mounting points are assymetrical, in addition to which the two case halves only fit together one way round. For the orientation of the board and non-board mounted components, check the photographs and diagrams given. Note that some of the unused board mounting pillars on the base and lid sections of the suggested case must be cropped off to allow room for the volume control potentiometer and switch.

No layout for the pick-ups is given as the constructor is the best person to ascertain this from the most comfortable position of the hand of the player. However, ensure that no pick-up is to be located over one of the circuit board mounting pillars before beginning cutting. If the lid of the case is covered with masking tape or self adhesive plastic film to prevent damage to it during construction, the outline of the player's hand may be marked out directly onto this. An effective method of cutting the holes for the pick-ups is to use a sheet metal punch of diameter 20-23mm. In order to attach the earth wire to the brass plate, a small notch must also be made in the edge of the hole, either by drilling a small hole of diameter less than 3mm adjacent to the edge of the punched hole or simply filing. The earth leads themselves must be soldered very cleanly to the brass rims of the pick-ups before they are installed. This is shown in the photograph. The brass area is a large heat sink and so therefore requires quite a lot of heat but the minimum of solder. The pick-ups





Soldering to the piezo transducers.



IC1 pinout

TR1 pinout (viewed from below)

can then be fixed in place using a cyanoacrylate adhesive, such as Loctite Super Glue 3. Soldering to the silvered electrode must be done with the minimum of heat for a fraction of a second, or destruction of the silvering begins to occur. Connections from inputs to pick-ups are neatly made with ribbon cable. The inputs are numbered from the highest pitch to the lowest. Since hum was a slight problem in the prototype (cured by sticking earthed aluminium foil inside the case and earthing the metal panels), the constructor may find screened leads to the pick-ups reduce the problem.

Assuming completion of all other mechanical work on the case, the circuit board may now be screwed down and hardwired to the pick-ups, potentiometer, battery and switch.

The artwork on the case of one prototype was made by cutting the shape of the hand out of coloured, self-adhesive plastic. Another case was sprayed using a matt black cellulose aerosol, then a mask in the shape of the player's hand was cut with a scalpel from an outline drawn on self-adhesive film initially applied to the case. Finally, fix the sponge rubber 'Trimseal' pads over the pick-ups and screw the case together to complete construction. The presets may need to be adjusted to damp the resonance of some of the drums (anticlockwise) and peak the resonance of others.

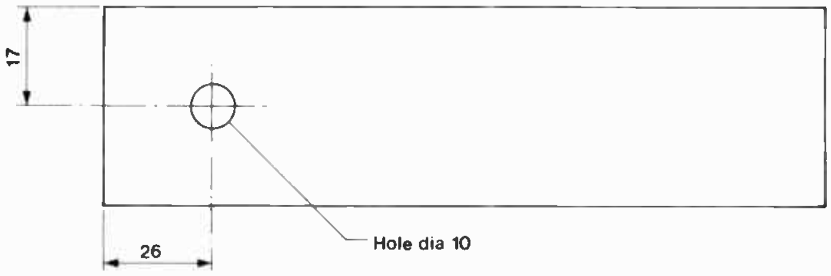


Figure 3(a). Drilling of case side panel for volume control.

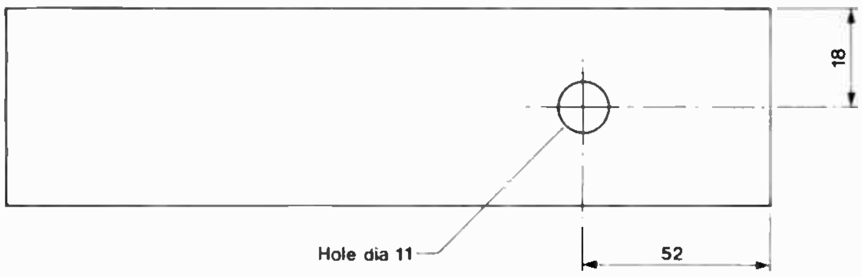


Figure 3(b). Drilling of case side panel for jack socket.

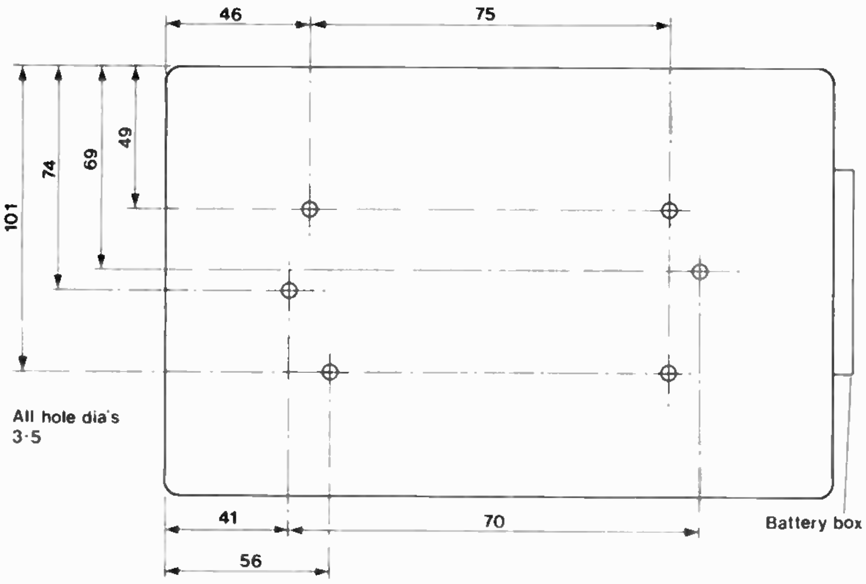


Figure 3(c). Drilling of case bottom moulding for adjustment of presets.

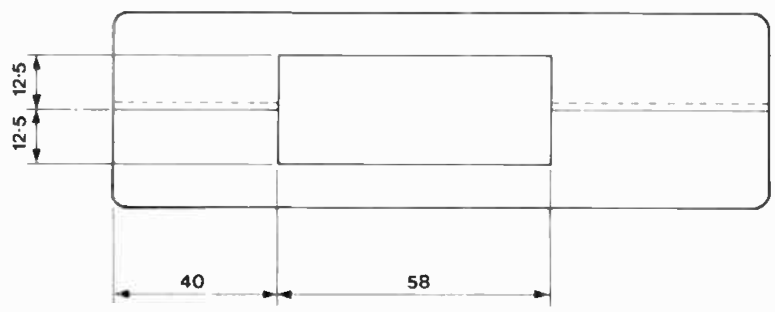
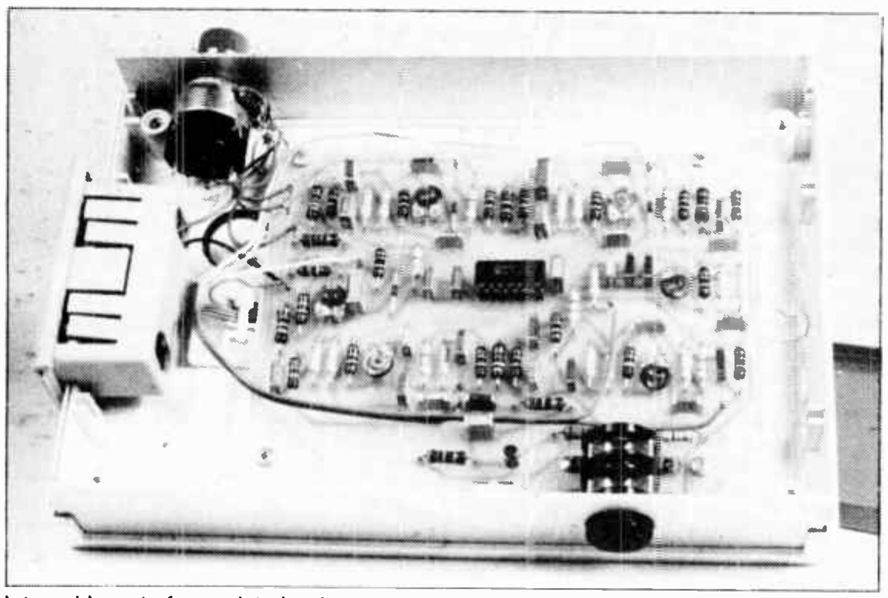
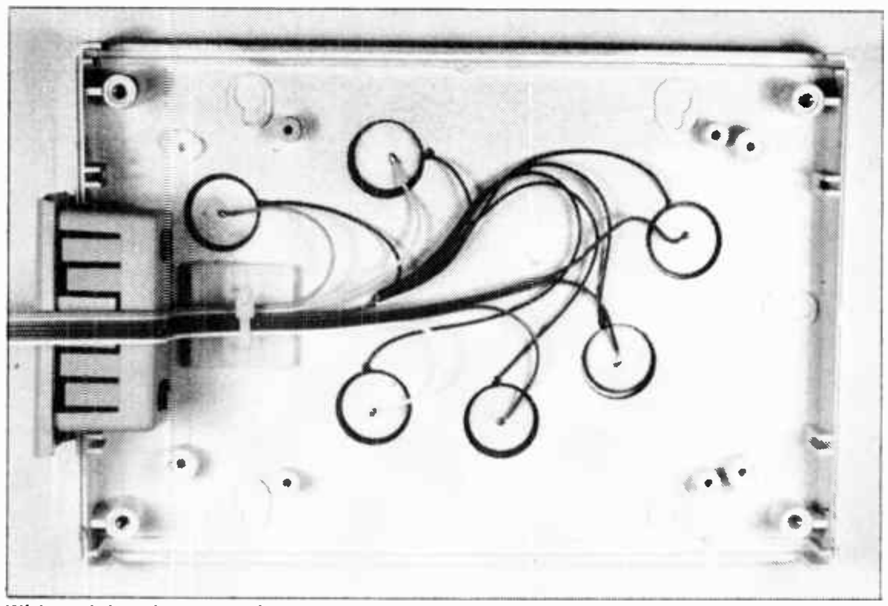


Figure 4. Battery box aperture. The cut-outs must be made in the top moulding end with the groove joint and bottom moulding end with the tongue.



