

# electronics today

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## CIRCUIT DESIGN - A BLACK ART? 2 major features inside to break the spell!

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...AUDI...MPUTING...MUSIC...RADIO...ROBOTICS.

The RIAA equalisation is introduced by applying the negative feedback via a network with a frequency dependent impedance. However, since this stage relies on the presence of negative feedback to ensure a satisfactorily high input impedance, the input impedance will vary as a function of frequency. The cartridge, however, must be loaded by a constant resistance if cartridge impedance interaction is to be avoided. Furthermore, since the negative feedback is coupled to the complex output impedance of the cartridge via the base-emitter junction of Q1, the negative feedback and hence the frequency response of the stage can be affected by the cartridge itself. As a result this type of stage can suffer badly from cartridge impedance interaction.

## Several In

In the development of this design several input stage configurations were tested for noise, distortion and cartridge impedance interaction. When a medium-priced moving magnet cartridge was connected to a stage like that in Fig. 3, severe cartridge impedance interaction was evident. The frequency response of the preamplifier peaked above 2 dB at 13 kHz and rolled off rapidly above 15 kHz. The same cartridge when connected to our exhibited quite a good frequency response to beyond 20 kHz, and the frequency response graph obtained was identical to that when a FET buffer amp was placed between the cartridge and the input stage, indicating almost total lack of cartridge impedance interaction.

This is a result of the use of the separate linear gain stage formed by IC1 (Fig. 1) to isolate the cartridge from the RIAA equalisation.

The preamp conforms to the proposed RIAA equalisation in Fig. 2. The 75  $\mu$ s and 7950  $\mu$ s time constants are obtained by passive RC filters at the output of the first stage. Resistors R5, R6 and capacitor C3 form a simple 6 dB/octave low-pass filter with a -3 dB point at 2122 Hz, and

$$t = \frac{1}{2\pi f} = \frac{1}{2\pi(2122)} \doteq 75 \mu\text{s}.$$

Capacitor C4, together with resistors R7 and R8, form a 6 dB/octave high-pass filter with a -3 dB point at 20 Hz, which is equivalent to a 7950  $\mu$ s time constant. The two remaining time constants are introduced into the negative feedback of IC2 and are formed by the values of resistors R9, R10, R11 and capacitor C6.

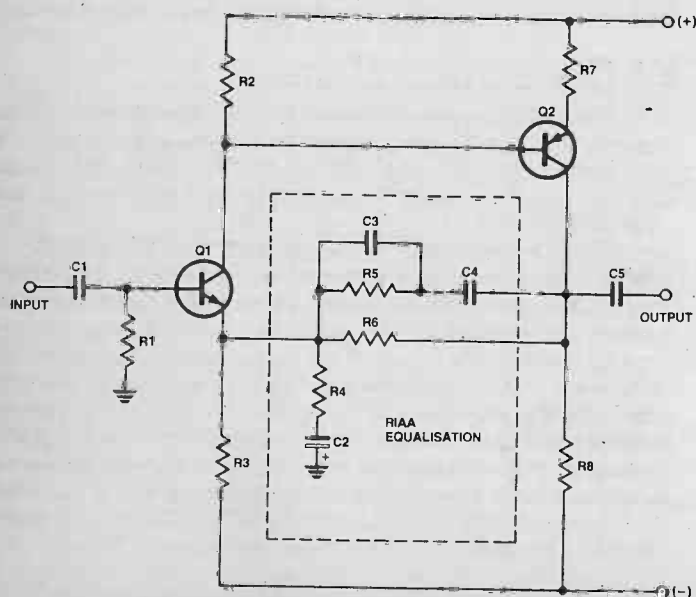


Fig. 3 Typical moving magnet input stage found in medium-priced amplifiers.

This method of generating the RIAA curve offers a number of advantages over the more conventional method.

Firstly there is low interaction between the different time constants, so that the RIAA curve can be optimised for a particular cartridge more easily by changing the resistor or capacitor values slightly. If the 75  $\mu$ s time constant is included in the negative feedback stage, the gain of the stage must decrease to unity at a suitably high frequency, so the stage must be compensated for unity gain to prevent instability. The NE5534AN is internally compensated for gains of 3 or above, so no additional compensation is required.

## Stage Fright

Another advantage of the two-stage approach is that the total gain necessary in the MM stage can be divided between the two stages, so more negative feedback is available for each stage. This will have the effect of decreasing non-linearities in the stages, provided the stages conform to the criteria for the avoidance of SID (slew-induced distortion) and amplitude overload. Fortunately, in the case of a phone input stage, both of these are limited by the recording medium. The RIAA standard sets a maximum recording velocity of 25 cm/sec, and most cartridges have output levels around the 1 mV/cm/sec figure. So maximum output levels from such a cartridge will be put in the order of 20-30 mV. Even the highest output cartridge produces signal voltages usually in the 5 mV/cm/sec range. Combining a worst case of, say 5 mV/cm/sec with the maximum allowable recording velocity of 25 cm/sec yields an output voltage of 125 mV. To ensure that the input stage cannot be overloaded we simply set the gain of these stages so that this maximum input signal cannot drive the output of the input stages into clipping. The NE5534AN is capable of driving to within 2 V of the supply voltage, so a supply voltage of  $\pm 15$ V gives the desired gain of around 75. We have divided this gain between the two input stages so that the first stage has a gain of 8.3 and the second stage a gain of 9 in the midband region (the actual gain of the second stage is of course a function of frequency due to the RIAA equalisation).

As a result the total harmonic distortion of this MM input stage is well under 0.001%. The actual measured distortion using an HP3580A spectrum analyser was around 0.0005% at 1 kHz. (At these distortion levels even the best distortion analysers are practically useless, since the distortion is well below the level of noise.) Similarly, intermodulation distortion (IMD) was measured at well below the 0.0001% figure.

## Noise

Another very important parameter for both MC and MM input stages is noise performance. Since an op-amp is used as the first stage of the MM input amp, we have only limited control over the noise performance of the stage. It is therefore essential that the op-amp used have excellent noise performance. In order to predict the necessary noise performance for a moving magnet input stage we must look at the sources of noise within the cartridge itself.

It can be shown from the laws of thermodynamics and statistical mechanics that every resistor generates noise. This noise is a result of the way nature works and is not caused by imperfection in a practical resistor (ie a perfect resistor will still generate thermal noise). This noise must be added to any signal dropped across the resistance. The equation for thermal noise is:

$$\bar{e}_n = \sqrt{4kTR\Delta f}$$

where

k = Boltzmann's constant

T = temperature in absolute units (K)

$\Delta$  = noise bandwidth (brickwall bandwidth)

R = resistance in ohms

$e_n$  = average noise voltage



# FEATURE : Pickup Amp Design

This equation predicts that thermal noise is raised by increasing resistance temperature or the bandwidth of the measuring equipment. So the frequency response of the apparatus used to determine thermal noise must be quoted if the figure is to be meaningful. Furthermore, the  $\Delta f$  here refers to a 'brickwall frequency response', not the usual half-power bandwidth, although for many purposes this is sufficiently accurate. To overcome this problem noise performance is often quoted in the form of total equivalent input noise and expressed in units of  $nV/\sqrt{Hz}$  ( $1 nV = 10^{-9}V$ ). This is justified by the equation for thermal noise, i.e.

$$\text{since } \bar{e}_n = \sqrt{(4kTR\Delta f)}$$

$$\text{then } \bar{e}_n = (\sqrt{\Delta f})(\sqrt{(4kTR)})$$

$$\text{or } \frac{\bar{e}_n}{\sqrt{\Delta f}} = \sqrt{(4kTR)}$$

So the ratio:  $\frac{\bar{e}_n}{\sqrt{\Delta f}}$

depends only on temperature and resistance, and this is just what we want. In order to get from this figure to an actual total equivalent noise figure we simply multiply by the square root of the bandwidth.

Most moving magnet cartridges have a coil resistance around 500 ohms. This resistance will generate thermal noise, so the cartridge itself limits the best possible signal-to-noise ratio. Using the equation for thermal noise we obtain for the noise of the cartridge:

$$\begin{aligned} \bar{e}_n &= \sqrt{(4 \times 1.37 \times 10^{-23} \times 290 \times 500)} \\ &\frac{\text{Hz}}{\text{Hz}} \\ &\text{(assuming temperature of resistor is } \\ &\approx 290K). \\ \text{i.e. } \bar{e}_n &= 2.8 \times 10^{-9} \\ &\frac{\text{V}}{\sqrt{\text{Hz}}} \\ \text{i.e. } \bar{e}_n &= 2.8 \text{ nV}/\sqrt{\text{Hz}}. \end{aligned}$$

We can express this in more familiar terms by converting the cartridge noise figures into a signal-to-noise ratio figure. In audio we can regard the bandwidth in question to be around 20 kHz, i.e.  $\sqrt{\Delta f} = 140$ , and  $140 \times 2.8 \text{ nV}/\sqrt{\text{Hz}} = 392 \text{ nV}$ . If the average output level of the cartridge is around 5 mV, the signal-to-noise ratio is given by:

$$20 \log \frac{5 \times 10^{-3}}{392 \times 10^{-9}} \approx 82 \text{ dB}$$

This figure represents the best signal-to-noise ratio possible with most moving magnet cartridges, since this is due to noise generated within the cartridge itself. A well-designed input stage should approach this noise figure as closely as possible without sacrificing performance in other equally important parameters such as distortion and frequency response.

The noise generated by an active device is determined by a number of factors, the most important of which is the current flowing through the device. However, since we have elected to use a high-quality operational amplifier for the input stage, we have no control over device current. All we can do is choose a low-noise op-amp and avoid degrading its noise figure as much as possible. The NE5534AN has a recommended equivalent input noise voltage around  $4 \text{ nV}/\sqrt{\text{Hz}}$ , only 3 dB above the noise generated by the cartridge itself! In order not to degrade this figure we must keep all resistances in series with the cartridge as low as possible. Any additional resistance will generate a ther-

mal noise voltage of its own, which must be added vectorally to that generated by the cartridge. From the basic equation of thermal noise generated by two individual resistors  $R_1$  and  $R_2$  for example, we obtain:

$$\begin{aligned} \bar{e}_{n1} &= \sqrt{(4kTR_1\Delta f)} \text{ and} \\ \bar{e}_{n2} &= \sqrt{(4kTR_2\Delta f)} \end{aligned}$$

Here we assume that both resistances are at the same temperature. Since these noise voltages are not correlated (ie they consist of 'randomly' changing voltage) we add them using the vector sum:

$$\text{i.e. } \bar{e}_{nT}^2 = \bar{e}_{n1}^2 + \bar{e}_{n2}^2$$

where  $\bar{e}_{nT}^2$  is the square of the total equivalent noise voltage.

$$\text{Therefore } \bar{e}_{nT}^2 = 4kT\Delta f(R_1 + R_2)$$

$$\text{or } \bar{e}_{nT} = \sqrt{(4kT\Delta f(R_1 + R_2))}$$

If  $R_1$  now represents the cartridge resistance and  $R_2$  the value of an added resistance equal to the value of  $R_1$ , we get:

$$\begin{aligned} \bar{e}_{nT} &= \sqrt{(4kT\Delta f(2R_1))} = \\ &= \sqrt{2} \sqrt{(4kT\Delta fR_1)} \end{aligned}$$

$$\text{or } \bar{e}_{nT} = 1.4 \bar{e}_{nT'}$$

equivalent to a 3 dB decrease in the signal-to-noise ratio.

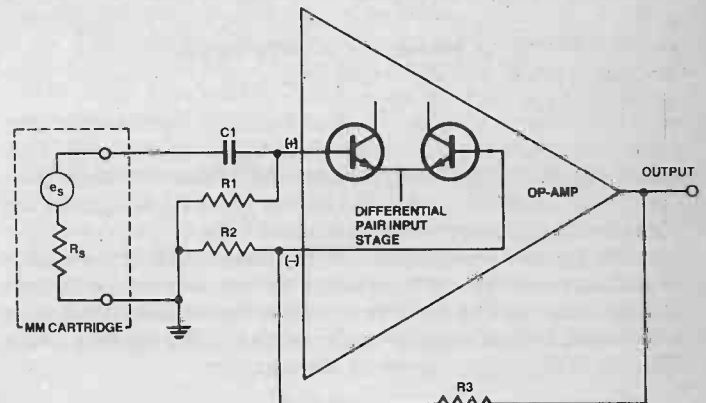


Fig. 4 Standard technique for connecting an op-amp to a signal source.

## Generation Game

Figure 4 shows the standard technique for connecting an op-amp to a signal generator such as a moving magnet cartridge. Most op-amps, and certainly the 5534, have input stages that consist of a differential pair, providing both inverting and non-inverting inputs.

The effective signal voltage generator of the cartridge is represented by  $e_s$  and the cartridge resistance by  $R_s$ . Resistor  $R_1$  in this case would be 47k, so that the cartridge would have the correct load resistance. (The input impedance of the op-amp is very high and can be ignored for this discussion.) Capacitor  $C_1$  prevents any DC current flowing through the cartridge from the non-inverting input. Since the combination of  $R_1$  and  $C_1$  forms a 6 dB/octave high-pass filter, the value of  $C_1$  would ordinarily be chosen so that the resulting  $-3 \text{ dB}$  point was well below the audio spectrum, around 5 Hz for example. This will occur when the impedance of  $C_1$  is equal to that of  $R_1$ , i.e. 47k. Since the reactance of the capacitor is given by the equation:

$$X_c = \frac{1}{2\pi fC}$$

we have:

$$C = \frac{1}{2\pi f X_c}$$

In this case  $C = \frac{1}{2\pi \times 5 \times 47 \times 10^3}$

$$\div 6.77 \times 10^{-7} \text{ Farads.}$$

So to obtain an adequately flat frequency response a suitable value for C1 would be 680 nF (0.68uF), which is convenient.

When noise considerations are taken into account, however, this value is entirely unsuitable. The increasing impedance of C1 at low frequencies, while not sufficient to cause gross frequency response errors, will seriously degrade the noise performance of the stage. At sufficiently low frequencies the impedance seen by the non-inverting input will be simply the value of R1. Using the equation for thermal noise given earlier, we can calculate the resulting signal-to-noise ratio. Since

$$\overline{e_n} = \sqrt{(4kT\Delta f R_1)},$$

$$\overline{e_n} = \sqrt{(4 \times 1.37 \times 10^{-23} \times 290 \times 20 \times 10^3 \times 47 \times 10^3)} = 3.87 \mu V,$$

only 62 dB below 5 mV.

Furthermore, since the input stage is a noise generator, a low source impedance is necessary to minimise the resulting noise at the output of the op-amp. To overcome this problem we increase the value of C1 so that at worst its impedance at, say, 3 Hz is comparable to that of the cartridge ie

$$C = \frac{1}{2\pi \times 3 \times 500} = 106 \times 10^{-6} F.$$

So a value around 100 uF should suffice. Notice that this capacitor would have to be an electrolytic or tantalum. Tantalum capacitors are not recommended, however, since their capacitance can be modulated by the input signal, producing considerable distortion at low frequencies.

The value of resistor R2 must also be low, so that the source impedance of the inverting input of the op-amp can be kept as low as possible. The limitation here is due to the minimum load impedance allowable on the output of the op-amp. Since the gain of the stage is given by the equation:

$$A_v = \frac{R2 + R3}{R2},$$

the ratio of R2 and R3 is determined by the desired voltage gain. At the same time, however, the total resistance R2 + R3 represents the load on the output stage of the op-amp. Since this must not be less than a certain specified resistance, determined by the individual op-amp used, a minimum value of R2 is predicated. In the input stage, for example, the required voltage gain in the first stage is around 8.3, so:

$$\frac{R2 + R3}{R2} = 8.3$$

The NE5534AN has a measured minimum load impedance of 600 ohms, and for minimum distortion it is desirable to increase this slightly, for example to around 1k2. Therefore:

$$\frac{1k2}{R2} = 8.3 \text{ or } R2 \div 144R$$

A suitable value for R2 would be 120 ohms, making R3 1k0 to give the required voltage gain. Fortunately this value for R2 is low enough not to have significant effect on the noise performance.

Similar measures must be adopted around the second stage. At low frequencies the non-inverting input of IC2 (Fig.1) has an input source resistance determined by R7 and R8, ie around 8k. The noise performance of the second stage would be improved if this value could be decreased. Unfortunately this would entail increasing the value of C4, which is not practical since this capacitor must be a greencap if the preamp is to conform accurately to the RIAA curve. This is not really a problem, however, since the voltage gain in the first stage increases the signal voltage at the input of IC2 to around 40 mV for 5 mV input signal, ensuring a sufficiently good signal-to-noise ratio in the second stage.

## The Moving Coil Input Stage

The subject of noise performance is particularly important for a moving coil input stage. The moving coil cartridge works on exactly the same principle as the moving magnet. The signal voltages produced are the result of relative motion between a coil of wire and a magnetic flux. In this case, however, the magnet assembly is mounted rigidly to the cartridge body and the coils are mounted on the cantilever assembly; hence the name 'moving coil'.

In order for the total mass and therefore the inertia of the stylus/cantilever system to be kept to a minimum, the coils are made with very fine wire and a small number of turns. Typical output voltages for moving coil cartridges vary widely from one manufacturer to another, but a figure of 40 uV/cm/sec is probably a reasonable compromise. A gain of 25 is therefore required to boost this voltage to that of a typical moving magnet cartridge. Once again we can calculate the best possible signal-to-noise ratio for a moving coil cartridge based on its thermal noise. The coil resistance of a moving coil cartridge with an output of 40 uV/cm/sec would be approximately 20 ohms (although this figure can vary widely, typically 5-50 ohms).

From the equation for thermal noise we obtain:

$$\frac{\overline{e_n}}{\sqrt{\text{Hz}}} = \sqrt{(4kTR)},$$

ie

$$\frac{\overline{e_n}}{\sqrt{\text{Hz}}} = \sqrt{(4 \times 1.37 \times 10^{-23} \times 290 \times 20)}$$

$$\div 0.56 \text{ nV}/\sqrt{\text{Hz}}.$$

The total noise over a 20 kHz noise bandwidth is therefore:

$$0.56 \text{ nV} \times \sqrt{(20 \times 10^3)}$$

i.e:  $0.56 \text{ nV} \times 140 \div 78 \text{ nV}.$

Since the cartridge output voltage will be around 40 uV/cm/sec x 5 cm/sec, ie 200 uV for a recording velocity of 5 cm/sec, the resulting signal-to-noise ratio will be:

$$20 \log \frac{200 \times 10^{-6}}{78 \times 10^{-9}}$$

or around 68 dB unweighted.

This figure is only approximate, of course, but it is roughly correct and represents the best possible signal-to-noise ratio with a moving coil cartridge. The object is to design a preamplifier that will approach this noise figure and maintain a flat frequency response, low distortion and constant resistive input impedance. At these noise levels we cannot use an NE5534AN in a circuit like the MM input stage. The total equivalent input noise in that case was around 4 nV/ $\sqrt{\text{Hz}}$ , ie 560 nV. The resulting signal-to-noise ratio would be only 51 dB with respect to an input signal of 200 uV.

In order to achieve a satisfactory noise performance it is necessary to look at the various sources of noise in bipolar transistors and decrease the total equivalent input noise through optimum biasing of the input stage and choice of the first transistor.

## Noisy Thermals

One source of noise in the transistor is of course thermal noise. We saw before that to minimise thermal noise it was necessary to ensure a low source resistance over as broad a frequency range as possible. In order to do this for the MC stage the total resistance in series with the source must be kept to a similar value to the source resistance, ie around 10 or 20 ohms, depending on the cartridge.

The problem is that the resistance of the base-emitter junction of most bipolar transistors, called the base spreading resistance, is usually much higher than this. One solution is to use a large number of low-noise transistors in parallel to form the input transistor, thus decreasing the base spreading resistance.

Another solution is to use a power transistor, such as a 2N3055, as the input transistor, and the results using this method can be quite good!

The third alternative and the one we elected to use in this design, is to make use of an exceptional matched pair produced by National Semiconductor. This device, the LM394, has a low base resistance, very low noise and high  $h_{FE}$  of around 500.

Another source of noise in bipolar transistors is shot noise or base current noise. This is a white noise generator (ie the average amplitude of the noise current is constant with frequency), but the noise is increased if emitter current is increased. The base resistance, however, is also a function of the current flowing in the emitter, and is given roughly by the equation:

$$r_b = \frac{26}{I_E(\text{mA})}$$

The resistance of the base decreases with increasing emitter current, so noise voltage produced by thermal noise across the base resistance is decreased by increasing the emitter current. In a bipolar transistor, therefore, we have two distinct sources of noise, one increasing with the emitter current while the other decreases. For this reason an optimum emitter current exists which represents the best compromise between these two noise sources. With an LM394 operated from source resistances typical of moving coil cartridges, the optimum emitter current is around 8 mA, much higher than would normally be used in an input stage. The result, however, is a very low value of input noise for source resistances around 10 ohms.

The complete circuit diagram for the moving input stage is shown in Fig. 5. The collectors of the LM394 are connected to the input of an NE5534, which functions as a high-gain differential amplifier providing adequate open loop gain to ensure low distortion and a flat frequency response when negative feedback is applied. The input choke is used to minimise the stage's susceptibility to RF noise.

The input impedance of the stage is determined by the parallel combination of R1 and R2, around 65 ohms for the values shown. This should be suitable for most moving coil cartridges, but is easily changed if required. The DC operating point of the LM394 is determined by the constant current source formed by Q1, Q2, R3 and R6. So the current in resistor R2 is determined by this constant current source and the DC current gain of the LM394. Hence the value of R2 can be increased, in order to increase the input impedance, over a fairly wide range of values without affecting the operation of the circuit.

## Silent Coupling

Once again the input coupling capacitor C4 is used to prevent DC current flowing through the cartridge. Capacitor C4 is

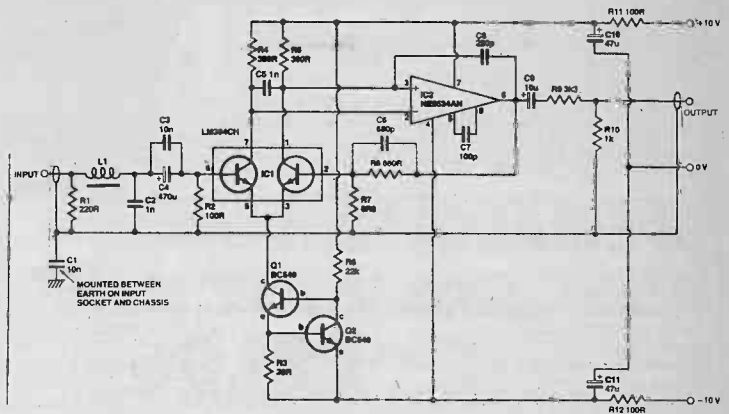


Fig. 5 Circuit of one channel of the moving coil input stage.

shunted by C3, a 10nF capacitor, so that the base of the first transistor in the LM394 is decoupled for RF, through C2. Capacitor C2 represents a shunt capacitance to ensure correct loading of the moving coil cartridge. The value shown should be suitable for most cartridges, but can be changed for optimisation with any particular cartridge.

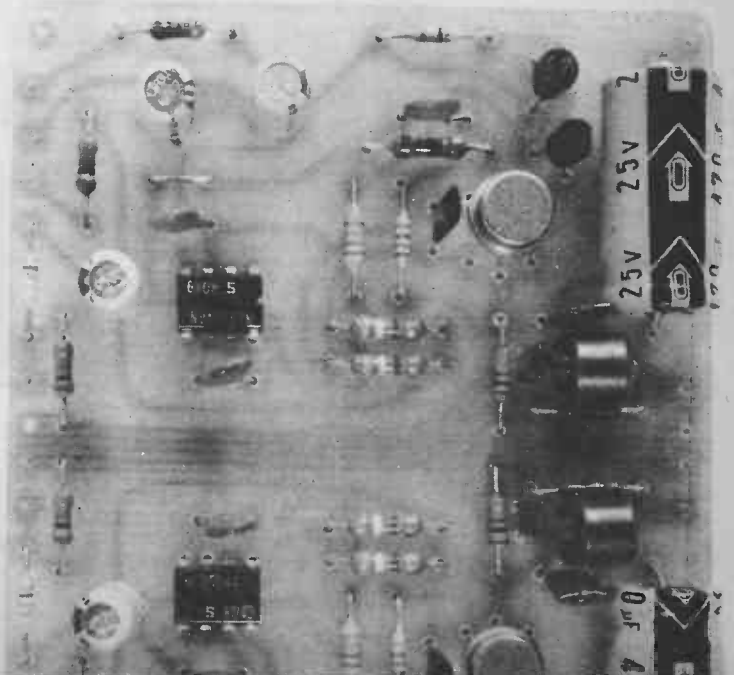
To prevent loading the 5534A, the feedback resistor R8 is kept above 600R, ie 680R. Resistor R7 effectively increases with the cartridge and must be kept as low as possible for best noise performance. The value of 6R8 chosen gives the stage a gain of around 100, which is too high. This is corrected, however, by a simple passive voltage divider at the output, formed by R9 and R10. Capacitor C9 doubles as a feedback isolation capacitor to ensure that reactive components in the load cannot cause a phase shift sufficient to cause oscillation.

The noise performance of the stage is extremely good. The total equivalent input noise was measured at 83 nV over a 20 kHz noise bandwidth. This equivalent to 0.6 nV/√Hz or a signal-to-noise ratio of 68 dB with respect to an input signal voltage of 200 μV. This might sound like only an average noise figure compared to that attainable with the moving magnet preamp, but it should be remembered that the noise generated by the cartridge itself is of this order of magnitude!

A final point worth mentioning here is that all the noise figures quoted in this article are flat or unweighted measurements!

ETI

Next month we put the theory into practice with a project based on these principles. Quite simply, it will be the best preamplifier input stage ever published.



# PARKING METER TIMER



Living in fear of parking tickets is a thing of the past now that the ETI Parking Meter Timer is here. It'll remind you to get back to the car before the wardens do. Design and development by Rory Holmes.

After much design research ETI have come up with the best parking meter timer that has ever been offered to the home constructor. It is a truly functional design, and due to a careful selection of circuit components it is built to fit an extremely small and readily available case which can be clipped neatly on to a key ring. No awkward miniature switches are required; a single pin head touch switch allows the selection of all four time periods, including on-off control and an alarm test option.

The time periods are arranged in 20 minute increments, giving 20, 40, 60, or 80 minute options to suit modern

parking meters. Three minutes are allowed for each 20 minute period for returning to the car, so giving more time the further away you go. The unit provides a high efficiency pulsed tone alarm using the Toko slim-line transducer.

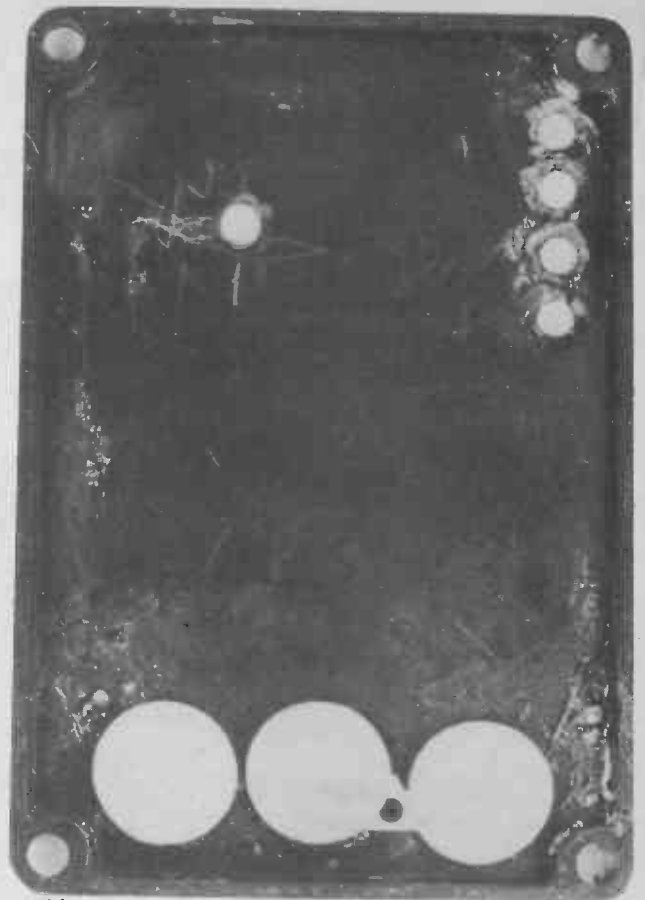
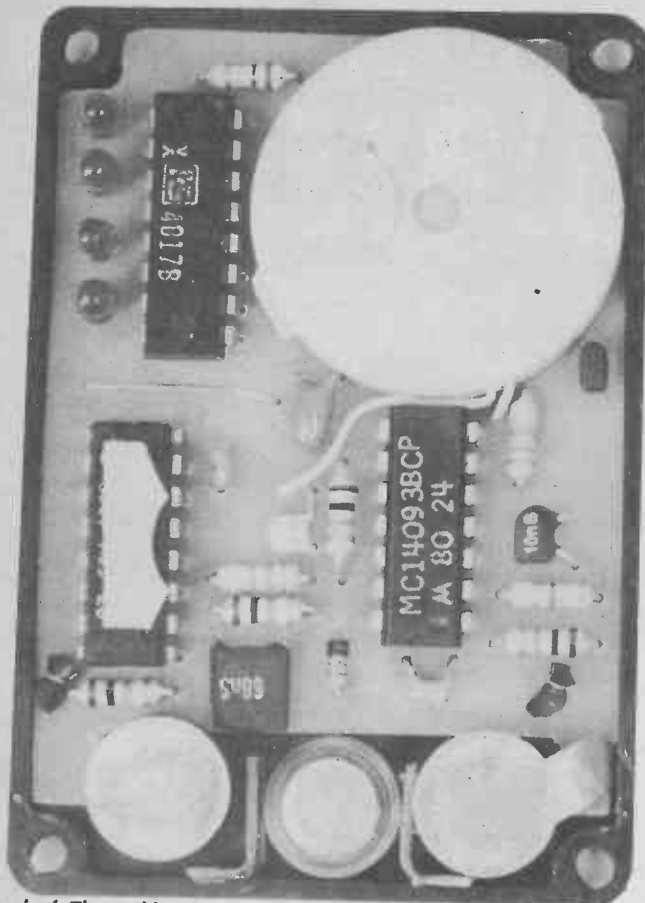
By the use of only three CMOS chips and special power control circuitry, the current consumption when not in use is completely negligible. Power is provided by three ordinary hearing-aid batteries (available from most chemists). The LED display for indicating the time period selected also includes automatic blanking for further battery economy.

## Construction

The Parking Meter Timer has been specifically designed to fit into a very small case. In fact it's not a standard case at all, but two lids from the smallest Vero potting box fitted back-to-back. We're trying to think of a project that uses two lidless boxes! The PCB is cut to fit inside the bottom lid with a cut-out to hold the three batteries in a line; see the overlay diagram and photographs.

