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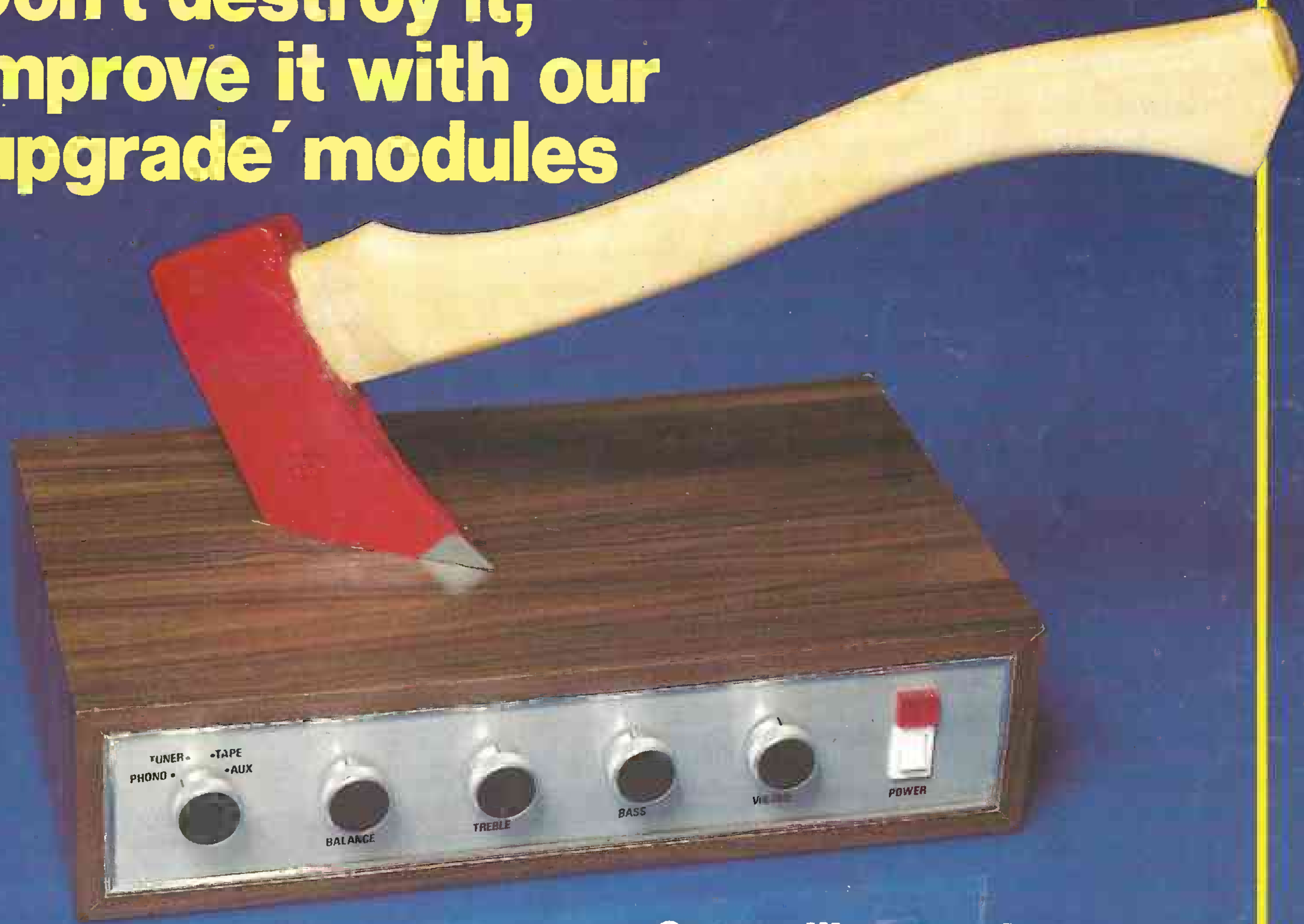

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FIRST REVIEW! NEW CRIMSON ELEKTRIK AMP!

..AUDI ... TING....MUSIC...RADIO....ROBOTICS..

TABLE 1

CK1010 Pre-amplifier

Frequency Response : 30 Hz – 40 kHz ± 2 dB (built-in LF filter at -12dB/octave below 20 Hz)

Pick-up input noise ratio : -70 dB (ref. 3 mV unweighted)
Moving coil input s/n ratio : -65 dB (ref. 200 nV unweighted)
Tape/tuner input : -80 dB (ref. 100 mV)
Pickup Overload : 37 dB (ref. 3 mV)
THD : 0.005%
RIAA Accuracy : ± 0.3 dB (20 Hz–20 kHz)
Price : see text

C1100 Power Amplifier

Power Output : 94 W (into 8 R)
141 W (into 4 R)
Power Bandwidth : 10 Hz – 40 kHz
THD : 0.005% (any power, 20 Hz – 20 kHz)
THD : unmeasurably low
S/N Ratio : -120 dB (ref. 94 W into 8 R)
Price : See text

As this is a kit amplifier, test results will depend to some extent upon the standard to which the kit is actually constructed. The results given herein are for a kit which has had its assembly optimised to obtain the best possible results and can thus be considered as pretty much as good as you'll get!

The Crimson delivered a good all-around performance as could be expected from any amplifier of its price—or indeed, of any unit costing twice as much. It is noticable that the output power increases some 3dB into low impedences, thus offering advantage both with difficult loads and under dynamic conditions. Only upon the power output did the Crimson fail to meet spec. It turned in 94 W into eight-ohms, but was capable of delivering 140+ both into 4 R continuously, or under dynamic drive conditions into 8 R. Thus the amp has an excellent "headroom" into eight ohms, for which I think we could forgive them losing six watts somewhere in the paper work.

Down to Sound

So after all that, what does the new Crimson actually *sound* like? In a word, good. I tried it with KEF 105 II loudspeakers most of the time, but wired in a pair of Wharfedale Laser 80s to see what it could do for a pair of less ambitious enclosures. Source equipment was a Thorens TD 160S/SME III with Dynavector Karat and Shure MV 30 HE cartridges, to test out the MC and MM boards respectively. In addition a Sony ST-J75 tuner provided some nice off-air signals, for when I grew fed-up of changing LPs!

As comparative amplifiers I had the Trio KA-1000 Sigma-Drive still at hand and a friend's specialist British unit, which clapped out halfway through the tests in a blaze of melting output stages and is thus best left un-named!

The Crimson proved to be a very controlled piece of hi-fi. It kept command of the bass signals superbly, never sounding loose. It has an excellent bass extension and when used with good speakers is capable of frightening levels of energy! The protection circuit never affected operation and is thus commended.

Mid-range is open and detailed and more revealing than is first apparent, since the amp is unkind to poor quality source material — as is most top flight hi-fi. Treble is replayed clearly and precisely with no signs of hardness, a complaint which was levelled at some earlier designs — unfairly in my view. Power is well maintained at all frequencies.

The dynamic range available through the KEFs was stunning. I had just laid hands on some dbx recordings (upon which more next month) and the Crimson really showed what it could do with these! A copy of the Police album 'Zenyatta Mondatta' was unbelievable in its dynamics. The 1100 coped with the challenge admirably. . . . while another amp committed hara-kiri in a somewhat spectacular fashion! The low noise-floor and sharpness of reproduction did the CL1010/1100 great credit.

Supply Procedure

I think the bass quality is due in no small part to the PSU design. The phono inputs too are worthy of special praise. The noise is low and there is good matching for the cartridge. The moving-coil board is worth acquiring, as it extends the range of the Crimson at a very reasonable cost.

Overall the performance is truly outstanding for a £250 amplifier and can be considered worthy by *any* standards of cost or absolute performance. There is little to criticise in the sound at all and nothing of substance amiss with the kit itself. Personally I think the volume attenuator should be on the front panel, as it is awkward to locate where it is and I would change the toggle-type tape switch for something a little more up-market! — and if that's all I can find to criticise in a design, it must be good!

Conclusions

It would seem then that Crimson have maintained their position at the top of the commercial kit-build field. There is no Oriental amplifier I know of that can better the sound of this combination overall at any price and only a few — such as the KA-1000 — are of comparable standard.

Power delivery is excellently maintained, with the result that if your speakers can take it, the 1010/1100 can go a lot louder than 100 W would suggest! Low impedances are handled smoothly and easily, so there will be no problems with 'nasty' loads.

I can say no more than that for £250 it is a bargain and one that becomes the reference point for kit amplifiers from now on.

Details from: Crimson Elektrik, 9 Claymill Road, Leicester LE4 7JJ.

Record Time

Next month we're taking a look at dbx disc replay in more detail, as many readers had enquired about them, following Pete Green's report on the launch of their new decoder. We have also located a source of a large catalogue of these recordings and can now pass on supply information!

ETI

UPGRADING AMPLIFIER PSUs

Even hardened DIY types may quail at the thought of tinkering with their expensive commercial hi-fi, but an improvement can often be made simply by beefing up the power supply. Phil Walker discusses PSU requirements and offers a ripple monitor to check whether your supply is up to the job.

Now that Christmas is over and we're looking forward to spring-cleaning, perhaps it is time to blow the dust and pine needles out of the hi-fi power supply corner. Is it really up to the job or is it the mud at the bottom of the audio pond?

What's in a power supply? Or what SHOULD be in a power supply?

Here we examine what steps we must take to determine exactly what is necessary to get the best results out of your audio system.

In order to get anywhere we must first decide or find out what power output is required and into what load impedance we are going to put it. For an existing amplifier these are already defined, so don't be greedy!

From these facts we can determine the voltage and current flowing in the load, thus making the estimate of the power supply requirements. From basic theory we have:-

$V = I \times R$ (Ohm's Law)
$P = V \times I$ (Power Law)
(V = volts, I = amps, R = ohms, P = watts)
From this we get:-

$P = V^2/R$ or $V = \sqrt{(P \times R)}$
also
$P = I^2 \times R$ or $I = \sqrt{(P/R)}$

To get an estimate of the peak voltage required we approximate the signal to a sine wave. For this particular case the peak voltage is $\sqrt{2}$ (or 1.414) times the RMS value given by the above equations.

To go further now we must know what sort of output stage your amplifier has. In general there are three main types.

Split Rail

Each power rail supplies the load voltage and current for approximately half the cycle (Fig. 1.). The voltage required from each rail is:-

$$V = \sqrt{(P \times R)} \times \sqrt{2} + X$$

(X is a factor to account for practical conditions — see later)
The average current required from each rail is:-

$$I = \sqrt{(P/R)} \times \sqrt{2} \times 0.6 \times 0.5$$

(The 0.5 factor is because the load current flows for half the output cycle only; the 0.6 factor is because the average of a sine wave is about 0.6 of the peak value) A reasonably safe approximation for the current is:-

$$I = \tfrac{1}{2} \times \sqrt{(P/R)}$$

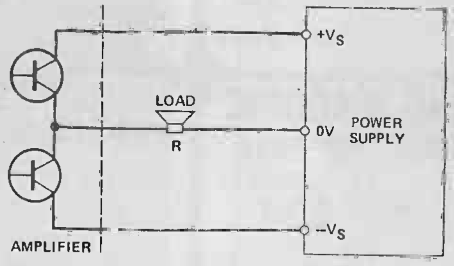


Fig. 1 Split rail power supply configuration.

Single Rail

The power rail supplies current on the positive half cycle of the signal only (Fig. 2). This drives the load while at the same time recharging the output capacitor C which has previously supplied the negative half cycle of the output signal. The positive end of C in the diagram normally rests at about half the supply voltage when no signal is applied. The supply voltage for this configuration is the same as the total for the split rail version and the voltage and current values found there apply.

Bridge

In this configuration the power supply provides the output current over the whole of the cycle. Therefore the average current is:-

$I = \sqrt{(P/R)}$ (approximated as before)

The supply voltage is effectively switched so that the positive and negative half cycles of the signal are drawn from the same supply rails. Therefore we have:-

$$V = \sqrt{(P \times R)} \times \sqrt{2} + Y$$

(Y is a factor to account for practical conditions — not the same as X above.)

What About X and Y?

Well, these are just factors which are included to make up for unavoidable imperfections of practical semiconductor devices. Bipolar and field-effect transistors always drop some volts when conducting and their driving circuits often require even more.
As a general approximation:-

for bipolar transistors	X = 4	Y = 8
and for MOSFETs	X = 8	Y = 16

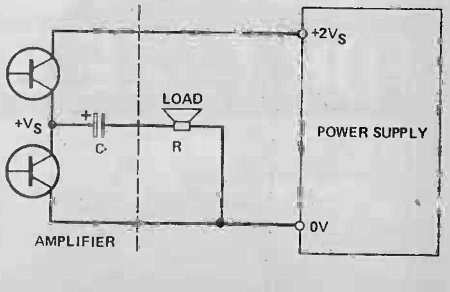
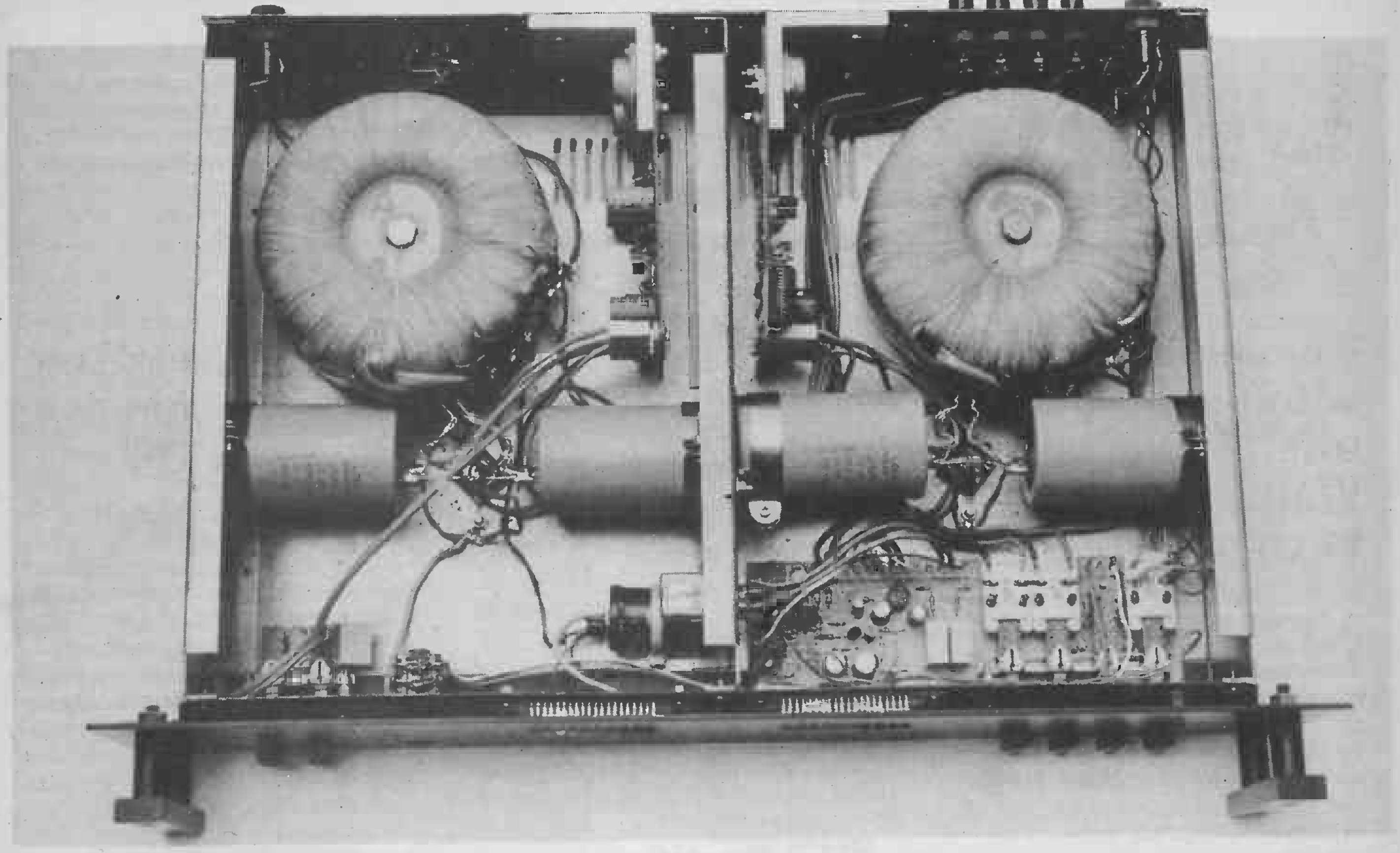


Fig. 2 Single rail configuration.