Internal layout of completed unit.



Wiring of the piezo transducers.

PARTS LIST

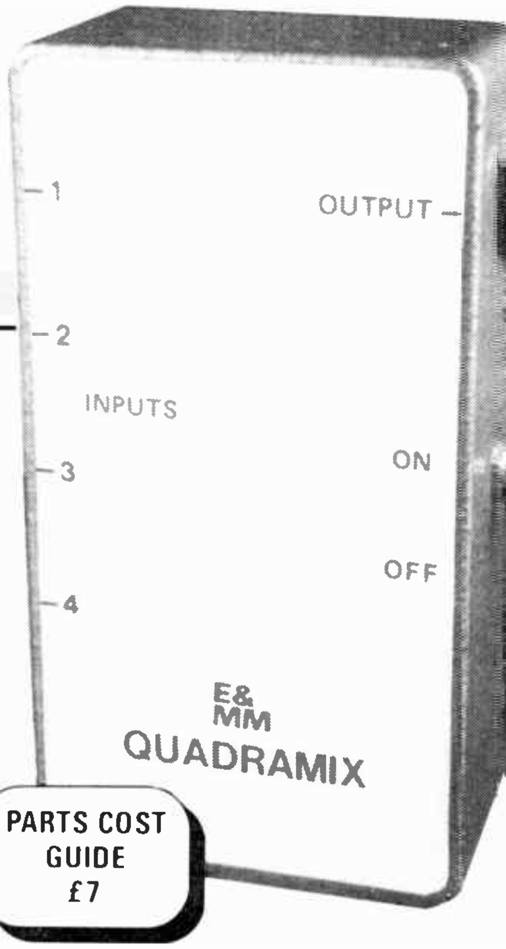
Resistors - all 1% 0.4W metal film			
R1,38,57,67	470k	4 off	(M470K)
R2	180R		(M180R)
R3,13,23,33 42,52,62	10k	7 off	(M10K)
R4	270R		(M270R)
R11,14,17,21 24,27,31,34 37,41,43,46 51,53,56,61 63,66	180k	18 off	(M180K)
R12,15,16,22 25,26,32,35 36,44,45,54 55	68k	13 off	(M68K)
R18,28	150k	2 off	(M150K)
R47	680k		(M680K)
RV1	Switched pot. 4k7, linear		(FW41U)
RV11,21,31 41,51,61	470k Hor. S-min preset	6 off	(WR63T)
Capacitors			
Electrolytic			
C2	47u, 10V axial		(FB38R)
Polycarbonate			
C1,3,52,53	100n	4 off	(WW41U)
C11,32,33	15n	3 off	(WW31J)
C12,13,36	4n7	3 off	(WW26D)
C14,15	12n	2 off	(WW30H)
C16,26	1n5	2 off	(WW23A)
C21,24,25	27n	3 off	(WW34M)
C22,23	10n	2 off	(WW29G)
C31,34,35	39n	3 off	(WW36P)
C41,42	22n	2 off	(WW33L)
C43,44	68n	2 off	(WW39N)
C51	33n		(WW35Q)
C54,64	150n	2 off	(WW43W)
C61,62	47n	2 off	(WW37S)
Semiconductors			
IC1	CD4069UBE		(QX25C)
TR1	BC108C		(QB32K)
D1	BZY88C5V6		(QH08J)
D41,51,61	1N4148	3 off	(QL80B)
Miscellaneous			
XL11,21,31,41, 51,61	Piezo transducer 27mm	6 off	(QY13P)
SK1	Jack socket		(HF90X)
	14 pin DIL Socket		(BL18U)
	Knob K15		(HB36P)
	Verobox type 201		(LL05F)
	PP3 Battery holder		(XX33L)
B1	PP3 Battery		
	Hexadrum PCB		(GA32K)
	Rubber disc 27mm	6 off	(QY16S)
	⅜" nut for RV1 stem		
	Veropins type 2145	14 off	(FL24B)
	10-way ribbon cable 1m		(XR06G)

Note: A complete kit (LW85G) of all the parts listed (excluding batteries) is available from Maplin Electronic Supplies Ltd at a price of just £18.95 inc. VAT and P&P.

Note: The parts list and ready-made circuit board have the component reference number prefixed by the number of the drum circuit in which it is used, e.g. drum 1, R3 becomes R13.

QUADRAMIX

Robert Penfold



Most mixer designs are fairly complex and expensive to construct, but have a great many useful facilities and features. However, there are occasions when the most basic of mixers is all that is needed, with a gain control at each input not even being necessary. An example of such a situation would be when using a few of E&MM's very popular 'Syntom', 'Synwave', and Hexadrum projects, with the outputs fed to a single amplifier. The relative output levels of the effects units could be adjusted using the output level control on each of these sound making projects, and all that is needed is a basic mixer circuit to combine the four outputs and prevent any interaction between the output level controls.

The 'Quadramix' is a basic four-into-one mixer which has unity voltage gain from each input to the output. The input impedance is 100k at all four inputs and the output impedance is low so that the unit also acts as a buffer amplifier. The noise level of the circuit is too low to be of any consequence, as is the distortion level, provided the input signal is kept below the clipping threshold of approximately 6 volts peak to peak — more than enough for most musical instrument outputs. Power is provided by a PP3 size 9 volt battery which has an extremely long life since the current drain of the circuit is only about 2mA.

Basic Mixer

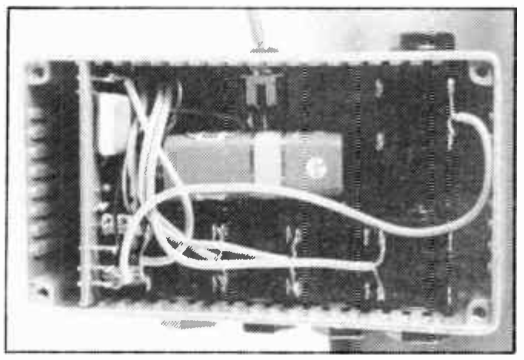
Modern mixer designs are invariably based on an operational amplifier used in the configuration shown in Figure 1, and this design is no exception.

If we ignore input 2 and RB for the time being, the circuit is a straightforward inverting mode operational amplifier circuit. Due to a negative feedback action, the circuit stabilises the inverting (-) input at the same potential as the non-inverting (+) input. The latter is normally biased to the 0V rail in a circuit having dual balanced supply rails, or at half the supply voltage if a single supply is used. With no input signal, the output is at the same voltage as the two inputs, and this optimises the output voltage swing before the onset of clipping.

If RA and RC have the same value, an input voltage will produce an identical change in the output voltage, but a change of opposite polarity. For example, an input voltage of +2 volts would produce +1 volt at the inverting input if we assume no change in output voltage. This is caused by a simple potential divider action across RA and RC, and in practice it would result in the output swinging negative in order to balance the input potentials. This state of balance would be achieved with the output 2 volts negative of its quiescent level, since a potential divider action gives a potential at the inverting input which is half way between the voltages at input 1 and the output. The circuit thus acts as an inverting buffer amplifier.

With an input to input 2 as well, the circuit operates in much the same way, but the output must now respond to the sum of the two input voltages. If input 2 is also at +2 volts, for example, the output would have to be 4 volts negative in order to counteract both input signals and maintain the balance of the input voltages by a potential divider action. With input 2 (say) 2 volts negative, the two inputs would counteract one another and the output voltage would remain at its quiescent level.

This configuration is known as the 'summing mode', and it obviously provides the required mixing action. Although only two inputs are shown in Figure 1, it is possible to have any desired number of inputs with an extra input resistor being used at each additional input. An important property of this circuit is the constant voltage produced at the inverting input, and what is termed a 'virtual earth' is formed here. This isolates the inputs from one another so that changes at one input (such as connecting or



Internal view of the Quadramix.