The board should be assembled first, checking component orientation from the overlay and carefully observing the following points. When mounting the components on the





Overleaf: The parking Meter Timer being modelled by the lovely Sonia. Provided you're careful with the construction, everything can be fitted into the lid of a small Vero potting box . . .

. . . with a second lid completing 'case'. Countersink the holes for the LEDs as shown, so as to get a neat finish, and remember to insulate the battery sides from one another.

board it is essential to crop the leads very close to the tracks before soldering, to reduce the resulting solder lump. Apart from C3 and C5, all the passive components need to be mounted flat against the board. The LEDs (high efficiency 2 mm types) must have their leads bent apart slightly before they are pushed through the PCB holes. They are soldered in (observing polarity) with a 1 mm gap between the LED base and board. The piezo transducer is supplied with two mounting flanges; these need to be cut off before the transducer can be stuck to the PCB. This should only be done after the surrounding components are soldered in and with the PCB in the case, to ensure that the transducer clears the case rim. The two leads are then soldered to the board at the points marked on the overlay. Sockets must not be used on the ICs (there isn't enough room under the lid), but remember the static precautions for the CMOS chips.

### Make Contact

The touch contacts are formed by two ordinary pins soldered vertically into the PCB at the points marked. The pin heads should be above the component side and at a height to just protrude through two holes drilled in the front lid. The timing resistor R2 should initially be left off the PCB (its

exact value is set during testing). The batteries used are ordinary 1V4 hearing-aid types; they are held in place by the PCB cut-out and the two case halves. We used suitably bent pieces of copper contact strip soldered to the PCB for connecting our batteries so they could be removed. Two additional copper strips were glued to the inside of each lid, to connect the batteries in series when sandwiched by the lids. An alternative method is to solder the batteries permanently in circuit with thin wires. Remember that the sides of

the batteries must not touch each other; spacers made of cardboard or plastic will keep them apart.

### Testing

Initially a 2M0 pot should be wired with flying leads to the PCB connections for R2, and set at about 1M0 (half travel). 4V2 can now be applied to the circuit (at the supply pads by the battery cut-out), either

### PARTS LIST

#### Resistors (all 1/4 W, 5%)

R1,8	4M7
R2*,3,4,7	1M0
R5	6M8
R6	100k
* See text	

#### Capacitors

C1	68n ceramic
C2,3	2n2 ceramic
C4,5	10n ceramic
C6	470p ceramic
C7	3u3 16 V tantalum

#### Semiconductors

IC1	4060B
IC2	4017B
IC3	4093B
D1	1N4148
LED1-4	2 mm red LEDs

#### Miscellaneous

TX1	PB-2720
PCB (see Buylines); Veropins; three off 1V4 hearing-aid batteries; two off potting box lids (Vero order ref. 75-1413E).	

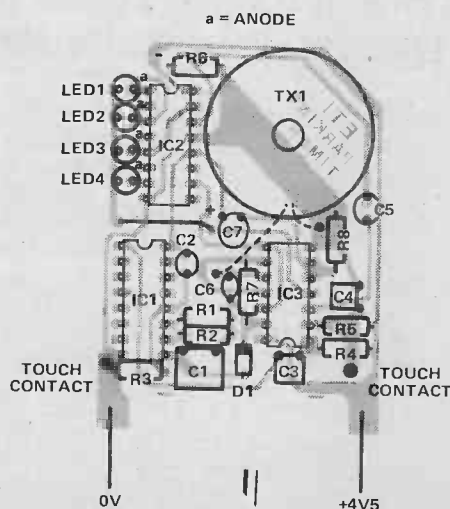


Fig. 1 Component overlay for the ETI Parking Meter Timer. You can't use sockets, so take care when soldering the ICs.

# PROJECT : Parking Meter Timer

## HOW IT WORKS

from a variable supply, or by wiring three spare batteries together. The LEDs should now illuminate sequentially each time the touch pins are bridged by two fingers (remember they will turn off after about 20 seconds). On the fifth touch the alarm should sound, and on the sixth the unit is in the off state. The basic time period can now be set for exactly 17 minutes. Connect a voltmeter to the  $Q_9$  output of IC1 (pin 13); after turning on to the first time period the meter will be at 0 V. It should switch to 4 V after 1 minute 4 seconds, and then back to zero after the same amount of time. By adjusting the pot and checking with a stopwatch, this time can be set exactly. The pot is then disconnected and its resistance measured; a standard resistor closest to this value can then be soldered into the PCB.

The case is assembled with adhesive pads stuck inside the lids to firmly sandwich the PCB and batteries. Holes must be drilled for the LEDs and we recommend that these are countersunk from the inside of the lid. A hole drilled directly above the transducer will increase the volume. 6 BA bolts about 5/16" long are ideal for securing the case halves; if one bolt at the transducer corner of the case is left out, the timer can then be fitted directly onto a key ring!

The heart of this circuit is IC2, a 4017 decimal counter decoder; it is used for a number of different functions in a rather unusual way. Each of the decimal decoded outputs goes high in succession as the counter receives clock pulses on pin 14. The '6' output (pin 5) is wired to reset pin 15, thus the counter will have six different states and each clock pulse will shift it to the next state, counting from 0 to 5 and back to 0 again. The 5 output is not shown on the circuit diagram because it is not actually used; however, the divide-by-10 output on pin 12 (which is high for states 0 through 4 and low for states 5 to 9) is used, and in this configuration for IC2 it will only be low for state 5.

The counter is used both for counting the timing pulses generated by IC1, and selecting and setting the number of count periods before the alarm goes off. IC1 (CD4060B) provides the timing pulses for IC2; it consists of an internal RC-controlled oscillator followed by 14 stages of binary division. The oscillator is set to a frequency of 4 Hz by R2 and C1. When IC2 moves from state 5 to 0 (ie when switching on), IC1 is powered up from the divide-by-10 line of IC2 and it receives a positive reset pulse to pin 12 from C2 and R3. It then starts counting and provides a negative-going edge on the  $Q_{12}$  output every 17 minutes. This  $Q_{12}$  output is passed as a negative pulse through C3 and R4 to one input of the NAND gate IC3a; the output of IC3a goes briefly high and a slightly shorter positive pulse is applied through C5 to the clock input of IC2. Thus IC2 moves one count state every 17 minutes.

When IC2 reaches the fourth state the pin 10 output goes high and is used to enable the alarm. The alarm is a standard gated astable built around IC3a and b, driving an acoustic transducer. One input from each gate is wired to the '4' output of IC2. When this line goes high the oscillator is enabled and produces a tone pulsed at about 4 Hz. This modulation is derived from the master timing clock via D1, which swit-

ches the oscillator on and off. If left unattended it continues sounding for a further 17 minutes until IC2 receives another clock pulse from the timing generator IC1, at which point IC2 moves to state '5', turning off the alarm and powering down the rest of the circuit (the cycle can only be started again with a clock pulse from the touch switch).

The touch switch, built around IC3a, will also clock IC2 to select the number of 17 minute periods before the alarm sounds. IC3a is a NAND Schmitt trigger, with both its inputs normally held high by R4 and R5. Its output is thus low and C5 is discharged through R6. The pin 2 input is taken low when the touch pins are bridged by the skin resistance; this switching action is debounced by C4, and the output of IC3a will now go high, supplying a brief positive pulse through C5 to clock IC2. Thus the touch switch is used to turn the circuit on and initially set the number of time periods; thereafter clock pulses are supplied by the time generator.

IC2 also acts as the LED display driver for indicating the time period selected. The LED anodes are connected to the decoded output lines of IC2. When a state is selected via the touch switch, its output line goes high and applies a positive voltage to the corresponding LED. The cathode driving is arranged so they will only remain on for about half a minute after IC2 switches out of state '5'. As IC2 switches from 5 to 0 the divide-by-10 line goes from low to high and the positive end of C7 (which was previously discharged) is taken to the positive rail. This puts a high logic level on IC3d's inputs which drives its output low, thus providing a current-limited ground return for the LED which is selected. Meanwhile C7 is slowly charged up via R8 providing a long time constant of 20 seconds. When the voltage at the negative end of C7 goes below IC3d's switching threshold, the output of the Schmitt gate will revert high again thus extinguishing the LED.

## BUYLINES

ETI

Apart from the PCB, which is available from our PCB Service as usual (see page 98), there should be no problems obtaining any of the parts for this project from the usual mail order sources.

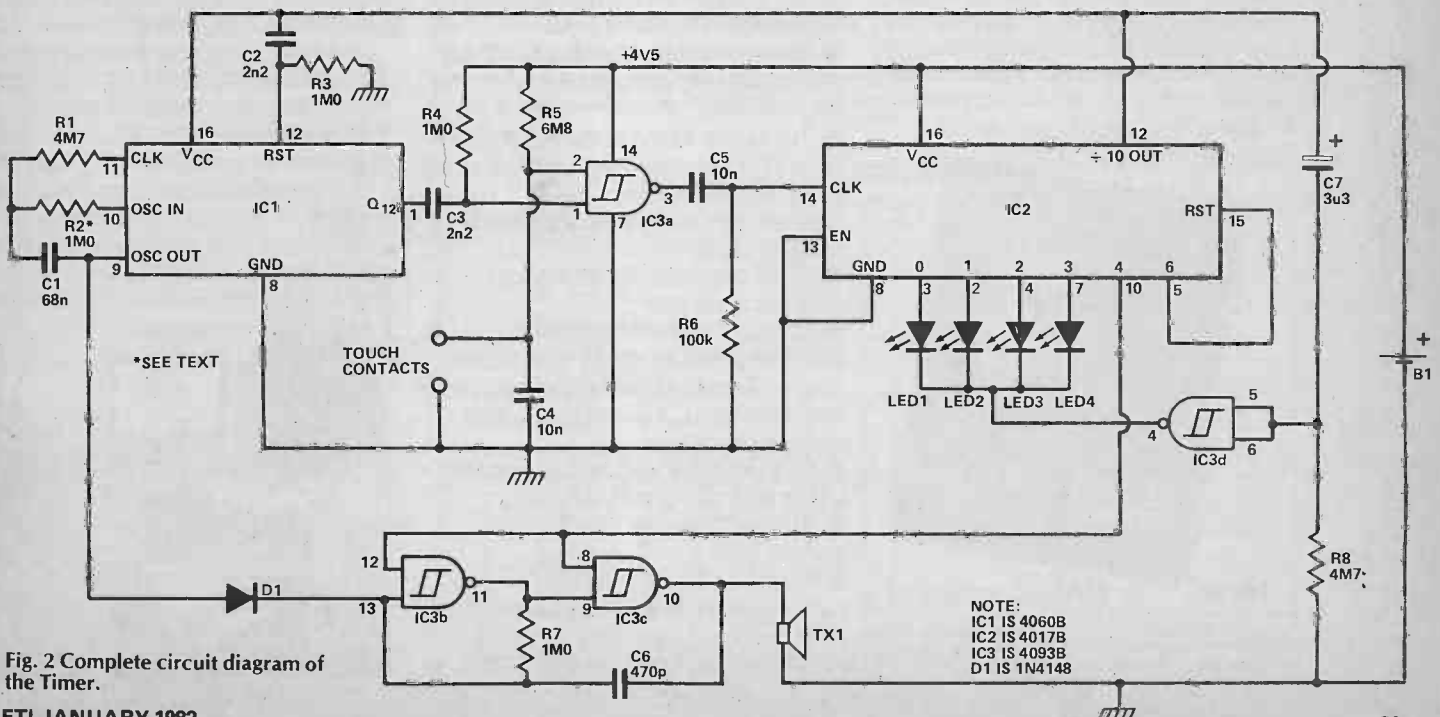
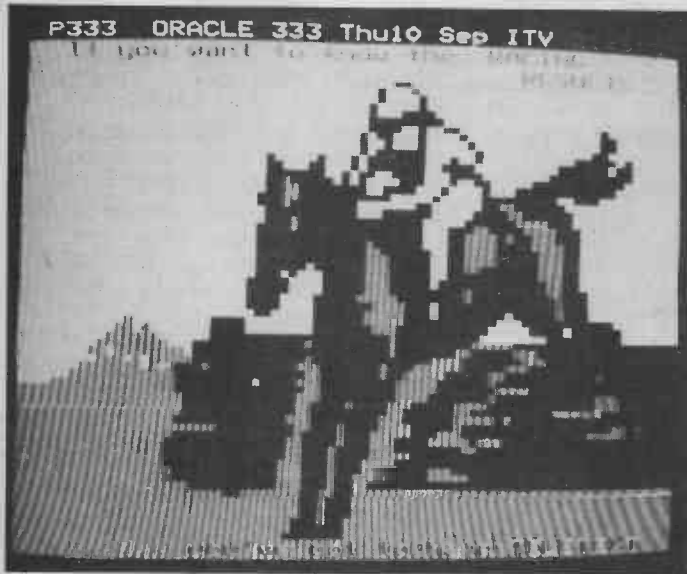


Fig. 2 Complete circuit diagram of the Timer.



# TELETEXT EXPLAINED

Take a trip inside your TV with Vivian Capel and find out how the Teletext information makes its way onto your screen.



Over the last few months there has been much publicity for the Teletext services of the BBC and ITA, called Ceefax and Oracle respectively, although the first experimental transmissions began as long ago as March 1973. At first, the two services differed in technical details, but commendably, these were mutually resolved and in the following year, April 1974, the standards were unified and a service was launched in September of the same year.

As most readers will already be aware, the service consists of magazines of up to 100 pages, each of which can be selected by the viewer by means of a keypad. Once selected, the page can be held on the screen indefinitely, yet can be updated if additional information is added in the transmission. Though primarily of written material, pages can also contain rudimentary graphics and illustrations. Lettering can be of different colours and sizes.

The most remarkable feature of Teletext is that all this can be transmitted along with the normal television service without interfering with it in any way and without using extra bandwidth. Thus it is compatible with existing receivers, most of which can receive Teletext with the addition of a suitable decoder.

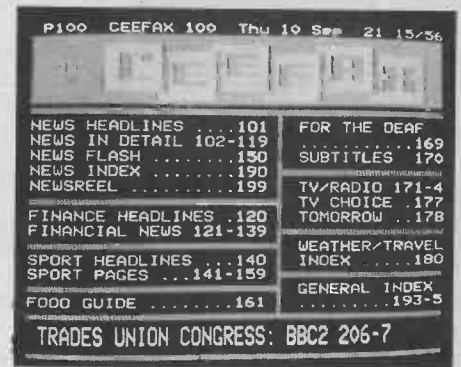
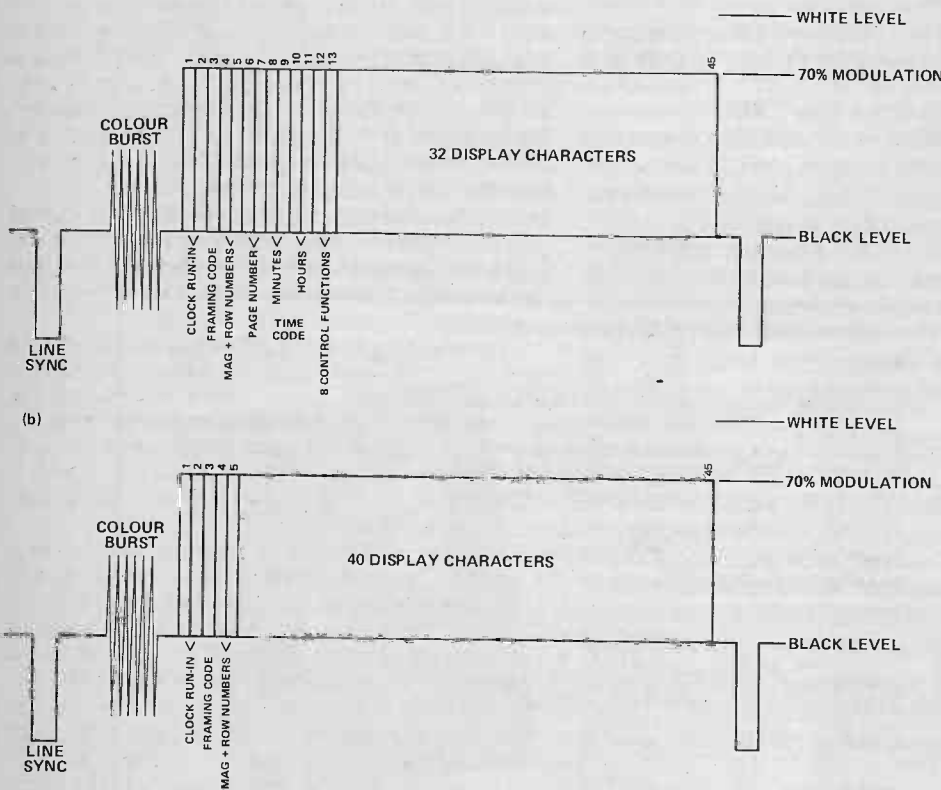


Fig. 1 (a) Composition of a TV scanning line showing the make-up of the 45 Teletext bytes for a header row. The first 13 bytes are control bytes as indicated. (b) A normal Teletext row. The first five bytes are the same as for the header, and are followed by 40 display characters.

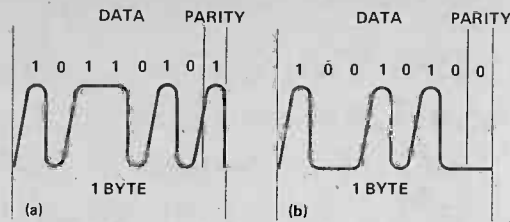


Fig. 2 (a) An eight-bit character byte of raised cosine pulses. Seven bits carry the data code, the eighth being a parity protection bit. If the number of bits at logic 1 is even, the parity bit is also 1 to give a total odd number. In (b), the number of parity bits is already an odd number, so the parity bit is 0.

So how is it done? Well, the 625 lines that are transmitted to make up each complete television frame are not all used to convey picture information. The frame is divided into two fields which are interlaced so that the scanning lines of each fall between those of the next. This greatly reduces flicker. However, there is a gap or margin at the top and bottom of each field which totals 25 lines; it is made up of  $22\frac{1}{2}$  lines at the top and 3 at the bottom of the first field, and 22 at the top and  $2\frac{1}{2}$  at the bottom of the second. If this seems odd, remember that the bottom of the first field runs on to the top of the second and the bottom of the second to the first of the next frame, thus there are always 25 lines between fields.

The purpose of these blank lines is to ensure that the flyback lines from bottom to top of the frame are blanked out and also that the time bases have settled down nicely after the start of a new scan before any actual picture information appears. They do, however, offer the opportunity of carrying extra information which is not directly intended for reproduction on the viewing screen. For example, lines 19 and 20, and 332 and 333 on alternate fields carry certain transmitter test signals.

It is on the previous lines (that is 17 and 18) and 330 and 331, that the Teletext signals are carried. If the height of a normal TV picture is reduced so that black bands appear at the top and bottom, the Teletext signals can be clearly seen as a row of dots and dashes which are continuously changing form.

## Teletext Pages

The pages are not transmitted as a normal TV picture by means of a video signal with each part of the displayed page being successively built up by the scanning lines. This would take not two, but all of the lines which are transmitted, and furthermore, it would not be possible to freeze a page, anymore than a frame of a normal TV programme can be frozen.

Instead, each character is transmitted as a separate and complete signal consisting of an 8-bit binary word or byte. The 1 or high bits correspond to about 70% of peak-white modulation of the carrier, while the 0 or low bits are represented by the black level. Bit pulses are not square in form and do not need to be; actually an extended video bandwidth would be necessary to handle square waves of the bit frequency which is 6.9375 MHz. The waveform is described as a raised cosine.

Each page row consists of 45 bytes, and as 5 bytes are required for synchronizing and address purposes in a normal row, this leaves 40 for the display characters. There is a maximum of 24 rows to a page so this means that up to 960 characters can be displayed on each page.

Following the identification coding for page one, each of the 24 rows are transmitted consecutively, whereupon the code for page two follows succeeded by its 24 rows, and so on until all the pages in the magazine have been transmitted. Then page one is transmitted again and all are repeated as if on an endless belt.

When the user keys a particular page the decoder waits until the appropriate identification code comes round again, and then loads all the following information into a memory bank until the code for the next page is received, when it ceases storing the signal. From the memory, the data signal is fed to the alphanumeric generators which produce a video signal to reproduce the characters on the receiver's display tube. The generators will continue to scan the memory circuits until the unit is switched off or another page is selected. Thus the page is displayed continuously, irrespective of what is being transmit-

ted subsequently. But there is a provision whereby information on the same page can be changed if following versions of the page are altered with fresh information.

## Access Time

So we have a bird's eye view of the basic principles. We will now take a closer look at the details. First, the access time. This is the time that elapses from when the page is keyed to when it actually appears on the screen. It is obviously a matter of chance - like waiting for a bus - it depends on whether the one you want is due or has just passed. So what is the maximum time you may have to wait?

One complete page row is transmitted during each TV line, so as two lines per field carry Teletext signals, a full 24-row page is transmitted during twelve fields, which at a field frequency of 50 Hz is 0.24 seconds. If 100 pages are contained in the magazine, the complete magazine cycle will take 24 seconds. So if you have just missed the required page, that is the maximum time it will take for it to come around again.

However, while each row takes the same time to transmit - whether full or not - the blank spaces each requiring a character byte, this is not the case with the pages. Any that have less than 24 rows are transmitted more quickly. Also, not all of the 100 pages are used in each magazine at present. Therefore the complete cycle will be rather less than 24 seconds and, of course, you will not just miss the chosen one every time. So average access time should be under 12 seconds.

Extra pages or magazines mean longer access times as long as the transmitted data is restricted to two television lines. It is possible to use more lines (up to 16 in each field) and this would extend the capacity to eight times its present maximum without adding to the access time. It is even possible to send different versions of the same page under differing time codes to give further extension. As there are now three, and shortly four, channels available, the potential of information available via teletext is quite considerable.

## Coding

Each normal row consists, as we have seen, of 45 bytes. The first two of these is a series of alternating 1 and 0 bits there being 16 in all. Thus a pulse train at the bit frequency of 6.9375 MHz is produced, and this serves to synchronize the decoder's data clock which detects and recovers the individual bits. Readers who are familiar with colour TV principles will recognise this as being similar to the colour-burst occurring after each line sync pulse that keeps the colour oscillator in step.

One form of clock circuit is a high-gain tuned circuit resonating at the bit frequency and which is set into oscillation by the incoming stream of pulses. The phase of the clock output wave form is delayed so that the peaks occur about the middle of each bit.

The third byte is known as the Start or Framing Code. As each byte in the sequence follows its predecessor without a break, the decoder must have some means of determining where to start. Again referring to television as an example, the scanning lines must all start at the same point, and to ensure this, line sync pulses are inserted. Without them, the screen is filled with a meaningless jumble of lines which can be produced if the line-hold control is misadjusted.

The Framing Code performs the same function, defining the start of the next byte, so keeping the detector circuits in step until the end of the row when, after the next clock run-in bytes, a further framing code byte is received. This byte takes the form 11100100, which is so chosen that it cannot be confused with any character byte even if one bit has been wrongly received.

The fourth and fifth bytes carry codes to identify the magazine number which is also the hundreds digit of the page number, (pages 200-299 are in magazine No. 2; pages 300-399, magazine No. 3 and so on), and also the row number. The latter enables rows to be missed to give spacing without having to waste transmission time by including a row of spaces. Thus if row 10 is signalled after row 6 there will be three spaces without transmitting three empty rows.

## Header Row

The above description is for normal rows, five sync and control bytes followed by 40 character bytes, but the first or header row has additional information. This incidentally is termed row 0, the following ones being numbered 1-23.

After the five bytes which are the same as for normal rows, come bytes six and seven which carry the code for the page number in units and tens respectively. Next follow four bytes which convey a timing code; minutes units, minutes tens, hours units, and hours tens. The time thus indicated is the nominal transmission time for that page and may not necessarily be the actual clock time. It really is a form of reference which enables different editions of the same page to be sent at different times. At say one a minute, there could be over 1,000 editions of a particular page in a day, each identified by its time code. Some decoders have the facility of storing a particular page for future display, so pages that had been superseded by later editions could still be recalled if they had been so stored. Further possibilities whereby this code can be used to provide further facilities are in the course of development.

This makes 11 control bytes so far for the header row and thus brings us to the last two, 12 and 13. There are 16 bits available here, and eight are used to control the same number of functions. When the appropriate bit is at 1, the particular function is activated. The remaining eight are protection bits which we shall deal with in a moment.

The eight functions controlled by bytes 12 and 13 are as follows.

**Erase Page.** Should the information contained on any page be significantly different from that contained on a previously transmitted page bearing the same number, the former page will be erased to avoid confusion. This will be done with the bit set at 1.

**Newsflash.** When the newsflash page is keyed, the viewer watches an ordinary programme which is mixed with the blank Teletext page. If a newsflash is transmitted, it appears boxed and superimposed over the normal picture. When the appropriate bit is set to 1 in the transmission the newsflash appears. The viewer can wipe the flash using a control on this keypad, but if he stays tuned to the page, the next flash will appear in due course.

**Subtitles.** This is another page to be superimposed on a normal TV picture. Certain programmes are subtitled, and when this page is selected, the presence of a 1 bit in this position brings up the subtitles in a box.

**Header Suppression.** The rest of the header row after the 13 control bytes consists of 32 characters which always follow the same format for each page and are the only ones in the row that are visible. They contain the page number displayed (as distinct from the page code which identifies the page for the decoder), the originating service (Ceefax or Oracle), the date and day and finally the clock time. For some pages such as a newsflash, it may be desirable to suppress the header, and a 1 bit here will do just that.

**Update Instruction.** Where part of a page has been updated from the previous transmission, it may not be transmitted in its entirety, only the new portion. This control bit instructs the memory to replace the old rows with the new.

**Interrupted Sequence.** Some pages are transmitted more often than their natural sequence, such as the Index and others deemed in greatest demand, to reduce their access time. Also some pages such as the subtitles are granted priority, as a change of subtitle may be required more often than the normal page cycle. This bit ensures priority handling of the data, and also suppresses discontinuous page numbers.

**Inhibit Display.** If for any reason a page has become unintelligible, this control bit will inhibit its display.

**Rolling Pages.** If material is too much to be contained in a single page, rather than take up another numbered page, a second, third or even more sub-pages can be displayed in sequence with each page being held for a specified time, say a minute. In such

```

P315 ORACLE 315 Thu10 Sep ITV
1627384950 CROSSWORD 938
ACROSS
1. He is after study (5)
2. He sounds incentive (6)
3. She's jewel to mum (5)
4. She's in network (6)
5. She poles round USA (5)
6. She's a jewel (5)
7. She says she is ! (6)
8. He says OK (5)
9. She forms as dirt (6)
0. She forms a rash (5)
CLUES DOWN
1. Take gas off artist (5)
2. All across, as men ? (5)
3. Bath e.g., in-land (5)
4. Boat, bust leg: laugh (6)
5. Fly VII? Abuse (6)
6. Difficult ? (6)
7. A by rail for birds (6)
8. Poets in Lombard St. (5)
9. Dress or - mortis ? (5)
0. Tree, left roguish (5)
SOLUTION TO 92A
    
```

cases the headers of sub pages are not needed, and the control bit initiates the next sub-page. An example of this type of material is the football results. The user can hold any sub-page if desired.

## Protection Codes

It is obvious that a signal depending on a string of pulses and spaces is vulnerable to error. A momentary break in the signal where there should be a pulse could be interpreted as a 0 instead of a 1 by the decoding circuits. Also an interference spike could trick the circuit into reading a pulse or 1 instead of an 0.

In the case of the characters, this could result in a completely wrong character being produced resulting in confusion or even a misleading message. To avoid this, each character byte includes what is known as a parity bit. The code to produce the range of characters required needs only seven binary code bits out of the eight in each byte.

The eighth becomes a 1 if there are an even number of 1 bits already in the byte, but becomes a 0 if the number of 1 bits are odd. In this way the 1 bits are always odd in number. The decoding circuits count the number of 1's in each byte and if they are an even number it is obvious that one has been lost or there is one too many, so the entire byte is rejected. Thus a blank space occurs which is better than a wrong character. This is termed odd parity.

While it offers a good degree of protection against error, it is not foolproof; two errors could occur in a byte to give the required odd number of 1 bits. In the case of the characters, this could be confusing if a wrong character was thereby displayed, but it would not be disastrous. Such an error in any of the address codes could cause complete mayhem! Hence, a greater degree of protection is required for these than for the characters.

A method is used that was devised by R.W. Hamming of the Bell Telephone System of America in 1950. In the eight-bit byte, only four bits are used to carry the signal, the other four are parity check bits. This is why, as described earlier, the eight functions determined in the header row require two bytes, numbers 12 and 13 totalling 16 bits. Eight of those are Hamming protection bits.

These protection bits are interleaved with the data bits so that bits 2,4,6, and 8 carry the data, while bits 1,3,5, and 7 are the parity ones. This provides a check on every single data bit which

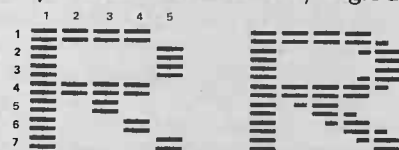


Fig. 3 The 7 x 5 matrix on which characters are built, with and without character rounding.





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Small, black, and brilliant p.29



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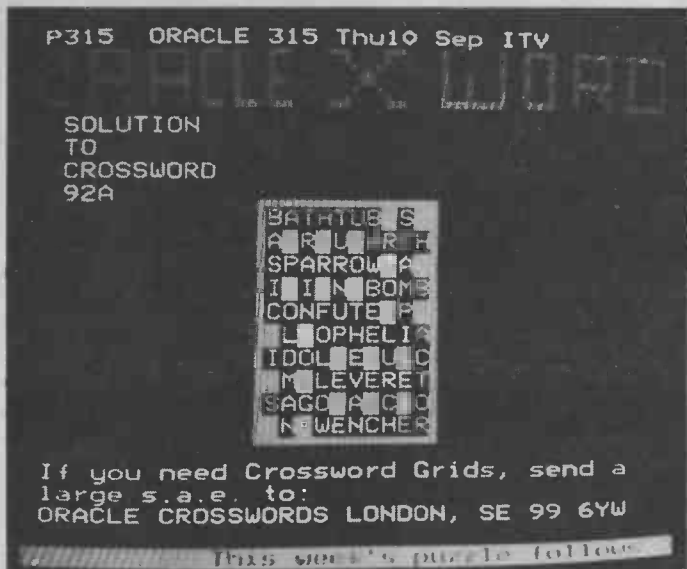
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if incorrect, cannot only be detected but corrected. Thus several errors in the same byte can be put right. So a high degree of error protection is achieved although at the expense of bit capacity.

## The Characters

Normal characters occupy seven field scanning lines in the reproduced picture; however, some lower case letters that have tails such as q, y, p, g, and j, need an extra two lines below normal. One line is allowed between rows, so this makes ten lines in all. As the picture is made up of two interlaced fields, the total number of lines is twenty. As the character cannot be produced simultaneously but must be built up by the line scan, the character generator must consult the memory many times before the character is complete.