Most of the space in this 100 W power amplifier is taken up by the power supply components — and quite right too!

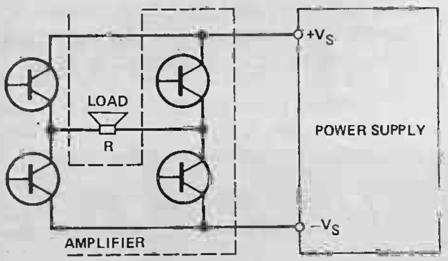


Fig. 3 (Left) Power supply configuration for a bridge amplifier.

What About Class A?

Up to now we have only considered class B or similar output stages but some people may well have class A amplifiers. If this is the case, the power supply will have to be rated to provide a continuous current $\sqrt{2}$ or 1.4 times as great as found above.

The Power Supply Transformer

Now that we know the voltage and current required, we can determine what transformer will be required. As a general rule the current rating of the transformer secondary will be the same or slightly greater than the required DC output.

The voltage rating will be $1/\sqrt{2}$ (0.707) times the required rail voltage plus a volt or two to make up for rectifier losses. The configurations shown in Figs. 4 and 5 are the most usual and will cater for most needs.

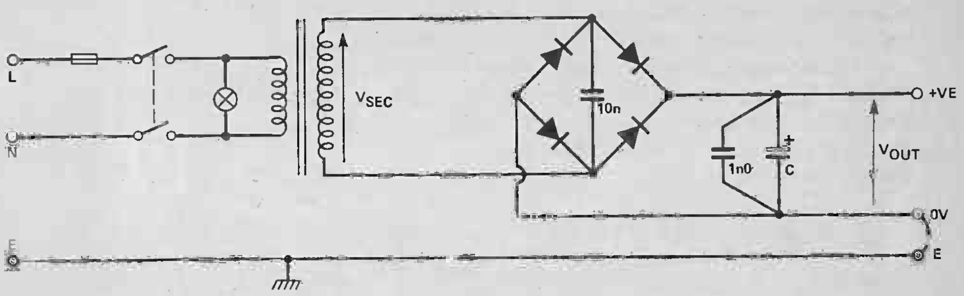


Fig. 4 Power supply for single rail and bridge amplifiers.

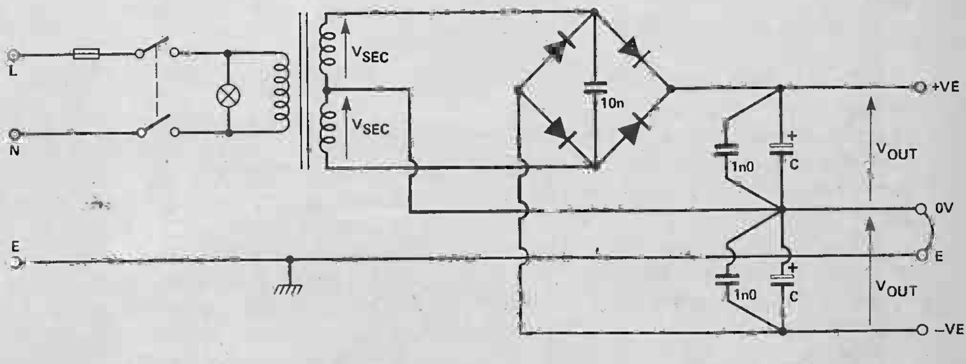


Fig. 5 Power supply for split rail amplifiers.

For single rail & bridge amplifiers:

$$V_{sec} = (V_{out} + 2)/\sqrt{2}$$

For split rail amplifiers:

$$V_{sec} = (V_{out} + 1)/\sqrt{2}$$

The Smoothing Capacitor

The main thing to bear in mind about this component is that the larger its capacitance and the better its quality, the better it will work.

The effect of this component is to reduce the ripple voltages due to the rectified mains and signal frequency currents flowing through it. If either of these currents produce an excessive voltage, the performance of the ampilifer is likely to be impaired.

An approximation to the mains derived ripple voltage developed across the capacitor can be found from the basic expression for capacitance:-

$C = Q/V$
(C in Farads, Q in Columbos, V in Volts)
Also $Q = I \times t$
(I in Amps, t in seconds)
Therefore:-

$$C = \frac{I \times t}{V}$$

For 50 Hz mains input the maximum time between capacitor charges is 10 ms. (Assuming full wave rectification.) The amount by which the capacitor discharges between charges is the ripple voltage and is dependent on the load current. Rearranging the last equation we have:-

$$V = \frac{I \times t}{C} \quad (t = 10^{-2} \text{ for 50 Hz, full wave rectified})$$

From this it is obvious that C should be as large as possible to keep the ripple small. For example, if the load current is 2 A and we wish to know what the ripple voltage is with a 10,000uF capacitor, we have:-

$$V = \frac{2 \times 10^{-2}}{10{,}000 \times 10^{-6}} = 2 \text{ Volts (peak to peak)}$$

This about the figure required for a reasonable compromise between cost and effectiveness.

The capacitor should have a ripple current rating of at least three times the DC output current.

The Rectifier

Whether this is to be a single unit or separate diodes, the working voltage for each section must be a minimum of 1.5 times the *total* secondary voltage of the transformer and for preference threetimes it. It must have a current rating at least equal to the load current and a surge capability of:-

$$I_{surge} = \frac{V_{DC} \times C}{t}$$

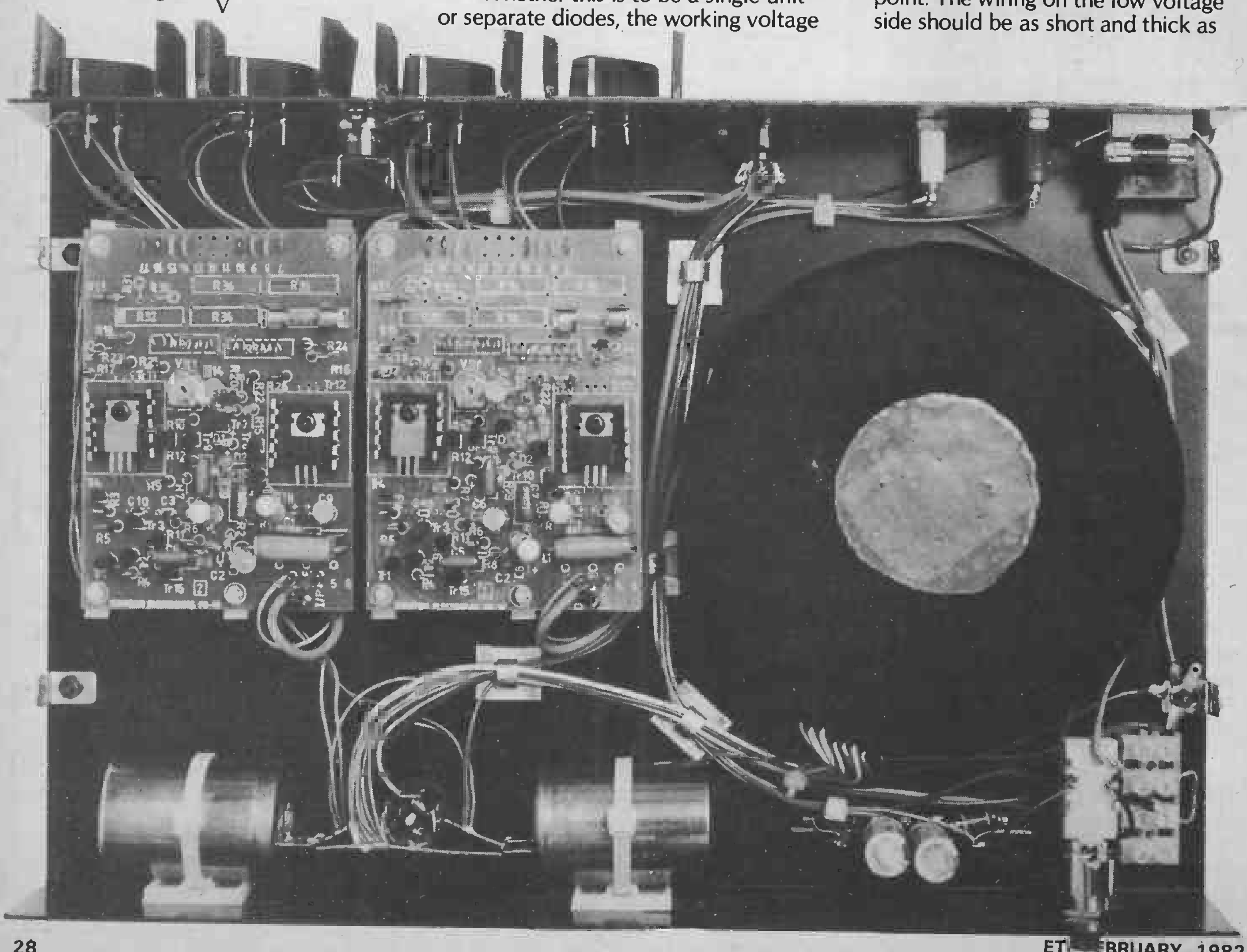
(I_{surge} in Amps, V_{DC} in Volts, $t = 10^{-2}$)

Bear in mind that this device may require some form of heatsink when supplying large currents.

Other Points

The rest of the power supply is mainly up to the individual but some things are worth mentioning.

The mains wiring should, of course, be completely safe with no exposed connections accessible even with the covers removed. A suitable slow-blow fuse, switch and indicator lamp should be included. The mains earth wire should be securely connected to all the metalwork and provision for connection to the amplifier brought to a convenient point. The wiring on the low voltage side should be as short and thick as



practicable. The possibility of earth loops must be avoided as far as possible.

The mains input to the transformer may be shunted with a capacitor of 10nF or so, to clean up any high frequency noise on the mains. *These must be made for this purpose — 250 V AC working or better.*

The rectifier bridge may be shunted with a 10nF capacitor of suitable voltage rating to reduce switching noise. It is recommended that the main reservoir capacitors also be shunted with non-electrolytic capacitors of 1uF or so to reduce their high frequency impedance.

For the best results each amplifier should be fed from a separate supply so that loading on one is not coupled to another.

For the purist, each power rail would be fully regulated and a proper mains filter would be used to eliminate any remaining mains-borne interference.

Power Supply Ripple Monitor

The purpose of this circuit is to continuously monitor the ripple voltage on a capacitor and light an LED if that voltage exceeds about 2 V peak to peak. This will show up those peaks in your programme material which cause distress in the power supply and which may be detracting from the overall clarity of reproduction.

Construction

Construction is very simple and straightforward, provided care is taken to ensure that the polarity of each semiconductor is correct.

The PCB is laid out with two circuits in mirror image to allow stereo pairs of amplifiers or single split rail amplifiers to be accommodated easily.

If desired, the polarity of all the semiconductors may be reversed using a BC212L transistor instead of the BC182L.

PARTS LIST

Resistors (all ¼W, 5%)

R1	1k0
R2	10k
R3	See Table 1

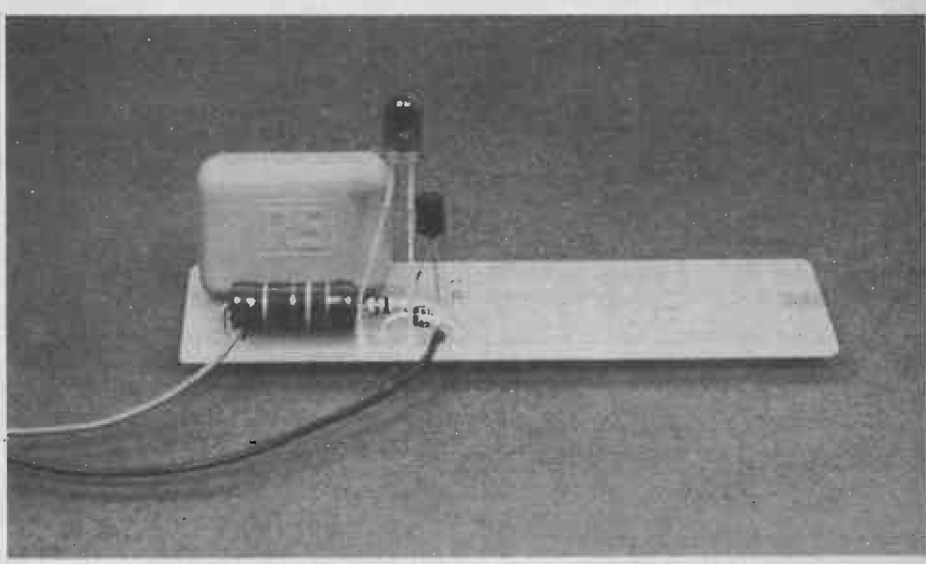
Capacitors

C1	1u0 250V polycarbonate

Semiconductors

Q1	BC182L
ZD1	5V6 400 mW zener
ZD2	27v 400 mW zener (required if supply voltage over 35v)
LED1	red LED

PCB (see Buylines)



The dual ripple monitor PCB with one circuit built onto it. For use with two mono single rail amps, you can saw the board in half - or use two boards for a stereo split rail amplifier.