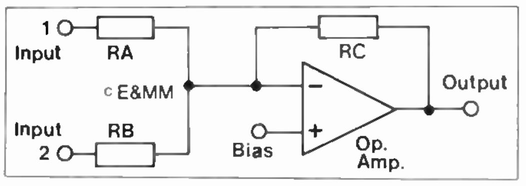
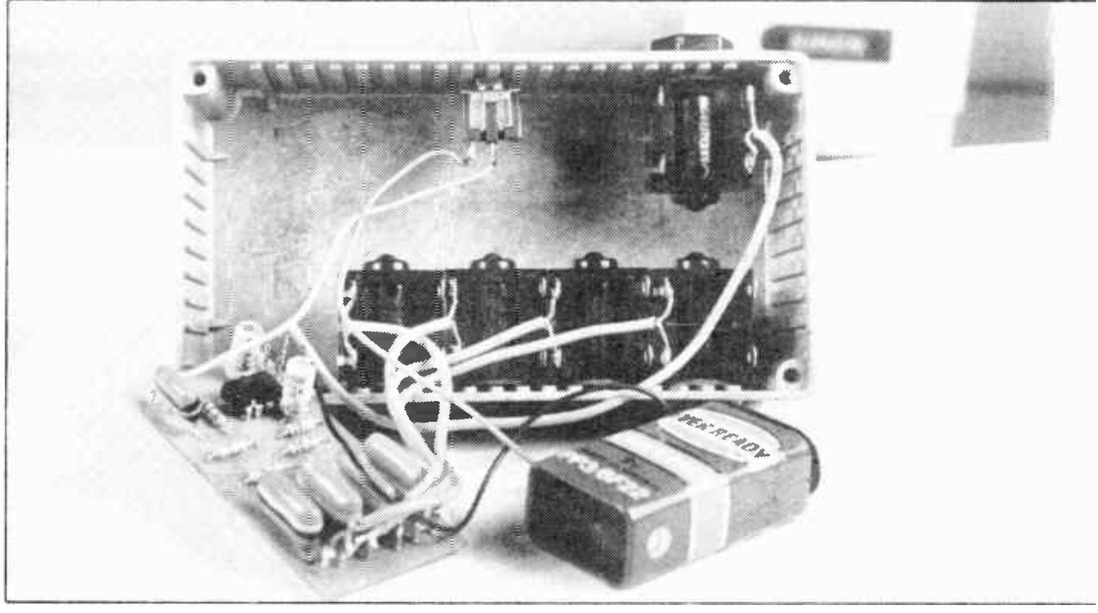


Figure 1. The basic operational amplifier mixer circuit.



Internal view with components removed.

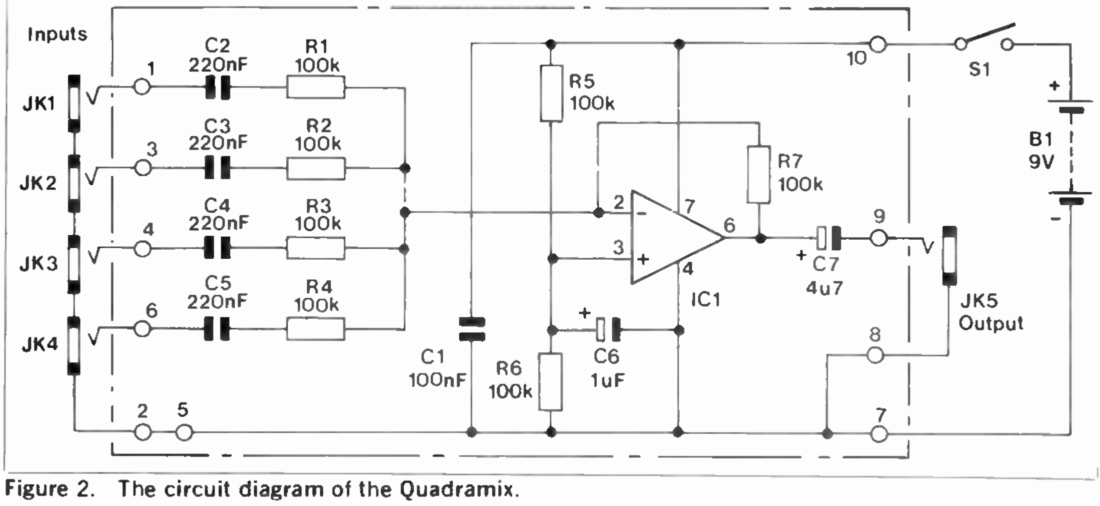
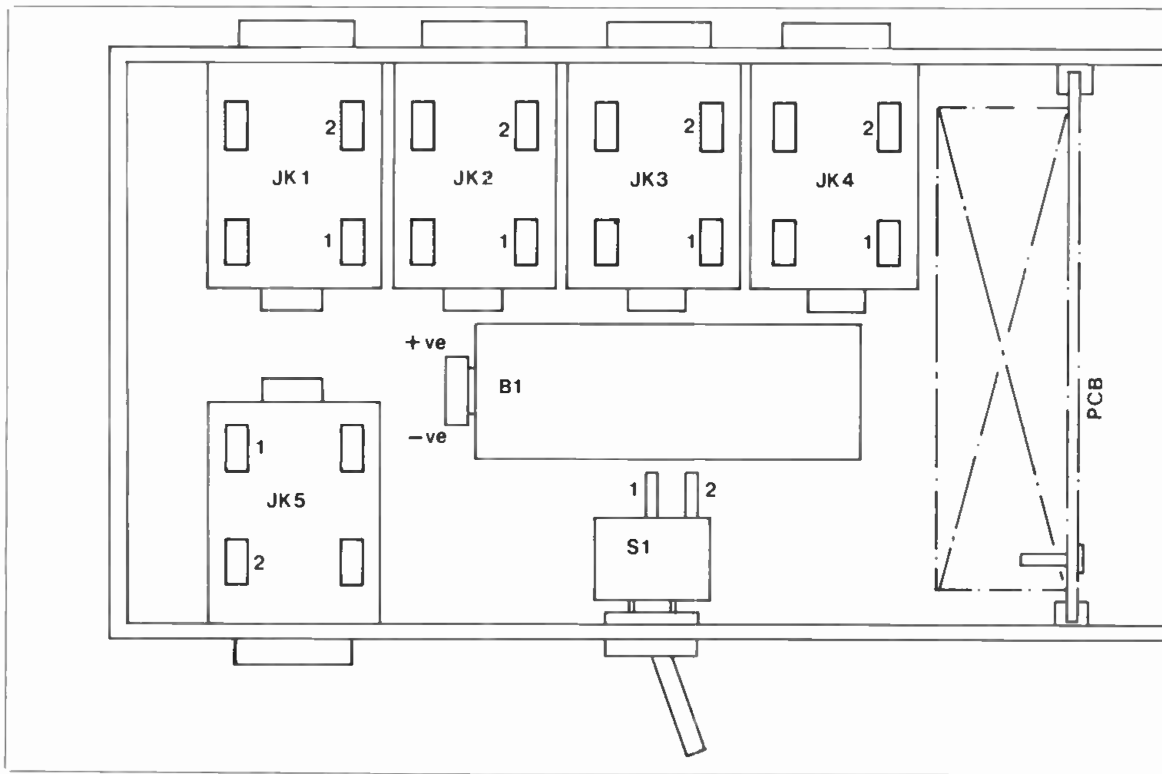


Figure 2. The circuit diagram of the Quadramix.



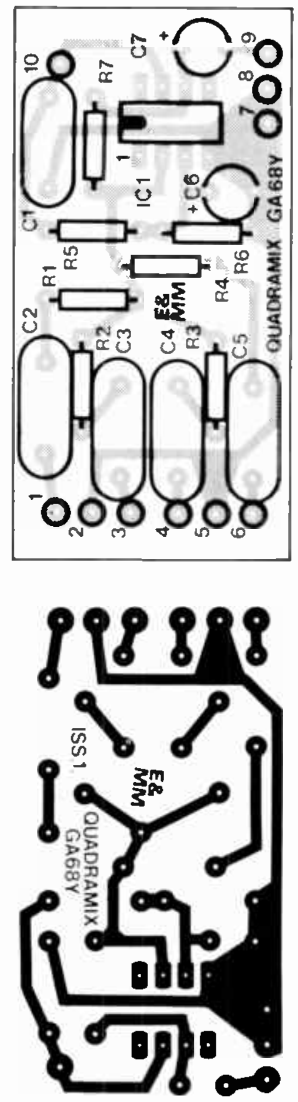


Figure 4. Construction details of the Quadramix: socket and battery wiring, plus component positions and PCB track layout.

disconnecting a piece of equipment) have no affect at the other inputs.

The Circuit

The full circuit diagram of the 'Quadramix' is shown in Figure 2, and has obvious similarities with the basic configuration of Figure 1. One obvious difference is that four inputs are provided in the practical circuit and this necessitates the use of four input resistors (R1 to R4). A DC blocking capacitor is also used at each input, and these are C2 to C5. The non-inverting input is biased by R5 and R6 since a single supply rail is used. C6 filters out any noise which might otherwise find its way to the non-inverting input due to stray coupling.

C7 provides DC blocking at the output and C1 is a supply decoupling capacitor. The circuit only has one control, and this is on/off switch S1.