Horizontally, the characters are made up of five segments or dots. So with the seven lines height, this forms a matrix of 35. A sixth dot is allowed for spacing between the characters. The time taken to scan one line of one character is 1 us which divided by the total of six dots, gives a dot frequency of 6 MHz, the normal maximum video bandwidth.

Unlike lettering appearing in a normal TV picture, such as captions, the form and appearance is not fixed at the programme source, only the type of character. Form is determined by the character generators in the decoder and so can vary,

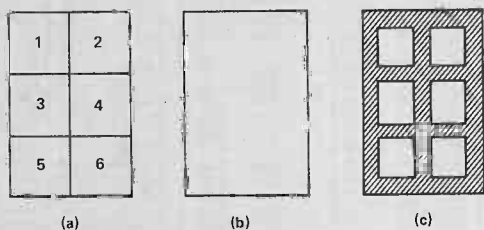


Fig. 4 (Above) (a) Graphics are built up on a 3 x 2 matrix numbered as shown - any position may be illuminated. (b) When the units are coded for contiguous display, they join with those adjacent. Here all six are on to give a completely illuminated rectangle. (c) When coded for separate display, they are isolated from their neighbours as shown here.

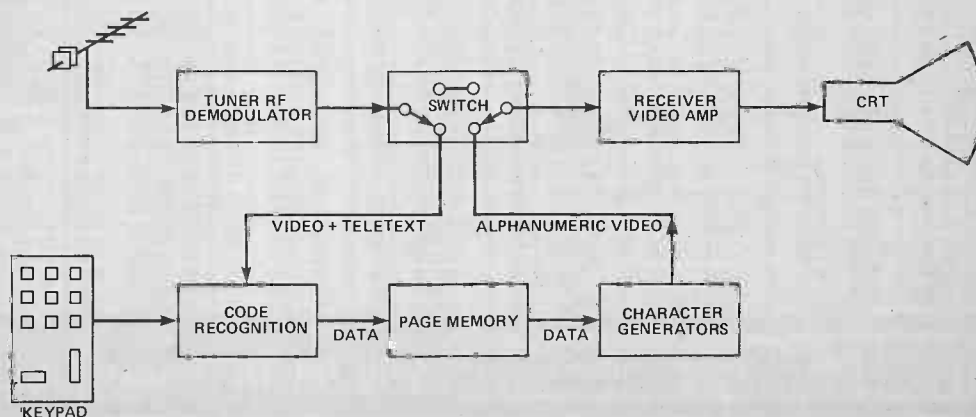


Fig. 5 Simplified block diagram of a Teletext receiver.

some decoders have a double-height. Another improvement that is becoming more common is the rounding of edges which otherwise present a stepped appearance. In the case of the double height, the instruction to do this is built into the character code, but a decoder without this facility just ignores it and reproduces at normal height.

A total of 94 characters can be displayed including upper and lower case letters, numerals, the fractions  $\frac{1}{4}$ ,  $\frac{1}{2}$ , and  $\frac{3}{4}$ , punctuation signs, and various others that are commonly found on typewriter keyboards; plus spacing in background or display colour.

## Graphics

In addition to alphanumeric signs, basic illustrations can be produced, being built up from simple graphic building blocks. For these, a matrix three units high by two wide is used, the graphic block being the same dimension as a character.

These are 64 possible ways in which the six units in the graphic block can be illuminated, from all on to all off, so these make up the 64 graphic symbols. In addition, the units illuminated can either be contiguous, that is appearing as one unbroken illuminated segment, or separated from the adjoining ones. So for example, if all six in the block are on in the contiguous mode, it will appear as a continuous rectangle of light, but if they are in the separated condition, it will appear as six small blocks in the larger one which will look like a dark grating. The required mode is conveyed by part of the code.

Finally, we will look again at the character coding, which you remember consists of a single byte of seven data bits and one parity bit. The first three bits carry the colour information. They give seven combinations and these correspond to seven different colours; red, green, yellow, blue, magenta, cyan, and white. The fourth bit, in combination with the first three, signals steady display or flashing, start or end of a box and normal or double height for alphanumeric characters. When graphics are being displayed, the same four-bit combination gives concealed display, contiguous graphics or separated graphics, black background, new background, hold or release graphics. There are eight possible combinations when the fourth bit is set to 1, but not all are used.

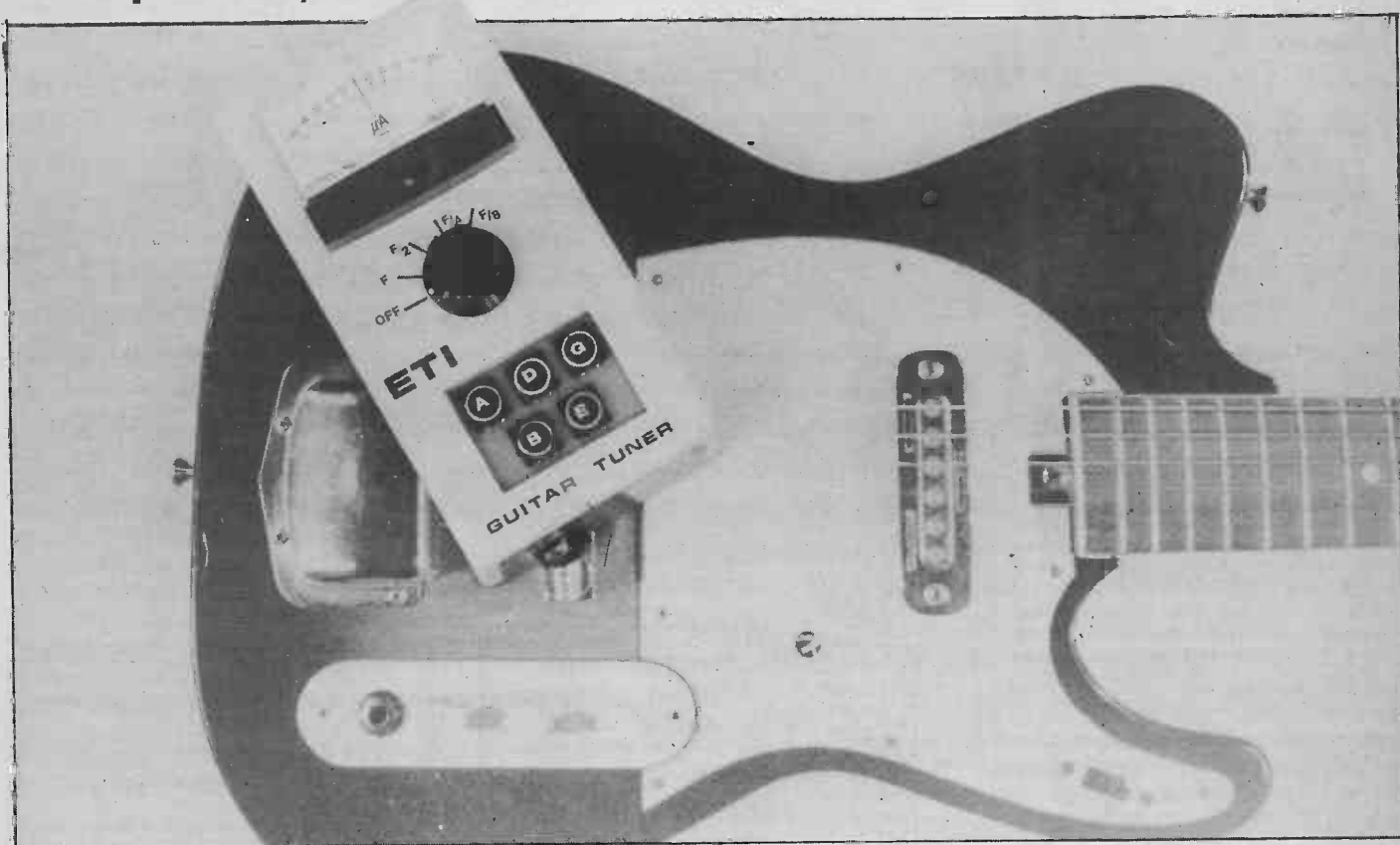
The remaining three bits select either alphanumerics or graphics and the specific character. So seven bits are utilised, the eighth being, as we have seen, the protection parity bit.

Add-on decoders designed for use with an existing set contain a tuner, IF and demodulator circuits so that the aerial is plugged straight into a socket on the unit. A PAL encoder and RF modulator enable the output to be applied to the aerial socket of the receiver which is tuned to an unused TV channel. The decoder tuner is then used for normal TV viewing.

This extra circuitry increases the cost and, in theory should add a degree of noise to the normal TV picture as the signal passes through two tuners and a modulator instead of the receiver's single tuner. In older sets employing tuners that tended to be noisy, there could actually be an improvement as the Teletext unit may have some gain through its tuner and modulator and so act as a preamp.

# GUITAR TUNER

Roadies, perfectionists and the tone deaf can all benefit from this project. Our Guitar Tuner is both versatile and accurate. Design and development by Brian Brooks.



Musicians who wish to play together have to ensure that their instruments are tuned to the same fundamental pitch. If they aren't, the resulting sound is pretty awful. Listening to the radio these days can give the impression that some groups actually *like* to play this way — nevertheless, we feel that the majority of musicians out there would be grateful for something that does away with the need for a tuning fork (or pitch pipes) and a good ear for pitch. Look no further — this is the project you've been waiting for.

The ETI Guitar Tuner is quick, simple, versatile and highly accurate. Unlike other designs, reference frequencies are provided for all six strings of a guitar, and are selected by momentarily pressing one of the push-buttons. The note chosen is then synthesised by a frequency generator chip and remains latched until another pitch is selected. This overcomes the major disadvantage of the tuning fork, in that the note does not die away

quickly. An octave selector switch allows the Tuner to be used with bass guitars or other instruments (anyone for synthesisers?), and even acoustic instruments can be tuned, by using a high impedance mike to provide an input.

Although it is the ideal project for anyone with a musical bent, or bent music, the people who will benefit most from this project are roadies, who often have to quickly tune a number of instruments before a concert while loud music is being played over the PA. Since the Guitar Tuner uses a meter to indicate beats visually, this daunting task becomes a piece of cake.

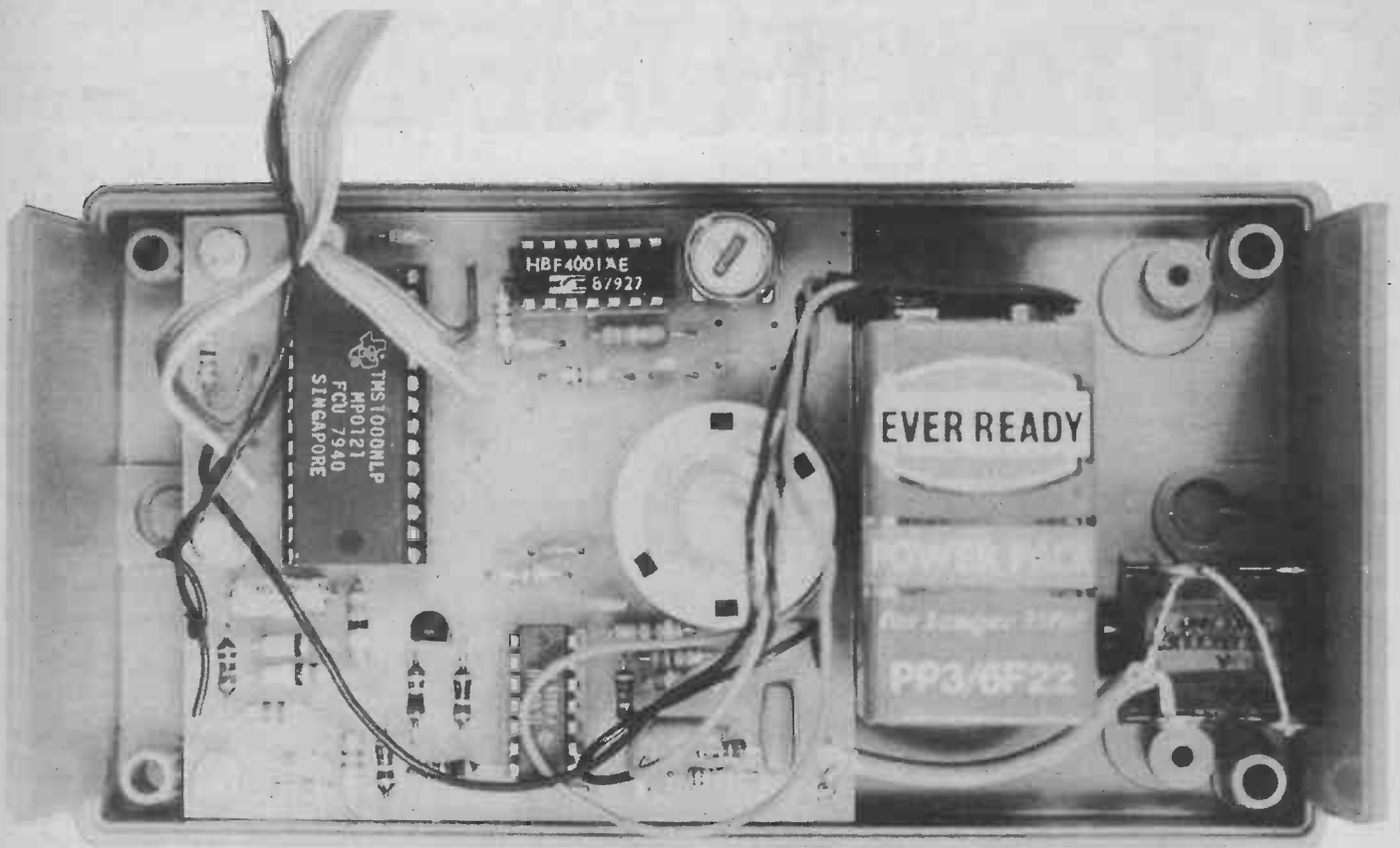
## Construction

This is quite straightforward if the overlays are followed carefully. Start with the main board, soldering in the low-profile components first — wire links, resistors, diodes and IC sockets. Insert Veropins at the points where interwiring will take place later. Follow with the capacitors and the FET, and

finally fit SW1 and L1. The tags on SW1 must be cut off so that it will fit the holes in the PCB. Check that all the polarised components are inserted the right way round; the coil has an asymmetrical pinout and will only fit one way. Now insert the ICs into their sockets, being *unbelievably* careful with IC1 — remember that this is an A-series *unprotected* CMOS chip and it will commit suicide at the slightest hint of static, so DON'T TOUCH THE PINS.

The completed board can now be fastened to the case bottom using short self-tapping screws. Although the PCB is only supported at one end (see photos), it is quite rigid enough. Mount the jack socket at the opposite end of the case to the PCB and link it to the input Veropins with a short piece of screened cable. Note in the photographs how two pins of the jack socket are connected to the screen of the cable — the socket is a shorting type so that if an instrument is not plugged in, the input is grounded to prevent noise pick-up.





Cut holes in the lid for the meter and the shaft of the rotary switch — the easiest way to accomplish the latter task is to smear ink on the tip and replace the lid to leave a drilling mark on the inside. Don't worry about missing slightly; the knob is wide enough to hide a multitude of sins!

Now solder the five push-buttons into place on the smaller PCB, making sure that the small dot on top is orientated as shown on the overlay, and solder the ribbon cable to the pads at the edge of the board. Make a rectangular cut-out in the lid of the box to allow access to the push-buttons (placing it far enough to one end to avoid fouling the rotary switch shaft) and fasten the keyboard inside the lid with double-sided tape or sticky pads. The corners of the board may be cut at an angle to fit between the moulded pillars.

After bolting the meter into place, the final interwiring can be completed. Solder the other end of the ribbon cable wires to the pins around IC2 — checking carefully that they're in the right order — and make the connections to the meter. All that remains now is to solder the battery leads, connect the two PP3s and fasten them (using sticky pads again) in the space between the main PCB and the jack socket.

## PARTS LIST

### Resistors (all 1/4W, 5%)

R1	1k
R2,3,8	100k
R4,12	10M
R5,10	1M
R6	220k
R7	8M
R9,11	47k
R13,14,18,20	4k7
R15,16,17,19	10k
R21	3k3

### Capacitors

C1,2	10u 16 V PCB electrolytic
C3	220p ceramic plate (low temp. coeff.)
C4,5	1n0 ceramic plate
C6	10u 25 V axial electrolytic
C7,9	100n polyester
C8	470p ceramic plate
C10	220p ceramic plate

### Semiconductors

IC1	4001A or 4001UB
IC2	TMS1000
IC3	TL064CP
Q1	2N3819
D1-3	1N4148
ZD1	12 V 400 mW zener

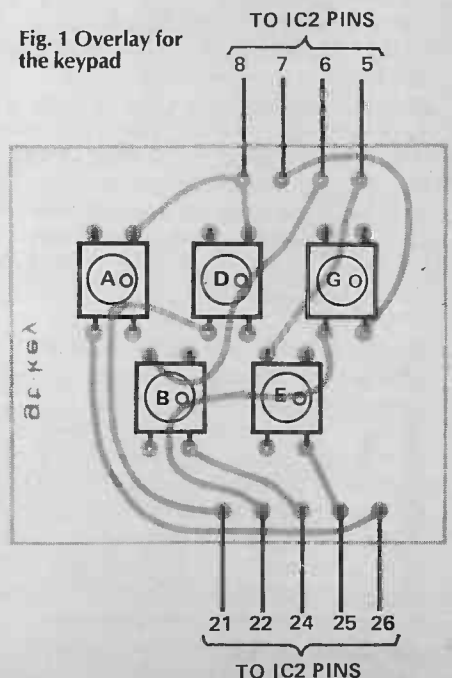
### Miscellaneous

L1	3.5 mH
SW1	2-pole 6-way rotary switch
PB1-5	push-button switches
SK1	1/4" mono jack socket
M1	100 uA centre zero meter
Two off PP3 batteries and clips; PCBs (see Buylines); IC sockets; cable; case (Vero order ref. 202-21030); knob to suit.	

## TABLE 1

NOTE	OCTAVE	FREQUENCY (Hz)
E	F/4	82.4
A	F	110.0
D	F	146.8
G	F	196.0
B	F	246.9
E	F	329.6

(E's are 2 octaves apart)



## BUYLINES

A complete kit of parts is available from Magenta Electronics Ltd, 135 Hunter Street, Burton-on-Trent, Staffs DE14 2ST. The kit costs £28.98 (including VAT), plus 40p p & p. Parts are available separately if required, as are the PCBs, which cost £4.48 for the pair. Alternatively you can obtain the boards from our PCB Service; details on page 98.

## HOW IT WORKS

The reference oscillator built around IC1a and b must be very stable, so a high-Q coil and capacitor design is used rather than the more familiar R/C astable. The IC is used here in the linear mode, so the A-series or unbuffered CMOS chip is better. Furthermore, this IC has its own stabilised 12 V supply, derived from R1-ZD1 and decoupled by C1.

IC2 is a frequency synthesiser chip, producing top octave notes A, D, G, B and E. These are selected by closing the appropriate push-button in the keyboard matrix driven by IC2; each note latches when the key is pressed so momentary action only is required. SW1b selects either the top octave (frequency F), or the three octaves below it, F/2, F/4 and F/8.

The remainder of the circuit requires a rail at half-supply voltage and this is achieved using potential divider R2-R3 and voltage follower IC3a. C2 provides decoupling for this rail. IC3b and c amplify the input signal; the input impedance is 47k, suitable for use with a guitar or high impedance mike if acoustic instruments are being tuned. IC3d is a synchronous rectifier and is used to compare the input frequency with the internal reference. When two signals are in phase the output of IC3d is positive, and when they are out of phase it is negative — the output swinging between the two at 90° and 270°. Q1 is driven from the reference to alternately invert the gain of IC3d at the appropriate pitch. Meter overload protection is provided by D2 and D3.

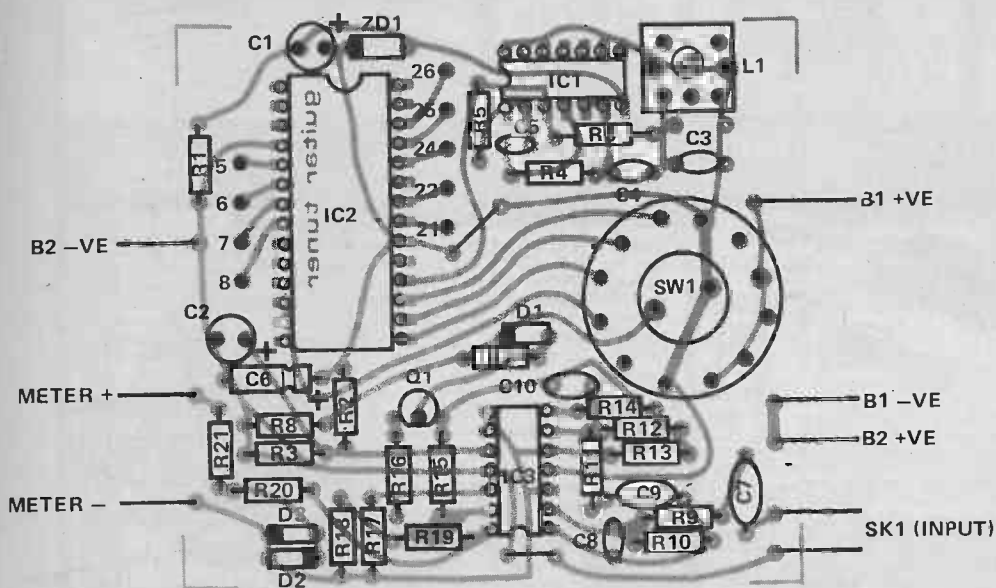
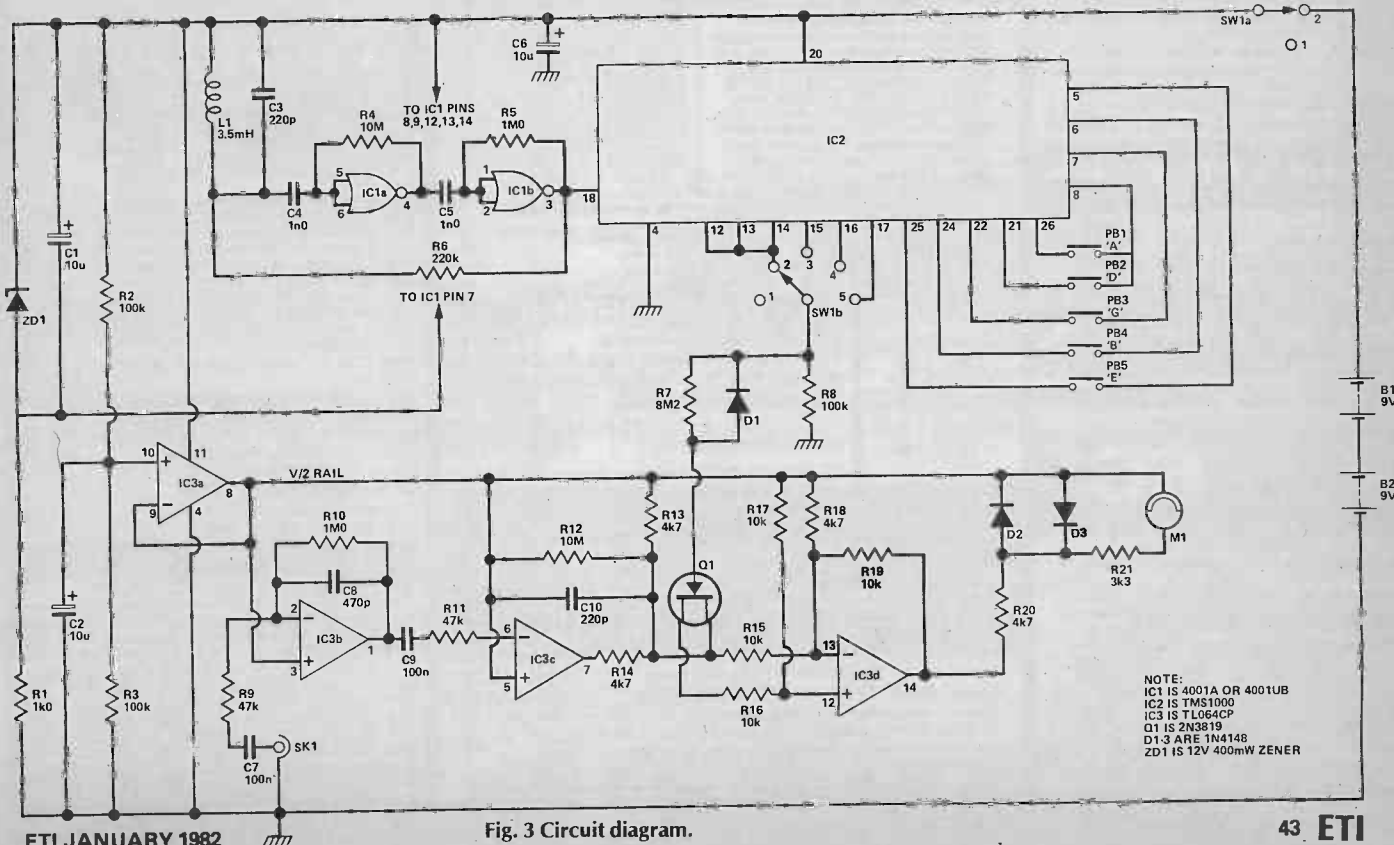


Fig. 2 Component overlay for the main PCB.

## Setting Up

To calibrate the Guitar Tuner, select any of the frequencies in Table 1 and tune the coil while measuring the output of IC2 (pins 12-14) with an accurate frequency meter. Alternatively you may feed in a known original frequency to the jack socket (using a mike and a tuning fork, an accurate signal generator, an atomic clock or whatever else you happen to have handy) and adjust the coil for zero beats on the meter. Once set, the oscillator will stay very accurate regardless of the battery state or the ambient temperature.

To use the Guitar Tuner, plug your electric guitar (or microphone) into the jack socket, press the button corresponding to the string you want to tune, select the correct octave with the rotary switch (for example, use F/4 for the bottom E string), and strum the note. If you're way off tune the meter won't respond at all, but as you tune the string closer to the right pitch, the needle will start to register the beats by oscillating rapidly about the centre position. This oscillation will slow down as you approach the reference until the needle stops at zero, at which point the instrument is perfectly tuned.



NOTE:  
IC1 IS 4001A OR 4001UB  
IC2 IS TMS1000  
IC3 IS TL084CP  
Q1 IS 2N3819  
D1-3 ARE 1N4148  
ZD1 IS 12V 400mW ZENER

# AUDIOPHILE

This month Ron (Sherlock) Harris discovers that the Trio Sigma-Drive is not something you build into a spaceship, and reveals the Mystery of the Missing Letters. Read on, gentle reader. . .

One of the latest ideas to attempt to raise the height of hi-fi is that of Sigma-Drive. It is the brainchild of Trio, who have an enviable reputation for amplifier design anyway. They now have a range of models incorporating the new panacea and I'm taking a look at the most powerful of these this month, the KA-1000, which boasts some 100 W per channel, separate PSU and a recording facility which will delight the tape enthusiast.

## Total Sum Drive?

As any mathematician worth his slide-rule will tell you, 'Sigma' means 'total sum of', (but for no reason other than that it is the definition assigned to it by mathematicians.)

The principle of Sigma-Drive is thus to take into account the *total* load presented to the amplifier and to use this to control the circuitry such that the loudspeaker accurately follows the signal input.

Trio's own literature is somewhat confusing, making much ado about feedback, back-EMF, speaker wires, non-magnetic designs etc. etc. — and totally clouding the issue! As far as I can see what Sigma-Drive *actually* does is to extend the feedback loop of the amplifier to include a voltage derived from the speaker terminals. This is then used as an error-correction signal to alter the drive from the output stage to the speaker. In practice this will raise the damping factor of the amp significantly, since the output appears to be a constant-voltage source, ie zero-impedance. But as nothing in life is perfect — I dare say even Stevie Nicks has off-days — the output impedance appears to be extremely low, rather than actually zero. The measured damping factor of the KA-1000 is somewhere between 1500-2000, even at low frequencies.

This in turn means that the effect of the cables connecting amp and speaker is drastically reduced, even eliminated. Supercables are thus rendered superfluous, as the amplifier is correcting the output *after* the cable!

The logical extension of all this must be a tri-amp system using Sigma-Drive such that the control is applied to the voice-coil itself.

In this manner all external influence would, in theory, be corrected by the amplifier. Any plans Trio?

## Non-Magnetic What?

The KA-1000 also boasts a 'non-magnetic' PSU design to minimize the distortion which Trio say is induced into circuits by the proximity of magnetic current carrying materials. The power amp is designed to also have an exceptionally wide open-loop response, such that it is highly stable and has a fast rise-time.

As you can see from the diagram, the preamp section is well endowed with facilities; most of which hide beneath a very attractive smoked glass flap when not required. LED indication is also provided, in case you're not exactly sure what you've just pressed, presumably.

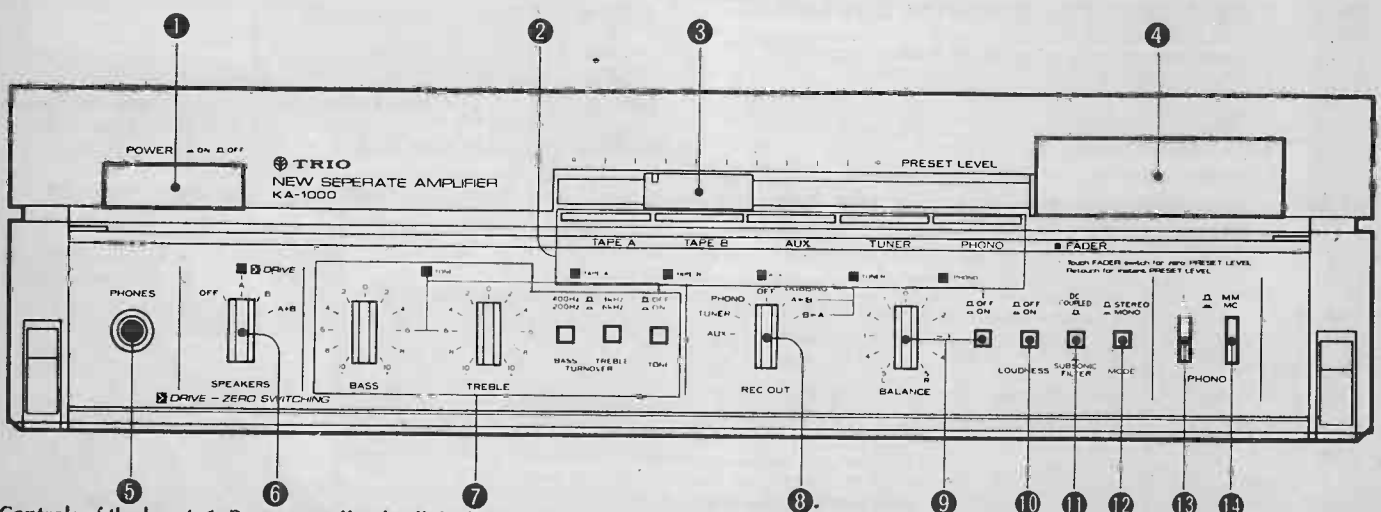
The volume control is a little unusual in its operation. For a start it is a horizontal slider and is labelled 'pre-set' level. Next to it is a comparatively huge rectangle, which glows with a blue light, marked 'fader'.