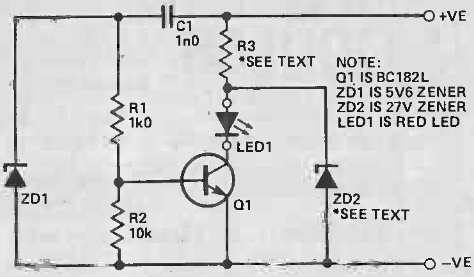


Fig. 6 (Left) Circuit diagram of the ETI Ripple Monitor.

HOW IT WORKS

The circuit is connected across the reservoir capacitor of the power supply. C1 is normally charged to about the same voltage as the reservoir capacitor and is also discharged by the same amount between charges. If the subsequent recharge is great enough, the current which flows through R1 and R2 will be enough to turn on Q1 hard and illuminate the LED. ZD1 is a zener diode which provides protection for Q1 base at switch-on and also a convenient discharge path for C1 in normal operation.

R3 limits the current in the LED and in conjunction with ZD2, limits the voltage reaching Q1 collector to a safe value.

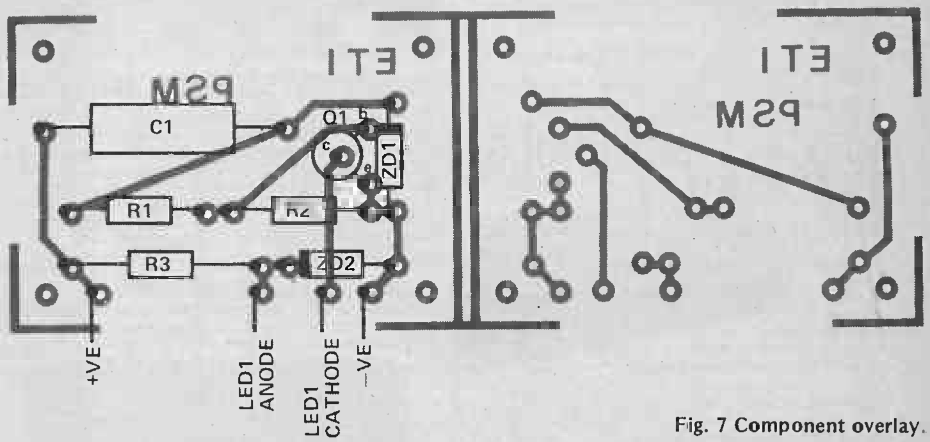


Fig. 7 Component overlay.

TABLE 1

Supply Volts.	20	25	30	35	40	45	50
R3	1k2	1k5	1k8	2k2	3k3	3k9	3k9

R3 must be a 1 W resistor.

For more brightness reduce R3 to approx ⅔ of the value shown.

ETI

DESIGNER'S NOTEBOOK

Don Keighley discusses the advantages and disadvantages of capacitively-coupled and direct-coupled preamplifiers in tape playback systems. A new IC from National Semiconductor now gives the designer greater freedom of choice.

To most engineers, direct-coupling an audio tape playback head (cassette or reel-to-reel) to its playback preamplifier seems incongruous. Direct-coupling implies that there is a possibility of direct current flow through the head which will cause erasure of parts of the recorded signal — consequently most tape preamplifier designs have a capacitively-coupled input stage. An example of this is shown in Fig. 1.

The playback head in Fig. 1 is shown connected to the preamplifier by a length of screened cable (with the screen earthed): the input preamplifier stage is shown within the broken-lined box. Capacitor C1 prevents direct current flow from the preamplifier to the playback head which would otherwise occur via transistor-biasing resistor R1.

The value of this coupling capacitor depends primarily upon the input impedance of the preamplifier and can be calculated from the formula:

$$C1 = \frac{1}{2\pi f_o R}$$

where f_o is the low-frequency corner frequency and R is the preamplifier input impedance (assumed to be purely resistive in this example). For a low-frequency corner of, say, 10 Hz and an input impedance of 50k, the capacitor value should be 320nF.

Now, if the preamplifier had a flat frequency response (ie equal gain at all frequencies in the audio band) this capacitor value could be OK. However, tape preamplifiers have an increased gain at low frequencies of about 25 dB (ie an 18 times increased response at low frequencies). Input noise, which increases with low frequency anyway, thus causes high output noise levels after amplification. Without resorting to the use of a less noisy preamplifier and tape head (both of which will be more expensive), the only other way of reducing this noise is to increase the value of the coupling capacitor. Increasing the capacitor, say, 200-fold (to about 64uF) will reduce output noise to an acceptable level.

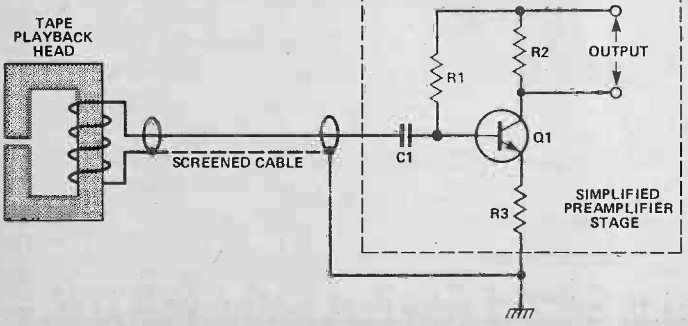


Fig. 1 Simplified details of a capacitively-coupled tape preamplifier input stage showing bias resistor, coupling capacitor and tape playback head.

Problems

The upper limit of the capacitor value is set by the turn-on time of the preamplifier — at the instant of switch-on this capacitor is fully discharged but must charge up to the preamplifier input voltage via the input impedance. If we assume that the input impedance is, as in the previous example, 50k, then it will take about two time-constants (a useful rule-of-thumb!) for the capacitor to charge up to about 90% of the final voltage.

$$\text{Now } 2\,CR = 2 \times 64 \times 10^{-6} \times 50\,000$$
$$= 6.4 \text{ seconds}$$

In most applications this time is too long and so a trade-off between time and output noise must be made — not a satisfactory arrangement.

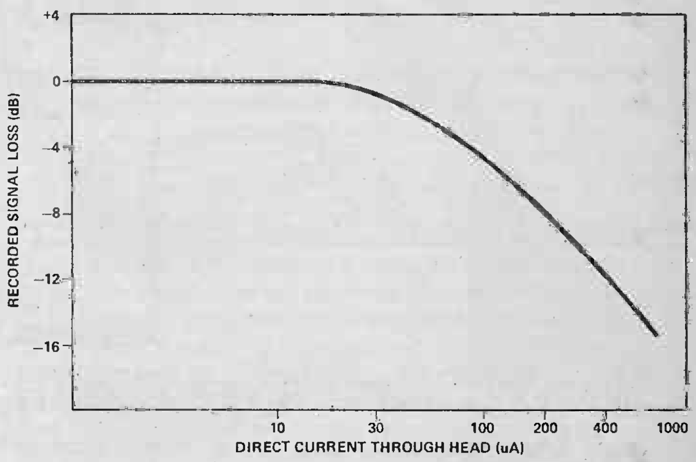


Fig. 2 Graph of approximate tape signal loss as a function of direct current through the tape head during playback, for 100 plays. The pre-recorded signal on the tape is a 10 kHz sine wave.

The most effective way to get a significant improvement in signal-to-noise ratio is, in fact, to direct-couple the playback head to the preamplifier. But this causes erasure of signal — or does it? Figure 2 shows a graph of direct head current against recorded signal loss, for a recorded signal frequency of 10 kHz which has been replayed 100 times. Severe signal loss does not occur unless the direct current through the playback head exceeds 30 uA or so. As long as this current is kept much lower, say an order of magnitude less than 30 uA, no signal loss will occur at all.

National Semiconductor has released a preamplifier IC (the LM1897) for use in such a tape playback system, which ensures that the direct current through the playback head is, typically, as low as 0.5 uA. A simplified circuit showing the IC's

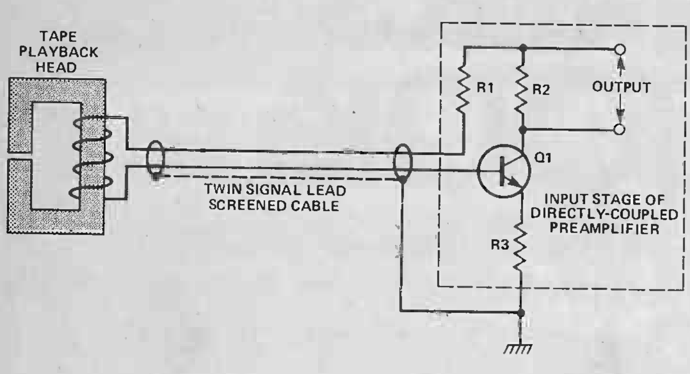


Fig. 3 Simplified details of a direct-coupled tape playback system. The transistor bias resistor is now coupled to the transistor base via the playback head. With a low bias current the DC voltage across the head is low (<1 mV) and the input stage is virtually balanced.

method of connection is given in Fig. 3. The quasi-balanced arrangement of the circuit means that screened cable with two twisted signal leads and an earthed screen can be used, thus decreasing capacitive hum pickup and crosstalk (from the other channel in a stereo system).

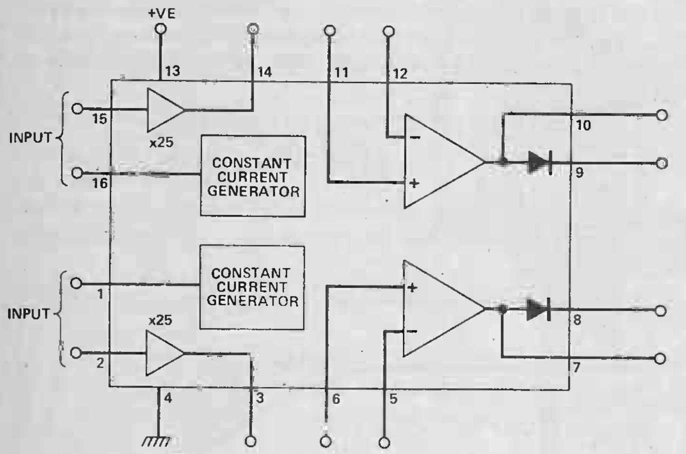


Fig. 4 Internal block diagram of the LM1897, showing leadout pin numbers.

An internal block diagram of the whole IC is shown in Fig. 4. Constant current generators provide the bias current for the input stages of the preamplifier (the input stages are shown as x 25 amplifiers) and they set the bias current for each amplifier to about 0.5 uA. Each input stage is followed by an operational amplifier, to allow equalisation and further amplification (with the use of feedback networks) before the final output. The output signal is taken either via a diode (for diode switching applications) or straight from the operational amplifier.

IC specifications are good and include a power supply rejection of 105 dB; distortion figures of, typically, 0.03%; a wide gain bandwidth of 76 dB (5600 times) over a 0-20 kHz bandwidth; and a power supply range of 4-18 V DC.

The circuit of a typical stereo tape playback system is shown in Fig. 5. Equalisation is that of a standard audio cassette system for ferric cassette tapes (ie corner frequencies of 50 Hz and 1.3 kHz). Reducing R1 to 33k changes the upper corner frequency from 1.3 kHz to 2.3 kHz to give the required equalisation for chromium dioxide tape. Circuit gain is 200 at a frequency of 1 kHz, but can be increased or decreased by reducing or increasing resistors R1 and R2 equally. Altering these two resistors equally will not affect the response of the circuit — just the overall gain.

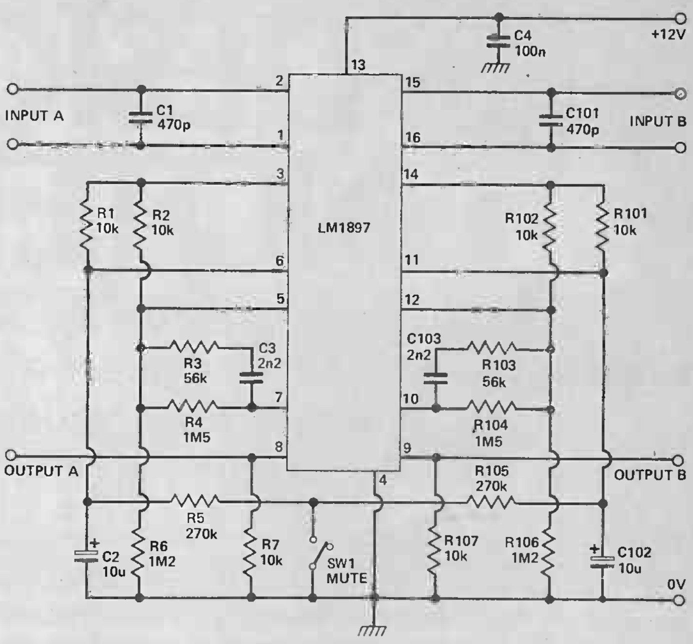


Fig. 5 Circuit of a typical stereo tape (ferric) playback system. To adjust circuit equalisation to suit chromium dioxide tape, see text.

Resistor R7, along with switch SW1, provides an optional muting facility. The value of this resistor should, at all times, be at least 10 x R1. In a stereo application the equivalent resistor of the other channel can be coupled to the same single-pole switch.

Other Responses

Of course, the IC can be used for applications other than tape playback systems. If we remember that the frequency response of the circuit is dependent on components in the operational amplifier's feedback loop, then it is a simple job to change the response. Figure 6, for example, shows the circuit for a microphone preamplifier which has a flat frequency response.

The circuit is capacitively-coupled to allow its use with a microphone, or other signal source, which is referred to ground — some sources *cannot* be directly coupled due to their nature. Slower turn-on time and increased low-frequency noise are the consequences (although noise is not such an important consideration as it is in a tape playback system because of this circuit's flat response). Performance is still good and its low power consumption and supply voltage make the circuit ideal for battery operation.