Construction

A suitable housing for the unit is a diecast aluminium box having approximate outside dimensions of 120 by 65 by 40mm. The four input sockets are mounted on one side of the case with the output socket and on/off switch on the other. The positioning of these, especially the four input sockets, is quite critical as there is not a great deal of excess space inside the case. Figure 3 shows the positioning of the on/off switch and sockets, and it is strongly recommended that this layout is copied accurately. The mounting hole diameters are correct for the specified components, but note that other components might need mounting holes of slightly different diameters.

Details of the printed circuit board and wiring of the unit are shown in Figure 4. Construction of the board is quite simple, but it is easier if R2 and R3 are soldered into place before C2 to C5. Veropins are used at the points where off-board connections are made to the board.

Screened leads are used to connect the board to the input and output sockets. Provided the specified case is used, the completed printed circuit board is slotted into the lowest set of guide rails in the case, once all the wiring has been completed.

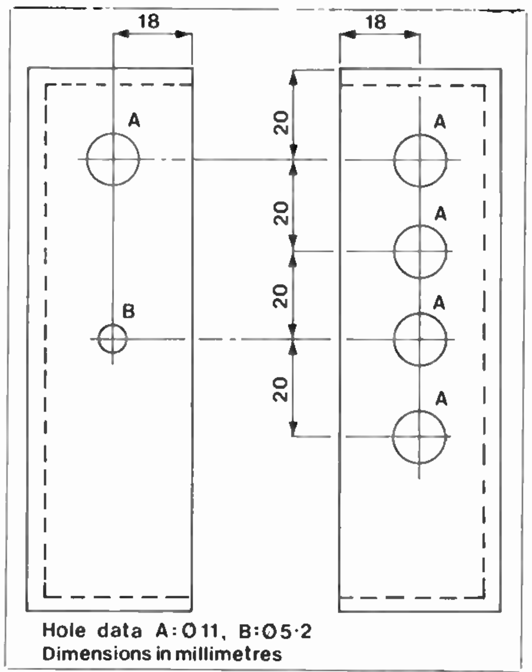


Figure 3. Drilling of the mounting holes in the case.

There are suitable spaces for mounting holes in the printed circuit board if a different case is used.

The battery fits into the space between S1 and the input sockets. A piece of foam material can be glued to the removeable base panel of the case so that the battery is held firmly in place when the base panel is screwed in place.

The unit is connected to the other components in the system using normal screened audio connecting cables. As the circuit has unity voltage gain, a fairly high input impedance, and a low output impedance, it should not produce any problems with incompatability when it is added into a system. In fact, it can be used as a buffer amplifier in situations where a relatively high impedance signal source is driving a fairly low input impedance and loading effects are producing poor results.

As described here, the unit is only suitable for mono operation, but for stereo operation it is merely necessary to use two units, one in each stereo channel. If more than four inputs are required the circuit could easily be modified to have any desired number of inputs, as explained earlier. Alternatively, two 'Quadramix' units connected in series will accommodate seven inputs.

E&MM

QUADRAMIX WIRING CHART

From	To	Remarks	
JK1/1	PCB/1	conductor	screened lead
JK1/2	PCB/2	screen	
JK2/1	PCB/3	conductor	screened lead
JK2/2	PCB/2	screen	
JK3/1	PCB/4	conductor	screened lead
JK3/2	PCB/5	screen	
JK4/1	PCB/6	conductor	screened lead
JK4/2	PCB/5	screen	
JK5/1	PCB/9	conductor	screened lead
JK5/2	PCB/8	screen	
B1/+ve	S1/2	Battery clip leads	
B1/-ve	PCB/7		
S1/1	PCB/10		

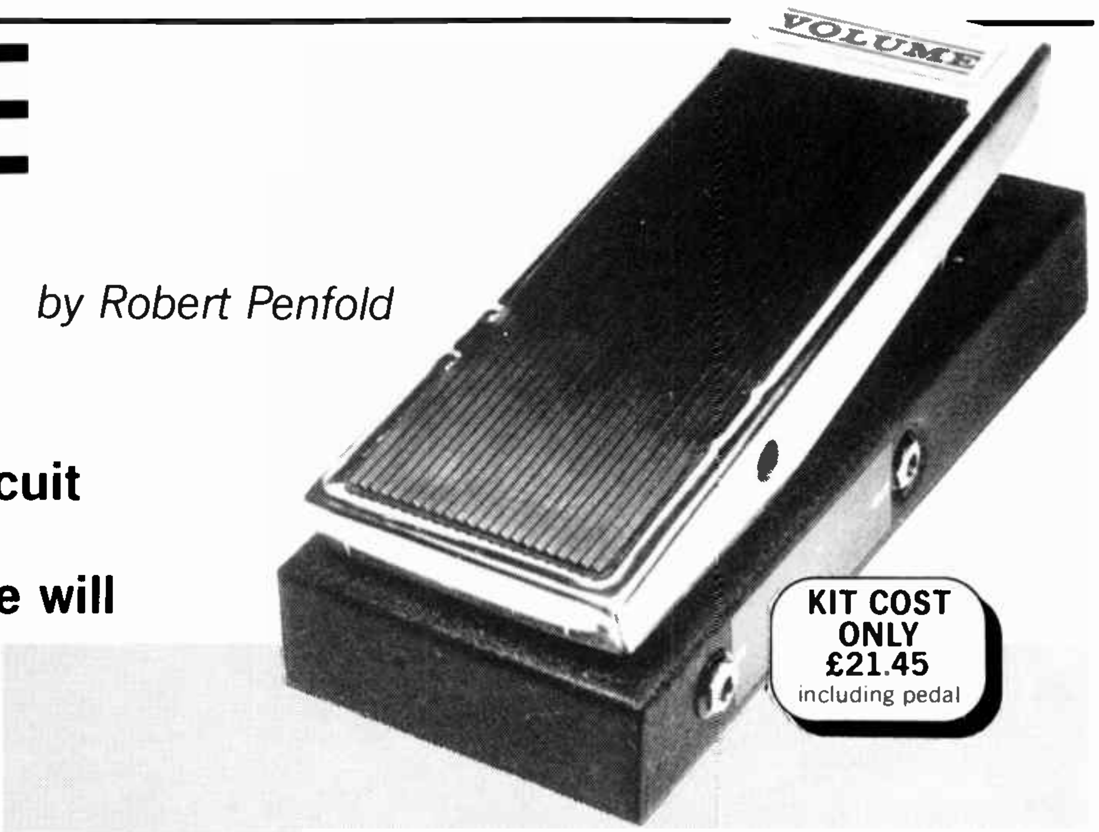
QUADRAMIX PARTS LIST

Resistors - all 1% 0.4W metal film unless specified

R1-R7	100k	7 off	(M100K)
Capacitors			
C1	100n polyester		(BX76H)
C2-C5	220n polyester	4 off	(BX78K)
C6	1u 50V min. p.c.m.		(YY31J)
C7	4u7 25V min. p.c.m.		(YY33L)
Semiconductor			
IC1	LF351		(WQ30H)
Miscellaneous			
S1	SPST min. toggle		(FH97F)
B1	PP3 size 9 volt battery		
SK1-SK5	Standard (¼ in.) jacks		(HF90X)
	PCB		(GA68Y)
	Battery connector		(HF28F)
	Case		(LH71N)
	Veropins		(FL23A)
	Connecting wire		(BL00A)
	Screened cable		(XR15R)

VOLUME PEDAL

by Robert Penfold



★ Unique Hall effect circuit
★ Noiseless operation
★ Low output impedance will drive long cables

A conventional volume pedal consists of an ordinary potentiometer connected in the usual volume control fashion, and operated from the foot pedal via a rack and pinion mechanism. This system works very well, but with a lot of use the potentiometer's track can become worn with consequent noise being generated as the pedal is operated.

The problem is overcome in this pedal, which uses a magnet and a Hall effect device instead of a potentiometer. As the pedal is depressed the magnet is brought closer to the Hall effect device, and the increased magnetic field is converted into an increase in voltage.

The input signal is passed to the output by way of a voltage controlled attenuator (VCA), and, like a volume control, this can provide a level of attenuation of anything from zero to around 80dB. However, it is of course controlled by means of a voltage applied to its control terminal.

The output voltage of the Hall effect device is slightly too high in terms of its quiescent level, and too low in terms of voltage change produced by the varying magnetic field, and so the device cannot directly control the VCA. A level shifter and low gain DC amplifier are therefore used to process the output of the Hall effect sensor and give a suitable control voltage for the VCA.

Pre-emphasis (treble boost) at the input of the VCA and de-emphasis (treble cut) at the output are used to give a slight improvement in the signal to noise ratio of the unit. The ratio is actually about 80dB, and the background noise should be completely insignificant provided the unit is not used with a very low level signal. The circuit can take a maximum input level of about 2 volts RMS at most frequencies without serious distortion being produced. The circuit has an input impedance of about 50k and an output impedance of approximately 350 ohms.