Apparently, you are supposed to set the level you want with the 'pre-set level' and then touch the fader to drop this to zero — for changing inputs, records, etc. Touching the fader again restores the pre-set level — hence the name.

In practice, I'm afraid I found the slider stiff and awkward and would have preferred a nice normal rotary. (I'm just an old-fashioned city-boy really!) The fader is a neat idea, and is easier to operate than the usual '-20 dB' mute switches which abound on Eastern facias.

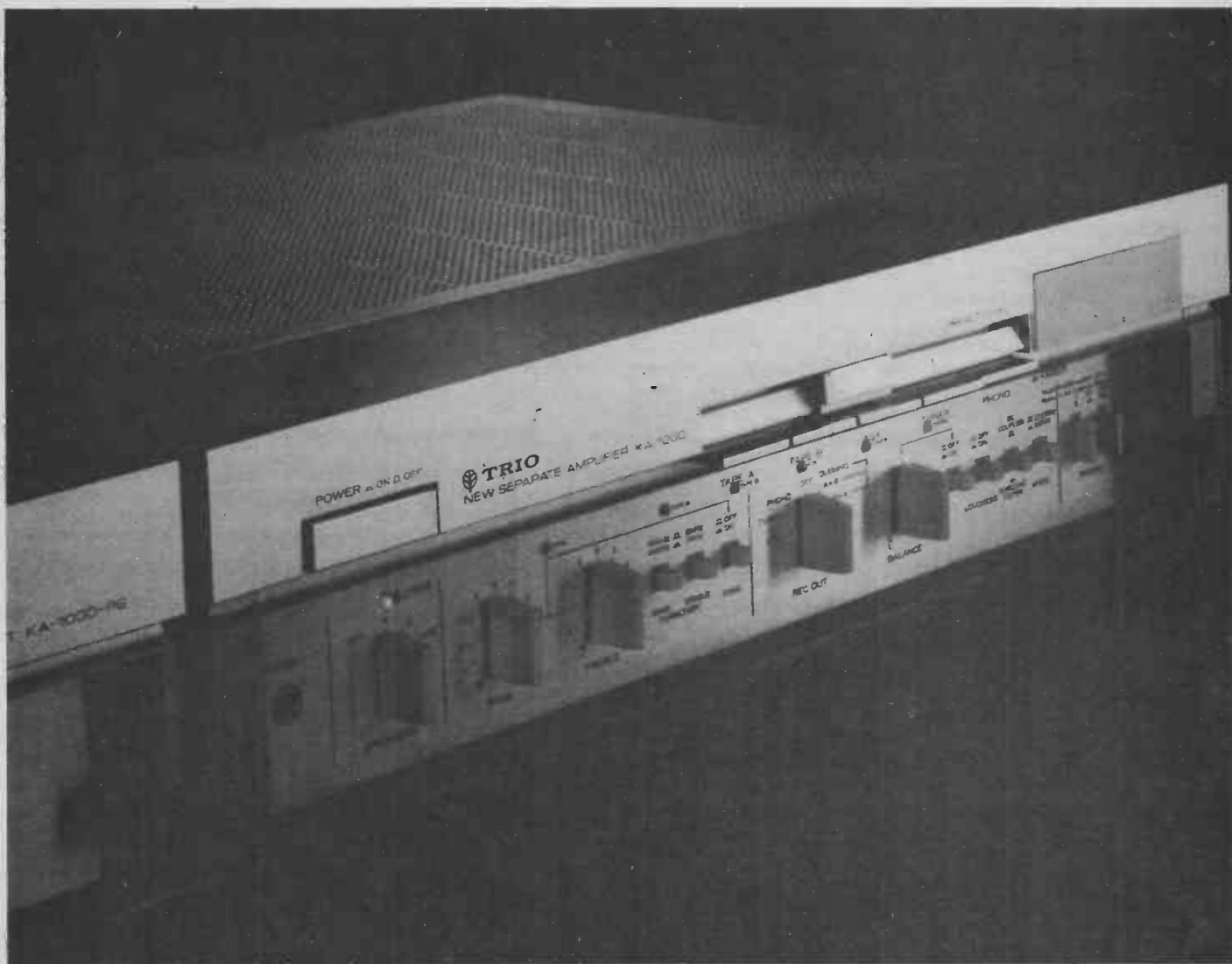
## Control Tones

Sigma-Drive is only available on the main speakers and the selector switch for combining, or cancelling, the two pairs pro-



Controls of the beast: 1. Power on-off, relay linked to the PSU. 2. Input select bank: note this does not control Tape Out! 4. Fader switch: takes the volume down to zero (and back) at a touch. Handy for changing records, etc. 3. Preset Level: a volume control in disguise. 5. Phones: about the only standard thing on the unit! 6. Speaker select: note that Sigma-Drive is only available on A. 7. Tone: can be switched for turnover-and out completely. 8. Record out: sets the input which controls the Tape Output. Independent, so you can play a record and record from the radio! 9. Balance: switchable out would you believe? 10. Loudness. 11. Subsonic Filter. 12. Stereo switch. 13/14. Phono input select and MC/MM gain select.





vided for will thus act as a comparator — should you want to see what difference Sigma-Drive actually makes.

Tone control turnover frequency is variable and the whole circuit can be switched out, as can the balance control (!) for 'straight-wire-with-gain' freaks.

Tape facilities allow you to record from one source while listening to another and dubbing in either direction between two tape decks is provided for.

The PSU has no controls on it at all, being operated by the mains switch on the front panel of the KA-1000 itself.

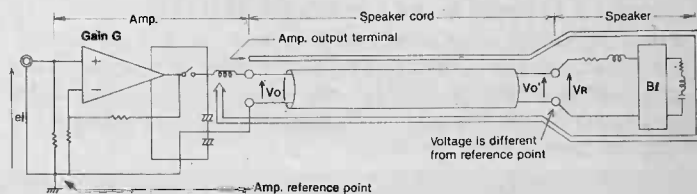
### Test Drive Time

The Trio did well under test, as you'd expect, and gets ten out of ten for engineering. The distortion measurements are well-nigh all noise and, as a simple amplifier, it is difficult to fault the KA-1000. I could not duplicate Trio's figures for S/N on the phono input though, but 76 dB weighted is a good figure nonetheless. I also found this input a little insensitive and it took some driving in moving-magnet mode to obtain full power (around 4 mV in fact).

The moving coil input returned figures of 66 dB for noise and sensitivity of about 200 uV (for 100R). Almost exactly to spec and good results in themselves.

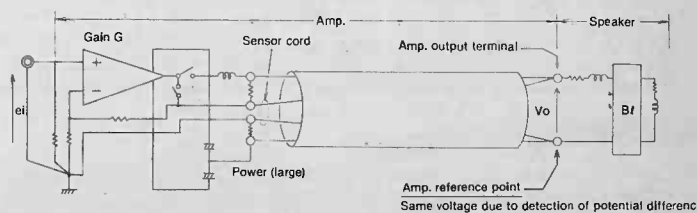
Peak power delivery into 8R proved to be in order of 170 W with an incredible slew-rate of 140 V/us! Checking the frequency response confirmed the rise-time measurement, as the amp is flat to around 300 kHz!

Table 1 gives a run down of the rest of the specs and will interest the mathematically-minded.



Above: how the speaker and feedback are arranged in a normal amplifier. Note that due to the voltage drop down the cable, the damping factor is reduced at the speaker terminals from the value it had at the amp terminals.

Below: how Sigma-Drive overcomes the problem, fitting the speaker into the feedback loop with the use of the sensor terminals. Four wires to the loudspeaker are now required.



## Enter The Drag

With this amplifier you wouldn't be moving the hi-fi around too much. There are four wires to each speaker and, although Trio provide a 'Sigma-Cable' with these already in, it wasn't in the box when I received it. It would probably be too short anyway according to Murphy's Third Law of Installation, so I wasn't that upset.

Adding more leads to the set-up is about as much fun as a toothache on Sunday afternoon, and just as welcome. All through the leaflet Trio terrify the buyer with the consequences of connecting the Sigma-Drive terminals incorrectly and by the time I'd finished I was feeling quite nervous myself! (A large brandy was thus justified. . .).

Thankfully all was well and the amp clicked into operation first time. I was surprised at how distinctive a sound this amplifier has. It is clean, sharp, and very fast. You notice it first on bass guitar or drum sounds. The output simply gets from nothing to music quicker and more realistically. Undoubtedly the best Eastern amp I've heard.

Some may find it a little cold, I think, as it is most unforgiving of programme faults, or inferior equipment coupled to it! Most of my tests were conducted with my beloved Thorens 160S/SME III and Dynavector Karat at one end and KEF 105 II's at the other.

For a time I substituted Shure's new MV-30HE pickup, designed for the SME III, but I found the output was too low to drive the Trio successfully.

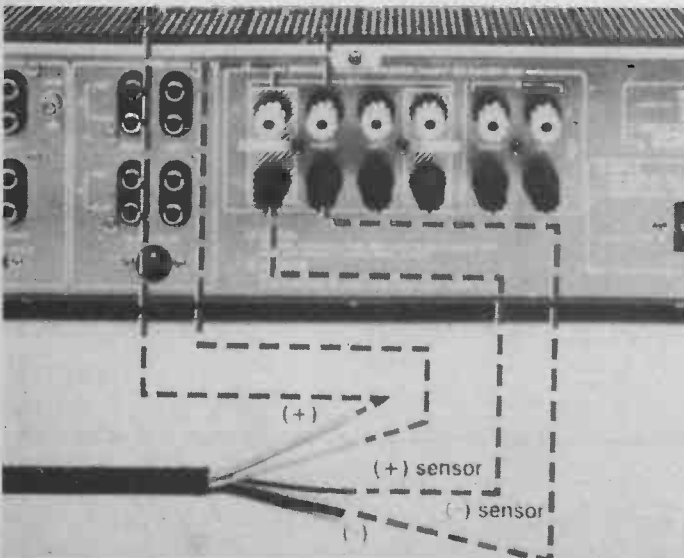
## Conclusions?

The only conceivable conclusion I can come to is that Sigma-Drive works. Switching it out produces a 'looser' bass sound altogether and the detail disappears to a large extent, as the frequency response appears to roughen up considerably.

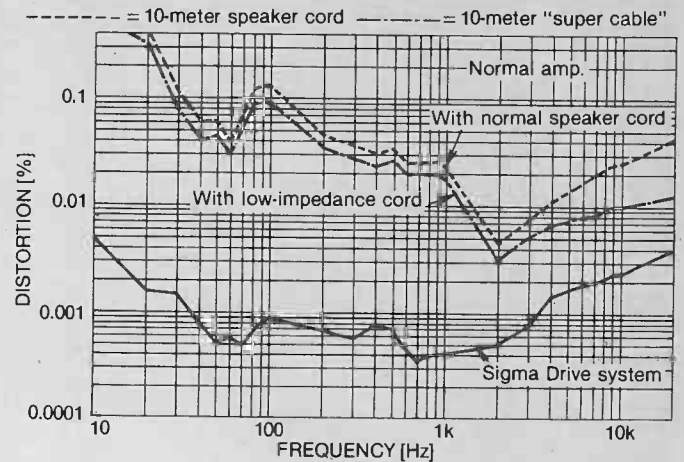
This makes it unlikely that people using a KA-1000 would connect two sets of loudspeakers — I know I wouldn't — as the difference in sound quality is very marked. With the Sigma-Drive 'on', the Trio is a fine sounding machine and one which is well worth the asking price for the engineering which has gone into it, and the sound quality it delivers.

## Accessory To Revelation!

Some months ago, I told the tale of our horrendous experience with the products of a well-known accessory company, without actually naming the firm concerned until such time as I'd received a reply from them to my irate epistles.

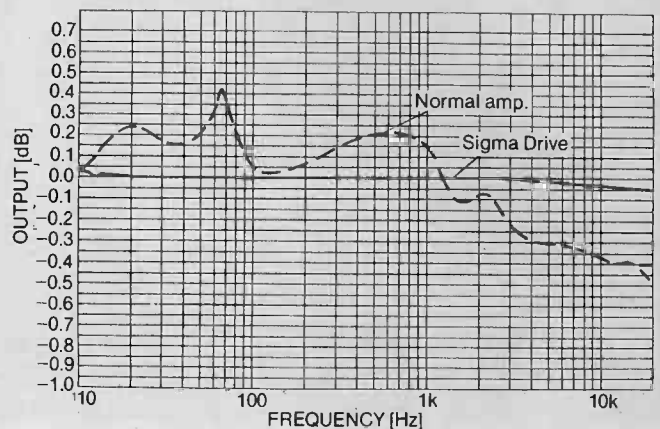


How all those wires fit on to a loudspeaker. Confusing isn't it?



Above: the frequency vs distortion characteristics of a normal low amp into eight ohms, 20 Hz – 20 kHz. Measured at the amp. out terminals.

Below: the effect of the above, using Sigma-Drive and normal drive methods. Flatter is it not? All effects of the cables have been removed totally.



Apparently, they had never received my correspondence at all, only becoming aware of the fuss when a little man strolled up to their exhibition stand armed with a copy of ETI! (Even though I never mentioned the name QED, most people had sussed out who it was I was raving about.) The MD rang me up immediately after the exhibition, and I have since received this letter expounding their side of the case:

To: Electronics Today International, Argus Specialist Publications Ltd.

Dear Mr Harris,

Further to our recent telephone conversation, and the subsequent receipt of your 'Audiophile' article, I would like to confirm that prior to the exhibition 'rumour' we were totally unaware that you were trying to contact us. I can assure you that all letters received by us are answered (including the ones asking for spare parts for wasting machines etc.).

Having established contact, however, I am at a loss to explain how the three products referred to in your article passed our inspection. Normally, each product is individually tested and such faults should be spotted. My only defence is that these must be isolated incidents because otherwise we would have been literally flooded with complaints and this has not occurred.

We estimate that the three products were probably manufactured at the end of last year. Since that time we have introduced a complete new range of electronic test gear to make it

easier for our staff to identify such faults and all products now carry the initials of the inspector.

In addition, I am pleased to confirm that all QED 3-way speaker switching units are now internally wired with QED 42-strand cable.

Finally, I would like to thank the reader of your magazine who informed us of the Audiophile article because he, more than anybody else, is a testimony to QED's reputation for both quality products and business integrity. As a user of several QED products, he felt that we should have the opportunity to 'clear our name'.

Assuring you of our best intentions at all times.

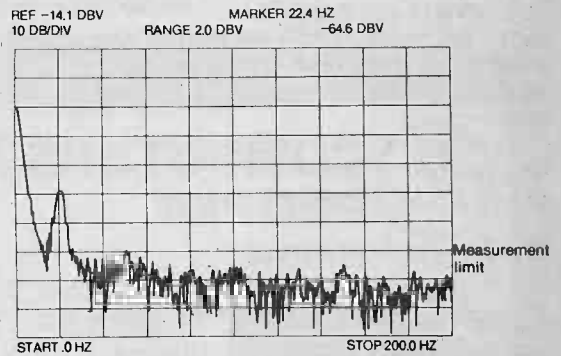
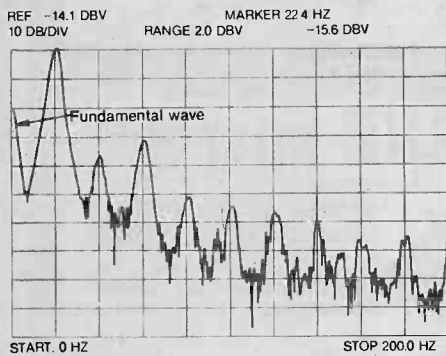
Yours sincerely  
 Bob Abraham  
 QED Audio Products

ETI

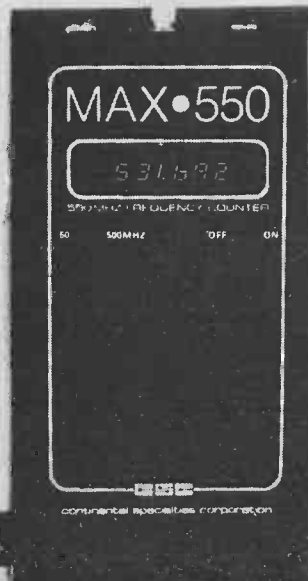
## Trio KA-1000 Amplifier

**Power Output (8 ohm load, 1 kHz): 103 W**  
**Power Bandwidth (100 W): 3 Hz — 320 kHz**  
**THD at 100 W output : 0.008%**  
**Intermodulation Distortion (100 W): 0.1%**  
**Rise Time : 140 V/us**  
**Peak Power Delivery (8 ohm) : 172 W**  
**S/N ratios (weighted) — moving magnet : 76 dB (sen. 4 mV)**  
**moving coil : 66 dB (sen. 200 uV)**  
**tape/aux/tuner inputs : 100 dB (sen. 150 mV)**

**Dimensions — Main amp : 440 x 123 x 375 mm**  
**PSU : 140 x 123 x 358 mm**  
**Typical Price : £530**



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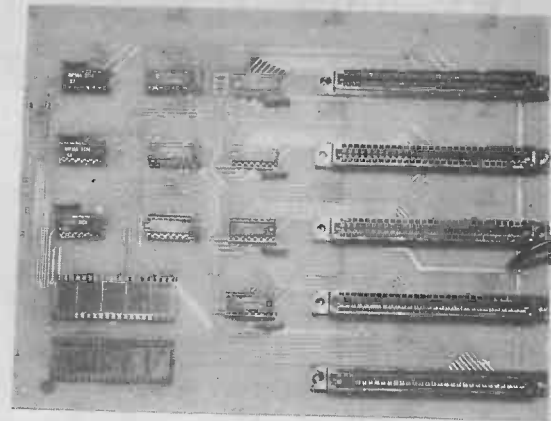
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# COMPUTER EXPANSION SYSTEM



Warbles, whines, pops, bleeps and explosions! Now your computer can produce all these sounds even when it isn't about to break down. In fact, clever programmers will be able to generate polyphonic music with this Sound Card — the second of our expansion boards. Design by Watford Electronics.

The introduction of the General Instruments' sound chip AY-3-8910 has opened up a whole new world of sounds other than the familiar electronically-generated squawks. This chip is capable of producing anything from explosions and laser gun blasts to three-note-at-a-time melodies.

The chip has an internal set of registers controlling the various functional blocks as shown in Fig. 3. There are three main channels (A, B and C), one for each note produced. The frequency of the note produced is determined by the value in the respective tone period registers. Two registers, an eight bit and a four bit, combine to give 12 bit resolution of the tone period for each channel. The frequency output as related to the value in the tone period register is given by:

$$F_t = \frac{F_{CLK}}{16 TP}$$

where

$F_t$  = output tone frequency

$F_{CLK}$  = input clock frequency to chip

TP = Decimal value in tone period registers

## BUYLINES

A full kit of parts for this project, including the double-sided plated-through PCB, can be obtained from Watford Electronics, 33/35 Cardiff Road, Watford, Herts, for £26.95 plus VAT. Note that only one AY-3-8910 is supplied in the kit.

The demand for the two boards published last month has enabled a price reduction to be made: the motherboard is now £36.50 plus VAT, the RAM card is £28.50 plus VAT.



Fig. 1 Pinout of the AY-3-8910.

The clock for the chips is generated using a standard colour burst crystal (3.579545 MHz), divided by two to give an  $F_{CLK}$  of 1.789725 MHz. Taking  $F_{CLK}$  as 1.7898 MHz, if we require an output tone of 440 Hz then

$$F_t = \frac{F_{CLK}}{16 TP}$$

$$F_{CLK} = 1.7898 \times 10^6 \text{ Hz}$$

$$F_t = 440 \text{ Hz}$$

$$TP = \frac{F_{CLK}}{16 F_t} = \frac{1.7898 \times 10^6}{16 \times 440}$$

$$= 254.23 \text{ (254 to nearest integer)}$$

So to generate on channel A an output tone of 440 Hz we would need to load the R0-R1 register with 254. The equation to split the value TP into the values for R0 and R1 is:

$$R1 + \frac{R0}{256} = \frac{TP}{256}$$

## HOW IT WORKS

The clock frequency is generated using a standard colour burst crystal and is then divided using a 4013 dual flip-flop set to divide by two. This gives a clock of 1.7898 MHz. The data inputs and selecting of the sound chips are provided by a 6520 parallel input/output chip. The eight data inputs DA<sub>0</sub>-DA<sub>7</sub> are driven directly by Port A (PA<sub>0</sub>-PA<sub>7</sub>) of the 6520. Data bus direction and function is controlled by BC1 and BDIR. These are driven by PB<sub>0</sub> and PB<sub>1</sub> respectively.

PB<sub>2</sub>, PB<sub>3</sub> and PB<sub>4</sub> are used to select IC5, 6 and 7 respectively. The analogue output from the chip is amplified by IC3, an LM386 half-watt audio amplifier. Power-on-reset for the 6520 is provided by R1 and C2. Reset for the sound chips if you get stuck in the middle of an 1812 overture is provided by R8, C9 and PB1.

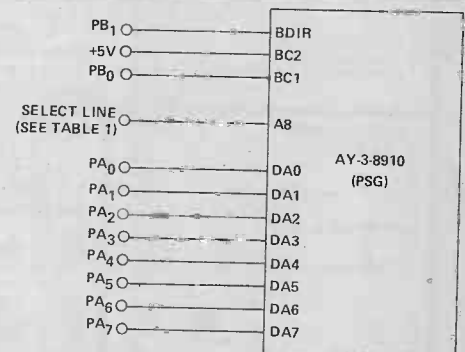


Fig. 2 How the programmable sound generator (PSG) is wired to the ports of the 6520. Table 1 refers.

## TABLE 1

BDIR	BC2	BC1	PSG	FUNCTION
0	1	0		INACTIVE
0	1	1		READ FROM PSG
1	1	0		WRITE TO PSG
1	1	1		LATCH ADDRESS

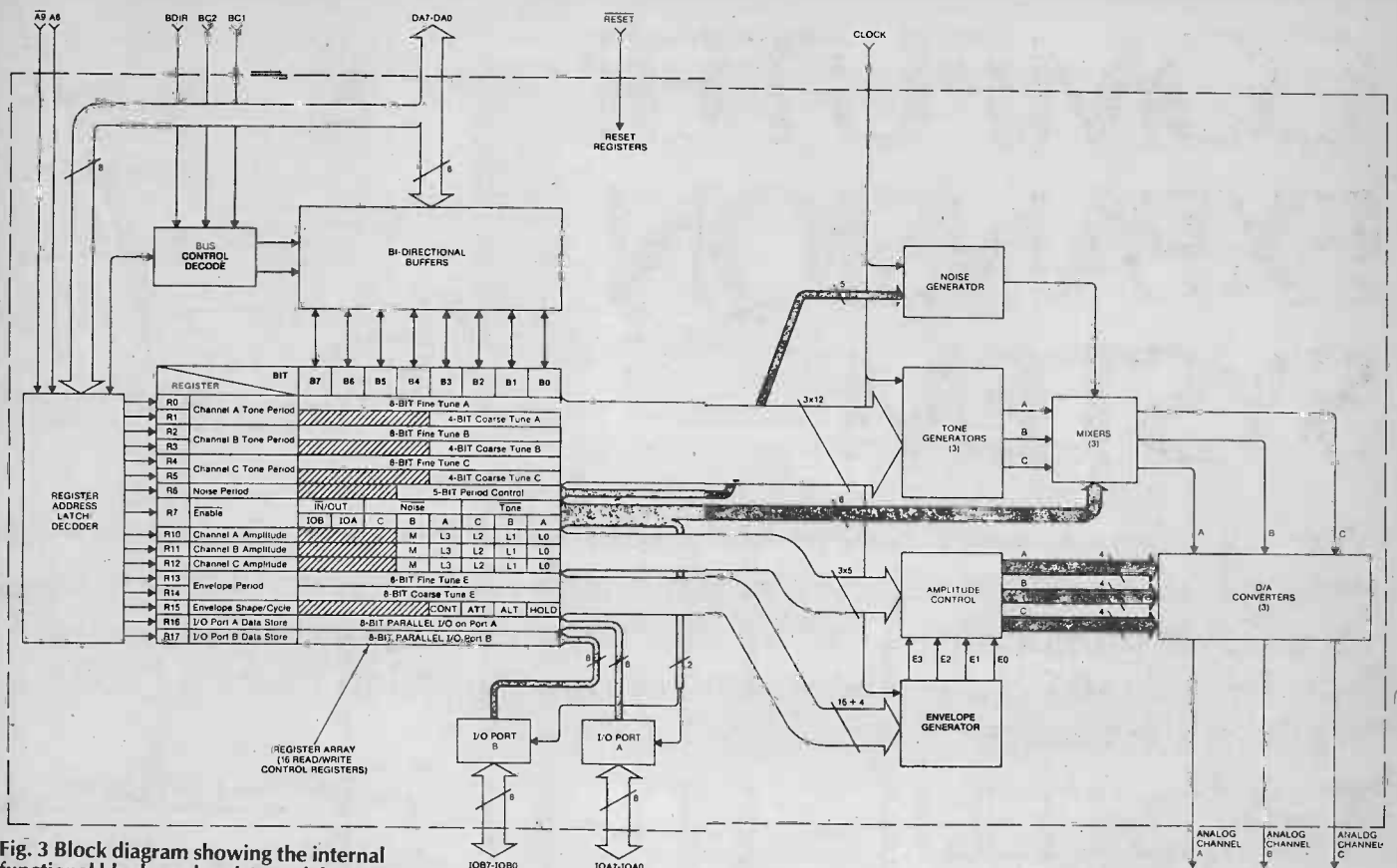


Fig. 3 Block diagram showing the internal functional blocks and registers of the AY-3-8910 PSG.

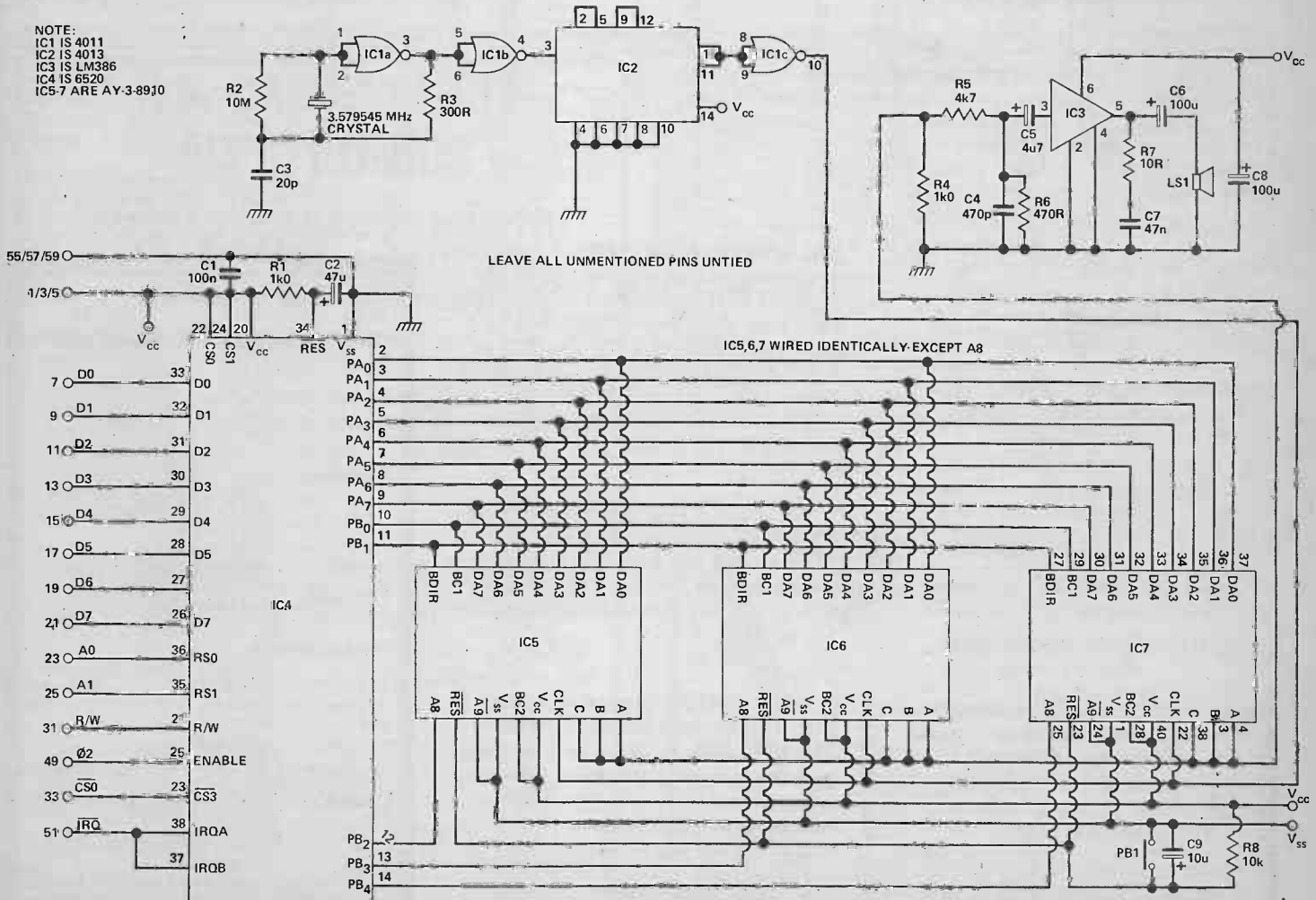


Fig. 4 Complete circuit diagram of the Sound Card.