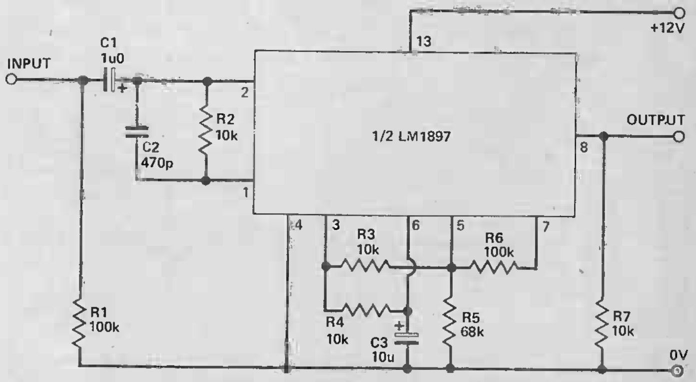
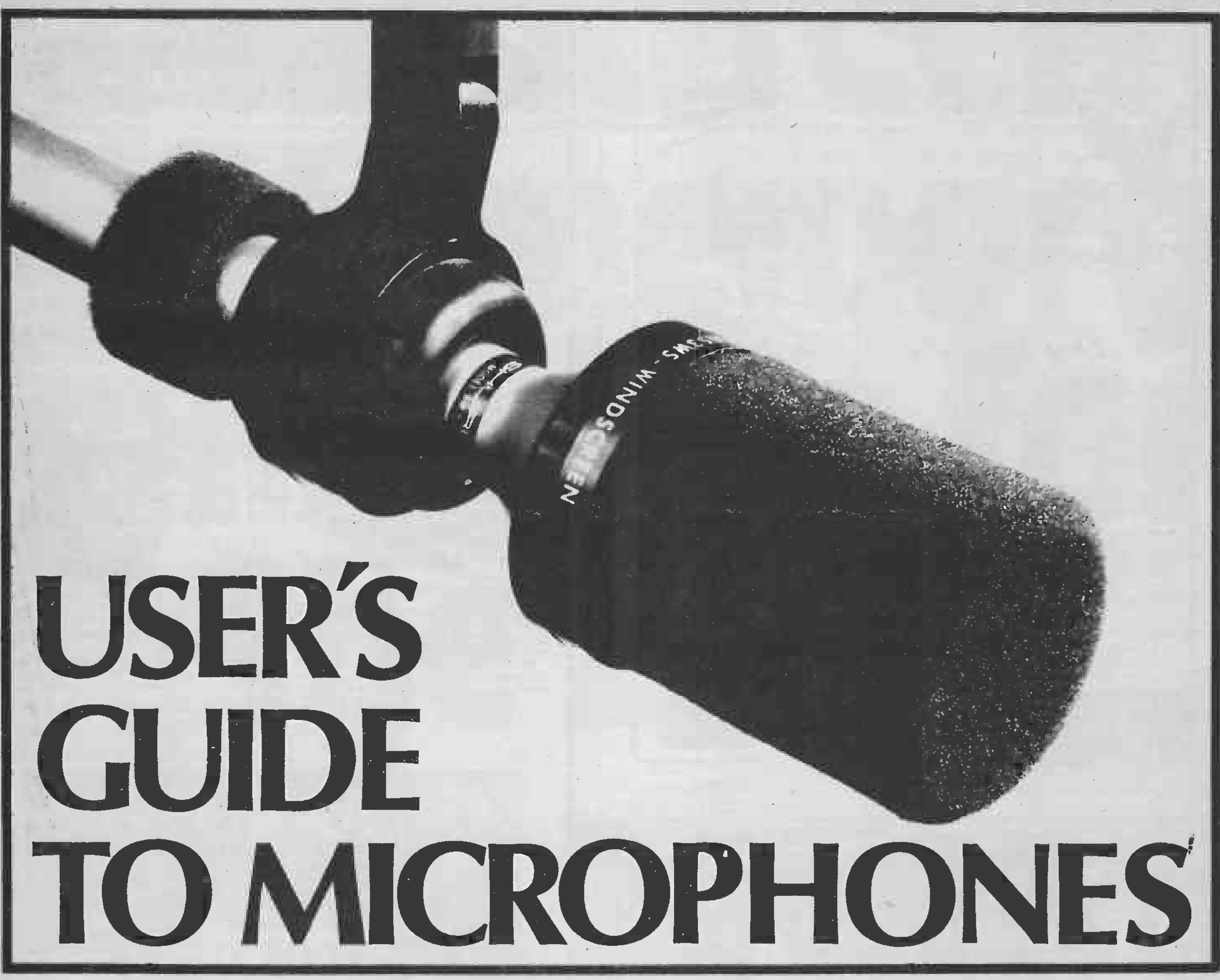


Fig. 6 A capacitively-coupled microphone preamplifier. The flat frequency response of the circuit means that low-frequency noise is not such an important consideration as in a tape playback system.

ETI



USER'S GUIDE TO MICROPHONES

So you think you know about microphones? After reading this article by Vivian Capel you will !

Along with the loudspeaker, the microphone is the most often used transducer in the world. In broadcasting, recording, noise measurement and communications; they come in all sorts and sizes. Millions are in use at any given moment. Yet unlike the loudspeaker, they are little understood except by the professional user. As with any other component, appreciation of the working principles and operation helps considerably in the choice of a suitable unit for a particular purpose and its subsequent utilisation. This article will explore these factors with a view to making the most suitable choice.

Transducer Types

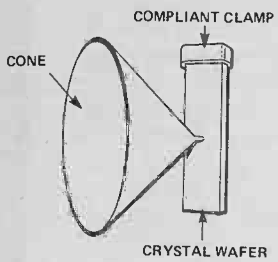
There is more than one way of converting sound into electrical energy, and most of the practical ones have been used in various microphone designs. Some of these are as follows:

Carbon: Carbon granules are packed between two plates, often also made of carbon, one of which is fixed and the other moveable. The moving one is linked to a diaphragm so that sound pressure waves acting on it alternately compress and release the granules. This produces a variation of resistance across the plates through the granules. A current is passed through from an external DC source, which varies according to the resistance fluctuations.

An output voltage can be obtained by a resistance/capacity coupling circuit or by means of a transformer. The transformer primary, DC source and microphone are connected in series, so that an AC output signal appears across the secondary.

The advantage of the carbon microphone is high output (it can drive an earpiece direct with no amplification) and it is also extremely robust. Disadvantages are poor frequency response due to the inertia of the moving system, non-linearity (and hence distortion) and noise produced by the granules rubbing over each other. Uses are therefore confined at present to telephone and other communications.

Crystal: Various natural and man-made substances such as rochelle salt (sodium potassium tartrate), quartz, tourmaline, barium titanate and lead zirconate titanate, exhibit the piezoelectric effect — they generate a voltage when subject to stress or strain. The basic crystal microphone consists of a thin



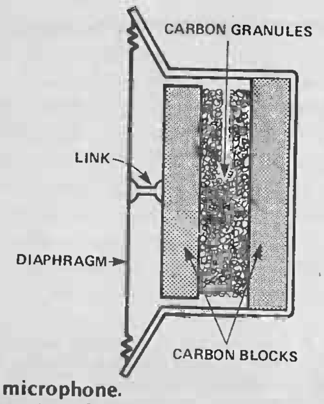


Fig. 1 (Left) Diagram of a crystal microphone.
Fig. 2 (Right) Diagram of a carbon microphone.

wafer of crystal secured at one side but free at the other. A cone is mounted with its apex bearing against the wafer near its free side and the sound pressure variations cause sympathetic flexing of the crystal and the production of a corresponding voltage.

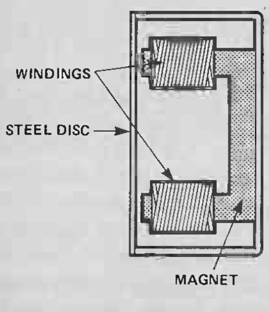
Variations of this are the bimorph, which has two wafers cemented together to give a push-pull effect, whereby one is stretched while the other compressed; this gives twice the output and partial cancellation of non-linearities. There is also the multimorph, or sound-cell, which has several crystal elements and dispenses with the cone; the sound activating the crystals directly. Output is lower, but cone resonance is eliminated.

The advantage of the crystal unit is that of a high output voltage without an external DC source; it is also relatively inexpensive. Disadvantages are its fragility; the wafers crack easily with physical shock and they are affected by humidity. Man-made crystals are rather better than the natural ones in these respects. Frequency response is uneven and poor at the top end due to inertia effects. The main disadvantage is the very high impedance (around 1M0) which results in severe loss of high frequencies over comparatively short screened cables due to capacitance effects.

When used (with short leads) to drive valve input circuits in cheap PA amplifiers and tape recorders, the high output and high impedance eliminate the need for an input transformer. They are also used to measure vibration in some industrial applications.

Moving-Iron: A steel disc is supported over the pole-pieces of a permanent magnet on which are wound a pair of coils. It vibrates as a result of the sound pressure wave and so the spacing between the disc and the pole-pieces varies. The magnetic field also varies therefore and the flux lines cut the windings, inducing an EMF. Variations include balanced and rocking armature types.

Advantages are robustness and that units can be made very small if required. They can also be used for machinery vibration tests which would destroy other types of transducer. Disadvantages are poor frequency response, pronounced resonance of the disc and distortion caused by unequal flux variation. They are rarely used other than for vibration or sub-miniature applications.



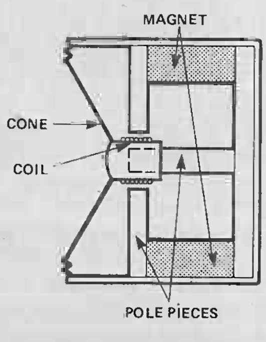


Fig. 3 (Left) Diagram of a moving-iron, or magnetic, microphone.

Fig. 4 (Right) Diagram of a moving-coil microphone.

Moving-coil: One of the most common types of transducer, the moving-coil microphone works in exactly the same way as a loudspeaker, only in reverse. The coil at the apex of the cone or diaphragm is immersed in a magnetic field between the concentric poles of a permanent magnet. Movement of the cone and coil induces current into the windings which is proportional to the sound pressure acting on the cone.

The advantages are high quality signals with extended frequency response and the ability to withstand hard usage. It is not easily overloaded by very loud sounds. The disadvantage is the comparatively high mass of the cone/coil assembly which puts a resonance peak in the frequency response between 2-5 kHz. This is a particular drawback for public address work where any resonance in the microphone response will initiate early acoustic feedback. With some microphones the peak can be 5-6 dB or even more, but the better models have it damped to around 2-3 dB. Double-transducer models are available, although these are expensive. They work like a speaker with a woofer and tweeter, one handling the low and the other the high frequencies. An electronic crossover ensures that only those frequencies within the designed range of each unit are supplied to the output. The resonant peaks of each are pushed outside of the pass range, so a flat response free from peaks is achieved.

With some applications the moving-coil resonance peak is considered an advantage, and gives what is sometimes termed a 'presence effect'. As the peak coincides with that part of the frequency spectrum at which most ambient noise occurs, it gives such ambience emphasis. So, background sounds are brought into on-the-spot recordings, interviews and the like to add realism. It can also brighten some instruments such as the piano and render certain vocalists with poor articulation a mite more intelligible. It will also emphasize a lisp, and give strings a wiry tone.

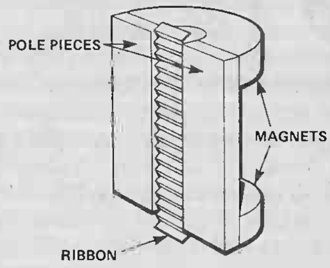


Fig. 5 Diagram of a ribbon microphone.

Ribbon: A corrugated ribbon is suspended edgeways between the pole pieces of a magnet so that as the air molecules impinge upon it, it moves back and forth and generates a current. In effect it is a single-turn moving coil without a cone. The impedance is very low, around 0.1 ohms and so a built-in matching transformer is required. The area affected by the sound wave is small and so the output is also low. Some increase in efficiency is obtained in some models by mounting a small parabolic wave-guide in front of the ribbon like a minature horn.

The advantage is that the ribbon has a very low mass, resulting in an extended high-frequency response and excellent response to transients. The resonance is high, near or beyond the normal range, so that the response is smooth over the normal audio spectrum. This makes it well suited for public address use. It has a warm, natural tone.

The low output is not a major disadvantage as modern designs give little less than many moving-coils and most input circuits possess sufficient sensitivity. Ribbons are rather fragile, though, and can be damaged by shock or by blowing into them (never 'test' any microphone by this means). They tend to be expensive but not more so than the better moving-coils. Bass is over-emphasized when used close to the lips and the explosive consonants p and b can have a shattering effect. Pop shields of foam rubber or other material are frequently used to minimise this.

Note that not all ribbon microphones are free from resonances. Some have a peak designed-in to satisfy the demand from some quarters for a presence effect!

Capacitor: The value of a capacitor depends in part on the distance between the plates. If one is fixed but the other can be moved by air pressure variations, the distance (and hence the capacitance) will be varied. Capacitor microphones consist of a plastic metal-coated diaphragm stretched over a circular shallow cavity in a metal casing. The back of the cavity serves as the fixed plate, while the diaphragm is the moving one. Aluminium or gold is used for the metal coating, and the diaphragm is sometimes embossed to give a required degree of elasticity.

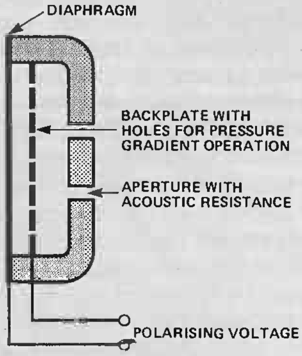


Fig. 6 Diagram of a capacitor microphone.

As the capacitance varies, current from an external source flows in and out of the device, and a signal voltage is developed over a series resistor. The applied voltage tensions the diaphragm, imparting rigidity, and so enabling it to move accurately in response to the sound pressure variations.

The diaphragm has a low mass which gives the advantage of a flat and extended frequency response, free from resonance peaks over its range, and an excellent response to transients.

A disadvantage is the need of a high polarising voltage, usually about 50 V. The capacitance of the transducer is small, around 20-30pF, and so are the variations. Thus the high voltage is required to produce usable output fluctuations. This means that the device has a very high impedance which would cause excessive HF loss over even a few inches of screened cable. To overcome this the preamp is included in the microphone case, receiving its power from the polarising supply.

This supply must be incorporated in the equipment input circuits; a means for conducting the current must also be provided through the cable. This can take the form of 'phantom' powering, in which one pole of the supply is conveyed along the signal wires through a real or artifical centre tap to the input transformer, and the other pole along the braiding. Another

Fig. 7 Phantom powering of a capacitor mike. The supply is sent via the centre tap on the transformers and the screening.

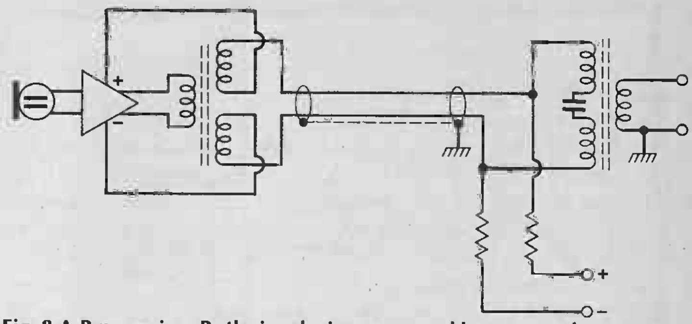


Fig. 8 A-B powering. Both signal wires ares used by means of capacitively-coupled split transformer windings. This avoids using the screen to conduct current.

method is A-B powering, where the two supply poles are carried by the two signal wires — these are DC-isolated from each other by capacitors between split windings on the internal output and equipment input transformers.