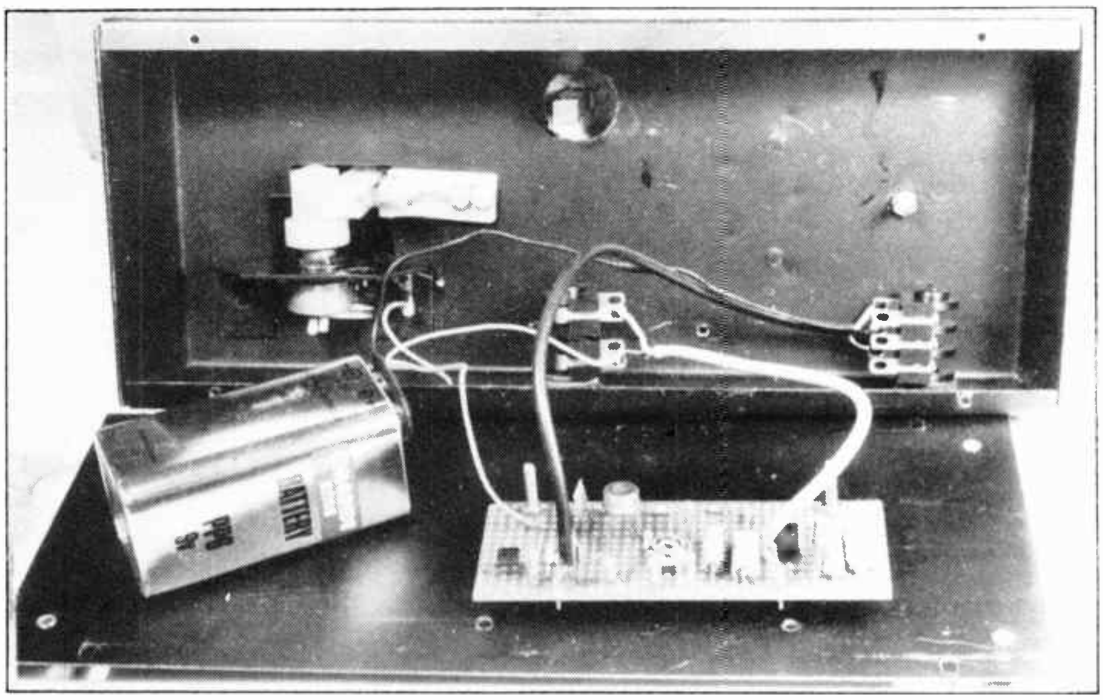
The Circuit

Figure 1 shows the complete circuit diagram of the Volume Pedal, and IC1 is the Hall effect sensor. This is not one of the usual Hall effect switches but is a linear device, of recent origin, which has differential outputs.

IC2 is an operational amplifier used in the inverting mode.
gives an output voltage swing that is sufficient to give a wide attenuation range from the VCA, and a single resistor and diode chain in the feedback network of IC2 is adequate to give the desired tailoring of the amplifier's transfer characteristic.

The VCA is based on IC3 which is the well known MC3340P device. This can provide a voltage gain of about 13dB, but in this application a maximum gain of only unity is required and this is achieved by using R7 and R8 to provide about 13dB of attenuation at the input of IC3.

The current consumption of the circuit is about 12mA, and it is switched on automatically by insertion of the input jack.



View inside the pedal.

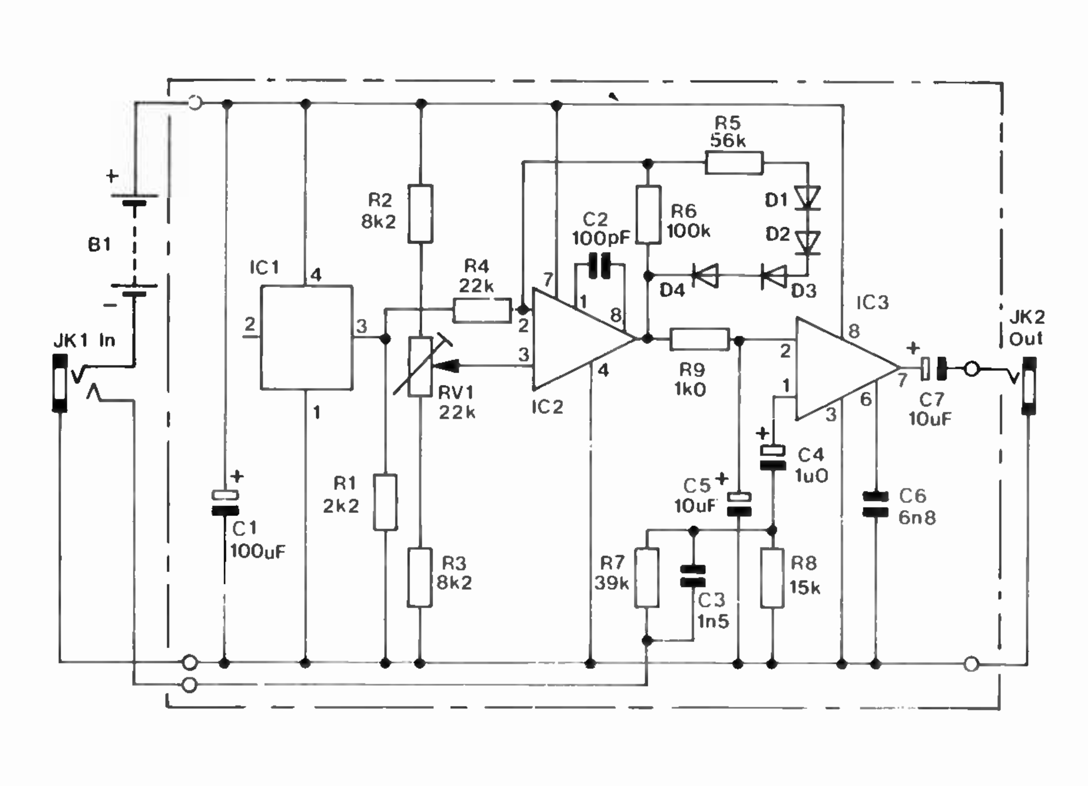


Figure 1. Circuit of the volume pedal.

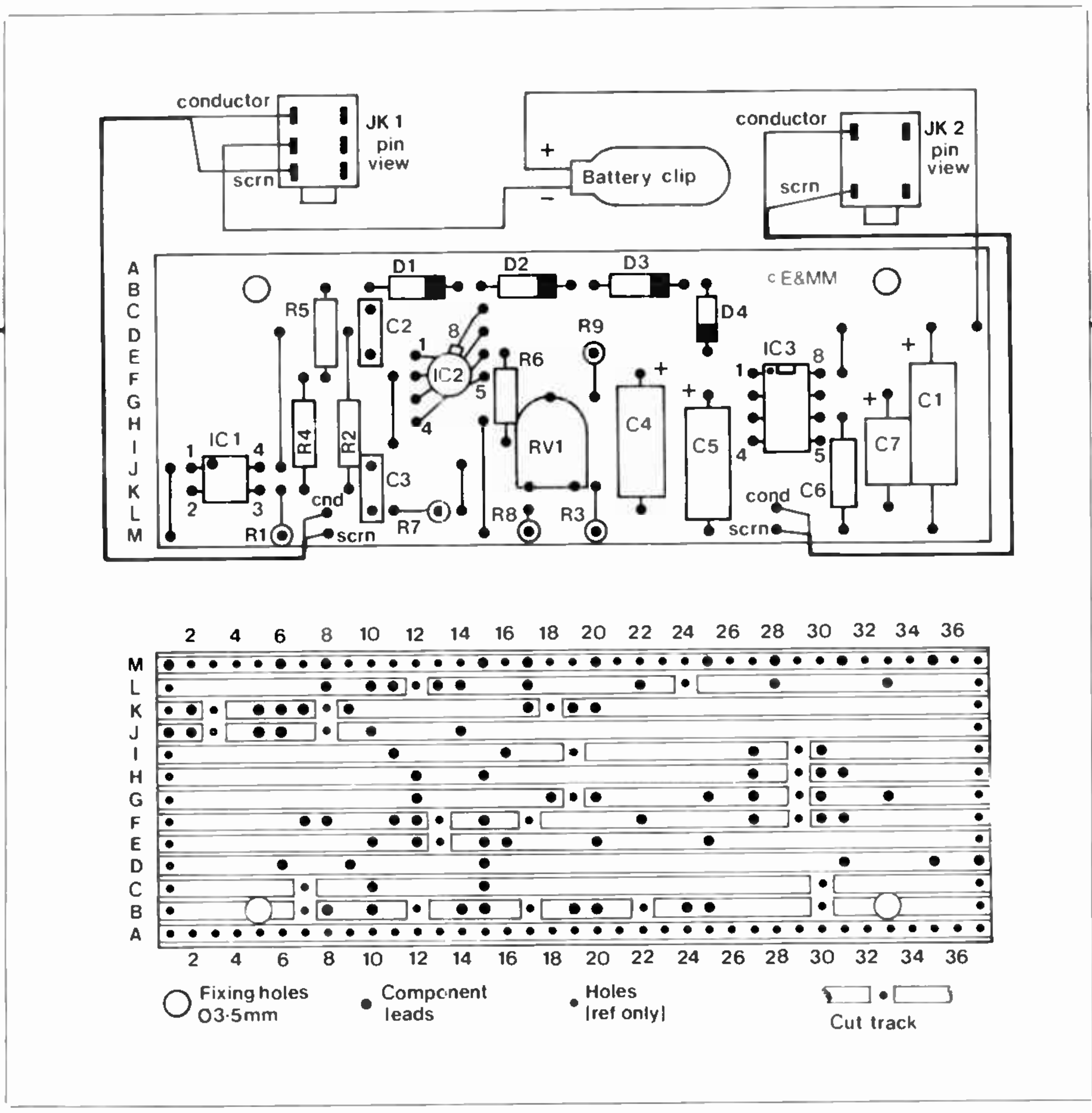


Figure 2. Veroboard layout and wiring of the volume pedal.