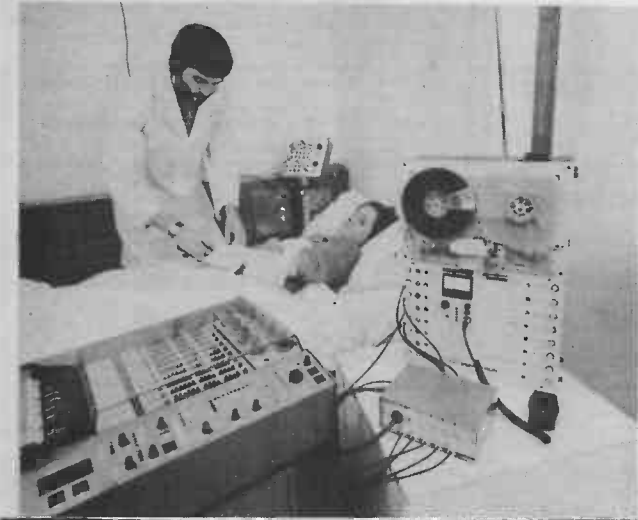
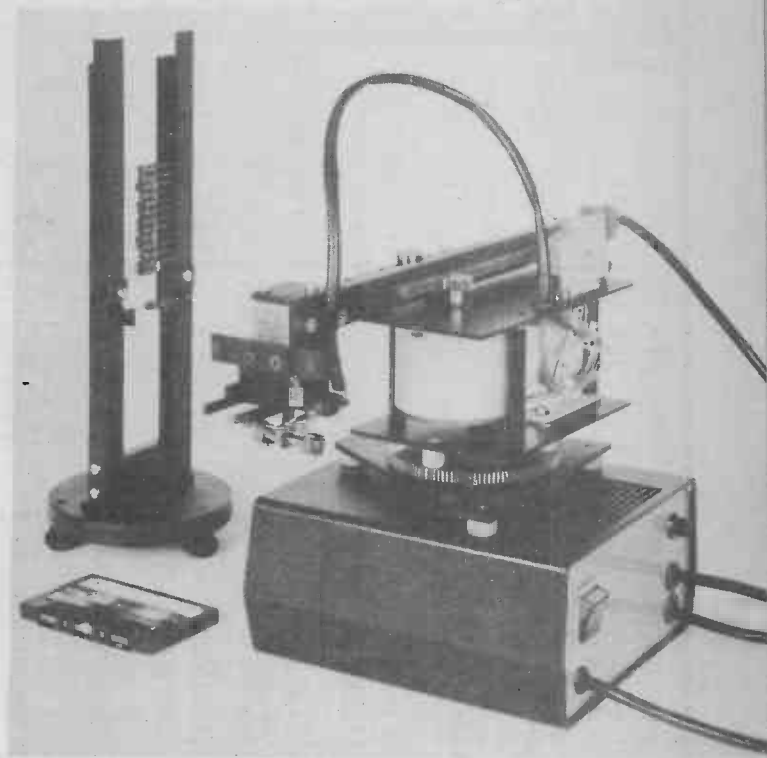
# DIGEST

## Teaching Robotics

A complete package for the introduction of robotics into engineering is being launched by Didatec. Using the microprocessors which are now available in most colleges and many schools, it is possible for engineering students to be introduced to the principles of robotics. As an add-on package, Robert, the intelligent robot, provides the basic requirements for this area of study. He is equipped with linear and rotary movement and an arm mounted gripper. Robert is not simply a 'pick and place' device: he can take resistors from a magazine of mixed units, measure them and sort them to a number of stations. Robert is designed to connect to most popular microcomputers, as the practical RAM area required is only 8K. Once he has been linked to the computer, he simply has to

wait for his instruction program to be loaded into memory from the cassette supplied, and he's ready for action! To help students understand the basic program, a large-scale flowchart is included together with a printout of the program. Robert's electronic interface covers the following requirements: decoding for eight address lines and one select line; flexible address location; data latches and power drivers for stepper motors and solenoid; A-D converter and drivers for resistor measurement and timing logic.

Familiarity with the basic operation will enable students to create their own robots in more advanced projects. For further information on this subject contact: Didatec Engineering Teaching Equipment, 19 Peel Street, Marsden, Huddersfield HD7 6BW.



## Brain Probe

Psychiatrists are using electronics to probe the unknown regions of the brain during sleep at the Institute of Psychiatry in London. Data from the findings is recorded on an SE Labs instrumentation tape recorder — the portable SE 8/4 — and then played back for in-depth analysis. The research team is attempting to make a contribution to the understanding of the pineal gland in the brain about which little is known, except that its malfunction might possibly predispose an adult to mental illness. Because the frequency of the electroencephalogram (EEG), at 0-20 cycles per second, is too low to be measured by an ordinary tape recorder, the Institute selected the frequency-modulated SE 8/4 to ensure precise reproduction of their results. Four of the machine's eight channels are utilised for monitoring sleep, and two for recording a voice commentary and time. The time channel puts regular blips on the EEG tape, allowing various stages to be logged. Any combination of FM and Direct data channels can be used and voice interrupt on one data channel is a standard feature. The FM replay modules are capable of directly driving galvanometers in an SE Labs UV oscillograph. A built-in calibrator permits system check-out and signal monitoring without the use of other test equipment. SE Labs (EMI) Limited is based at Feltham, Middlesex and is a member of the Thorn EMI Group.

## A Question Of Microchips

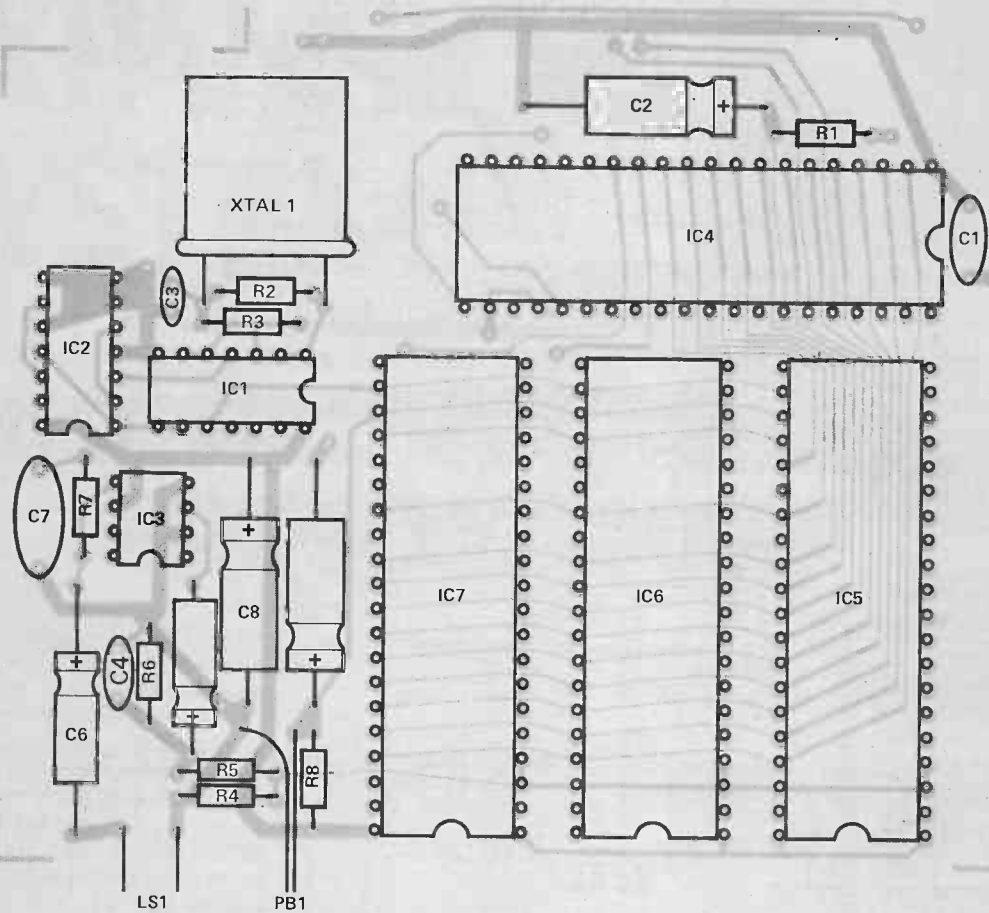
Up until now, no-one has known the extent of the use of the silicon chip in industry in Britain. So, the Policy Studies Institute has undertaken a nationwide survey of 1200 manufacturing establishments in order to discover the hard facts and figures of what is going on in this key new technology. A full report on the survey is to be published in the autumn, but because of the immediate interest in the findings, the most significant figures are being released in four interim reports, the first of which is 'Microelectronics In Industry: Extent Of Use'. The other three are: 'Advantages And Problems', 'Awareness and Government Support' and 'Manpower and Training'; all at £5.00 each. The survey was designed by PSI and carried out by Makrotest Ltd. Interviews were undertaken by telephone in January and February 1981 and the coding and first computer analysis were done in March to June 1981. The sample was made up of 1200 manufacturing establishments, with 200 in each of six size bands and the distribution within each chosen to reflect the national distribution. The size and representativeness of the sample and the unusually low refusal rate (under 13 per cent) suggest a high degree of reliability in the findings. For further information on this subject contact: Jim Northcott or Petra Rogers at the Policy Studies Institute, 1/2 Castle Lane, London SW1E 6DR

## Microelectronic Medicine

HED Electronic systems Ltd have undertaken the marketing and service support in the UK of a new technologically advanced digital sphygmo-manometer; an instrument for accurately measuring blood pressure. The makers of this new instrument, ASULAB SA — one of Switzerland's leading watch manufacturers — decided some years ago to apply their knowledge of volume quality production in microelectronics to wider fields, including medical instrumentation. This led to a period of research and development in collaboration with two leading Swiss clinics and ASULAB has now concluded 12 months of trials with

their sphygmo-manometer. Combining modern microelectronics techniques with data storage and liquid crystal display, the compact unit affords simple one-handed operation without use of a stethoscope, automatically measuring systolic and diastolic pressures and pulse rate. This makes it easy to use for people who must regularly monitor their own blood pressure. The unit is priced at £96 plus VAT and is supplied in a robots fitted case, complete with pressure cuff with built-in microphone. Further details from HED Electronic Systems Ltd, Victoria Avenue, Harrogate, North Yorkshire HG1 1DX.

Fig. 5 Component overlay for the Sound Card. This can be plugged in to any of the motherboard sockets that is convenient.



Since  $TP = 254$ , then

$$R1 + \frac{R0}{254} = \frac{254}{256}$$

and so  $R1 = 0$ ,  $R0 = 254$ .

Now that may seem a complicated route to generating a desired tone but it is easily incorporated into a single BASIC line.

$TP = 1.7898E 06 / (16 * F)$   
 $R1 = INT(TP/256) : RD = TP - 256 * R1$   
 where  $F =$  desired tone in Hertz.

From a BASIC program we have the ability to generate chords and tunes based on up to three notes at a time. More sophisticated software can include proper timing of notes and rests, trills, arpeggios, and so on.

The output of the three tone generators is taken to the input of a mixer block along with the output of a

noise generator. The mode of operation of the mixer block is governed by the contents of R7. Individual amplitude of each tone generator is controlled by R8, R9, and R10 (R10, R11 and R12 in octal). Overall amplitude and envelope shape control is possible using the contents of R11, R12, and R13 (R13, R14 and R15 in octal).

R13 and R14 (octal) are responsible for the envelope period or duration. Durations of less than 1/5000 of a second to 6 or 7 seconds are possible. R15 (octal) sets the envelope shape (see Fig. 6).

The Watford Electronics sound card has provision for up to three AY-3-8910 sound chips (one is supplied with the kit). This means that complex tunes can be simulated. For budding arcade gamers the sky's the limit — you can cram your program with explosions and 'invader' noises. The card, when constructed, will simply plug into any of the sockets on the motherboard.

## PARTS LIST

### Resistors (all 1/4 W, 5%)

R1,4	1k0
R2	10M
R3	300R
R5	4k7
R6	470R
R7	10R
R8	10k

### Capacitors

C1	100n ceramic
C2,5	4u7 16 V axial electrolytic
C3	20p ceramic
C4	470p ceramic
C6,8	100u 16 V axial electrolytic
C7	47n polyester
C9	10u 16 V axial electrolytic

### Semiconductors

IC1	4011B
IC2	4013B
IC3	LM386
IC4	6520
IC5,6,7	AY-3-8910

### Miscellaneous

XTAL1	3.579545 MHz crystal
PB1	non-locking push-to-make switch
LS1	8R speaker
PCB (see Buylines);	eight pin DIL socket; two off 14 pin DIL sockets; four off 40 pin DIL sockets.



# PROJECT : Computer Expansion Part 2

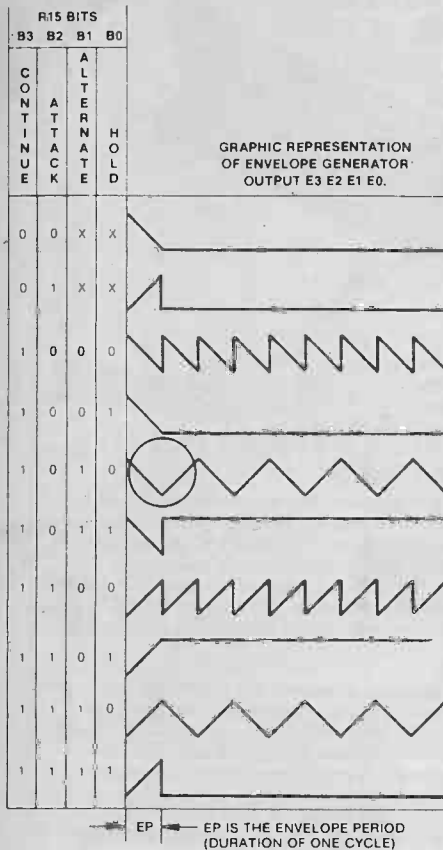


Fig. 6 (left) Register R15 in the AY-3-8910 controls the envelope shape of the sound produced. Here are the waveforms possible.

Watford Electronics can supply a data manual for the AY-3-8910 sound chip to purchasers of the kit for £2.50. The manual contains examples of register values for explosions, racing cars and so on. **ETI**

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# VOLTAGE~CONTROLLED POTENTIOMETERS

Normally pots are used to control voltage, but as Keith Brindley explains, the TDA 1074 uses voltage to control pots. This circuit design feature should put new life into your hi-fi.

In the June issue (pp 70-74) we saw how two of the chips in Mullard's range of voltage-controlled audio ICs could be used for signal switching and fixed-slope filtering in remote, touch, or computer-controlled preamplifier applications. The next stage in such a preamplifier will consist of volume and tone circuits.

In an ordinary, manual preamplifier these functions are provided by potentiometer control — the pot simply acting as a variable potential divider of the signal. Inevitably, because the pot is mounted away from the PCB (or at best, on it), a loop is formed through the pot which tends to pick up interference. Techniques such as screened cabling, PCB mounting of pots and so on reduce the amount of interference pickup, but only to a limited extent. Electronic potentiometers, however, can create a further, significant reduction in interference, since they are voltage-controlled and have no interference loops.

Mullard's IC, the TDA1074, can act as four voltage-controlled pots ganged into two completely separate double electronic pots. Use of the IC thus allows the active controls to be at PCB level, and coupled with good board design this means that few or no interference loops will be formed. Control of the 'wiper' position of the pots is by DC control voltage, making them an ideal choice in the volume and tone control stage of a remote, touch, or computer-controlled high-fidelity preamplifier.

## Go For The Pot

Mullard's principle of voltage-controlled potentiometers is quite straightforward: the position of the 'wiper' of a potential divider within the IC is controlled electronically by a DC control voltage and the output from this wiper feeds an inverting op-amp. Figures 1 and 2 show how this principle can be used in two ways. In both configurations we can divide the potential divider into two parts:  $\alpha$ , and  $(1 - \alpha)$  where  $\alpha$  is the ratio of

resistance to one side of the wiper and  $(1 - \alpha)$  is the ratio of resistance to the other side.

Inserting imaginary values of resistors ( $R1 = R3 = 10k$ ,  $R2 = 1M\Omega$ ) into Fig. 1 we can calculate the gain ( $G$ ) of the circuit. By inspection, when  $\alpha = 1$ , ie when the wiper is at the far right of the potential divider,

$$G = - \frac{R3}{R1 + R2} = - \frac{10k}{10k + 1M\Omega} = - \frac{1}{100}$$

The negative sign is required because we are using an inverting amp. Similarly, when  $\alpha = 0$ ,

$$G = - \frac{R2 + R3}{R1} = - \frac{1M\Omega + 10k}{10k} = - \frac{1}{100}$$

So the range of gain in this imaginary example is approximately  $\pm 40$  dB.

The gain of the circuit of Fig. 2 can also be calculated by inserting imaginary resistor values ( $R1 = R4 = 10k$ ,  $R2 = R3 = 1M\Omega$ ).

When  $\alpha = 1$

$$G = - \frac{R4}{R3} = - \frac{10k}{1M\Omega} = - \frac{1}{100} = -40 \text{ dB}$$

And when  $\alpha = 0$ ,

$$G = - \frac{R2}{R1} = - \frac{1M\Omega}{10k} = - \frac{1}{100} = +40 \text{ dB}$$

(Once again the outputs are inverted). So in this imaginary example, the range of gain is also  $\pm 40$  dB.

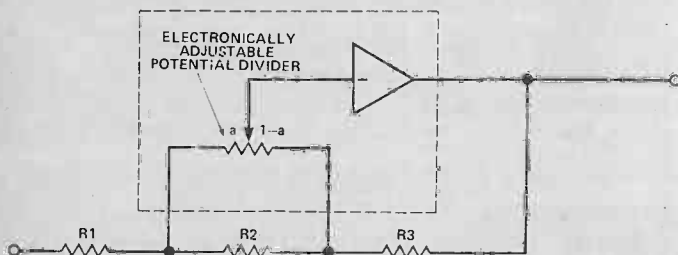


Figure 1. One of the two basic ways in which the gain block (the part of the circuit shown within the broken lines) of the TDA1074 can be used as a voltage-controlled potentiometer.

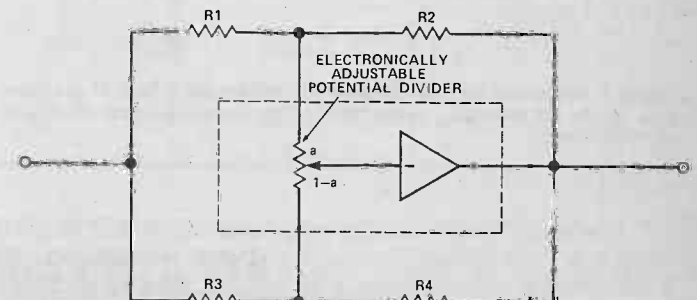


Figure 2. The second way in which a gain block of the TDA1074 can be used to form a voltage-controlled potentiometer.

## Setting The Tone

These two examples show how voltage-controlled amplifiers/attenuators can be easily made. Their frequency responses will be level. In contrast, the frequency responses of tone controls are *not* level — the circuit will have different gains at different frequencies. For example, turning the treble control up in an amplifier system increases the amplitude of the higher frequency components in the applied signal; turning the control down decreases the amplitude.

The circuits of Figs. 1 and 2 can be adapted to form variable-slope filters such as tone controls, simply by replacing one or more of the resistors in the circuits with capacitors. Of course, a capacitor has a 'resistance' (correctly speaking, a reactance) which varies with frequency, so the gain of the circuit will also vary with frequency. Replacing all resistances with  $Z$  values (where  $Z$  can be the resistance of a resistor or the reactance of a capacitor, both measured in ohms) the gain of the circuit of Fig. 1, at any one frequency, will vary between the limits

$$G = -\frac{Z_3}{Z_1 + Z_2} \text{ to } \frac{Z_2 + Z_3}{Z_1}$$

depending on the position of the potential divider wiper.

Similarly, the gain at any one frequency of the Fig. 2 circuit will vary between the limits

$$G = -\frac{Z_2}{Z_1} \text{ to } -\frac{Z_4}{Z_3}$$

depending on the position of the wiper. In other words, the circuits can be used to form voltage-controlled variable-slope filters. Such filters will be discussed later in the applications section.

Figure 3 shows a simplified internal circuit of the TDA1074 built up using the basic op-amp stages of Figs. 1 and 2. Op-amps 1a and 1b form one double-ganged pot, whose output is at  $V_{SS}/2$ . Decoupling/smoothing capacitors are required from pins 1 and 8 for this voltage.

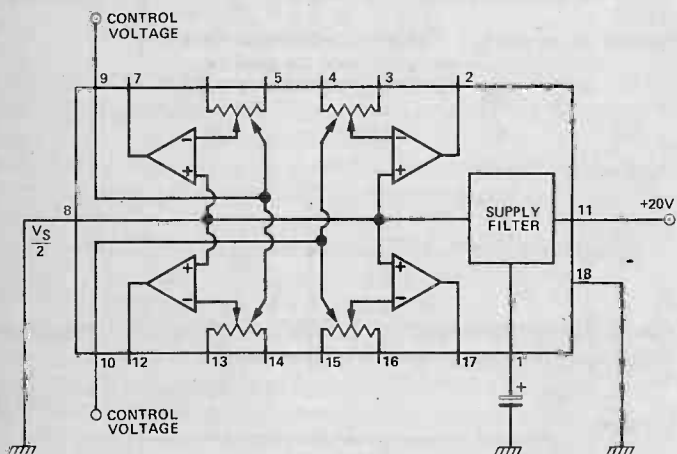


Figure 3. Simplified internal diagram of the TDA1074. Four of the basic gain blocks are internally connected as two, double-ganged, electronic potentiometers.

Maximum control voltage range (applied directly to pin 9 or 10) is  $\pm 1$  V of half-supply (eg using a supply voltage of say, 20 V, the control voltage range is 9-11 V) but most gain change occurs within  $\pm 200$  mV of  $V_{SS}/2$ . The most convenient way to derive a suitable control voltage range of 9V8 to 10V2 is by using a voltage divider from the power supply and the output from pin 8 (the filtered  $V_{SS}/2$  supply). Fig. 4 shows the idea.

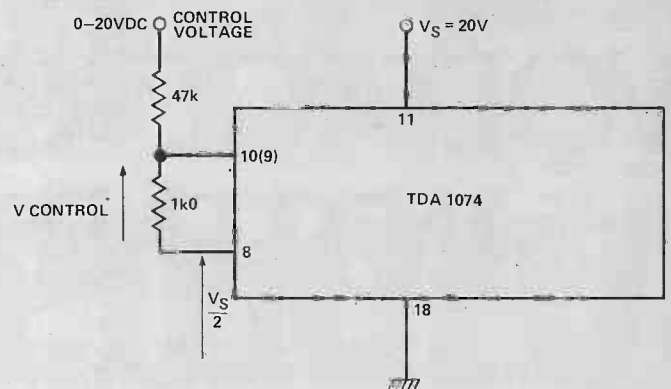


Figure 4. A simple potential divider circuit of only two resistors means that a control voltage range of 0 — 20 V DC can be used. Other voltage ranges can be selected by a suitable choice of resistors.

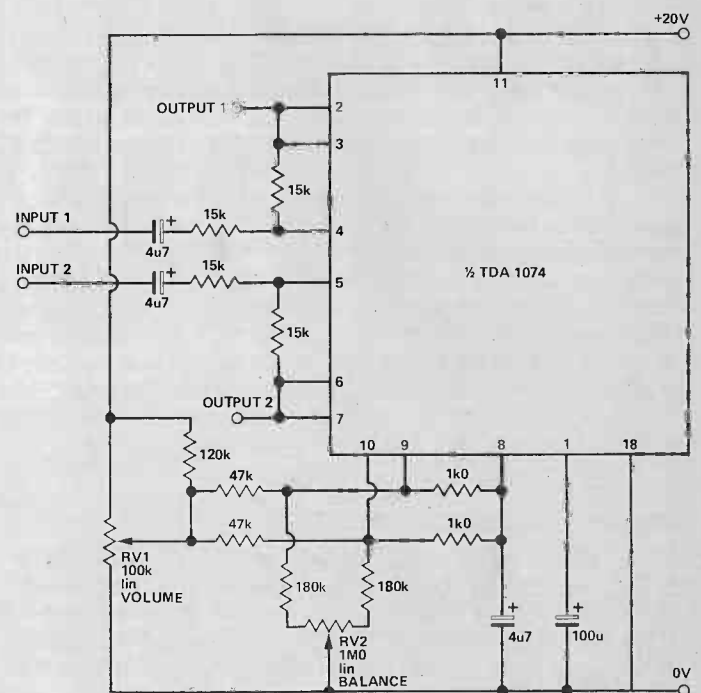


Figure 5. Stereo volume and balance controls obtained with only one half of the TDA1074.

## Applications

Volume and balance controls can be made by straightforward adaptation of the gain block circuit of Fig. 1. By having no resistance for  $R_3$  the maximum value of gain becomes  $R_2/R_1$ , and the minimum, 0. If  $R_2 = R_1$ , as in Fig. 5, then the circuit acts as volume control with a range of zero to unity gain.

Balance between two parallel audio channels is most easily achieved by adjusting the ratio of DC control voltages between the two. In Fig. 5, pot RV2 reduces one control voltage down toward 0 V more than the other, depending on the position of its wiper.

A superior balance control is achieved by separating it from the volume control into its own circuit. Figure 6 gives the circuit with suggested component values. At a control voltage of 10 V, the two halves of circuit each have unity gain. At the extreme ranges of the control voltage, one channel will have a gain of about 2 (+6 dB), as opposed to 1/30 (-30 dB) for the other.

## Mixing it

The basis of a high-quality voltage-controlled stereo mixer is shown in Fig. 7. The standard gain block is used to increase the level of one signal whilst decreasing the level of the other. At an input control voltage of 10 V the gains of the circuit are the same. Stages can be cascaded if required.

Maximum gain of each input is defined by the ratio of feedback resistor to input resistor on that input, eg  $R_2/R_1$ ,  $R_4/R_3$ ... Unity gain is thus obtained when  $R_2 = R_1$ ,  $R_4 = R_3$  and so on.

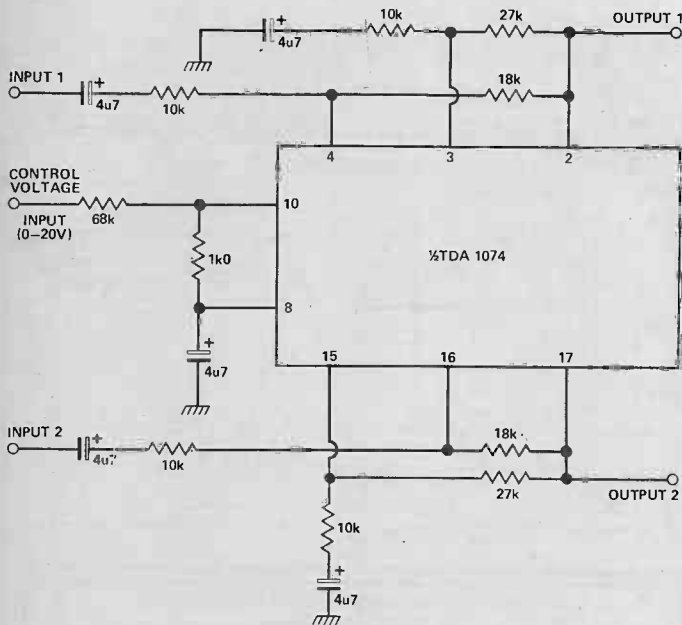


Figure 6. Superior balance control. A control voltage input of 10 VDC gives equal signal gains from both channels.

A variable stereo image width control is shown in Fig. 8. This can be used in place of a stereo/mono switch if fully variable control of signals is desired between the two extremes of stereo (complete separation) and mono (complete crosstalk). The effect is produced by feeding a controlled portion of the input of one channel to the input of the other. Varying the control voltage alters the amount of this crosstalk so maximum and minimum separation occurs.

## Voltage Controlled Filters

By replacing certain resistances with reactances as explained previously, bass and treble tone controls can be formed. The treble controls in Fig. 9 are, in fact, adaptations of Fig. 2 with capacitors added (in parallel with  $R_2$  and  $R_3$  of Fig. 2), forming frequency dependent 'potentiometer'. Similarly, the bass controls in Fig. 9 are taken from Fig. 1 (with a capacitor in parallel with  $R_2$ ).

Frequency response curves of the whole circuit are given in Fig. 10. Maximum cut and lift of the controls are seen to be about  $\pm 14$  dB at 60 Hz and 10 kHz and are completely variable, electronically, between these extremes.

Finally, to show that the TDA1074 can be used in other voltage-controlled filter applications, a presence control is given in Fig. 11. Presence is an effect where amplification of frequencies around 1 kHz takes place with little or no amplification of other frequencies. The effect is used mainly in live music work, where it can apparently boost (give presence to) the level of a singer's voice, compared with the backing music. Frequency response curves for the circuit are shown in Fig. 12.

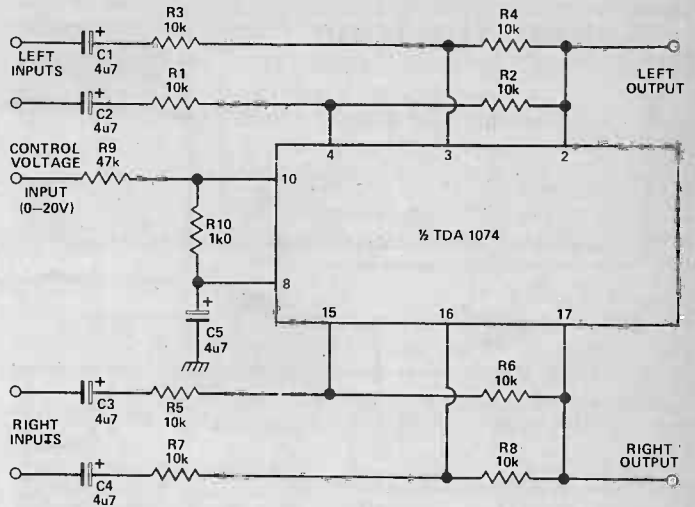


Figure 7. Voltage-controlled stereo mixer application using the TDA1074.

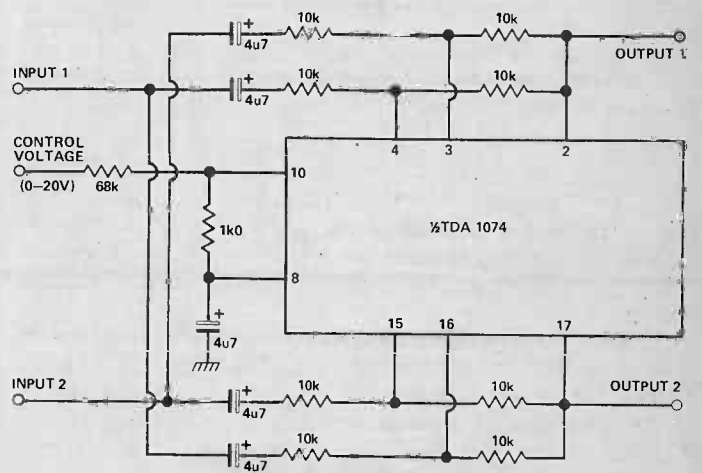


Figure 8. A completely variable stereo/mono control. A full stereo output is obtained when the applied control voltage is 20 V DC. Mono output occurs with a control voltage of 0 V DC.