A further disadvantage is that the microphones and their ancillary equipment are expensive. This is why they are usually found only in broadcast and recording studios, or other professional applications.

There is another method of utilising the advantages of the capacitor principle without requiring polarising voltages; this is by using it to tune a small FM radio frequency oscillator and demodulator. The capacitance variations produce frequency modulations which are demodulated to produce an AF output signal. This system is used with Sennheiser capacitor microphones.

Electret: A comparative newcomer to the scene, the electret is a capacitor microphone but with the polarising charge permanently implanted in the diaphragm. The diaphragm is placed between the plates of an air-spaced capacitor which is charged from a high-voltage source. The diaphragm is heated and then allowed to cool, whereupon the charge is fixed and permanent.

It needs to be thicker than the normal capacitor microphone, which increases its mass and curtails its transient and frequency response. An internal preamp is still required, but this can be a simple two-transistor design operating from a single internal 1V5 cell.

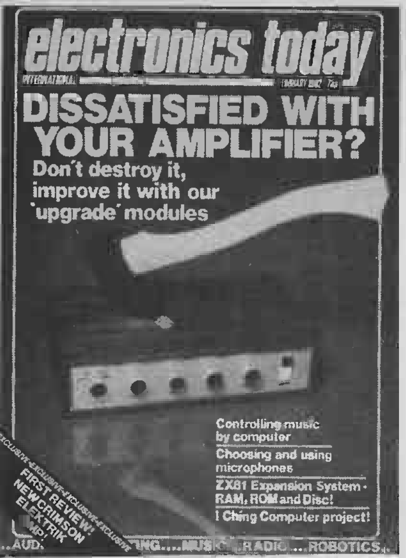
The advantages are a smooth extended response which, although not to the standard of powered capacitor models, is very good and better than most moving-coils. The diaphragm resonance is high (typically 8-10 kHz on the better models), and unlike the mid-range resonance of the moving-coil, this can be tamed with a little treble cut on the tone control. Electrets are light and can be made quite small, making them ideal for the tie-clip units which are favoured by many speakers and entertainers. In this case the battery cell can be a pen-torch HP7 housed in the microphone plug, or a small mercury cell in the microphone itself. The current drain is very small (a fraction of a milliamp) and a long battery life can be expected.

A major advantage is cost, as electrets are very inexpensive. Unfortunately this had encouraged many makers to produce poor specimens which can give disappointing results. As with capacitor mikes, they are affected by high humidity and can be overloaded by very loud sound sources. Given a good make, the electret has much to commend it for general use and has been found particularly useful for public address work, where the absence of a mid-range resonance enables feedback to be more easily controlled.

Directional Response

The manner in which a microphone responds to a sound field differs according to the operating principles and, of course, is important for choosing the correct type for a particular task. task.

Omnidirectional: When the back of the diaphragm of a microphone is sealed, there is an isolated body of air trapped in



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Miking-up made easy p.38

the enclosure, just as in a sealed speaker cabinet. This exerts a fixed pressure against which the diaphragm works. The sound pressure wave is alternately greater and less than the pressure exercised on the back of the diaphragm by the trapped air, so the diaphragm moves backwards and forward. Now sound pressure (like barometric pressure) exerts force in all directions, not only in the direction of sound propagation. So, just as a barometer will indicate pressure of air irrespective of how it is mounted, the pressure microphone will react to sound whether pointed toward the sound source or not.

It is therefore said to be omnidirectional, responding to sounds coming from all directions. This characteristic only holds good up to the freqeuency where the wavelength is greater than the diameter of the diaphragm. Above this (at

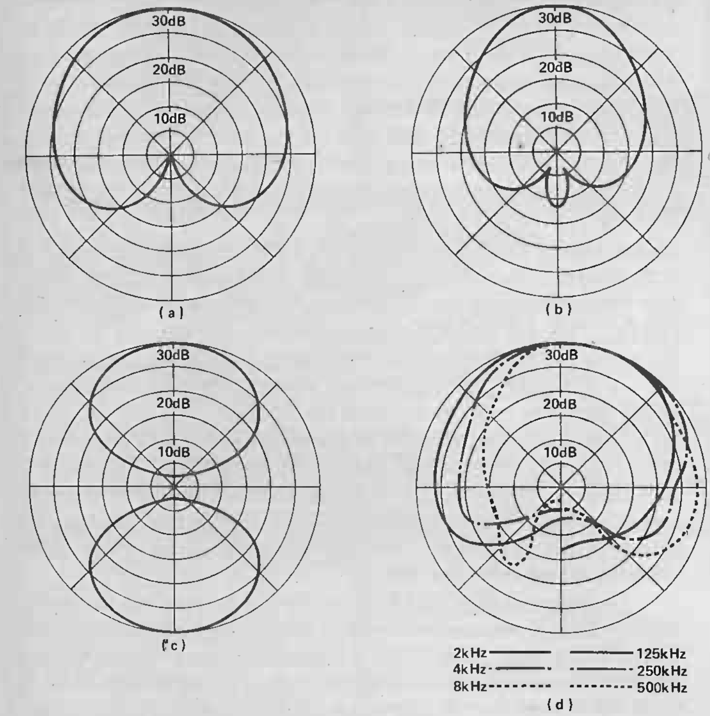


Fig. 9 Polar diagrams of nominal directional response. (a) cardioid; (b) hypercarioid; (c) figure eight; (d) polar response varies with frequency; plots are given for six different frequencies.

shorter wavelengths) the microphone begins to exhibit directional properties. When pointing at 90° from the sound source there are interference effects across the diaphragm, with different parts being affected by different portions of the sound wave at the same time. It can therefore only respond to a limited extent, proportional to the resultant pressure over the whole area.

When the microphone is pointing in the opposite direction to the sound source, the diaphragm is in an acoustic shadow for these short wavelengths, as all objects cast a sound shadow behind them (due to diffraction) when their size is greater than the wavelength of the sound. Hence the microphone response is very low at such frequencies in that position. To give some idea of the frequencies involved, at 1" the frequency is 13.5 kHz, at 1.5", 9 kHz. Most modern microphones have diaphragms smaller than 1" so the effect only occurs at the upper limits of the audio spectrum.

Cardioid: If an aperture is made in the chamber behind the diaphragm so that it is no longer sealed but is open to atmospheric pressure, the mode of operation is changed. Sound pressure is exerted on both sides of the diaphragm which, if equal, would result in zero movement and output.

However, access to the rear through vents at the sides of the microphone is restricted so that the pressure is not equal. Furthermore, the direct action of air molecules moving back and forth along the axis of propagation imparts energy to the front of the diaphragm when the source is situated at the front. Therefore part of the resulting motion is due to the velocity of the air molecules, so the microphone is often termed a velocity type. Pressure differences also are involved so it can be known as a pressure gradient microphone.

When the sound source is to the side of the microphone, similar pressure differences exist between the front and back of the diaphragm but there is no direct impingement of air particles on the front of the microphone. In fact, there can be limited velocity effect on the back of the diaphragm due to diffraction around the edges of the vent. Hence the output is much lower, with sounds coming from the rear of the microphone being almost completely cancelled and producing very little output.

The unit is therefore directional, favouring sounds arriving from the front and progressively rejecting others as the angle from the front increases. When plotted on a circular graph (in which the concentric circles represent the output level) the response resembles a heart-shape — hence the term cardioid.

Hypercardioid: If the acoustic resistance offered by the vents is modified, so that pressure exerted on the back of the diaphragm is increased, the response to sounds arriving from the side is reduced even further, and the directivity increases. The polar response is therefore narrower and becomes a hypercardioid. One drawback is that the increased rear sensitivity often more than cancels the effect of sounds coming from the rear on the front of the diaphragm. This results in a negative lobe at the rear.

Figure Eight: Ribbon microphones are equally sensitive to sounds coming from the front or rear unless the back of the instrument is deliberately restricted by pads. As they are velocity devices, the response to sounds coming from the sides is zero. The polar diagram with its front and rear lobes therefore appears like a figure eight. Some ribbon models have been designed to give a cardioid or hypercardioid polar response by using sound guides at the front and suppression at the rear.

Gun Microphone: These use an interference tube in conjunction with a pressure gradient transducer to give a highly directional forward response. The tube is open at the end and has a series of slots or holes running along one side.

When pointed at the source, pressure waves enter the end of the tube and the side holes to arrive simultaneously at the transducer diaphragm. When the source is to one side, the sound follows various paths of different lengths — the shortest

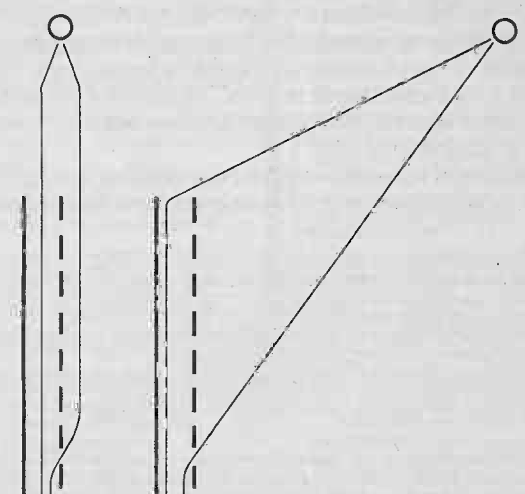
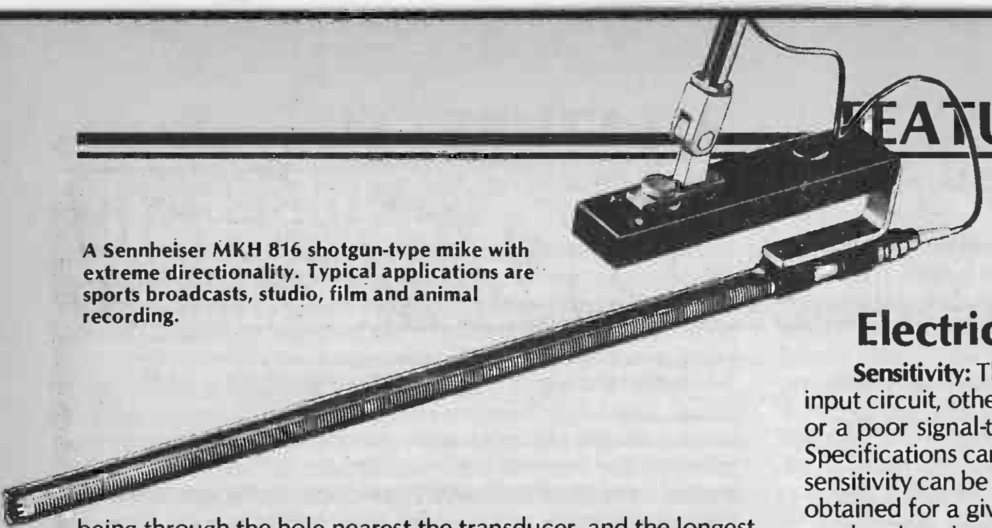


Fig. 10 (a) Sound enters gun microphone from the end and side to arrive at the diaphragm simultaneously when the source is the front. (b) With a source at the side, sound paths differ in length. When the difference is half a wavelength cancellation occurs. This happens over a range of frequencies due to the different spacing of the holes.



A Sennheiser MKH 816 shotgun-type mike with extreme directionality. Typical applications are sports broadcasts, studio, film and animal recording.

being through the hole nearest the transducer, and the longest being via the end of the tube. If the difference between these is half a wavelength, they will cancel and there will be zero pressure exerted on the transducer.

Obviously this happens at only one frequency, but as there is a series of holes there will be cancellation effects for all frequencies within the range of the tube. Cancellation also occurs when the path difference is 1½, 2½, 3½ (and so on) times the wavelength. Of course there are multiple paths, not just the two that provide cancellation, but being spaced between the critical pair these tend to balance out and produce secondary cancellation.

The tube is only effective at frequencies above the half-wavelength value of its total length — below this it reverts to ordinary hypercardioid operation or whatever the transducer mode is without the tube. Some gun microphones have omni transducers, which seems odd since there is no directivity at all below the tube frequency range. Omnis can be identified by having no vents behind the diaphragm. The half-wavelength frequency of a 2 ft tube is around 280 Hz. Directivity increases linearly with frequency.

Frequency Response

Frequency response is often quoted between two limits, such as 30 Hz-18 kHz. This is useless as it does not specify the levels at which the given limits are measured; they could be at −3 dB, −20 dB or any level in between. Furthermore, they give no indication of the flatness or otherwise of the response. An improvement is to give level limits such as 30 Hz-18 kHz ± 3 dB. This means that the limit frequencies (and all in between) do not vary by more than ± 3 dB.

Even this is not entirely satisfactory as it does not reveal whether deviations are sharp peaks or broad plateaus, nor their frequencies. The only certain method is the response curve, but even here there are pitfalls. Take a careful look at the vertical calibration; if the divisions are denoting large steps, the curve will look a lot flatter than it really is. The divisions should be in 1 or 2 dB steps, but 4 or 5 dB increments are not unknown and so can be misleading.

The actual response required depends on the application, but for most, smoothness is more important than extent.

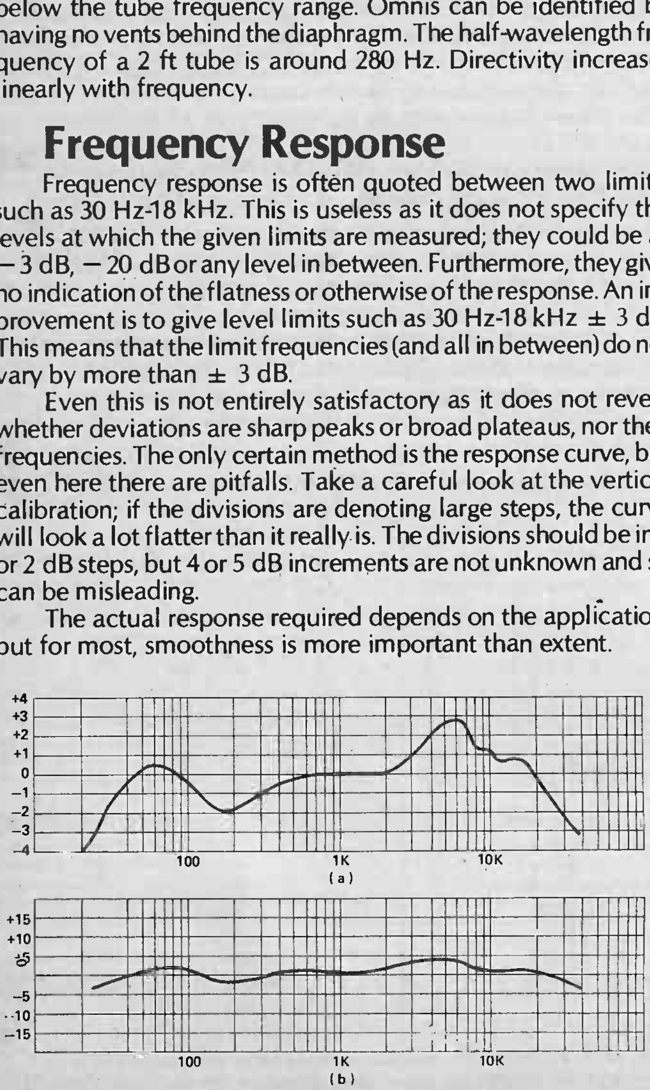


Fig. 11 Which microphone would you prefer, (a) or (b)? A close look at the vertical calibration shows that the frequency response is the same for both. Always check calibration when comparing response curves.