Construction

A 0.1" matrix stripboard measuring 37 holes by 13 copper strips accommodates practically all the components, and Figure 2 provides details of this board and wiring of the unit.

IC2 is a CMOS device and requires the normal MOS handling precautions to avoid possible damage by static charges.

The pedal itself is a modified Maplin volume pedal. The output lead should be removed, but retain the potentiometer to give friction to the pedal mechanism. Open out the recessed hole on the right hand side of the case to 11mm, and drill another the same size nearby to accommodate the jack sockets.

The magnet passes through a hole about 18 to 20mm in diameter which is made in the top panel, midway along and about 75mm out from the side of the case on which JK1 and JK2 are mounted. The magnet cannot be fitted direct to the underside of the pedal as it is too short to reach down into the case. Therefore a piece of timber or chipboard about 18 to 20mm thick is glued to the underside of the pedal immediately above the hole in the top panel of the case, and the magnet is then glued to this. A good quality adhesive must be used, and an epoxy type is probably the best choice for this application.

However, it is essential to mount the magnet the right way up or it will produce an output voltage of the wrong polarity from the Hall effect device. Probably the easiest way of finding the correct polarity for the magnet is to first wire the component panel to the off-board components. Connect the battery, and connect the unit into a signal path. It should be possible to control the gain of the circuit by adjusting RV1, and this component is adjusted just far enough in an anticlockwise direction to severely attenuate the signal as it passes through the unit. Apply each end of the magnet to IC1 in turn, and note which end produces an increase in gain from the circuit. The other end of the magnet is glued to the piece of timber or chipboard on the underside of the pedal.

The component board is mounted on the base panel of the case, and it must be positioned so that IC1 is aligned reasonably accurately with the magnet. The easiest way of achieving this is to hold the component board in position with IC1 over the magnet, and then measure the positions of the two mounting holes in the board relative to the sides of the case, and drill the mounting holes in the base panel of the case accordingly. Spacers a quarter of an inch long are used over the mounting bolts for the panel so that when the pedal is in the down position the magnet is in close proximity to IC1. If necessary, one or two extra nuts or some washers can be used to give further spacing and reduce the minimum gap between IC1 and the magnet, but do not use so much spacing that the two come into contact when the pedal is fully in the down position as this could possibly result in damage occurring.

The setting of RV1 controls the maximum amount of attenuation that the unit can provide, and it will probably be possible to obtain a maximum level of around 90dB. This is more than is normally necessary, and a lower level of around 60dB should be more than adequate and would give more precise control of the attenuation level. The magnet will have a slight effect on the unit even with the pedal in the fully up position, and to allow for this RV1 should be adjusted for a few dBs more attenuation than is required (since RV1 cannot be adjusted while the base panel is in position).

PARTS LIST FOR THE VOLUME PEDAL

Resistors — all 1% 0.4W metal film unless specified			
R1	2k2		(M2K2)
R2,3	8k2	2 off	(M8K2)
R4	22k		(M22K)
R5	56k		(M56K)
R6	100k		(M100K)
R7	39k		(M39K)
R8	15k		(M15K)
R9	1k		(M1K0)
RV1	22k sub-min horizontal preset		(WR59P)
Capacitors			
C1	100uF 10V electrolytic		(FB48C)
C2	100pF ceramic plate		(WX56L)
C3	1.5nF ceramic plate		(WX70M)
C4	1uF 63V electrolytic		(FB12N)
C5,7	10uF 25V electrolytic	2 off	(FB22Y)
C6	6.8nF polycarbonate		(WW27E)
Semiconductors			
IC1	634SS2		(QR55K)
IC2	CA3130T		(QH28F)
IC3	MC3340P		(QH49D)
D1-4	1N4148	4 off	(QL80B)
Miscellaneous			
JK1	Stereo jack socket		(BW79L)
JK2	Mono jack socket		(BW78K)
B1	PP6 battery		
	PP3 battery connector		(HF28F)
	Bolts 6BA 1in.		(BF07H)
	Nuts 6BA		(BF18U)
	Spacers 6BA ¼in.		(FW34M)
	Volume pedal		(XY28F)
	0.1 matrix Veroboard		(FL08J)
	Large bar magnet		(FX72P)
	Wire, solder, etc.		

Note: A complete kit (LW88V) of all the parts listed (excluding batteries) is available from Maplin Electronic Supplies Ltd for just £21.45 inc. VAT and P&P

AUTO SWELL

by Clive Button

- ★ **Swell and sustain at the touch of a pedal**
- ★ **Accurate setting of rhythm and lead levels**
- ★ **For use with any amplified instrument**
- ★ **Battery operation**

KIT COST ONLY £10.90

A foot operated volume control, or swell pedal, is one of simplest effects pedals there is – it is also one of the most useful. The most common application is for reducing an instrument's volume during accompaniment playing, allowing it to be increased for a solo. Whilst it is easy to set a pedal at either end, i.e. minimum or maximum, a half-way setting can be difficult to duplicate accurately. The auto swell enables the player to set a consistent accompaniment level, and increase the volume (at a preset rate) by pressing a pedal. When the pedal is released, the volume reverts immediately to the lower level. Noise and wear problems associated with the pedal operated pot type of mechanism are also eliminated.

The unit can be put to a variety of uses and some of these will be suggested in the section on applications.

The circuit is essentially a voltage controlled amplifier, the gain of which is controlled by a variable rate ramp which is initiated by a foot switch. The advantages of this method over such devices as a conventional swell pedal or compressor unit will only really become apparent when you start to experiment with the possible uses it can be put to on guitar or keyboards.

The Circuit

The input is passed to IC1a inverting input via C1 and R1; this first op-amp increases the signal by a factor of four; its output passes the signal to the circuit comprising R3, C2, RV1 and TR3 (TR3 being the FET). This circuit is in effect a variable potential divider, the top half being R3, the bottom half being the series combination of RV1 and TR3 whose resistive value can be altered by adjustment of RV1 or control of the gate voltage on the FET. C2 is for DC blocking only. The output of this potential divider is passed on to the second op-amp IC1b via R4 and C3 and finally via C4 to the output. So what we have is an amplifier whose gain is variable and can be controlled by a DC voltage on the gate of the FET. R9 and R10 provide a mid-point voltage for op-amp biasing, decoupled by C6.

The circuit around TR1 and TR2 gives us our control voltage, by providing a variable speed ramp voltage; this is achieved in the following way: With S1 (a