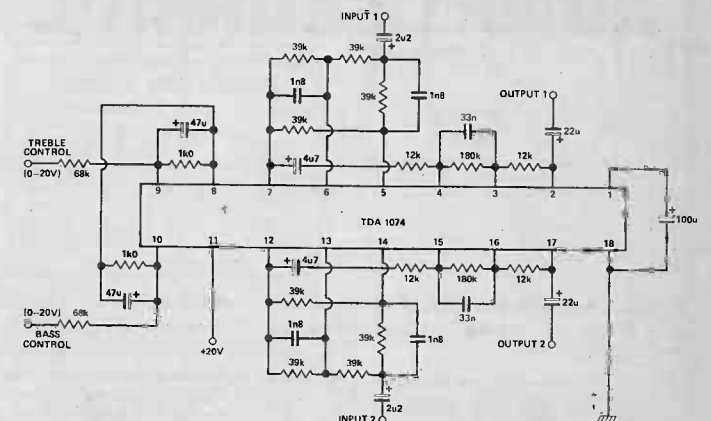


Figure 9. Bass and treble controls obtained by replacing chosen resistances from Figs. 1 and 2 with reactances.



# FEATURE : Voltage-controlled Pots

In conclusion, it is apparent that many more applications of this IC are possible and depend only on the designer's ingenuity. You can see from the basic gain block circuits of Figs. 1 and 2 how easy it is to make voltage-controlled amplifiers and filters by the simple choice of resistances or reactances.

Figure 11. Voltage-controlled presence control to boost frequencies around 1 kHz.

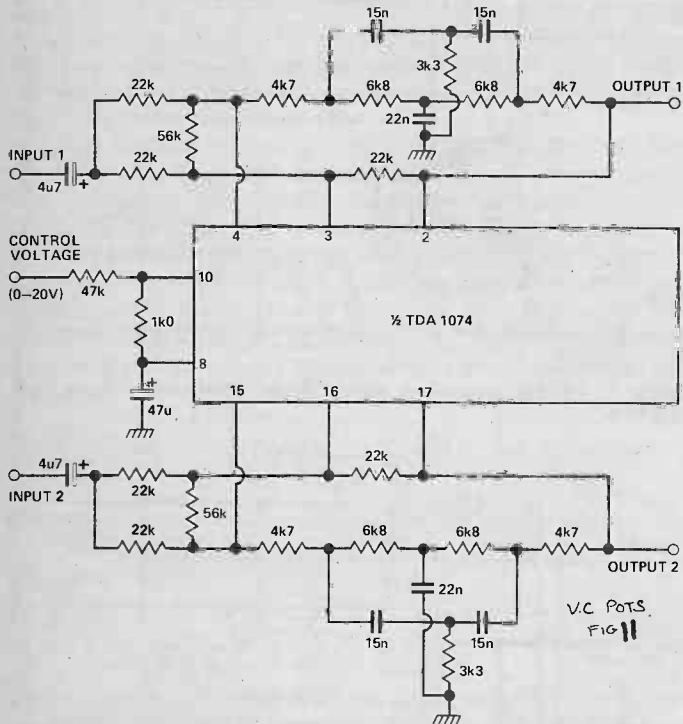


Figure 10. Frequency response curves obtained with the circuit of Fig. 9.

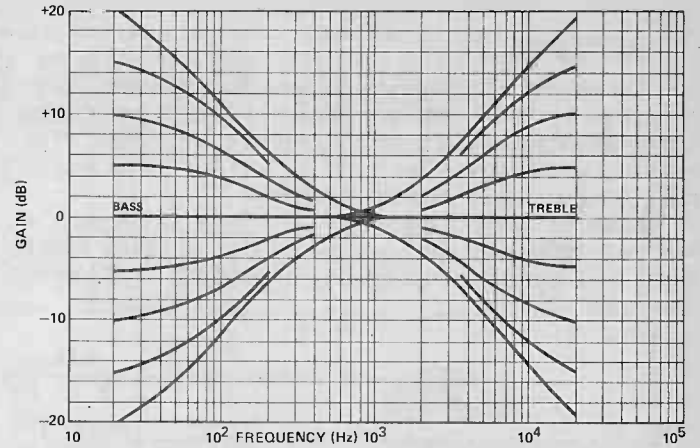
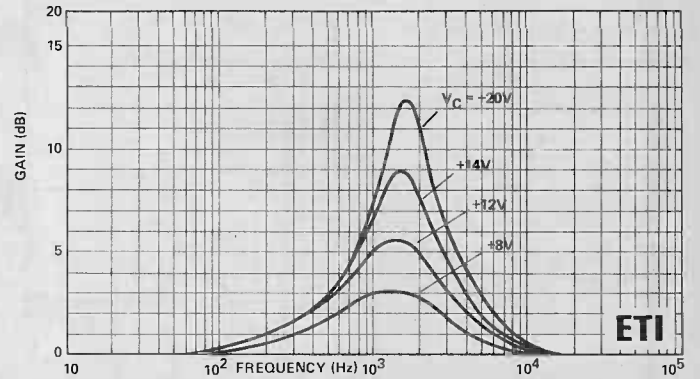


Figure 12. Frequency response curves of the voltage-controlled presence control circuit of Fig. 11.



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# DUMMY LOAD

Apart from a multimeter and perhaps an oscilloscope, a resistive dummy load of 4, 8 or 16 ohms impedance capable of dissipating up to 100 W is just about the most useful item of test equipment the audio enthusiast could have. Design by Andrew Kay.

The project staff at ETI have spent some considerable time over the past few years developing a variety of amplifiers. The fruits of these labours have been duly published and enjoyed by many readers. However, we've always lacked a decent dummy load for such work and have sort of *made do*. Whilst jury-rigging such things is in the finest traditions of electronic design and development, the (more than) occasional mishap is not just a frustrating interruption but often a decided nuisance.

In the meantime, a freelance associate of ours, Andrew Kay, had desired exactly the same thing — 'a decent dummy load'. He purchased a batch of one watt 1% resistors and made a 50 W dummy load. But, he figured, why not have a little more versatility and make two the same, allowing parallel and series connection to obtain a 4 ohm, 100 W dummy load or a 16 ohm 50 W dummy load as well as a twin 8 ohm 50 W dummy load enabling testing of both channels of a stereo amplifier at the same time! Frankly, we don't know why we didn't think of it earlier ourselves.

## Multi-resistors

By paralleling resistors of an appropriate value, one can obtain an effective resistance of the wanted value and wattage rating. Now, the cheapest, most common power rating for carbon film resistors is one watt (1 W). To obtain a 50 watt resistor, 50 would need to be paralleled. To obtain an effective resistance of 8 ohms, each 1 W resistor would have to have a value of 400 ohms. The nearest preferred value is 390 ohms. Fifty in parallel would give an effective resistance of 7.8 ohms which is about 2½% lower than the ideal 8 ohms. However, 49 in parallel gives an effective resistance of 7.959 ohms — less than ½% out. If you require the tolerance of your load to be within 1%, or better, then you'll have to use 1%, 1 W resistors. If you only require a tolerance of ±5%, then the common 5%, 1 W variety will do the job. Either

way, you're better off using 49 resistors so that the effective resistance of the load comes closer to the ideal 8 ohms.

The dummy load described here consists of two 8 ohm loads, which enables the testing of both channels of a stereo amplifier. The idea itself is not all new or original, having been used by radio amateurs for years to obtain resistive dummy loads for terminating radio transmitters while they are on test. The advantages of a dummy load for any kind of power source are:

- the power source (in this case an AF power amplifier) is presented with an ideal resistive load of the correct value
- the chances of damaging expensive loudspeakers during experimental phases of construction are eliminated
- completely silent 'full power' testing is made possible even for extended periods of time; which is great for public relations (and your ears.)

## Parallel Approach

Essentially each dummy load consists of 49 high stability 1% metal film resistors connected in parallel to give a terminal resistance of 8 ohms. Since the tolerance rating of the resistors is 1% the upper tolerance limit for the combination is 8.04 ohms and the lower limit is 7.88 ohms. The number of resistors to be bought was a compromise between the desire for a result of exactly 8 ohms and the need to keep the cost to a minimum.

Obviously, larger numbers of resistors could be used (say 70 x 560 ohms in parallel) and the reader can easily vary the circuit to suit the pocket

and availability of the resistors.

Separate terminal posts are provided for each load so that two separate 50 W sources can be terminated in 8 ohms each or the two halves may be connected in series to give a single 16 ohm 50 W load; and, last but not least, parallel connection of the two halves will result in a 4 ohm 100 W load. Because metal film resistors are used there are no inductive effects to worry about such as could occur if wirewound units were employed. The stray capacitances present are so low as to be insignificant.

## Construction

Construction is simple, if somewhat tedious. Lots of soldering is involved! The author used two ordinary household tin cans; one can has a lid (eg a coffee tin), the other is a smaller one of the throw-away type. The top and bottom of the smaller can were used as soldering planes for terminating the ends of the resistors, while the larger can was used to house the project with the lid carrying the terminal posts. Since the coffee tin is virtually leak-proof you could fill it with some kind of insulating fluid such as transformer oil and thereby increase the dissipation capability of the dummy loads.

Tin-plated steel is very easy to solder but the sharp edges are dangerous to careless fingers. Blank copper clad printed circuit board could be used instead but does not withstand heat as well as the plain metal sheet.

The arrangement of tin cans may not seem very glamorous but it is highly effective and very cheap — the whole cost for the project comprises about £4.50 for the resistors and about £1 for the terminal posts. The tin can housing can be spray-painted and the terminal posts labelled and marked to suit individual needs.

Before starting, choose a medium-sized tin with a resealable lid for the case and select a tin can of smaller diameter which will fit easily into the coffee tin. About 8 or 9 centimetres in



diameter should be fine for the smaller tin can. Using a can opener, remove the top and bottom of the smaller can and discard the contents (maybe you should eat the contents . . .). Also, discard the remaining cylindrical portion of the can! Mark up one of the tin-plated discs so obtained with a grid of 10 by 10 lines as shown in Fig. 1 to give 100 intersections. Allow a space of about 10 mm along one diameter as shown. This will allow the discs to be cut in half later. Clamp both discs together on to a drill bench or a block

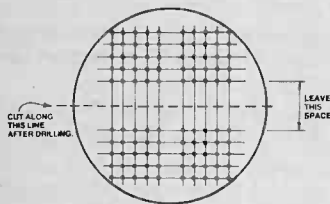


Fig. 1 Drilling and cutting details for the tin-plated discs obtained from a small can.

of wood, ensuring that they are exactly superimposed. Drill a hole on each intersection of the previously marked lines. Make the holes slightly larger than twice the diameter of the resistor leads; this will assist assembly later on. Take care that your hands are kept clear during drilling since if the drill bit grabs, the two tin discs will whirl around very much like a meat slicer, and almost as sharp! Only 98 holes are needed so don't get carried away.

When the holes are drilled, cut the two discs along the middle space left along one diameter so that you end up with four half discs each with 49 holes. Tin the area around each hole with solder and proceed with assembly.

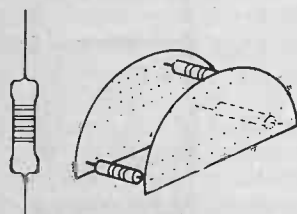
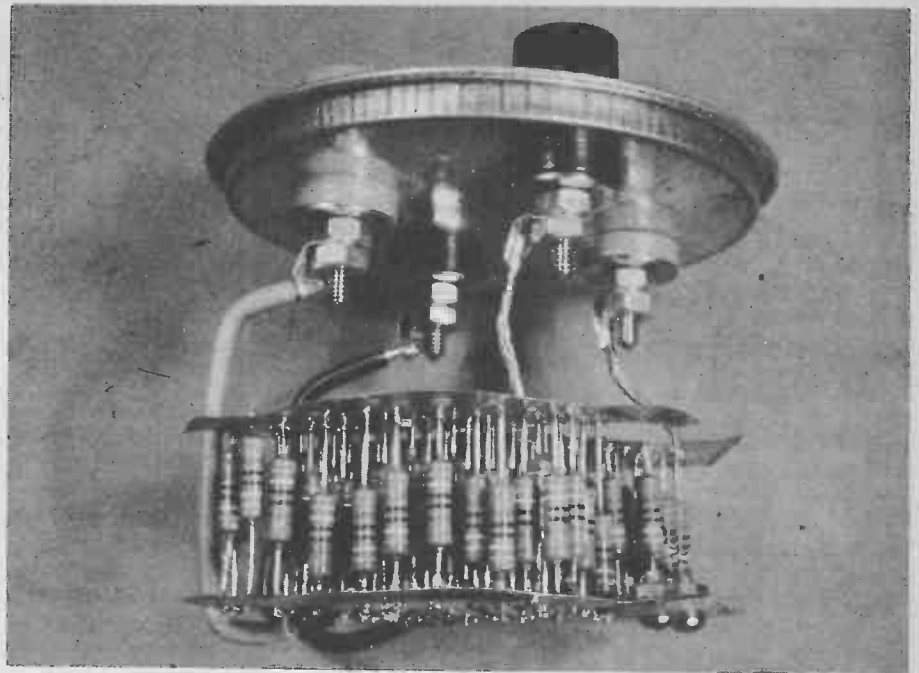


Fig. 2 Cut the resistor leads as shown at left and then solder three resistors to two half-discs as shown to make a rigid assembly.

## Lead Time

Trim all the resistor leads as shown in Fig. 2 so that one lead is longer than the other on each component. Take two matching half-discs and, using three resistors, assemble a rigid structure as shown in Fig. 2. Insert the resistors, one row at a time, in between the two tin plates with the leads poking through the holes. If you insert the longer lead of each resistor into its hole first, the other end should be short enough to allow manoeuvring into the hole in the second plate. After one row of resistors is in place solder all the



leads of that row on both plates, then proceed with the next row.

Repeat the assembly for the second half of the unit, then trim all excess leads flush with the surface of the tin plate.

Connect an ohmmeter between the plates of each load — the reading, believe it or not, should be pretty close to 8 ohms. Inspect all solder joints and resolder if the reading is not correct. Install the four terminals in the lid of the tin using one red and one black terminal for each half of the unit. Lay the two assembled resistor pads side by side as shown in Fig. 3. Using fairly stiff copper wire connect the upper plates to one terminal each. Use the same colour terminal for both plates as this will be important later if the loads are to be connected in series or parallel. Using the same sort of wire, but insulated, connect the lower sides of the resistor assemblies to the other two terminals. You should finish up with an assembly which will be supported under the lid of the tin and which is so positioned as to allow it to be inserted into the container for the lid to be sealed.

To prevent the two halves of the load from shorting together, install an insulating spacer between them using a scrap piece of copper-clad board or matrix board. If using the PCB material, ensure that enough copper is removed to insulate the two halves from each other. If using the matrix board, you will have to drill a couple of additional holes and use small screws to attach the spacer to the resistor assemblies.

Before inserting the assembled unit into the container, mark the lid to indicate which terminals are connected by the resistive pads.

To test the unit, connect each load across a known working amplifier or if this is not convenient, use a car battery (not more than 12 V) as the driving source. If using an amplifier, connect an AC voltmeter across the load under test. If you can use a sine wave generator to drive the amplifier, all the better. Adjust the amplifier volume control to give about 10 to 15 V across the loads. Check by feeling the resistors with your hand that they are in fact warming up. Increase the output of the amplifier until the voltage across the loads is about 20 V. This should result in the resistors getting quite hot after a couple of minutes.

If using a car battery, connect the two loads in parallel and connect the battery across them. Check the current drawn; it should be approximately 3 A with a 12 V battery.

When testing is satisfactorily completed, install the whole assembly into the container and press down the lid. If you plan to use the loads continuously, fill the container with insulating oil before assembling.

ETI

## BUYLINES

Every mail order supplier we know will sell you the resistors and terminal posts! It's worth phoning a few to get the best price on large quantities, however. Try Technomatic and Bi-Pak for starters.

## PARTS LIST

98 off 390R 1 W 1% or 5% resistors (or similar combination, eg 140 off 560R 1 W 1% or 5%). Use carbon film for low inductance. Four terminal posts, two red, two black. Two household tins — see text.

# TECH TIPS

## Salvaging Fluorescent Displays

A.G. Blewett, Ramsey

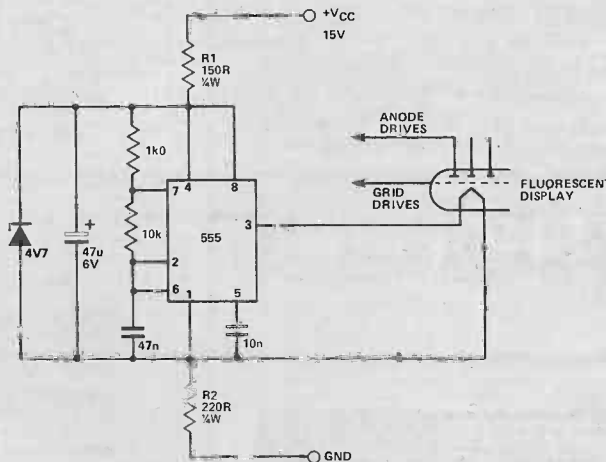
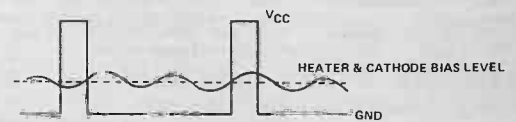
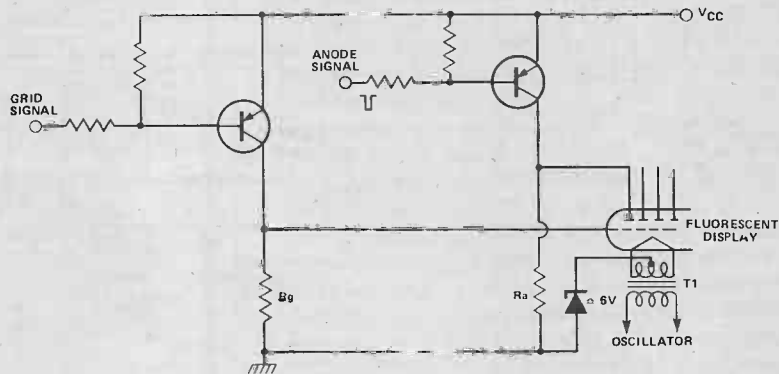
Multi-digit vacuum fluorescent displays (multiplexed operation) are widely available from 'junked' calculators and surplus sources. Unfortunately they are not as easy to use as LED displays, because they need an AC filament supply to ensure even brightness, and this supply needs to be DC biased, because the display operates just like a valve (remember them?). Figure 2 summarises the drive requirements.

In many calculators, this is achieved as in Fig. 1. A transistor oscillator feeds the isolation transformer T1, which supplies current for the filament. The centre-tapped secondary allows the filament (cathode) to be referenced to ground via a zener, so as far as the cathode is concerned the grids and anodes are negative when grounded through  $R_A$  and  $R_C$ . Taking these high via the transistors positively biases them, enabling the display.

Providing you don't need to conserve battery power, the circuit of Fig. 3 provides a convenient alternative to messing about with oscillators and ferrite transformers, and allows single rail operation.

The 555 timer is connected in the astable mode, and supplied from the 4V7 zener. It drives the display filament direct, because the relationship of R1 and R2 gives the correct bias levels to the display electrodes.

The resulting waveforms are summarised in Fig. 4. The 555 produces about 4 V peak-to-peak on load, which is 2



$V_{RMS}$ , just about right for correct filament operation.

Display brightness can be increased by increasing  $V_{CC}$ , but if this is done, the values and wattages of R1 and R2 will have to be changed as well.

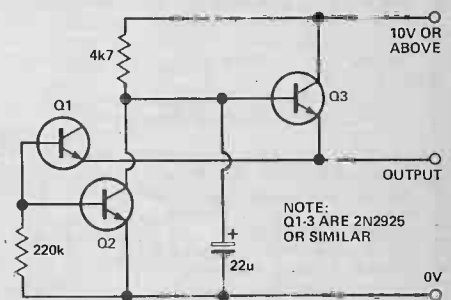


## Cheap Voltage Reference

A. Parker, Leicester

This circuit forms a cheap voltage reference, with the reverse base-emitter breakdown voltage of Q1 in series with the forward base-emitter voltage of Q2 providing the reference element. With this configuration the temperature coef-

ficients of the two base-emitter junctions almost cancel one another out and although separate transistors are shown for cheapness, a dual transistor such as the LM394 could be used to advantage. C1 is used to decouple the noise generated by the zener action of Q1 while Q2 is an emitter follower which, with the values shown, gives at least 20 mA. The reverse base-emitter breakdown voltage of silicon transistors usually lies around 5 V for RF types and up to approximately 9 V for audio types.



NOTE: Q1-3 ARE 2N2925 OR SIMILAR

Tech-Tips is an ideas forum and is not aimed at the beginner. We regret we cannot answer queries on these items. ETI is prepared to consider circuits or ideas submitted by readers for this page. All items used will be paid for. Drawings should be as clear as possible and the text should preferably be typed. Circuits must not be subject to copyright. Items for consideration should be sent to ETI TECH-TIPS, Electronics Today International, 145 Charing Cross Road, London WC2H 0EE.

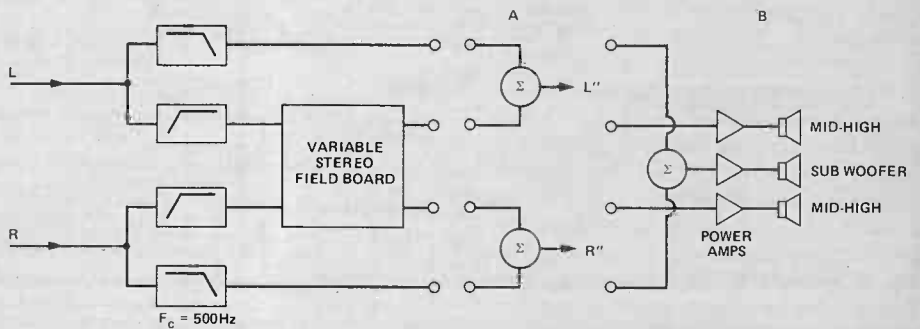
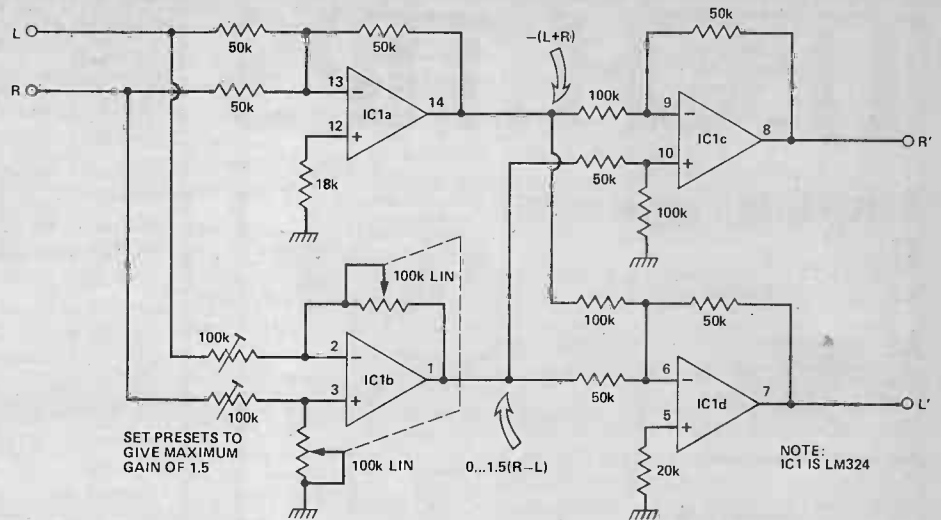


## Variable Stereo Field

F. Vanthuynne, Belgium

Here is a simple circuit, using only one IC, with which the usual stereo field can be varied from mono, over normal stereo, to 'superstereo'. The effect is controllable by only one (stereo) potentiometer: for the left channel from  $L' = (L + R)/2$  (mono) to  $L' = 2L - R$  (superstereo), and for the right channel from  $R' = (R + L)/2$  to  $R' = 2R - L$ . Unity gain is preserved through the entire control range. Pin numbers are given for the 324 quad op-amp, but this IC may be replaced by a lower noise type if desired. Resistors must have 1% tolerance because, if not, sounds in the centre of the normal stereo field may be attenuated in the superstereo position. The effect has to be connected between the preamp and power amp of a stereo system or, in the simplest case, one can use the tape output and monitoring input of a conventional amplifier or receiver.

The superstereo sensation may be increased by adding an active crossover filter as shown. In (a) the low bass part of the signal is bypassed; in the configuration (b), separate power amplifiers with individual gain controls are driven and low bass parts of both channels are summed, since sounds below 500 Hz have only poor directional characteristics.

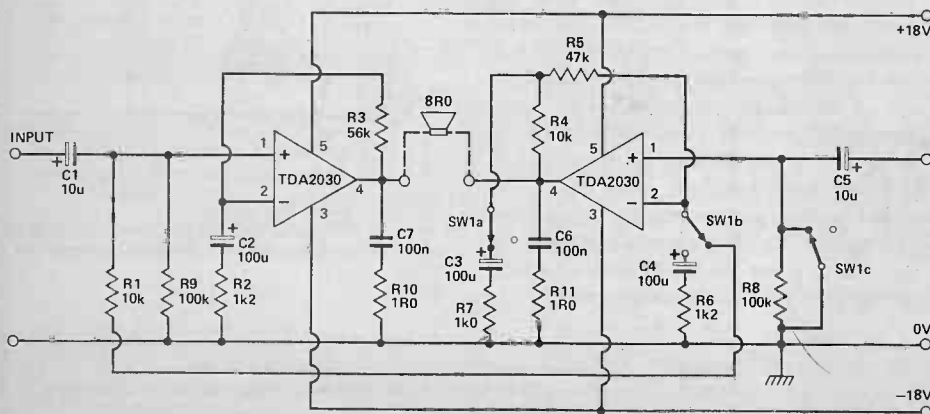


## Switchable Bridge Amplifier

J.P. Macaulay, Crawley

By using the TDA2030 IC audio amplifier it is possible to realise a hi-fi quality amplifier which will deliver 35  $W_{RMS}$  with less than 0.1% THD. This is done by using a pair of ICs in bridge and providing for use as a stereo amp by

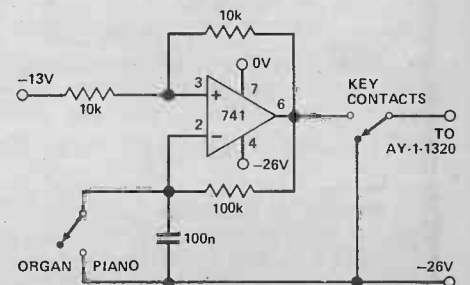
means of switching. The circuit is shown with the switching in the bridge mode. As can be seen, the amplifier requires a minimum of external components and these are only required to set the gain. The amps are housed in TO220 package and must, of course, be mounted on a heatsink, preferably one with a thermal resistance of  $3^{\circ}C/W$  or better.



## Organ Conversion For IC Piano

Martin Anderson, Oxford

This simple oscillator circuit overrides the decay characteristics of the AY-1320 touch-sensitive piano IC to provide the infinite sustain of an electronic organ. In effect, it simulates hitting the piano key with great rapidity, such that the repeated note merges into a continuous sound. The circuit is short-circuit-proof, and can be switched to revert to the original piano sound. If the frequency of oscillation is lowered, a vibrato effect occurs. Conversely, raising the frequency might cause breakthrough into the sound circuits, causing an unpleasant background noise. The values shown have been found to work quite satisfactorily.



## Sonic Measurements

Sonic Tape PLC are launching a first for Britain with their new electronic measuring device called (would you believe) Sonic Tape. This specially designed compact unit is battery operated and will, at the touch of the proverbial button, measure any distance from 2.5' to 80', within an accuracy of  $\pm 1''$ . The result can then be read on a large digital display. The unit is made in either a metric or imperial version. It works by emitting an ultrasound signal and the return echo is measured and displayed as distance. The company says that interest has been shown by architects, surveyors, builders, plumbers, carpet layers and designers; some of whom require more specialised units. The advantage of using a device like this is that only one person is needed to make simple measurements. The tape will be sold for £85 plus VAT and is available from Sonic Tape PLC, 5-11 Worship Street, London EC2A 2BH.



## TK — OK?

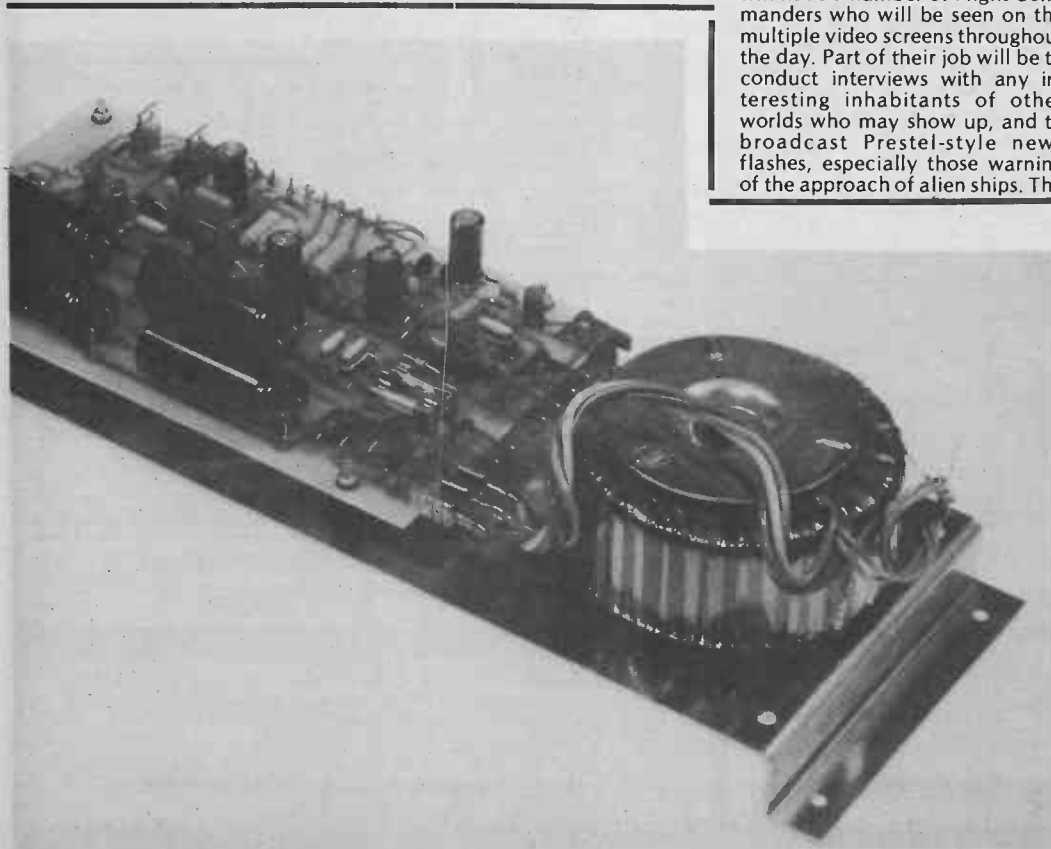
There is now a shortform catalogue available from TK Electronics. It includes lots of new additions to their range, eg a 25 A triac, 74LS TTL, microprocessors and associated chips and all at reasonable prices. Also included are details of their range of kits. All you have to do is to send an SAE to TK Electronics, 11 Boston Road, London W7 3S) and your copy will soon be winging its way to you.