Electrical Characteristics.

Sensitivity: The output must be sufficient to fully drive the input circuit, otherwise a power amp may not attain full power or a poor signal-to-noise ratio may result in other equipment. Specifications can be confusing due to the many ways in which sensitivity can be expressed — it stipulates the electrical output obtained for a given sound pressure, and both of these factors can be given in several different units.

Impedance: There are three commonly used impedance ranges; low (30-50 ohms), medium (200-1000 ohms, and sometimes also called low) and high (47 kilohms). High impedance screened cables result in HF losses over all but short lengths, while low impedance is vulnerable to resistive losses over long cable lengths and noise due to connecting plug contacts. It is the medium impedance, 200 ohms in particular, that is commonly used by professionals, and it is taking over from the previous standard of 600 ohms. Impedances in this range match well with most transistor input stages and can be fed without the use of a transformer.

Noise: When placed in a vacuum with no air actuation of the transducer, a microphone will still produce an output. This is due to thermal agitation of the wire in the case of moving-coil or ribbon units; for capacitor microphones it is caused by similar agitation of the diaphragm metalising, plus irregularities in the supply and noise in the preamp. Owing to the extra sources, the latter generate more noise than the former.

There are several ways of specifying microphone noise — the simplest is the actual output in microvolts. This may be weighted to take account of the nuisance effect that some frequencies have over others. The DIN 45-405 weighting curve, which peaks at between 3-4 kHz and rolls off on either side, is commonly used.

To be meaningful, the noise should be related to the sensitivity, as a given noise level will obviously be more obtrusive with a low signal output than a high. One way of doing this is to convert the noise voltage into equivalent sound pressure — the sound pressure that would produce the same output voltage in that microphone. This may be expressed as a decibel ratio, using the hearing threshold (0.0002 ubar) as the reference level. Alternatively it may be expressed in phons, which is numerically the same as the dB value.

Hum: Moving-coil or ribbon transducers are susceptible to hum when working in strong hum fields. Some have hum-cancelling coils built-in to minimise it. Occasionally the manufacturers a figure which indicates hum sensitivity, the unit being the microvolt per microTesla. The standard field density is 5 uTesla.

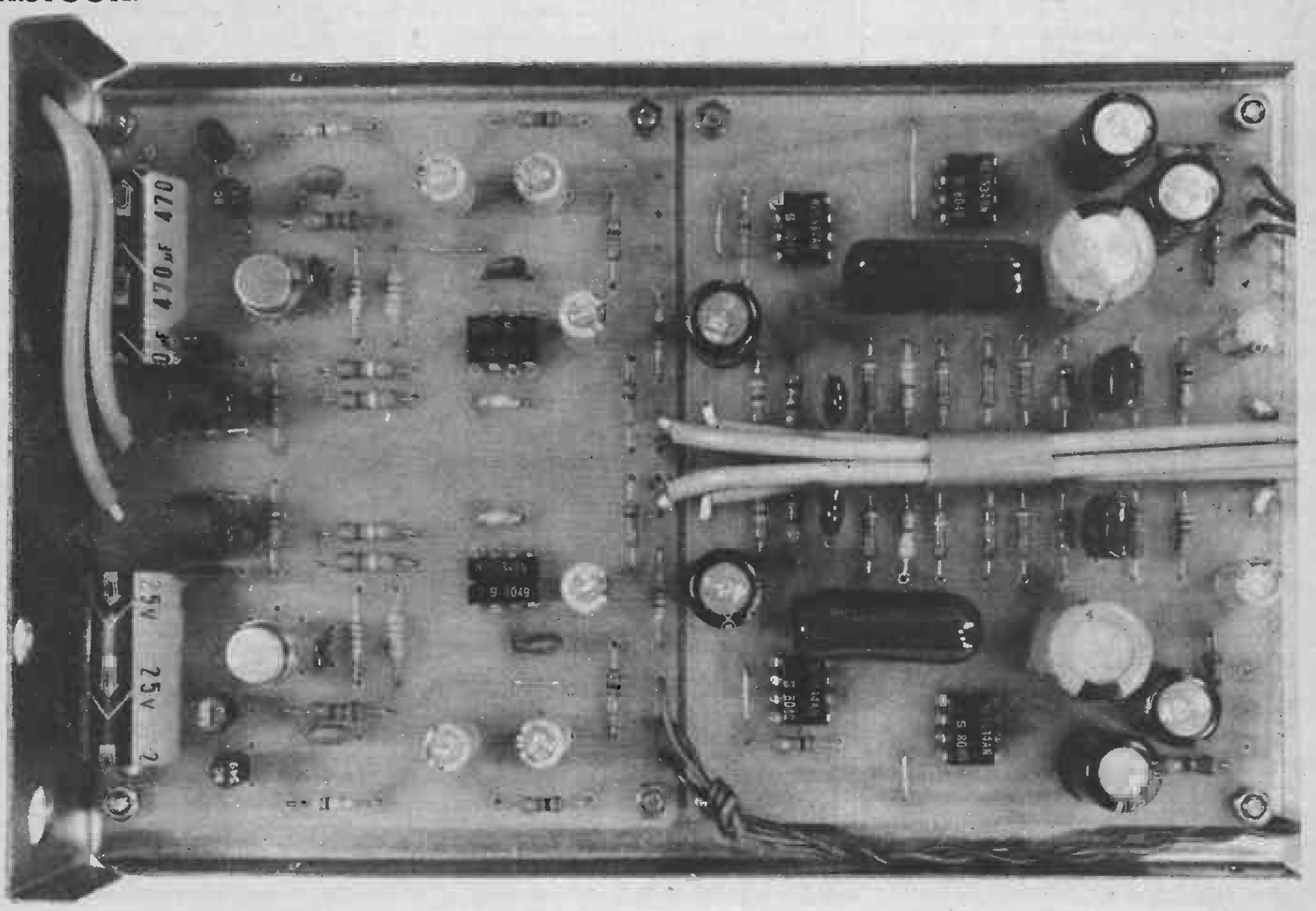
Overload and Distortion: Capacitor microphones are the most easily overloaded by loud sounds so a maximum sound pressure is often quoted, 200-300 ubars being a common value. Lower output models can handle larger volumes, some going to 500 and one even to 1,000 ubars. One model designed for instrumentation applications has an extremely low output that will accept up to 5,000 ubars, a level that would destroy human hearing.

Dynamic microphones (moving-coil and ribbons) are not usually given a maximum sound pressure rating, but they often have a level specified at which distortion will rise to a particular value. This is normally 0.5% THD (total harmonic distortion), but some of the cheaper ones use a 1.0% THD reference. The description usually reads SPL for 0.5% THD = 300 ubars (for example).

There is a tremendous range of microphones now available, making the best choice none too easy. However, if the acoustic and electrical requirements (together with the other features described) are isolated, the selection and subsequent use will be reasonably straightforward. **ETI**

HIGH QUALITY PHONO AMPLIFIERS

Last month we gave an insight into the design procedures for low level phono signals. This month the theory is put into practice with these outstanding input amplifiers. Design and development by David Tilbrook.



The preamp has been designed specifically to overcome the problem of cartridge impedance interaction. This has been achieved by separating the MM input stage into two separate 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 imedance 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.

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.

Equal Change

In an attempt to overcome bass problems 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

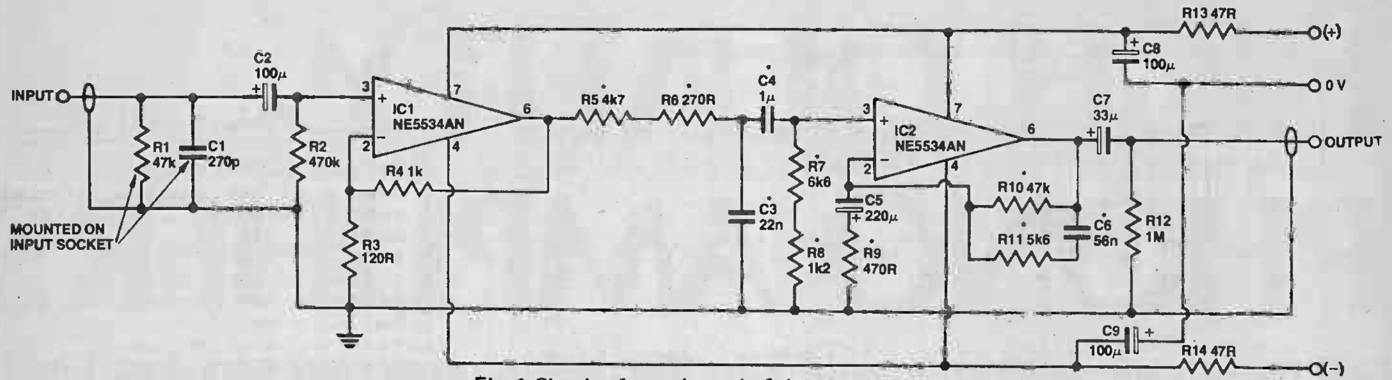


Fig. 1 Circuit of one channel of the moving magnet input stage. Note that the RIAA equalisation is incorporated in this stage.

* DENOTES COMPONENTS ASSOCIATED WITH THE RIAA EQUALISATION

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 is that severe cases of tonearm resonance or rumble generate distortion harmonics well above 20 Hz and into the audio spectrum. The only real cure is to remove the problem at the turntable or tonearm.

The preamp conforms to the proposed RIAA equalisation in Fig. 2. The 75 uS and 7950 uS 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

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

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 uS 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.

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

First there is a 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 uS time constant is included in the negative feedback of a 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. In the MM stage the gain of the second stage does not drop below 10; since the NE5534AN is internally compensated for gains of three or above no additional compensation is required.

HOW IT WORKS

MOVING-MAGNET STAGE

The input from a moving-magnet cartridge is connected to the non-inverting input of an NE5534AN via capacitor C2. R2 provides a DC current path to the input of the differential pair in the op-amp. The gain of this stage is determined by the ratio R4 to R3, which is around 8.3 in this case.

The resistor R1 provides a fixed resistive load necessary for best performance from an MM cartridge. Most cartridge manufacturers recommend that the input resistance be shunted by a certain amount of capacitance. This is the purpose of capacitor C1, the value of which should suit most cartridges. If you wish to optimise the value of this capacitor, don't forget to allow several hundred picofarads for the shielded cable capacitance.

The best way to ensure that the cartridge is loaded correctly is with a test record containing a square wave track, and an oscilloscope. With the correct cartridge load and a good tonearm/cartridge combination, a good square wave can be obtained.

The value of resistor R1 at 47k is effectively in parallel with R2, giving an input resistance of 43k, slightly below the 47k normally used for MM input stages. This is unimportant however, and will not affect performance of the cartridge. The important thing is that the value of this resistance remains constant over the full audio spectrum and beyond. In any case the value of the input resistance is easily changed by increasing the value of R1 to, say 56k instead of 47k.

The output of the first stage is fed to two 6 dB/octave RC filters which provide one half of the RIAA equalisation. Resistors R5, R6 and capacitor C3 form a first-order-low-pass filter set at the 75 uS time constant of the RIAA curve. At these frequencies (around 2122 Hz) the 1uF capacitor appears as a short circuit connecting R7 and R8 in parallel with the capacitor C3. This must be compensated when choosing the value of C3 to ensure the correct RIAA equalisation. Similarly C4, R7 and R8 form a low frequency high-pass filter set at 20 Hz (the 7950 uS time constant).

The output of these two filters is fed to the input of the second op-amp stage. The remaining RIAA equalisation is accomplished by the feedback loop around this stage. At frequencies below 500 Hz the 56n capacitor C6 has relatively high impedance. The voltage gain is therefore determined by resistors R9 and R10. At higher frequencies, where the impedance of C6 is less, both resistors R10 and R11 are in circuit. The capacitor C5 decreases the gain at DC, of the second stage to unity, ensuring a low DC offset at the output and therefore symmetrical output stage clipping.

The 1M0 resistor R12 ensures that the DC voltage on the output remains at 0 V. This is important so that operation of the selector switch following the stage will not cause thumps in the output.

Resistors R13, R14 and capacitors C8, C9 isolate the supply to the stage in order to decrease the effects of interactions between stages and to ensure freedom from 50 Hz ripple.

Moving Coil Input

The complete circuit diagram for the moving coil input stage is shown in Fig. 4. 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

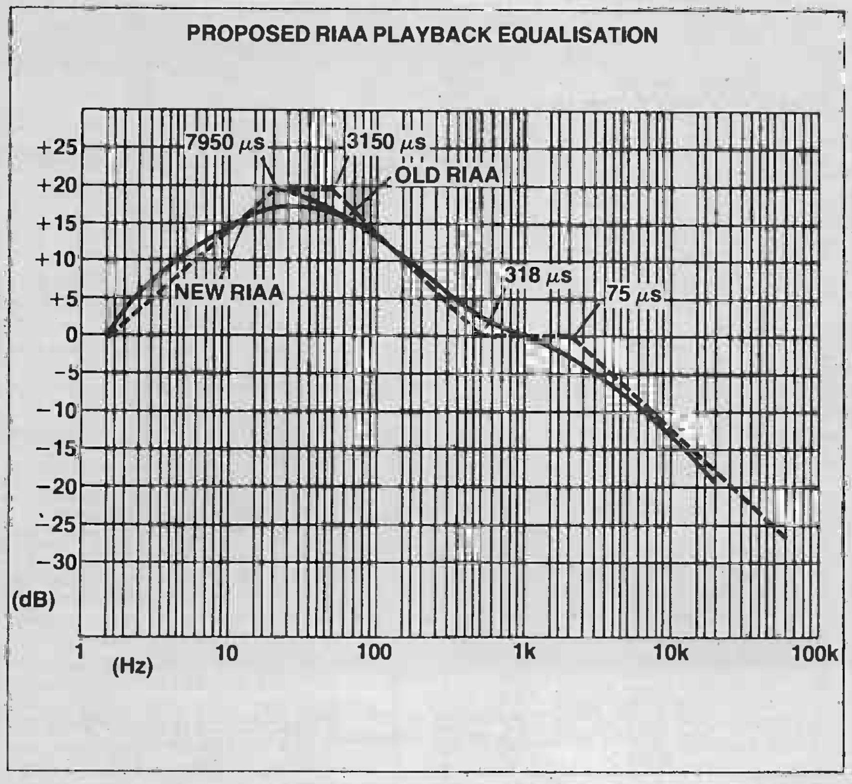


Fig. 2 Old and 'new' RIAA equalisation curves (solid line). The individual time constants (Bode plot—dotted lines) to produce the response are also shown.