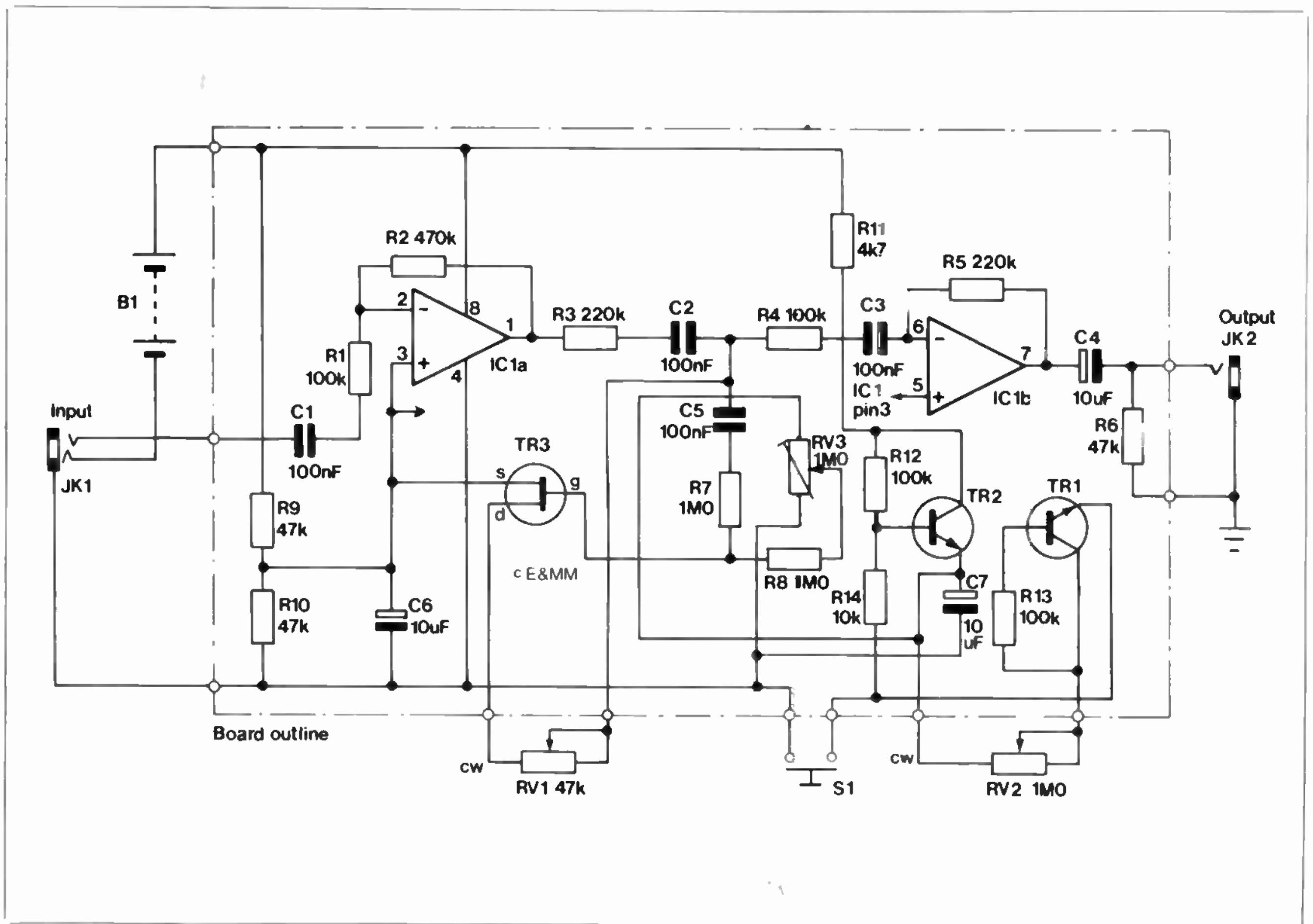


Figure 1. Auto-swell circuit diagram.

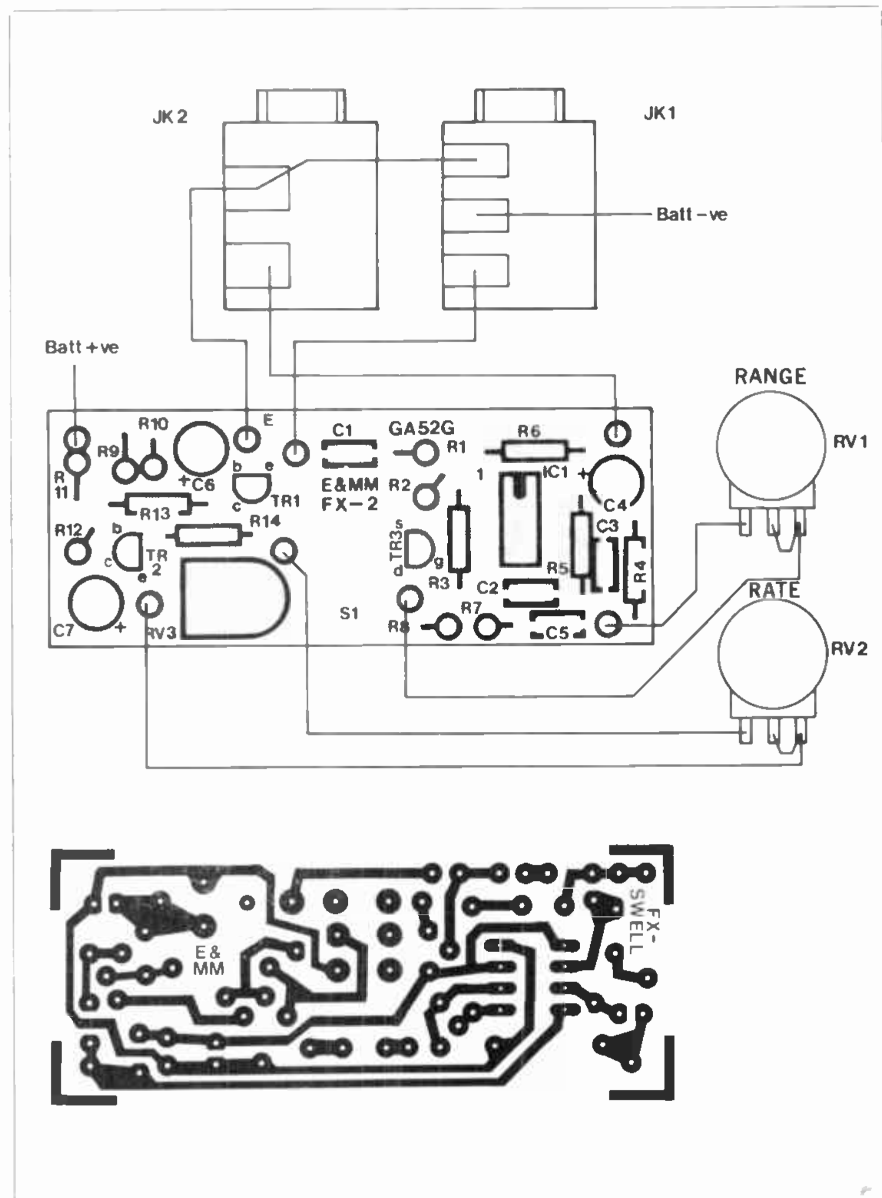
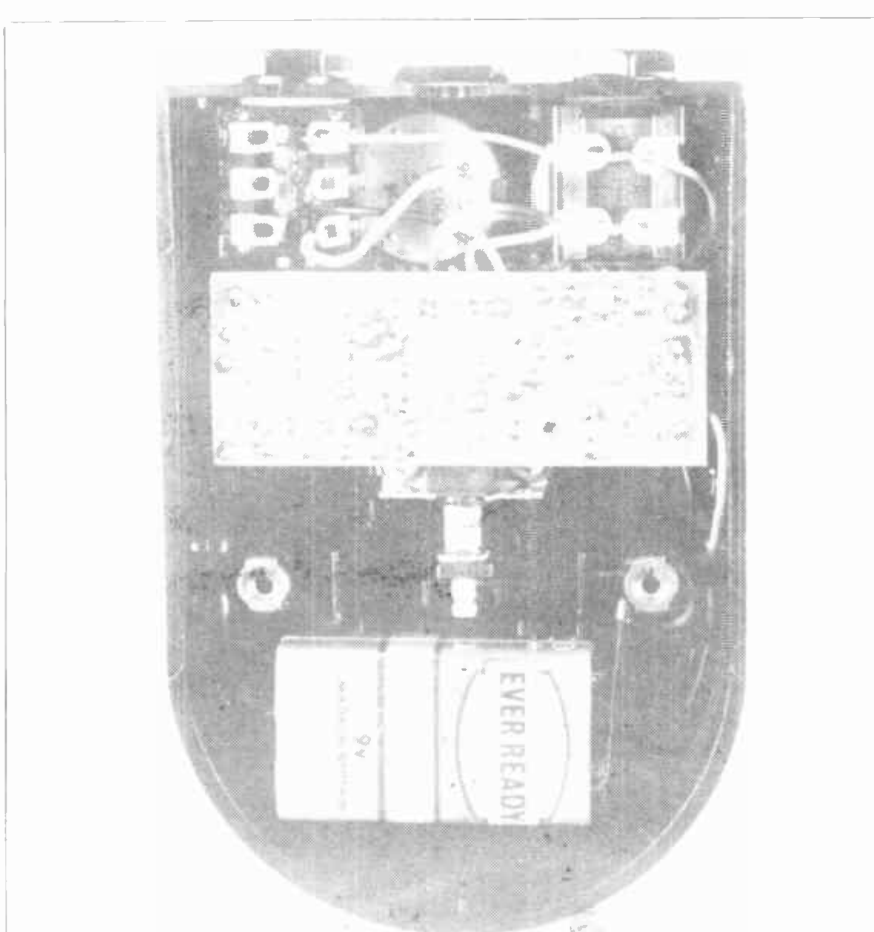


Figure 2. PCB track layout component overlay and wiring.



Internal view of auto swell.

single contact push-to-make momentary action switch) in its normally open position TR2 will be forward biased via R12 and will charge up C7 to positive supply voltage. When S1 is closed TR2 will be cut off via R14 and no longer supply a charging current to C7, however the circuit containing TR1 will be completed giving a linear discharge path to the voltage on C7, the rate of which can be controlled by RV2. This discharge ramp is passed on to the gate of TR3 via RV3 which is set for the gate voltage of the particular FET used. The circuit is powered by a PP3 9-volt battery that is switched on when the input jack is inserted.