## Burger Blast Off

News Digest is fast turning into Food Digest! Following in the 'foodsteps' of The CB Eyeball Bistro comes the 'Buck Rogers Burger Station', soon to be situated in Queen Street, Glasgow. It is planned to be a facsimile of the interior of a space vessel — to the standards of a 'Star Wars' movie set — in which an array of huge 70-inch video screens and computers will involve patrons in

the dramas of daily life aboard a space station. Charles Waldman, the entrepreneur who instigated this idea, has taken over the first floor of a five storey block at 25-37 Queen Street to house his eating Enterprise, as well as the top two floors to house his computer and video training centre needed to run it. It is expected to create about 100 new jobs, mostly for young people. The Burger Station will have a number of Flight Commanders who will be seen on the multiple video screens throughout the day. Part of their job will be to conduct interviews with any interesting inhabitants of other worlds who may show up, and to broadcast Prestel-style news flashes, especially those warning of the approach of alien ships. The

Station will have a main section to seat 180 people, and a 'shuttle' section seating a further 100, which will be built like a large spacecraft. Diners will be able to observe space ships docking at the station on the surrounding video screens. Food will be served by crew members in futuristic uniforms and a feature of the menu will be the biggest ice-creams in Britain, served in goblets with dry ice emitting eerie down-spilling vapours. There is little chance, however, of being served compressed food pills: it's more likely to be American burgers, Chicken Maryland and Pizzas — all at reasonable prices. We've heard of the Restaurant At The End Of The Universe, but this is ridiculous!



## Amplifier Module

From BK Electronics comes the first of a new range — the Model OMP 100, a 100 W RMS power amplifier module. It comes complete (ready built and tested) with integral toroidal transformer power supply and heatsink. A brief spec is: maximum output, 100 W RMS; loads, 4-16 ohms; frequency response, 10 Hz-20 kHz; sensitivity for rated output, 500 mV @ 10k and the outputs are protected against open and short circuit conditions. The unique feature of this module is that it has power and drive outputs provided to operate a matching LED VU meter. The price? £29.99 plus £2 p&p plus £6.50 for the optional VU meter. All prices include VAT and the unit is available from BK Electronics, 27 Whitehouse Meadows, Eastwood, Leigh-on-Sea, Essex.

# INFANT GUARD

Worried that your offspring will try out the 'sweeties' in your medicine cabinet? Build this project and let your mind rest in peace, not your kids. Design and development by Rory Holmes.



**T**his alarm is designed to protect a medicine cabinet from the inquisitive hands of young children, or indeed any cupboard that has important contents. There are certain requirements for an electronic alarm of this type. First, it should have a satisfactorily loud alarm sound to deter the child from meddling in the cupboard, and also to warn parents in the house that the cupboard has been opened. It must be possible to override the alarm during normal use of the cupboard and it should be easy to re-arm after it has either gone off, or been disabled for normal cupboard use. The alarm should be battery powered and therefore an extremely low stand-by current is required.

The ETI Infant Guard uses a high-efficiency piezo-transducer to give an intense high-pitched note with minimum current consumption. The circuit uses CMOS ICs and the current taken during stand-by allows a PP3 battery to last as long as its shelf-life. In order that the cupboard can be used normally, an override push-button is provided that can be mounted on the cabinet door; if the button is pushed while the door is being opened the alarm will not go off, and will remain disabled while the cupboard is used. Once triggered, the alarm can be switched off with the push-button, and

is reset for the next intruder by simply closing the cupboard again. This will also switch off the alarm.

The unit should be easy to install for any size or shape of cupboard with any number or type of doors. The usual method is to position microswitches or magnetic reed switches against all the doors, but this is a very awkward solution and far from universal.

It was decided to detect the amount of light in the cupboard with a phototransistor. When the cupboard is closed there is a very low light level (effectively dark); as soon as a door is opened the light level increases, which is detected and used to sound the alarm. The phototransistor protrudes through the self-contained alarm box, which is then simply placed anywhere in the cupboard, with only two wires leading to the override pushbutton. A small delay factor has been added in the alarm to prevent false triggering in the event of brief shadows covering the unit during normal cupboard use.

## Construction

Study the PCB overlay diagram first and then solder all components to the PCB noting their orientation. The phototransistor should be mounted with its full lead length above the PCB. The board can be mounted in the small Vero potting box using bolts or, preferably, sticky pads (remember to drill a hole allowing the phototransistor to protrude through the case). The push-button is wired to the appropriate points in the PCB by flying leads let in through a hole in the box.

We stuck our Toko transducer on the outside of the box with epoxy, and drilled a small hole for connecting the leads through to the PCB. The PP3 battery will also fit inside the box, which can then be assembled. (Take a close look at the internal photographs.)

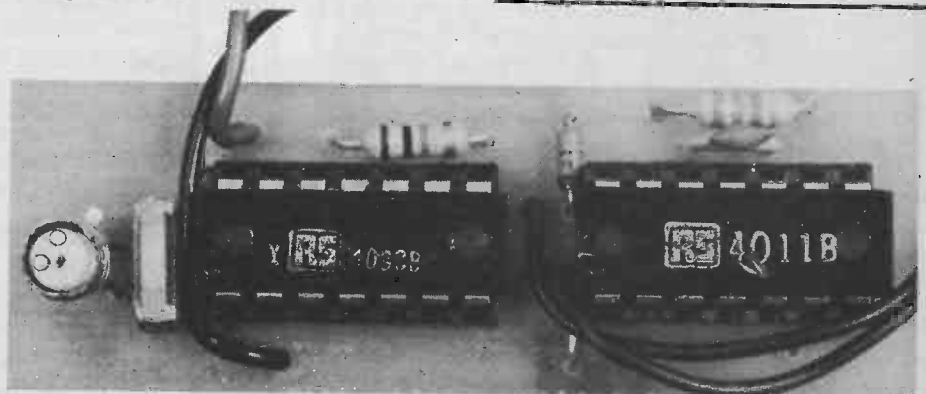
Incidentally, although a BPX25 phototransistor is given in the Parts List, we've found you can get better (and cheaper) results by sawing the top off a BC109.

When the unit is complete, the battery clip can be connected and the project should now be operational. Put the box in a drawer with the push-button hanging over the side; when the drawer is closed and then opened again the alarm will sound. The push-button will now override the alarm as described, and may be mounted on the actual cupboard in any desired position.

This project is simple and cheap and you might like to make a couple of them for use in different cupboards or drawers, screwing them permanently inside. Since nothing is mounted on the lids you might as well leave them off and mount the project with the opening against the drawer side. That leaves you with two spare lids — just the thing to house the Parking Meter Timer on page 29!

## BUYLINES

Bi-Pak, Watford and Technomatic are just some of the mail order companies who will be delighted to supply you with the components for this project. We will, in turn, be delighted to supply you with the PCB (see advert on page 98).



Close-up of the PCB for the Infant Guard project. Somewhat small, is it not? The peculiar-looking component is a BC109 transistor with the top sawn off.

# PROJECT : Infant Guard

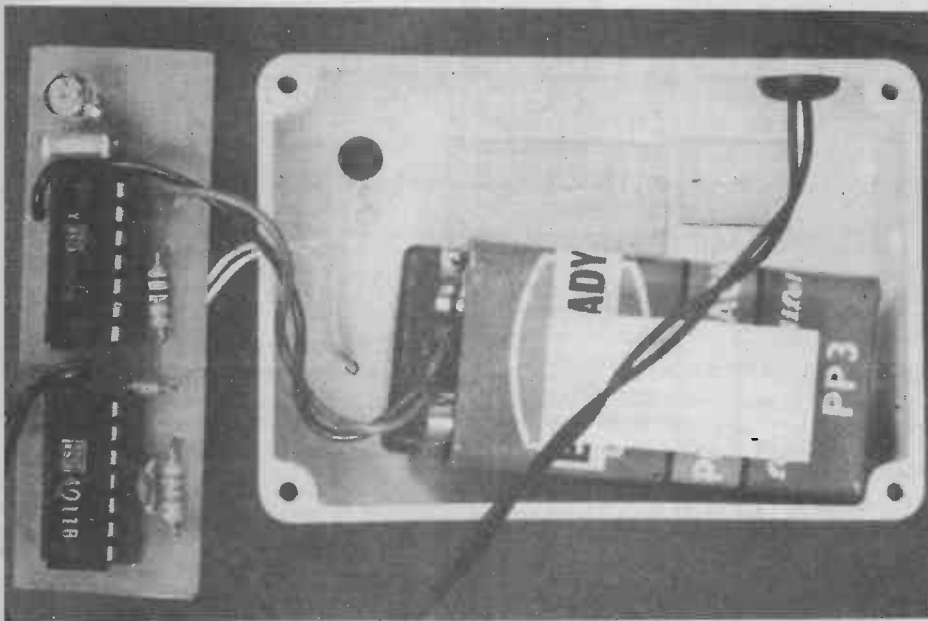
## HOW IT WORKS

The circuit consists of three sections, a light threshold detector, a latching circuit, and a gated audio oscillator for the alarm. The light detector uses a phototransistor Q1, which has a very low dark current that is effectively negligible. Thus in the dark R2 takes the inputs of IC1c to logic high and the output of the gate will be low. C1 is wired across the phototransistor to add a half-second time constant so that brief shadows (from hands in the cupboard) do not re-trigger the alarm when it should be silent. When any light falls on Q1 the current through it will increase and the inputs to IC1c are now taken low. The output of IC1c will thus go high when light enters the cupboard. Schmitt trigger NAND gates are used to provide clean switching thresholds and reduce the stand-by current consumption.

IC1a and b form a set-reset latch. In the dark condition the pin 13 input of IC1b will be held low and its output (pin 11) is therefore high. R1 holds the pin 8 input of the other gate (IC1a) high, making its output on pin 10 low; this holds the other input of IC1b low, thus keeping the latch output high even when the light detector provides a high input to IC1b.

In the light condition, then, both inputs of IC1d will be high and its output will go low which enables the oscillator circuitry. The latch can be reset to a low output by operating PB1 only when there is light (ie the cupboard is open). D1 is used to disable the C1/R2 time constant when PB1 is operated by discharging the capacitor C1. Thus when light is available and the alarm is sounding, pressing PB1 will cause the pin 6 input of IC1d to be latched low, turning off the alarm. The latch can only be set high again (re-arming the alarm) by removing the light.

The alarm oscillator uses the well-known gated CMOS astable configuration, built around IC2b and c. One input of each gate is driven from the output of inverter IC2a; thus when IC2a's input is low the oscillator is enabled, sounding the alarm. The output of this oscillator is wired to one end of the transducer and also to the input of inverter gate IC2d. The output of this inverter is wired to the other end of the transducer to drive it in a bridge fashion, providing four times the power and a greatly increased alarm volume.



## PARTS LIST

### Resistors (all 1/4W, 5%)

R1 1M0  
R2 5M6  
R3 33k

### Capacitors

C1 100n polycarbonate  
C2 10n ceramic

### Semiconductors

IC1 4093B  
IC2 4011B  
Q1 BPX25 (see text)

### Miscellaneous

TX1 PB-2720  
PB1 push-button  
PCB (see Buylines); PP3 battery and clip;  
case (Vero order ref. 75-1413E).

Fig. 1 Component overlay.

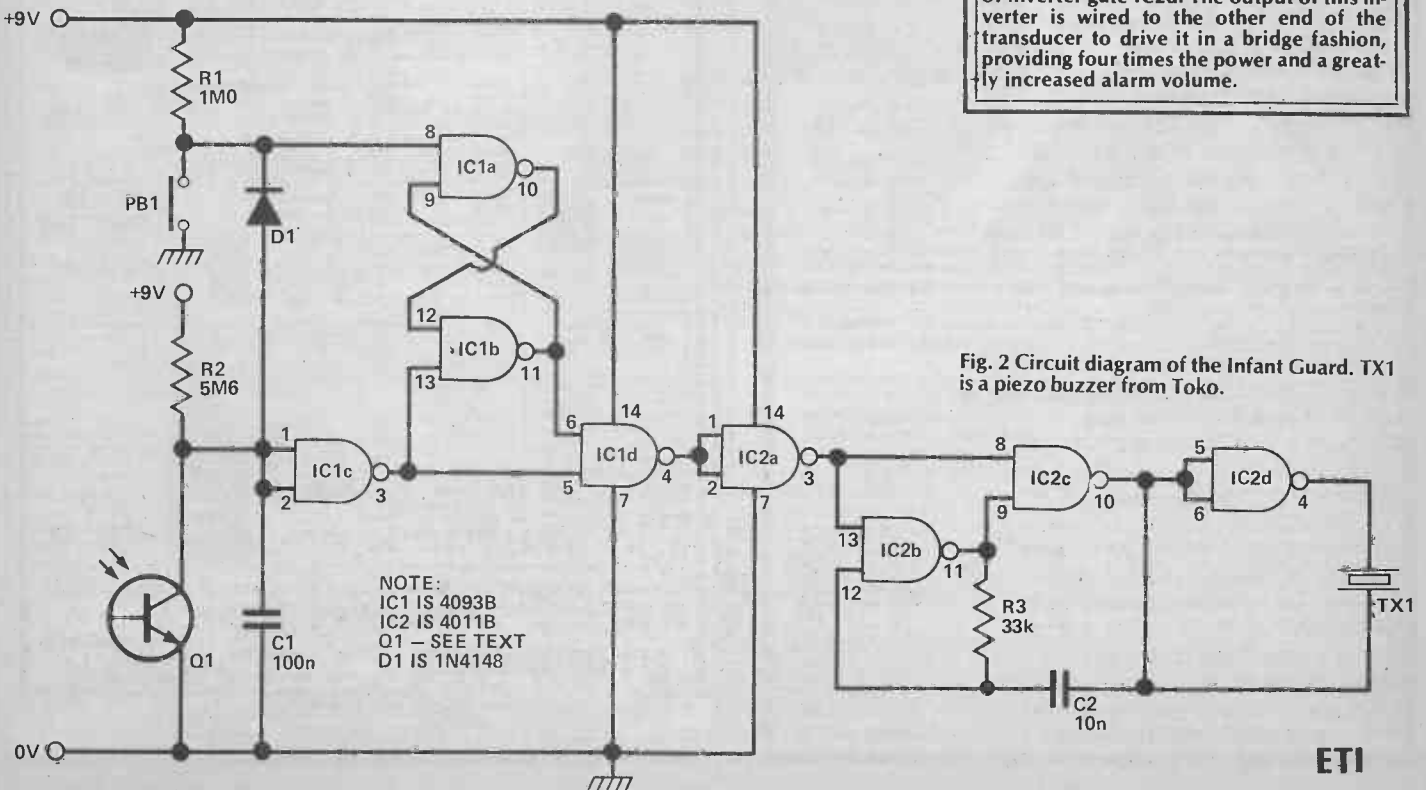
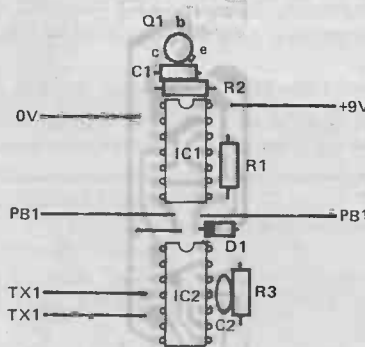


Fig. 2 Circuit diagram of the Infant Guard. TX1 is a piezo buzzer from Toko.



# ROBOTICS TODAY

The topic this month is 'Robodynamics and the Hobbyist', as H.W. Gleaves takes up our offer to use these pages for an exchange of ideas.

A robot is a complex system not unlike an automatic instrumentation system. It is a combination of a number of basic units, i.e. data processor (DP), mechanical superstructure, electronic signal processors, transducers and so on, the aim being to produce a machine that is general purpose in nature and capable of performing physical tasks in the same way that computers perform mental tasks. Since one cannot perform any physical task without some form of mental effort, it follows that any good robot will need to incorporate a computing ability.

A block diagram of a typical robot is shown in Fig. 1. Of course, not all robots will take this form — not all will possess a

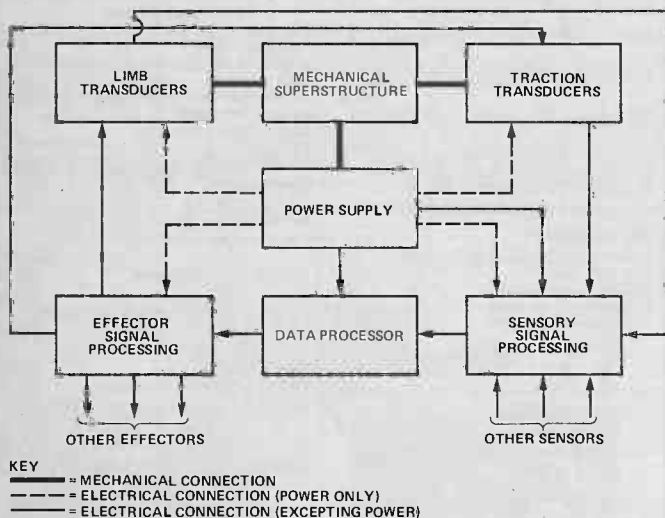


Fig. 1 Block diagram of a 'typical robot'.

limb, for example. However you design your robot it will, more than likely, take the form shown and therefore a brief description of the function of each sub-assembly (and guidelines for its design) will now be given.

## Mechanical Superstructure

This is required to support all of the internal components, such as motors, batteries, electronic circuit boards, and so on. It is responsible for holding the robot together! It also serves the often ignored function as a 'bed' or base for mounting the sensors and, as such, the geometry of the mechanical superstructure is very important, since the software used in the DP will also depend on this. Great attention should be paid to the design of the mechanics since, unlike an electronic circuit, modifications are (usually) not easily carried out.

I have found during my involvement with robots that the guiding principle in such a project is efficiency. Because the robot will have to depend on its own power supply, it is logical to optimise things so that it will obtain the maximum operating time from its batteries and the factors affecting this parameter are chiefly mechanical in nature. For example, the robot's centre of gravity should be designed to lie directly over the driving wheel axis, since wheel slip will be thus reduced.

## Power Supply

An ideal robot would contain two supplies — one for the chiefly electronic units, and another for the chiefly mechanical units. This is because the electromechanical parts apply a greater load to the power supply and the voltage drop introduced into the electromechanical supply could drastically affect the performance of the circuitry.

The type of supply used will be some form of battery, although in a number of cases an umbilical cable could be used. Lead-acid batteries are favoured, due to their good power-to-weight ratio, although nickel cadmium cells can be employed for units which require less power since these cells require less maintenance.

## Sensory Signal Processing

This is one of the most interesting aspects of robot design: its purpose is to provide conditioning of the signals from the multitude of transducers on the robot, i.e. analogue-to-digital (ADC) conversion. It will also consist of a switching pack, to enable the DP to access any one of the sensors. Many types of sensors can be added to a robot, eg ultrasonic ranging (discussed later), temperature sensing, and force sensing (also discussed later), each supplying data to the DP.

## Effector Signal Processing

The function of this particular unit is to convert the digital data from the DP into signals capable of operating the robot's effectors, i.e. its traction motors and its limb drive motors.

You must be aware that the robot must possess more sensors than effectors since it is an unwritten law that much more data goes into a system than comes out of it! Effector signal processing covers such things as digital-to-analogue conversion, and motor speed control.

## Data Processing

A more apt title for this device would probably be 'data converter', since that is its primary function in a robot. The object is that for every type of data entering the system, there should be corresponding data coming out. The relationship between data out and data in is complex, and is wholly determined by the software contained in the DP. It is a good idea to develop the software for your robot simultaneously with the hardware, since deficiencies in one aspect can be taken up in the other.

## Robot Sensors

A robot needs to have a lot of information entering it per unit time in order to make useful deductions about its environment. It is for this reason that I decided to try and make an accurate ultrasonic ranging system.

I knew the theory, since enough people had talked about it, but nobody seemed to be able to produce a working system good enough to use. I therefore embarked upon the design of a circuit which, in conjunction with the DP, would enable the robot to determine the range of obstacles in its path.

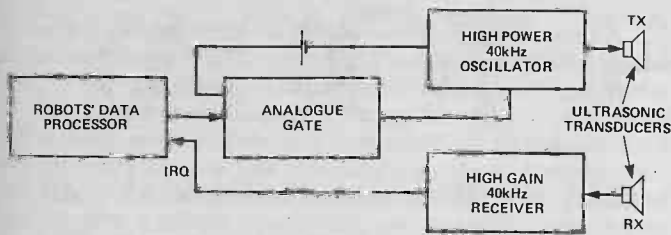


Fig. 2 Block diagram of an ultrasonic ranging system.

After much experimentation, a working system was put together and with it the robot could measure ranges up to a distance of nearly 2 m, to an accuracy of  $\pm 2$  mm.

The principle is described in Fig. 2. As can be seen, the high power ultrasonic oscillator has its supply wired in series with an analogue transmission gate and can be started and stopped by means of the logic control input. This input is connected to, and controlled by, the robot's DP. The high gain ultrasonic receiver is a combination of op-amps, designed to both amplify and filter the sound.

The whole assembly is controlled by the DP, which, as well as starting and stopping the transmitter, performs the additional function of counting (at a high rate) until sound reaches the receiver, whereupon it stops — the value counted being in proportion to the target's range. In the system presently in use, it is the 'interrupt request' input that is used, rather than just the 'interrupt', because the interrupt request, or IRQ, can be ignored by the DP and this prevents the DP being triggered when the ranging system is not in use.

## Interruption Control

The DP only activates the IRQ during the 'listening' period immediately following the switching 'off' of the transmitter. It is during this time that the burst of sound is travelling towards anything in the robot's path and if there is anything sufficiently close then the reflected sound energy will interrupt the DP. It will then start an interrupt service routine, the purpose of which is to inform the main program that a signal has been received and it is time to stop counting. This terminates with the range data safely stored somewhere in the RAM, ready for further processing. The software for the system can be grasped by reference to Fig. 3.

In the program, the most significant register is compared with a value called 'max'; this corresponds to the maximum range that either the system operates at, or that you wish to detect targets up to — it limits the value counted by the DP.

The circuits for both the receiver and the transmitter are shown in Fig. 4. In the prototype, one PCB was used for both circuits. Care should be taken to prevent the receiver being affected by the transmitter.

The system should be set up using a multimeter. First energise the 'send' line continuously, to give a constant 40 kHz, and place the transducers about three inches apart, facing in the same direction. Place a suitable target in front of the transducers, about six inches away. Using the meter, monitor the output of the receiver and adjust the sensitivity control until the output voltage drops to approx 0 V.

If the target is now moved away, the voltage should return to 5 V (approximately).

At this stage all that remains is to adjust the frequency control of the transmitter to obtain the maximum range, and this is done by performing the above procedure, but at gradually increasing distances. In the prototype a maximum range of about 4 m was achieved, but the positioning and orientation of the target were so critical that advantage could not be taken of this and the maximum usable range of 1 m was found to be entirely sufficient in any event.

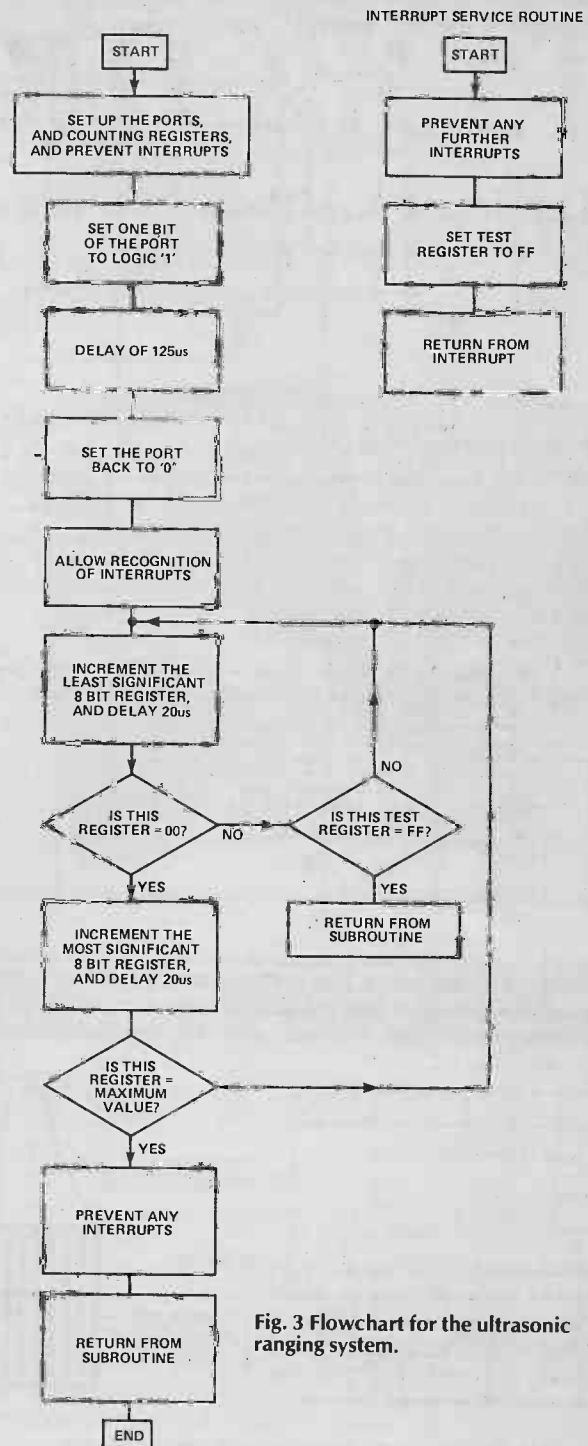


Fig. 3 Flowchart for the ultrasonic ranging system.

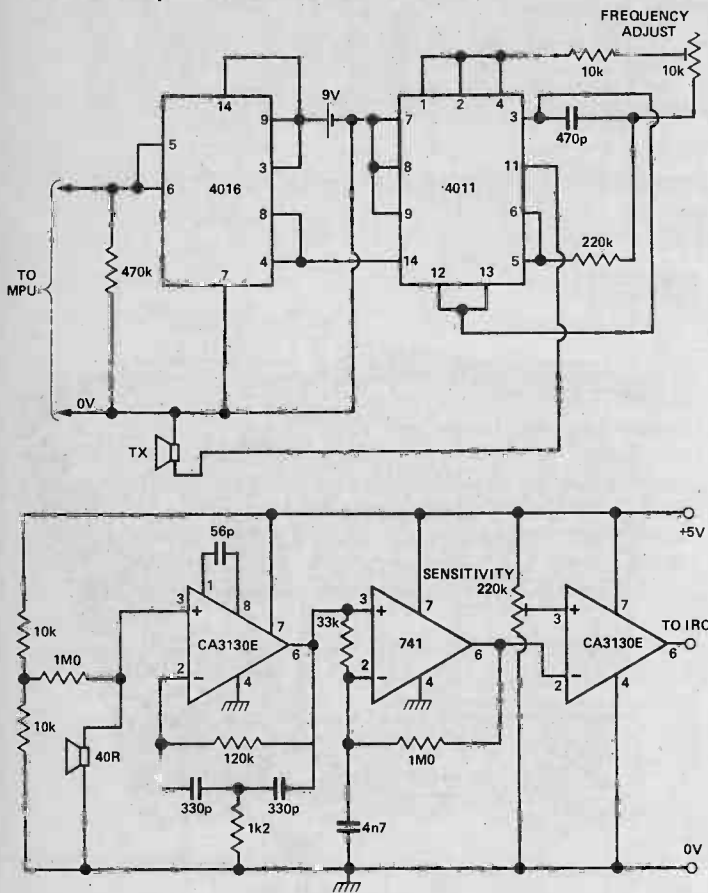
## Force Sensing

This is the ideal device to use in a robot hand or gripper, since with it the robot acquires the sense of touch. It is very sensitive, being able to detect even the slightest change in applied force, and it also has exceptionally good repeatability, although there is a 'recovery time', after the sensor has been compressed and then released, of about one second in duration.

The heart of the system is a piece of conductive foam of the type used with CMOS ICs. This foam is conductive — in fact the average piece has a resistance (between its two ends) of a few Megohms. The important point about this foam, however, is not the fact that it is conductive, but the fact that its resistance changes if a compressive force is applied to it (as it is compressed its resistance drops).

Using a simple circuit, a voltage can be derived that is proportional to the resistance of the foam, which is related to the

Fig. 4 Suggested circuit for an ultrasonic transmitter (top) and receiver (bottom).



force applied to the foam. This voltage can then be converted (using an ADC) to give the robot's DP accurate information about the applied forces. A simple circuit for you to experiment

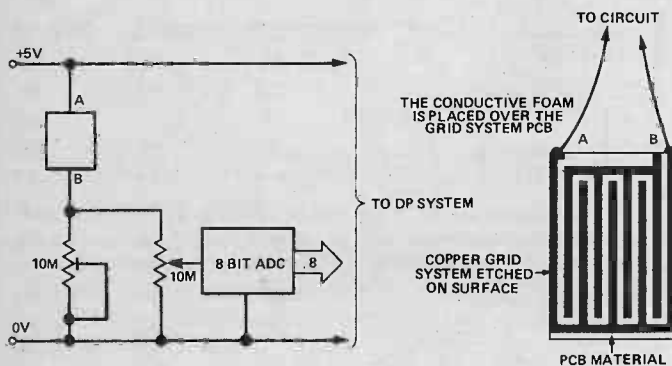


Fig. 5 Experimental force sensor.

with is shown in Fig. 5; here the data information sent to the computer increases as the force applied to the sensor rises.

## A Drunken Robot?

A major problem is a robot's inability to move in straight lines. This is brought about by the robot's drive wheels rotating at different speeds due to things like differences in instantaneous wheel load and friction. The combination of all these apparently minor effects, is to cause the wheels to rotate at unrelated speeds, a feature now known to be highly undesirable.

There are two approaches to solving this problem: computer intervention or use of a hardware lock circuit.

Both these ideas have been worked out on paper and the computer intervention method is preferred, because it is more accurate and uses the DP rather than hardware.

A block diagram of the computer intervention method of wheel control is shown in Fig. 6. The DP energises the 'sensor select' input and this has the effect of connecting one of the sensors to the tachometer circuit. (The tachometer could well be the one given in the November issue of ETI in the 100 Circuits Supplement.) The output from here will be a square-wave, the frequency of which depends on the angular velocity of the motor shaft. This frequency is converted into a voltage, and then into an eight-bit word, which is entered into the DP.

The DP can thus get an accurate digital representation of the angular velocity of each motor (and hence wheel). Using some specially written software these two numbers can be processed and another number computed, which is then applied to one of the eight-bit rate-multipliers. The output of these is fed to a series switch in the motor circuit. The general algorithm used by the computer will be to find the value of each motor speed and then, if these are different, proceed to slow down the faster of the two. (Trying to speed up the slower of the two could be disastrous if one of the wheels has become jammed somehow!)