PARTS LIST

MOVING-MAGNET STAGE

Resistors (all ¼W metal film, 5% unless stated otherwise)

R1,101	47k
R2,102	470k
R3,103	120R
R4,104	1k0
R5,105	4k7 1%
R6,106	270R 1%
R7,107	6k8 1%
R8,108	1k2 1%
R9,109	470R 1%
R10,110	47k 1%
R11,111	5k6 1%
R12,112	1M0
R13,14, 113,114	47R

Capacitors

C1,101	270p ceramic
C2,102	100u 16 V PCB electrolytic
C3,103	22n polyester
C4,104	1u0 polyester
C5,105	220u 16 V PCB electrolytic
C6,106	56n polyester
C7,107	33u 25 V PCB electrolytic
C8,9,108,109	100u 25 V PCB electrolytic

Semiconductors

IC1,2,101,102 NE5534AN

Miscellaneous

PCB (see Buylines); assorted mounting hardware; shielded cable.

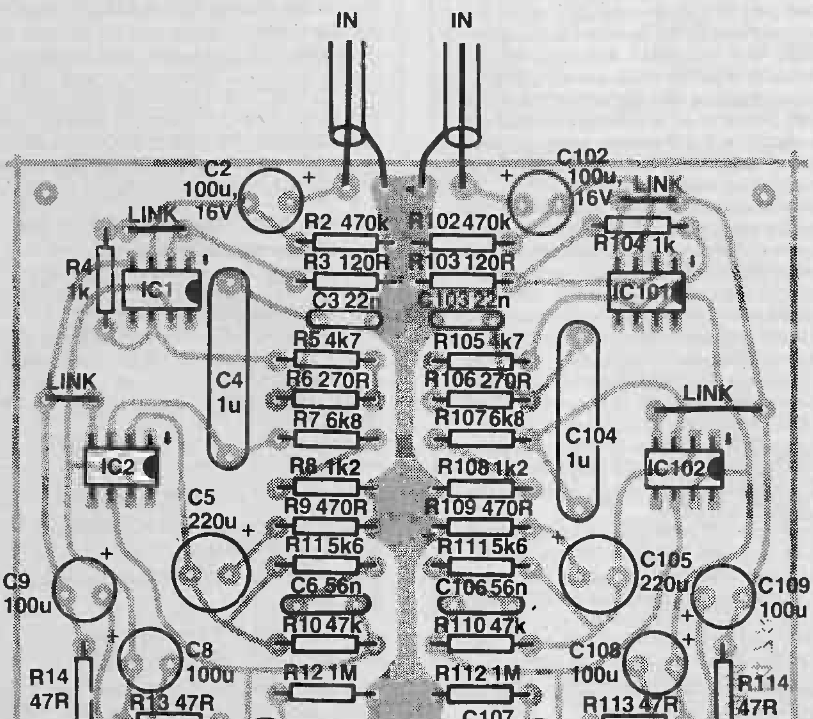


Fig. 3 Component overlay for the moving magnet stage.

impedance, over a fairly wide range of values without affecting the operation of the circuit.

Once again the input coupling capacitor C4 is used to prevent DC current from flowing through the cartridge. Capacitor C4 is shunted by C3, a 10n 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 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.

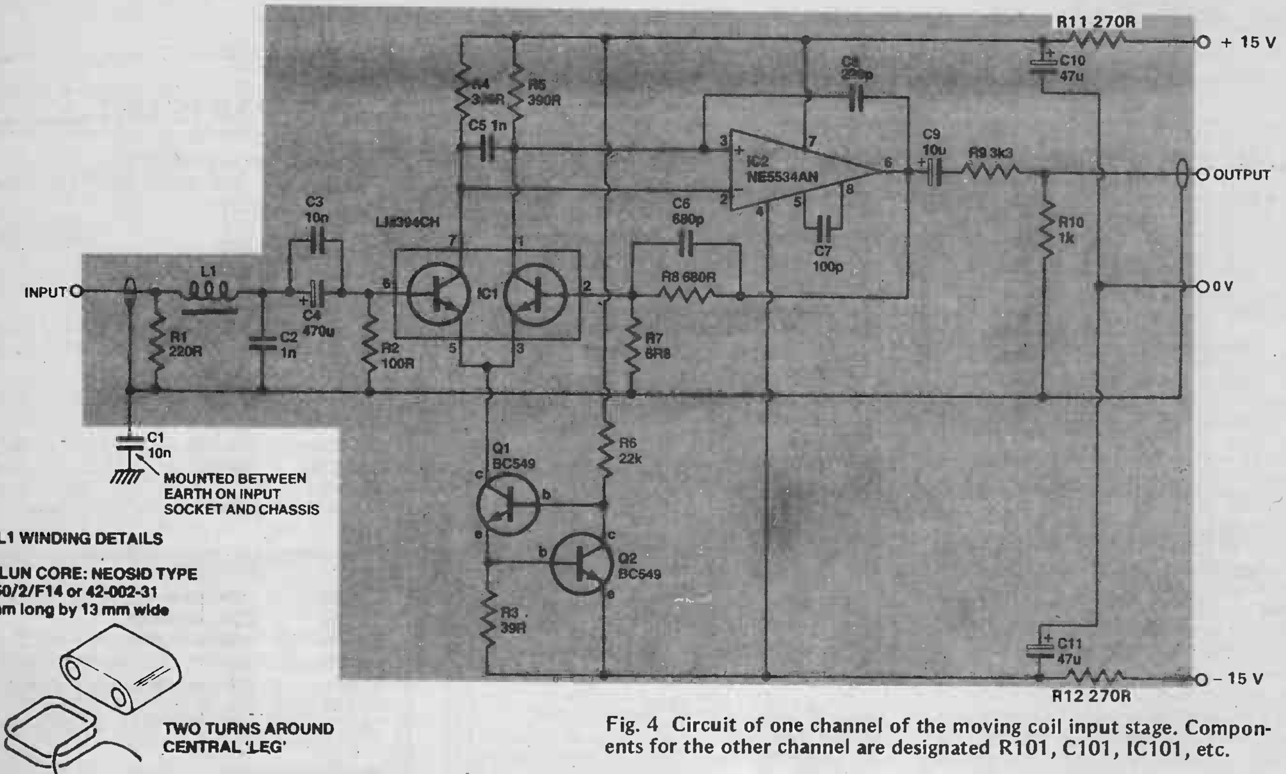


Fig. 4 Circuit of one channel of the moving coil input stage. Components for the other channel are designated R101, C101, IC101, etc.

HOW IT WORKS

MOVING-COIL PREAMP

The input from a moving coil cartridge is fed via L1 and capacitors C3 and C4 to the base of one of the transistors in the LM394, which functions as a differential input stage.

Q1 and Q2 form a constant current source, which stabilises the DC operating point and ensures a high impedance source to the emitters of the differential pair. The constant current source works by ensuring that a constant voltage is maintained across a fixed value of resistance. Resistor R3 is used for this purpose, with the base emitter voltage of Q2 expressed across it. If the current through R3 were to try to increase even slightly, the voltage on the base of Q2 would be increased, turning Q2 on harder. This causes the voltage on the collector of Q2 to decrease, decreasing the current through R3. So Q2 provides negative feedback acting to correct any deviations in the current flowing through the differential pair.

The collectors of the LM394 are shunted by the 1n0 capacitor C5. This decreases the gain of the first stage at high frequencies and helps to ensure stability (ie: freedom from high frequency oscillations).

The input stage is operated in full differential mode by connecting both collectors to inputs of the NE5534AN. If this is not done, the voltage gain of the input stage is decreased and the signal-to-noise ratio is degraded. Because differential pairs have two base-emitter junctions in the input circuit, their total equivalent input noise is inferior to that of a single transistor. However, since it is possible using a differential pair to obtain noise figures of the same magnitude as the thermal noise of the cartridge, the marginal decrease in the theoretically best signal-to-noise ratio is of little consequence. On the other hand the inherent linearity of a differential pair offers a significant advantage over a single transistor, improving both distortion and high frequency stability.

Capacitor C7 ensures stability of the op-amp by providing adequate compensation for the increased gain around the stage due to the differential pair. C9 provides DC isolation of the stage. The resistors R9 and R10 form a potential divider to decrease the signal level to that suitable for the MM input. If the particular moving coil cartridge used requires a different amount of voltage gain than is provided, the value of R9 can be changed accordingly. Replacing R9 with a short circuit (ie: a piece of tinned copper wire in place of the resistor on the circuit board) increases the voltage gain of the stage slightly over 100.

The two RC networks, R11, C10 and R12, C11 provide isolation of the supply voltage from other stages using the same power supply. This decreases interactions between stages, thereby improving crosstalk and the overall stability of the preamplifier.

BUYLINES

These low level input stages have been designed to deliver state-of-the-art performance — as we've not compromised on the design we suggest you don't compromise on the components. Both the MM and MC stages use high performance NE5534AN op-amps; a possible alternative to this is the TDA1034 op-amp. Accept no substitutes. The LM394 and NE5534 are available from Watford Electronics, as is the Neosid balun core; the latter item can also be obtained from Neosid Small Orders, PO Box 86, Welwyn Garden City, Herts AL7 1AS. Technomatic stock the LM394 and the TDA1034. The PCBs are essential to preserve the layout and earthing; a very necessary requirement if the full performance is to be achieved. Boards will be available from our PCB Service at the prices listed on page 81.

Construction

Construction of both boards is relatively straightforward, since almost all the components are mounted on the PCBs. Resistor R1 and capacitor C1 on the moving magnet board are intended to be mounted directly across the back of the input socket. Order of construction is not critical, although it is probably easier to mount small components first, followed by the larger components such as the electrolytic capacitors, ICs and transistors; these components will be damaged if the unit is powered up with them inserted incorrectly. Shielded cable should be used on all inputs and outputs. We have used mono shielded cable rather than the stereo type for ease of soldering.

The inductor on the input of the MC stage consists of two turns wound on a ferrite balun core, 6 mm long by 13 mm wide. We used the type given in the Parts List.

Each of the PCBs is a stereo input amplifier, with each channel sharing a common input earth track running down the centre of the board. The power supply wiring from each channel on the board can be connected in parallel, so only three wires (+,0,−) need to be brought out for power.

The input earth is *not* connected to the 0 V line from the power supply at any place on the PCBs. This means that without a separate 0 V connection added to the input stage they will *not* work. This has been done deliberately

SPECIFICATION

MOVING-MAGNET INPUT STAGE

Gain:	74, 1 kHz
Frequency response:	Conforms to RIAA Equalisation ±0.2 dB (This is the performance of the prototype. The actual figure obtained will be determined by the accuracy and longterm stability of the components used.)
Total harmonic distortion:	<0.001%, 1 kHz, 10 mV RMS input
Headroom:	>28 dB with respect to 5 mV RMS input signal ie:135 mV RMS max.
Noise:	Total equivalent input noise: 112 nV 'A', input shorted, 216 nV flat, input shorted.

S/N ratio:

	1 mV	5 mV	10 mV
Flat	73 dB	87 dB	93 dB
A-weighted	78 dB	92 dB	98dB

MOVING-COIL INPUT STAGE

Gain:	24
Frequency response:	7 Hz-135 kHz + 0, −1 dB
Total harmonic distortion:	<0.003%, 1kHz, 30 mV input
Noise:	Total equivalent input noise: 83 nV flat, input shorted. 42 nV 'A', input shorted 56 nV flat, after RIAA Eq, input shorted 34 nV 'A', after RIAA Eq, input shorted

S/N ratio of MC stage after RIAA Equalisation:

	60 uV	200 uV	500 uV
Flat	61 dB	71 dB	79 dB
A-weighted	65 dB	75 dB	83 dB

RESPONSE

Hz	IDEAL RIAA dB	MEASURED dB
2	−0.2	−0.2
4	+5.7	+5.7
8	+11.2	+11.2
16	+15.4	+15.4
20	+16.3	+16.3
30	+17.0	+17.0
40	+16.8	+16.8
50	+16.3	+16.2
80	+14.2	+14.2
100	+12.9	+12.8
150	+10.3	+10.2
200	+8.2	+8.1
300	+5.5	+5.4
400	+3.8	+3.7
500	+2.6	+2.6
800	+0.7	+0.7
1k	0.0	0.0
1k5	−1.4	−1.3
2k	−2.6	−2.4
3k	−4.8	−4.7
4k	−6.6	−6.6
5k	−8.2	−8.1
6k	−9.6	−9.6
8k	−11.9	−11.9
10k	−13.7	−13.8
15k	−17.2	−17.1
20k	−19.6	−19.5

PARTS LIST

MOVING-COIL STAGE

Resistors (all ¼ W metal film, 5%)

R1,101	220R
R2,102	100R
R3,103	39R
R4,5,104,105	390R
R6,106	22k
R7,107	6R8
R8,108	680R
R9,109	3k3
R10,110	1k0
R11,12, 111,112	270R

Capacitors

C1,3,101,103	10n polyester
C2,5,102,105	1n0 polyester
C4,104	470u 16 V axial electrolytic
C6,106	680p ceramic
C7,107	100p ceramic
C8,108	220p ceramic
C9,109	10u 16 V PCB electrolytic
C10,11, 110,111	47u 25 V PCB electrolytic

Semiconductors

IC1,101	LM394CH
IC2,102	NE5534AN
Q1,2,101,102	BC549

Miscellaneous

L1	Two turns on ferrite balun core, Neosid type 1050/2/F14 or 42-002-31

PCB (see Buylines); assorted mounting hardware; shielded cable.