Construction

Assemble the PCB in the usual way, following Figure 2. Also fit wires of the appropriate length to connect to the controls and jack sockets. The board is designed to fit in a Cliff CFS-1 pedal box which comes with a switch similar to Maplin's two pole latchswitch; this needs to be modified as follows. First, bend back the four tabs holding the switch in the frame, remove the switch and disconnect the wire that comes with the unit. Now is a good time to drill the pedal box to take the pots and jacks, whilst the fragile parts are out of the box. Next, the switch needs to be converted to non-latching operation: push the spring back and carefully remove the action lever (see Figure 3). Now solder the switch on to the board, and mount the whole assembly in the pedal. Note that the switch is now upside down in the bracket; because of the PCB you will only be able to bend two of the tabs over, but this should be sufficient to hold everything securely. Finally, wire up the sockets and controls: don't forget the battery negative lead, which connects to the centre tag on the stereo jack socket.

Setting Up

After the unit has been constructed RV3 will have to be set to suit the gate voltage of the FET. First of all plug the unit into an amplifier and check that with an input present, the output can be varied in level by the range control. If this is the case then set RV3 to its mid-point, set the rate control to fast and the range control to x2, adjust RV3 such that by pressing the switch you just get a doubling in gain at the output. The best way to do this is to adjust RV3 to the end that gives no alteration in gain at all when operating the switch, and gradually advance it until a doubling in level is achieved on operation of the switch, leaving it set at this point. This should ensure that the FET is working over its correct range and the range control is correctly calibrated. The unit is then ready for use.

Applications

The idea of this design was to produce a simple effects pedal that would give a gradual increase in gain to a guitar or keyboard signal at the touch of a foot operated control, the rate of the increase to be variable from almost instantaneous to around 3 seconds and the amount of increase variable from times two to times 10. The original requirement being to give guitarists the facility to apply sustain to individual notes within a solo by having a slow increase in gain that could be applied as and when required to counteract the natural decay in volume of the string. At this point I hear you all shout, "Why not use a compressor"? Well, the reason is that it cannot be applied to just odd notes here and there within a solo piece, and you also have to alter your technique to overcome the clipping effect to notes where you don't want it.

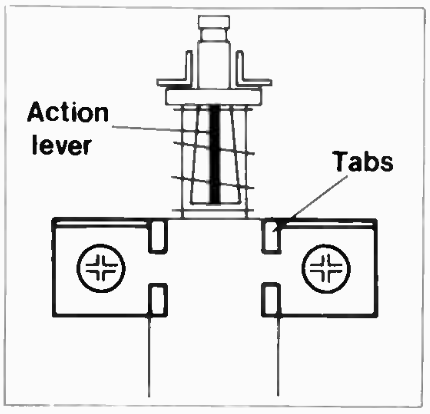


Figure 3. Switch modification details.

Although this was the main reason for designing the unit, it was soon found that it could be used to provide a vast number of other effects, and not only on guitar. I will mention a few of these and leave it to the constructor to discover the full potential of the unit. On guitar or keyboard it can be used with a fairly fast rate to produce violining or bowing effects by applying it at the start of each note played, or with a slightly slower rate the sound of a steel guitar can be imitated (very effective on chords). It can be used as a straight swell pedal with the range set to mid-way, or with the range set to x10 and the rate on slow a very intense crescendo effect can be obtained by playing a chord, applying the effect and letting it build in volume. With a range of x2 and instantaneous rate it can be used as a straight boost for guitars and with the range midway and the rate fairly fast you can make it sound as if you're playing backwards by applying it on each note and cutting it off sharply before playing the next note. (Like the guitar part on the Beatles' "Tomorrow never knows.")

Because the unit is in circuit all the time your instrument and amplifier settings will have to be adjusted to suit the way in which the effect is to be used, but as a guide you will find with the range control set to its mid-position you have a gain of unity, i.e. input signal equals output signal (that is with the switch in its normally open position).

PARTS LIST FOR AUTO SWELL

Resistors - all 1% 0.4W metal film unless specified			
R1,4,12,13	100k	4 off	(M100K)
R2	470k		(M470K)
R3,5	220k	2 off	(M220K)
R6,9,10	47k	3 off	(M47K)
R7,8	1M	2 off	(M1M)
R11	4k7		(M4K7)
R14	10k		(M10K)
RV1	47k log pot		(FW24B)
RV2	1M log pot		(FW28F)
RV3	1M hor. preset		(WR64U)
Capacitors			
C1,2,3,5	100n disc ceramic	4 off	(YR75S)
C4,6,7	10uF 35V PC electrolytic	3 off	(FF04E)
Semiconductors			
TR1,2	BC182L	2 off	(QB55K)
TR3	BF244		(QF16S)
IC1	LF353		(WQ31J)
Miscellaneous			
JK1	Stereo jack socket		(HF92A)
JK2	Mono jack socket		(HF90X)
	Knob	2 off	(YG40T)
	Knob cap (blue)	2 off	(QY01B)
	PP3 clip		(HF28F)
	PCB		(GA52G)
	Pedal switch box		(YK26D)
B1	PP3 battery		

Note. A complete kit (LW89W) of all the parts listed is available from Maplin Electronic Supplies Ltd for just £10.90 inc. VAT and P&P. The kit does not include batteries.

Car Digital Tacho contd from Page 13.

Calibration can be carried out against the mains frequency by using a transformer and bridge rectifier as shown in Figure 4 to provide a 100Hz signal. RV1 is adjusted to give a display reading of 30, or 36 if the mains supply frequency is 60Hz. The power supply section of an existing piece of equipment can be used in the same way if the rest of the circuitry is disconnected from the output of the bridge rectifier.

Fitting unit to car

After calibration, the unit is ready to be fitted to the car. It is impossible to give detailed fitting instructions for every car but the following notes may be helpful.

a) It is a good idea to try the unit in various positions for best readability, using adhesive tape, until you are satisfied.

b) Having decided on the best position use double-sided tape, adhesive pads or two pieces of velcro-tape, one glued to the unit and one to the car dashboard. All of these methods, of course, mean that the unit can be removed easily and the dashboard cleaned and left unmarked.

c) Alternatively, use self-tapping screws through on half of the case into the dashboard. This works well, but unless you can utilise existing screw holes you will be left with holes in the dashboard if you decide to remove the unit.

The three leads must pass into the engine compartment and it is important that they be protected by a rubber or plastic grommet. It may be possible to squeeze them through an existing cable entry or you may have to drill a new hole, but either way make sure they are protected.

Any suitable fused positive feed point and any adjacent earth can be used. Most cars have spare connecting points, usually with ¼" Lucar or blade type connectors for fitting auxiliary equipment, or you could use an in-line fuse holder and lamp fuse. The I/P lead must go to the 'CB' terminal on the iginition coil. If this is not marked and you cannot identify it from the owners manual ask your dealer. Coils have several types of connection, Lucar, screw terminals or push-on caps.

In any case make a professional connection using 'piggy-back' type Lucar connectors (which allow two connections on one terminal), solder tags or Scotchlok connectors. If in doubt consult your local automobile electricians. Fix the leads neatly to existing harnesses using tape or tie-wraps.

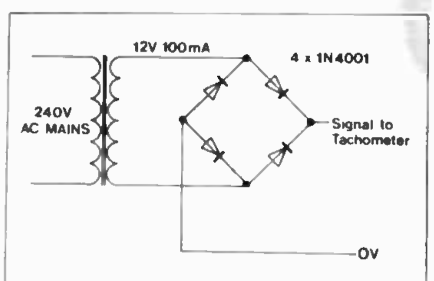


Figure 4. 100Hz calibration source.

PARTS LIST

Resistors — all 1% 0.4W metal film unless specified.			
R1,2	560R	2 off	(M560R)
R3	100k		(M100K)
R4-10	150R	7 off	(M150R)
R11	390k		(M390K)
R12	1k0		(M1K0)
R13-15	10k	3 off	(M10K)
RV1	100k Vert. S-min. preset		(WR74R)
Capacitors			
C1	100nF polyester		(BX76H)
C2	47nF polycarbonate		(WW37S)
C3	1uF 35V tantalum bead		(WW60Q)
C4	2n2 ceramic plate		(WX72P)
C5	1000uF 16V axial electrolytic		(FB82D)
C6	10nF disc ceramic		(YR73Q)
Semiconductors			
IC1	74LS221		(YF86T)
IC2	74C925		(QY08J)
IC3	NE555		(QH66W)
IC4	uA78L05AWC 5V 100mA regulator		(QL26D)
TR1-3	BC549	3 off	(QQ15R)
D1	4.7V 400mW zener diode		(QH06G)
D2	1N4001		(QL73Q)
D3	MAN 6740 Double-digit, common cathode display		(BY68Y)
Miscellaneous			
	8-pin DIL socket		(BL17V)
	16-pin DIL socket	2 off	(BL19V)
	Main PCB		(GA26D)
	Display PCB		(GA27E)
	1mm Veropins		(FL24B)
	Red display filter		(FR34M)
	Connection wire		(BL09K)

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Published by Maplin Electronic Supplies Ltd., P.O. Box 3, Rayleigh, Essex