It should also be possible to ensure that the above process does not result in a gradual reduction of the robot's speed, perhaps by speeding up the slow motor every now and then.

If the wheel had, in fact, been jammed or excessively loaded, its speed could be checked and, if no change had occurred, the faster motor could be slowed down as described above.

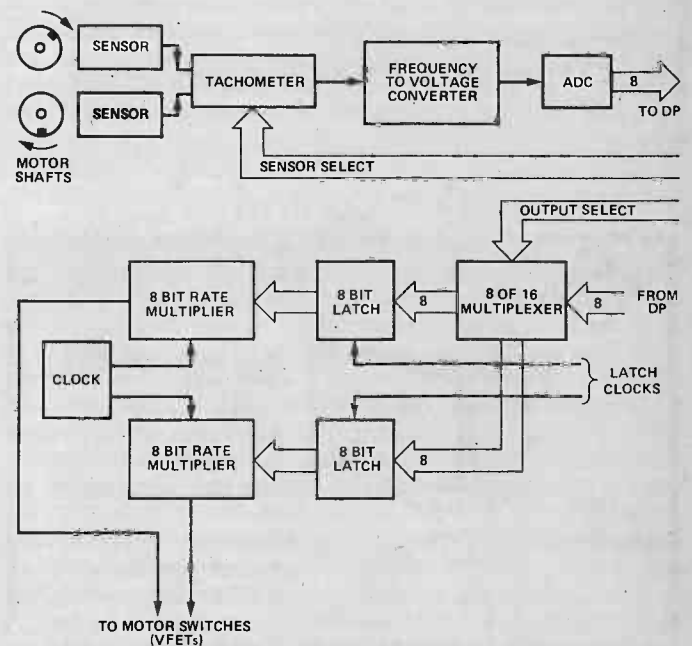


Fig. 6 Block diagram of motor regulation system.

## Conclusion

I hope that these ideas will give the readers of ETI something to begin experimenting with. Anyone who wishes to comment on any of the points raised herein can do so by writing to 'Robotics Today', 145 Charing Cross Road, London WC2H 0EE.

This new series will provide a stage upon which our readers may display their robotics achievements. It is intended to cover the practical application of robots in Britain today, be it at hobbyist level or in industry.

Readers in either category are invited to write to the editor of ETI, detailing their experiments, projects, application or usage of robotics. Any articles published will be paid for at commercial rates. It is also hoped to run an 'Ideas Forum' wherein readers can exchange views and ideas but that depends upon the response of our readers — you!

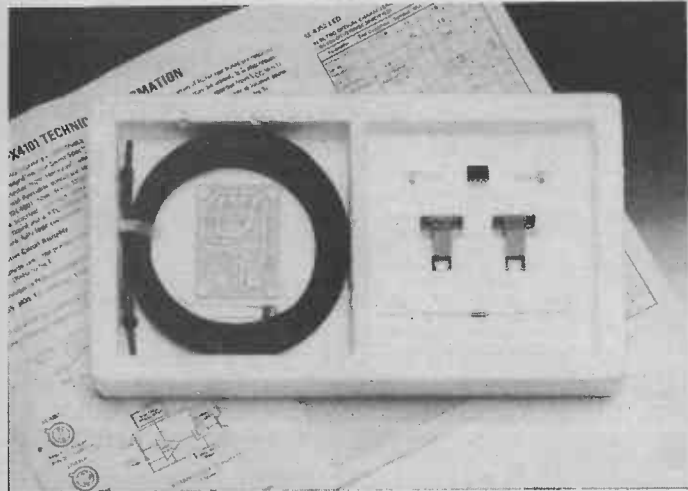
## Electric Motoring

If liquid fuel resources become very expensive or uneconomical, as we have been led to believe, it is possible that by the early part of the 21st century a large proportion of road transport vehicles will be battery powered. EASAMS Ltd has been awarded a contract as part of the 'forward looking' work of the Department of Transport. The study will be to find out the requirements of a refuelling network

for electric vehicles and is primarily concerned with the complex of stations and facilities needed to support both private and commercial electric vehicles. Study aspects will range over recharge opportunities when vehicles are parked for any length of time, supply locations for top-up connection, battery exchanges, capital outlay and the economics of operation, as well as the use and abuse of the mechanisms involved. The result of this study will assist the Department of Transport in formulating a future policy if it becomes necessary.

## Fibre Link

A new low-cost fibre-optic link kit has been added to the Honeywell Components Group's range of optoelectronic products. The kit is TTL/CMOS compatible, featuring the Sweet Spot LED, and comes complete with driver circuit, monolithic Schmitt trigger receiver, five metres of fibre and all connectors. Press-fit components and self-aligning connectors mean the kit is easily assembled without special tools or adhesive.



Typical applications for this kit are: intra-cabinet clock distribution, card-card data transmission and high voltage insulation. With simple modifications the kit can be used for RS232, RS422 and RS423 connection, hi-fi audio transmission, parallel-parallel ribbon replacement, and high-linearity analogue instrumentation links, among others. For further information contact Components Group, Honeywell Control Systems Ltd, Honeywell House, Charles Square, Bracknell, Berks, RG12 1EB.

## Soft Touch

A set of three small hand tools with soft, comfortably shaped self-opening handles has been added to the range of quality tools offered by Tele-Production Tools Ltd. The first tool is a flush cutting micro-shear with the cutting head angled at 45° for easy use on PCBs — a cut wire retaining clip is fitted as standard. Two fine-nosed pliers

are also offered, one 28 mm and the other 40 mm in length from the pivot. The idea behind the soft handles is that they will reduce operator fatigue. The tools are available from Tele-Production Tools Ltd at a cost of £10 per set or £3.75 individually (inclusive of postage and packing and VAT) from: Stiron House, Electric Avenue, Westcliff-on-Sea, Essex SS0 9NW.

## Powerful Aid

OK Machine & Tool (UK) Ltd's Elräck range now includes a universal three-pole electrical socket strip system which can be used on work benches, walls and instrument racks to greatly simplify multi-socket installation. The extruded strip is supplied in 25 m coils and has a 25 A, 110/240 V rating. Sockets, available for square pin (UK), two pin Schuko

and three pin round plugs, have a quick connect facility and are secured to the strip without additional fixing. This method ensures positive contact with the power rails, allows sockets to be repositioned and only permits sockets to be fitted in the correct way. A similar system is available for 19" front panels and has cable ducts with integrated socket strips. For further details contact: OK Machine & Tool (UK) Ltd, Dutton Lane, Eastleigh, Hants SO5 4AA.



## Pocket Precision

The D350 is an autoranging digital multimeter which is small and light enough to fit neatly in your shirt pocket. It comes from Micro-Data Systems and has a 3½ digit liquid crystal display which has both unit and function annunciators and fully autoranging operation. Functions include audible continuity and AC or DC current measurement to over 20 A. Power is from two U16 penlight cells, giving over 200 hours of operation. The multimeter is built around a single CMOS IC in a 64-pin flat-pack. This not only performs the A-to-D conversions and drives the triplexed display, but also contains the autorange switches, the AC to DC converter amplifier and the resistance measurement source. The D350 measures 5" x 2½" and is only ¾" thick. Weight including battery is only 4½ ounces. The price including accessories is £69 plus VAT, from Micro-Data Systems, Coach Mews, St. Ives, Huntingdon, Cambs.



## Charged Up

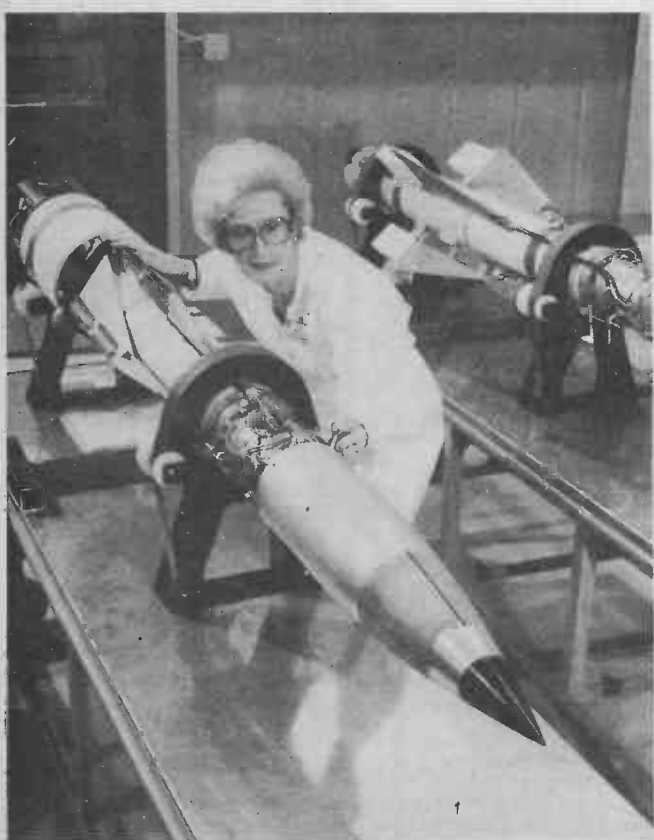
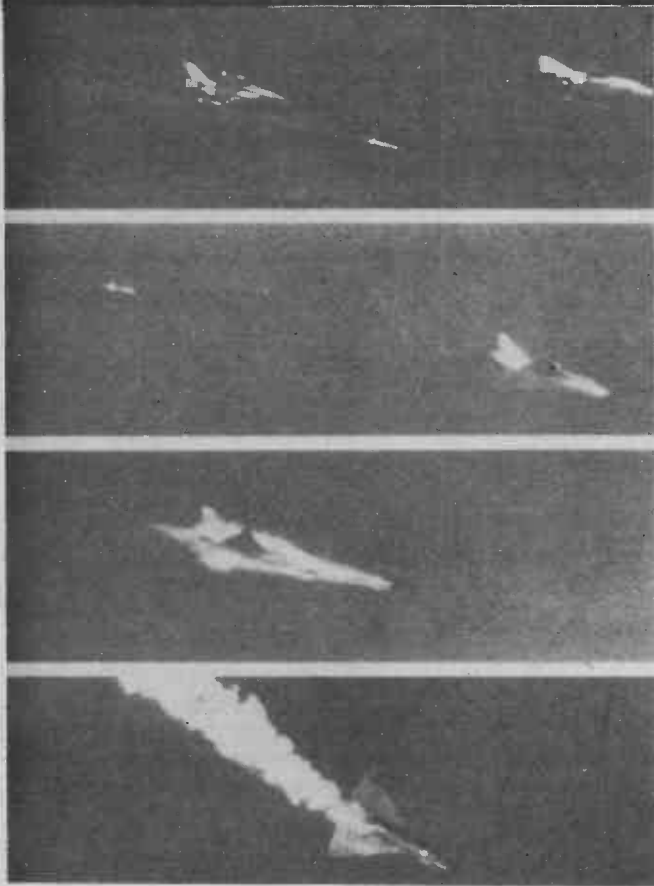
Now available in the UK is the Gould 'Again & Again' rechargeable battery system, offering a low-cost reusable alternative to expensive alkaline batteries for use in many applications including toys and games, radios and cassette recorders. This nickel-cadmium battery range includes all the popular sizes such as AA (HP7), C (HP11), D (HP2) and 9 V (PP3) batteries. An important feature of this system is the low cost of the battery charger — less

than £10 (excluding VAT) — and the charger will take all the batteries in the range. This means that for an outlay of around £15 you can have a set of batteries and a charger which will provide power for up to five years! As current consumption of the charger is so low, it will only cost you a few pence for each recharge cycle. For further information about stockists contact

Gould Battery Division, Raynham Road, Bishop's Stortford, Herts.



# Defence Digest



This new regular feature is devoted to defence electronics, its equipment techniques and application. Defence remains one of the largest growth areas in UK industry, with much of the real innovation and investment taking place there.

Defence Digest will thus act as a news (and views) section, containing up-to-date information and explanation of some of the happenings in the different sectors of the defence industry.

Companies with information and articles for these columns are invited to submit them direct to Defence Digest at our editorial address. Indeed, anyone with anything to say on the subject, be it information or opinion, is a potential contributor and should not refrain from putting pen to paper.

## Deadly Debut

Following our announcement of the Hughes Aircraft Company's Advanced Medium-Range Air-to-Air Missile (AMRAAM) we have pictures of it scoring a direct hit on a fighter aircraft drone target. In the first photo, the missile is launched from a US Air Force F-16 (left) escorted by a chase plane. In the second photo, the AMRAAM — its low smoke motor still burning — is guided by an on-board active radar seeker towards the tail of the QF-102 target. Next, the missile, after making a near dead-centre hit, passes through the target aircraft, setting it aflame even though the missile did not have a warhead. In the final picture, the QF-102 falls in flames over White Sands Missile Range, New Mexico, where the test launch took place.

## Defence At Greenwich

The picture shows the Parliamentary Under-Secretary of State for Defence (Procurement) Mr Geoffrey Pattie, MP, visiting Standard Telephones and Cables' South East London location at Greenwich on the 23rd September 1981. He was finding out more about the activities of STC's defence systems division. Mr Pattie inspected a new electronics manufacturing facility which is producing a range of subscriber and transmission equipment developed by STC for the British Army's system Ptarmigan. STC's defence systems division is the leading supplier to the Ministry of Defence of underwater cable systems



## Ready For Duty

One of the first production versions of the US Roland surface-to-air guided missile is shown here being inspected prior to delivery to the US Army from the Hughes Aircraft Company's manufacturing facility. To the rear is the aft section of another missile as delivered to Hughes by the Boeing Aerospace Company. Hughes then adds the guidance section it builds as the forward part. Both missiles have telemetry sections (the uncased portion) to provide test flight data in place of the warhead. After delivery to the Army, four of the first production missiles (including the one to the rear) were shipped immediately to a test range in France. There all four were fired successfully from a German fire unit, passing well within the lethal distance of simulated aerial targets. The firings demonstrated the interchangeability of Roland missiles and fire units between those built in France, West Germany and the United States. Hughes and Boeing, as associated prime contractors, are building the US Roland all-weather air defence system for the US Army under licence from Euromissile.

# PICKUP AMPLIFIER DESIGN

David Tilbrook takes us stage by stage through the wonders of modern design techniques for low-level signals, as he develops an ultra-fi circuit for moving magnet (and moving coil) designs.

Just as a loudspeaker represents a non-linear load to the output stage of a power amplifier, a moving magnet or moving coil cartridge represents a non-linear source impedance to the input stage of a preamplifier. This is the cause of many of the problems associated with any preamp.

Both moving coil and moving magnet cartridges generate electrical signals through the interaction of a coil wire and a magnetic field. The signal voltage produced is therefore proportional to the relative velocity between the coil and the magnet assemblies. This relationship is predicted by Faraday's law of induction, expressed mathematically as:

$$\varepsilon = - \frac{d\phi}{dt}$$

where  $\varepsilon$  is the signal voltage at any instant and  $\phi$  is the magnetic flux. The signal voltage produced at any instant is proportional to the rate of change of flux with respect to time,

$$\text{i.e. } \frac{d\phi}{dt}$$

The design of the cartridge must ensure that a linear relationship exists between the position of the stylus cantilever assembly and the magnetic flux. In this way changes in the position of the stylus give rise to changes in the magnetic field intensity. So the rate of change of stylus position with respect to time will be proportional to the signal voltage, i.e.:

$$\varepsilon \propto \frac{dx}{dt}$$

where  $\varepsilon$  is the signal voltage and  $x$  is the stylus displacement from its equilibrium position. This means that the waveform actually 'on' the grooves is not proportional to the signal voltage itself. Instead it is proportional to the integral of the signal waveform. If a square wave, for example, is to be produced from a record, the waveform as seen in the groove with a microscope will be a triangle wave.

Since the signal voltage is proportional to the velocity of the stylus, the signal slope is proportional to the acceleration of the stylus. In order for the high signal slopes to be reproduced accurately by the cartridge it is important to realise that the cartridge cantilever assembly and its associated suspension and magnet/coil system form a resonant mass-spring system analogous to a complex electrical series resonant circuit.

## Resonance

At one particular frequency, called the resonant frequency, the impedance of the cartridge will no longer be related linearly to the driving force on the stylus, and distortion results.

To overcome this problem two techniques are used simultaneously.

First the resonant frequency of the cartridge is moved to a frequency below the audio spectrum. Using the damped mass-spring model of a magnetic cartridge we can predict that the resonant frequency of the cartridge is moved to a frequency below the audio spectrum. Using the damped mass-spring model of a magnetic cartridge we can predict that the resonant frequency will depend on the mass of the stylus cantilever assembly and on the 'springiness' of the cantilever's suspension. This springiness is characterised by a number, often given the symbol  $k$ , called the spring constant. Spring constant is defined in terms of the force needed to bring about a certain compression or extension of the spring. Stiffer springs have a higher value for  $k$ . The spring constants, however, are so small that the numbers are hard to interpret. For this reason cartridge manufacturers usually specify this quantity by quoting the reciprocal of the spring constant,  $1/k$  called *compliance*. Stiffer suspension systems have lower compliance figures.

As stated earlier, the cartridge resonant frequency is a function of both the mass and the compliance of the cantilever and suspension system. The damped resonance mass-spring model of a magnetic cartridge predicts that the resonant frequency will be given by the equation:

$$f = \frac{1}{2\pi\sqrt{mC}}$$

where  $m$  is the mass of the cantilever/stylus system and  $C$  is the compliance of the stylus suspension system.

Note that the equation for the resonant frequency of magnetic cartridges has exactly the same form as the equation for the resonant frequency of an electrical resonance circuit, i.e.:

$$f = \frac{1}{2\pi\sqrt{LC}}$$

where  $C$  in this case is capacitance and  $L$  is inductance.

The equation predicts that the resonant frequency of the cartridge can be decreased by increasing either the mass or the compliance. Since the mass of the moving parts in the cartridge must be kept small so the stylus can respond quickly to changes in the record groove, the compliance must be increased until a suitably low resonant frequency is obtained. Most high-quality magnetic cartridges have resonant frequencies below 10 Hz.

The second technique used to overcome problems associated with this resonance characteristic is to decrease the  $Q$  of the system by damping the resonance with a suitable combination of mechanical and electrical losses. Mechanical damping is obtained by deliberately introduced friction within the cantilever suspension system. The cantilever suspension is

often terminated into a rubber mounting block for this purpose. The electrical damping comes about as a direct consequence of the law of conservation of energy. The cartridge is acting as a generator, delivering power to the input resistance of the preamplifier. Since energy is absorbed by this load resistance the  $Q$  of the cartridge resonance is decreased.

## Poles Apart

Until recently most stereo magnetic cartridges consisted of two fixed coils between the poles of a small magnet attached to the cantilever. Modulation of the record groove produces movement of the magnet, changing the magnetic flux and generating the signal voltage.

The coils usually have a large number of turns so that a reasonable signal voltage can be produced (typically in the order of 20 mV). The resistance of these coils usually ranges between 200-1000 ohms, but their impedance can be much higher, especially at high frequencies where the inductance of the coils becomes important. This type of cartridge is sometimes called a moving magnet cartridge to distinguish it from the more developed moving coil types. The relatively high reactive component of the cartridge impedance combined with the effects of the natural cartridge resonances makes it essential that the input impedance of the moving magnet (MM) input stage have well-defined characteristics if best performance is to be obtained from this type of cartridge. Most MM cartridges require a load impedance consisting of 47k of resistance shunted by several hundred picofarads. This capacitance is often provided by the shielded cable, but most cartridges require some additional capacitance across the MM input. In exceptional cases the input capacitance due to the shielded cable is too high.

In order to obtain the flattest frequency response possible from an MM cartridge it is essential that the load resistance be constant over the complete audio spectrum *and beyond*. For this reason measurements done on the input resistance of MM amps at one particular frequency (usually 1 kHz) are practically useless.

## Fall From Grace?

Many input stages exhibit a characteristic of falling input resistance at high frequencies. The input resistance of a bipolar transistor, for example, even with a small amount of emitter current, is insufficient to ensure a constant resistive load to an MM cartridge. The common two or three transistor phono stages of a few years ago often suffered badly from this problem, degrading the top end performance of an otherwise good MM cartridge. The problem occurs because all bipolar transistors have decreasing gain at high frequencies.

The most common method used to increase the input impedance of a bipolar input stage is through the use of negative feedback. The decrease in gain of the individual transistors in the stage at high frequencies decreases the overall open loop gain of the stage, which in turn decreases the amount of negative feedback available. Furthermore, the negative feedback is often applied at the emitter of the first transistor. The problem with this configuration is that the phase response in the negative feedback loop can easily be affected by the complex reactances of the cartridge and connecting cables, producing unwanted frequency response variations, or even instability in some cases.

All these problems come under the general heading of 'cartridge impedance interaction', and represent the most important single reason for the difference in sound between preamplifiers. Most preamps suffer from some degree of cartridge impedance interaction and in many cases the effects are pronounced.

In order to show how to overcome this major problem — and others — we will develop a full circuit design for an ultra-high quality pickup input stage, both MM and MC, and discuss this design at each stage.

## Action To Overcome

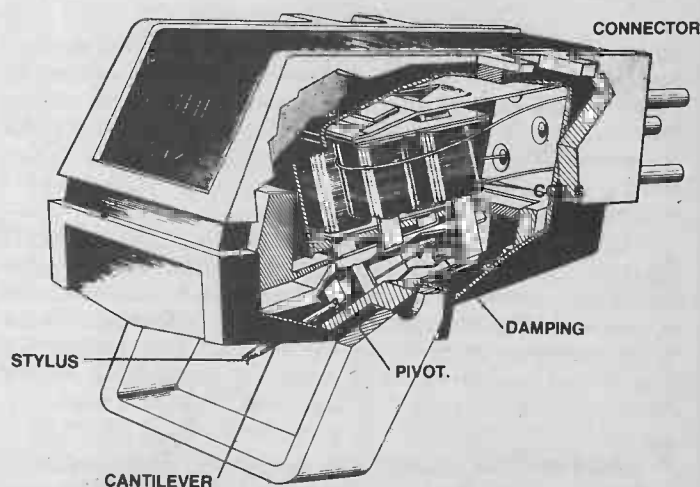
Overcoming cartridge impedance interaction can be achieved by separating the MM input stage into two active stages (see Fig. 1). The first stage consists of a single NE5534AN configured as a linear amplifier with a closed loop gain of around 8.3. The large amount of overall negative feedback increases the input impedance of the stage so that the measured input impedance is simply that of the 470k resistor, R2. Since the 5534 has a small signal bandwidth of around 10 MHz without additional compensation, the input impedance will remain unchanged over a very wide frequency range. The high input impedance of this stage would usually allow the input capacitor C2 to be conveniently small. However, for best noise performance the value must be increased substantially. This is covered in detail later in this article.

Capacitor C2 is necessary since it is not advisable to allow DC current from the first stage to flow through the cartridge. The value of C2 used here is 100uF, and this sets the lower -3 dB point well below 1 Hz. The upper -3 dB point of this stage is well above 100 kHz. An extended frequency response is necessary so that the accuracy of the RIAA equalisation is not affected by frequency response variations that might otherwise occur in the first stage.

## RIAA Equalisation

We said earlier that the signal voltage produced by a magnetic cartridge is proportional to the velocity of the stylus. If a low frequency signal is to be reproduced by a magnetic cartridge, large excursions of the stylus are necessary. If for example a 20 Hz square wave is to be reproduced by the cartridge then the cartridge must produce a DC voltage at its output for a period of 25 ms. In order to do this the stylus must move at a constant speed for this period of time, and therefore the waveform in the record groove is a triangle wave, as stated earlier.

Typical output voltages from moving magnet cartridges are in the order of 1 mV — 2 mV for a stylus velocity of 1 cm/sec. So if the peak voltage required on the square wave was, say, 10 mV, a stylus velocity of 10 cm/sec would be required for a medium-sensitivity cartridge, so the stylus must move at a constant speed of 10 cm/sec for a 25 ms time interval. The stylus therefore moves a total distance of 2.5 mm! On a stereo record the channels are cut in opposite walls of the record groove. If a low frequency mono signal is to be produced, both sides of the record groove force the stylus away from its equilibrium position, and a large vertical stylus excursion results. In the case



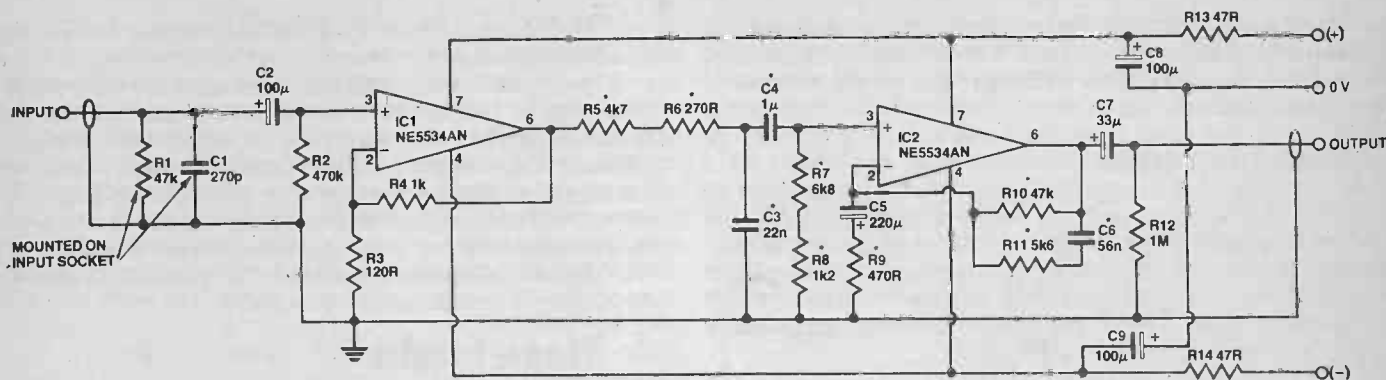


Fig. 1 Circuit of one channel of the moving magnet input stage.

of our square wave, the vertical excursion would be roughly 3.5 mm, which is simply not possible. The record would have to be as thick as most turntable platters!

Two measures are used to overcome this problem. First, the two channels are recorded on the record 180° out of phase, so that the large vertical excursion is replaced by a large horizontal excursion. Second, the low end of the frequency response is attenuated before the recording process, so the stylus excursions are decreased. The specific amount of low frequency attenuation is defined as that which would be caused by a first-order high-pass filter with a time constant of 318 us (ie the filter would be formed by an ideal resistor/capacitor filter, in which  $R \times C = 318 \text{ us}$ ). To convert from these time constants into frequency, simply apply the equation:

$$f = \frac{1}{2\pi t} \quad (t = \text{time constant})$$

This equivalent to a 6 dB/octave filter with a -3 dB point at 500 Hz. To prevent the low end from rolling off indefinitely a second 6 dB/octave filter is used to flatten the response again at 3150 us or 50 Hz. After this equalisation is applied, the stylus excursion of the 20 Hz square wave, for example, is decreased to around 0.3 mm, which is manageable.

Similar problems occur at very high frequencies. If we consider now a 20 kHz square wave at the same output voltage and hence the same recording velocity, the stylus now only moves a total distance of 2.5 um! Such minute distances are only a few orders of magnitude larger than the surface irregularities in the vinyl, so at these frequencies the signal-to-noise ratio is poor. To overcome this problem the top end is recorded at a higher level, which increases the stylus excursions and thereby improves the signal to noise ratio. The modifications to the recorded frequency response are referred to as RIAA pre-emphasis or equalisation (RIAA stands for Recording Institute Association of America), and must be corrected for by the input stage. The RIAA playback equalisation must boost the bass end and attenuate the treble end of the audio spectrum to return the overall frequency response to that of a linear system.

## Down And Out

Since the low end is amplified most of all by the RIAA playback signal any turntable rumble or cartridge/turntable resonances will be amplified. Modern power amps are quite capable of delivering full power to a pair of loudspeakers at 10 Hz or below, so appreciable amounts of subsonic content can be fed to the loudspeaker. This is potentially dangerous to the bass driver and decreases the clarity and accuracy of the low end.

In an attempt to overcome this problem the RIAA has proposed a change to its playback equalisation curve. The extreme bass frequencies are attenuated on playback by the addition of another time constant. This takes the form of a single-pole RC filter with a time constant of 7950 us, ie a -3 dB

point of 20 Hz. Since the frequency response is already flattened by the 3150 us time constant, this new time constant gives a 6 dB attenuation rate below about 20 Hz. The resulting RIAA playback equalisation is shown in Fig. 2. Note that there are four time constants associated with the proposed RIAA equalisation: 7950 us, 3150 us, 318 us and 75 us. These are shown on the Bode plot, which is the dotted line in Fig. 2. It should be emphasised, however, that the introduction of this low frequency time constant is not sufficient to remove severe cases of turntable or tonearm resonance. Some preamps incorporate multiple-order subsonic filters that offer a very fast roll-off below 20 Hz. The problem with this, however, is that severe cases of tonearm resonance or rumble generate distortion harmonics well above 20 Hz, into the audio spectrum. The only real cure is to remove the problem at the turntable or tonearm.

Many different techniques are used to give the preamp the desired equalisation. The most common is to include the RIAA equalisation circuitry into the feedback loop of the first stage. Fig. 3 shows a very simple MM input stage of the general type often found in medium priced amplifiers.

Transistor Q1 functions as a standard common emitter amplifier offering a voltage gain that is determined by the total impedance from its collector to earth divided by the total impedance from its emitter to earth. Transistor Q2 is a PNP transistor but functions in an identical manner. The product of their two voltage gains is called the open loop gain of the stage. If a current path is now made available from the output of Q2 back to the emitter of Q1, the voltage gain will now drop to a new figure called the closed loop gain. This is negative feedback, and it has the effect of decreasing the distortion and increasing the input impedance of the stage.

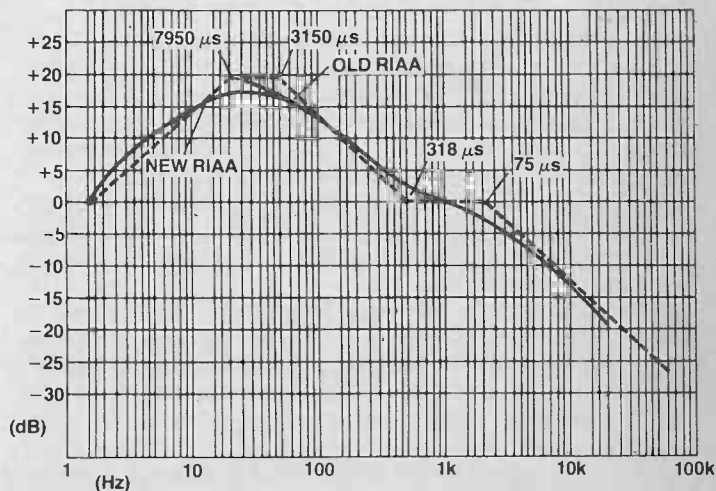


Fig. 2 Old and 'new' RIAA equalisation curves (solid line). The individual time constants (Bode plot - dotted lines) to produce the response are also shown.