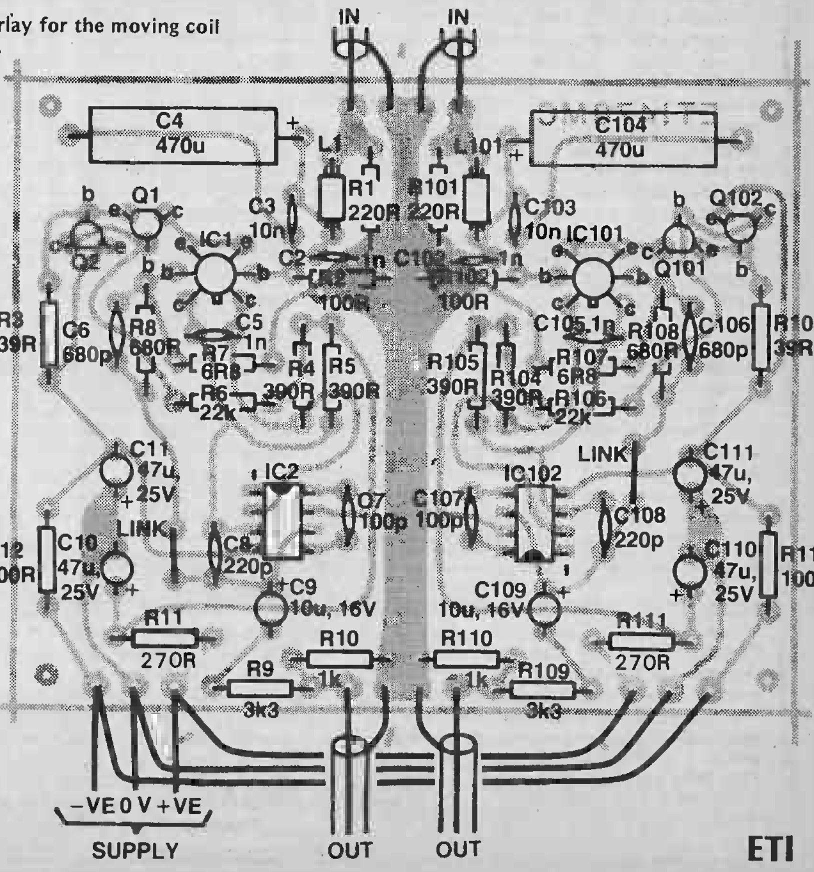
to ensure that hum present on the earth line, due to supply bypass capacitors for example, cannot modulate the signal earth, producing hum in the output. The 0 V line on the boards is, in fact, a separate supply bypass earth line and is not equivalent to the signal earth. A separate wire should be run from the centre point (0 V point) of the power supply used to signal earth at the input sockets.

Both boards should be mounted in a *steel* box which can be mounted as a unit inside the main preamp chassis. This greatly improves the rejection to 50 Hz magnetic fields generated by nearby power transformers or 240 V cables.

Powering Up

No setting-up procedure is required for either stage, but make a final check of all components before applying power to the unit.

Fig. 5 Overlay for the moving coil input stage.



COMPUTER-CONTROLLED LIVE MUSIC

Computers have been making inroads into the music business for some time now, but Peter Finbarr-Smith shows how even small groups can take the whole concept much further.

Music has always been popularly regarded as a purely aesthetic activity, with the musician condescending to operate the instrument simply because there had to be some way to make his music audible to the lesser creatures who provided his meals! Along with this idea goes the thoughts of some 'purists' that there is no place at all for electronics in 'serious' music. On the practical side, however, it's hard to beat a craftily applied wah-wah pedal as an emotional shifter, just as Segovia (or John Williams) can make strong matrons weep (or so I'm told), by wiggling a finger on a guitar string!

Of course, the answer lies in technique — a method of producing a specific effect in a listener by means of a trick, whether mechanical or electronic. Nevertheless, the 'old school' make a hasty distinction and throw their hands up in horror when a musician 'wiggles his finger' electronically! But whether the old school like it or not, they cannot halt evolution, in music or in anything else, and with the microprocessor very firmly with us, evolution will accelerate.

Whoever it was who said that the microcomputer would affect all of society was guilty of understatement. With some of the 'big name' groups turning to total computer-control of live shows it would seem that, in the near future, if you haven't got a

PHOTO BY KATHRYN MACFARLAND

microprocessor inside your guitar then you might as well not bother with strings either! In fact, while it may not quite come to that in the immediate future, computer-control (of the equipment, not the music) is neither particularly difficult nor, these days, incredibly costly. Anyone owning one of the many brands of 'personal' computers can start to think seriously of some of the possibilities of equipment control, plus, perhaps, computer generated sound effects.

This article is intended to show the scope of computer-control when applied to amplifiers and auxiliary equipment such as mixers, echo-units and (for the really advanced!) synthesisers and to show how it is done. At least, some thinking about new possibilities in music should be sparked-off.

Programmed Pots

To begin with, a computer can control volume, treble and bass, plus any other controls, on any or all amplifiers, plus parameters on other equipment used in a live performance or in a recording studio, with an accuracy which is unrepeatable by manual means. The computer can be small and easily carried, with a minimum of setting-up. In fact, once it has been programmed, setting-up is very much faster than before; probably the only adjustment necessary will be for the public address level, which is likely to be affected by the acoustics of the venue.

Although rehearsals, often conducted at a leisurely pace, can allow careful adjustment of equipment for special effects during a number on stage, there is usually very little time to make careful adjustments — especially when there are full mixers and synthesisers to take care of. The actual results are often a compromise, leaving the musicians disappointed, and the roadies (yet again) accused of sabotaging a carefully rehearsed effect. A computer, however, can re-set the whole lot quicker than you can say 'you're sacked', and can give results identical to those obtained at rehearsals. Apart from the obvious benefits of this arrangement, it is unbeatable for live recording as well saving a lot of time 'on the night'. Lights can be controlled, not only for on-off, but also for the exact level of brightness required for a difficult colour mix. Of course, once it's in the computer's memory, it never changes — although it can be changed very easily when necessary.

Data Chains

Here's how it's done. For convenience, we shall consider a set-up consisting only of amplifiers. The explanation also applies to the use of other equipment. The computer to be used for the system is set to the mode which would normally pass information to a cassette for storage, or to a printer. But now these items are replaced by the amplifiers. The computer doesn't know the difference (not so smart, eh?), and so is quite willing to send data (information) this way, even if it seems to go nowhere at all! Figure 1 shows how the system is connected, both for a small to medium set-up (a) and for a large system, which might also include such items as recording equipment and lights (b).

A cable is connected at one end to the computer's tape or printer socket, its other end going to a special socket on the back of the first amplifier in the 'chain'. Inside the amplifier the data goes to a decoder/command circuit. The data is also 'beefed-up' a bit and sent on its way out of a second socket via a similar cable to the next amplifier in the chain. In this way the data is sent to each of the amplifiers, and ends up in the last unit in the chain. Other units may be added at any time by simply joining them onto the chain.

For larger systems, another gadget is added into the chain. This is called a 'digital comparator' and both the computer output and the last amplifier in the chain connect to it. This device now issues the data to the units, and there are no prizes for guessing what it does! It takes a sample of the data leaving the computer and compares it with the data coming back from the last unit. If there is any difference between the two (usually caused by a broken cable or a disconnected plug) then a red light illuminates on the comparator's front panel. Computers using the printer socket for data output will halt the program until the fault has been fixed. The comparator usually gives its warning during set-up, with the lamp lit until the whole system has been connected, as faults of this nature rarely happen during a performance.

It will be seen from Fig. 1 that there is a second cable: the clock line. The cable comprises two wires, data and clock, inside a screen (rather like a thin guitar lead) and the screen keeps interference out, as well as preventing the fast computer signals from radiating electrical noise. The clock line is used to synchronise the data signals, and will be described later in the article.

Inside Story

Figure 2 shows a block diagram of a computer-controlled amplifier. It will be noted, after a quick glance, that the main amplifier is not actually connected in any way to the special electronics. The keen do-it-yourselfer will immediately realise that if he were to obtain the decoder/command board, then a slave amplifier would complete the unit! Quite correct. It should not be long before such boards are available, and normal amplifiers could then also be modified, possibly with a switch marked 'CC' (Computer Control) and 'N' (Normal), fitted to the front panel of the amplifier. However, a commercial product would either present the decoder/command board in a 'black box' for connection to a slave amplifier, or would consist of a unit combining both the board and an amplifier.

Getting back to what happens inside, it will be seen in Fig. 2 that the clock and data lines enter into and exit from the data receiver. In fact only the data enters the receiver, the clock being used to merely operate the input gate, ensuring that each bit of the data word enters the receiver at the correct time. A bit is either a '1' or a '0', and (usually) eight of these form a data word. More about that later!

As the amplifiers are in a chain, with each passing data to the next, then there has to be some means by which each unit 'knows' which commands are intended for it. The decoder takes care of this by looking at the first word in a group of words sent from the computer. If the correct address (the unit's personal name) is represented by the word, then that unit will accept the commands which follow immediately. As the data flows around the amplifier chain at the speed of light, it can be said that all units receive the data at the same time. The actual delay during the 'passing on' process occupies only millionths of a second.

Having accepted the first data word as the correct address, the receiver then passes each of the following words to the registers associated with each control on the amplifier. The registers are small memories capable of holding a single word (whereas the main computer memory can hold thousands of

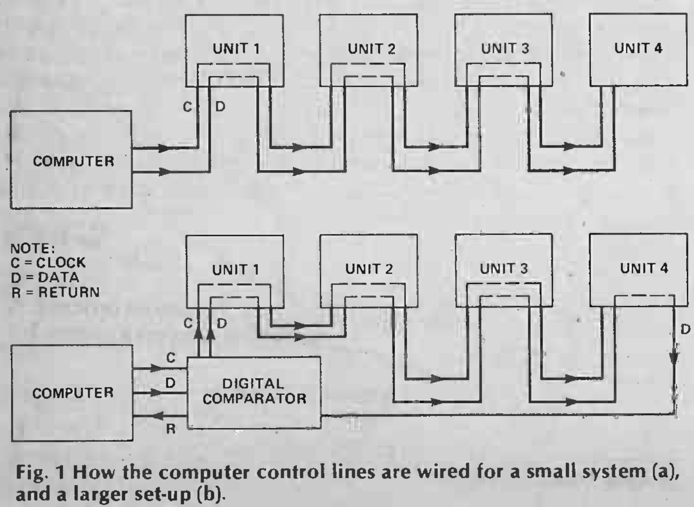


Fig. 1 How the computer control lines are wired for a small system (a), and a larger set-up (b).

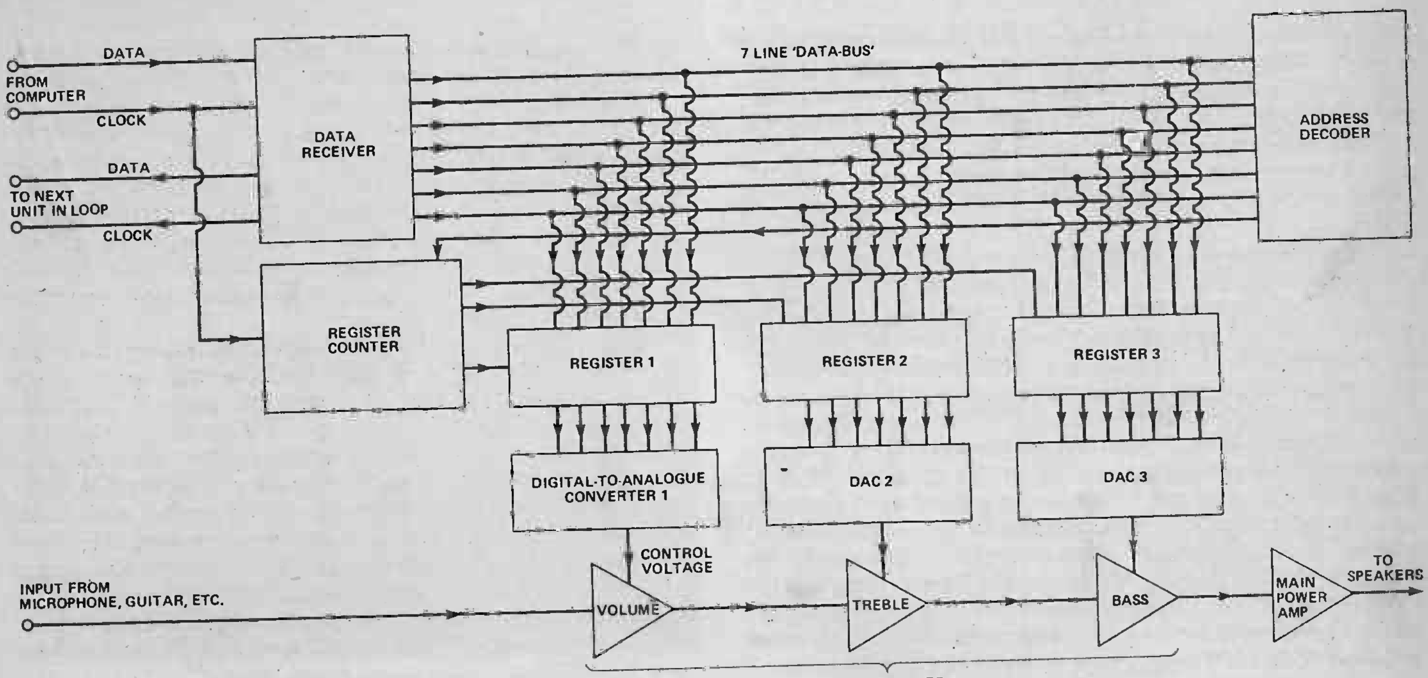


Fig. 2 (Above) Block diagram of a computer-controlled amplifier.

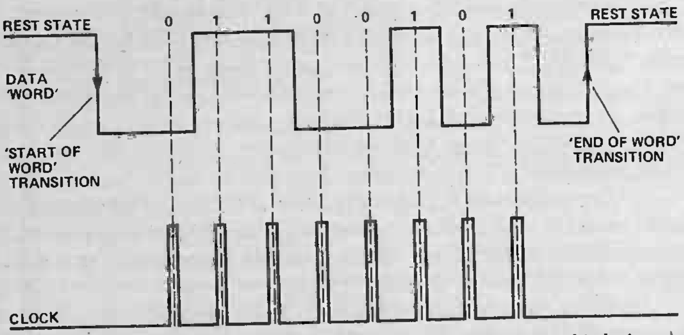


Fig. 3 Typical data and clock signals when an eight bit word is being transmitted by the system. The data is clocked in to the registers by the pulses in the clock signal, shown here as a separate waveform. Some systems may combine the clock and data into one signal.

words). Each word stored in the registers — consisting of eight bits, either 1s or 0s — is then converted into a voltage. The voltage is then applied to the appropriate part of the preamplifier, Volume, Treble or Bass, and these 'controls' are set until the computer sends the next batch of data. This usually happens at the end of a 'number', unless the controls need to be altered during the number for special effects. Some of these effects can be really startling, as when the computer has control of both tone and volume, an instrument can be made to virtually 'talk'! In other situations, the controls on a particular amplifier may not need alteration for extended periods. One obvious occasion when alteration of volume is needed is when a guitar changes from rhythm to lead, for example, when it may be necessary to go into the melody fast, without wasting time twiddling knobs!

Hi, Lo, Hi, Hi, Lo. . . . .

Having looked briefly at what goes on inside the amplifier, we shall now take a look at the clock and data signals. These are shown in Fig. 3. When the signal is high, it is said to be a 1, when it is low it is said to be a 0. The data line is normally high when in the rest state, to minimise the risk of interference producing false data. The level then changes to low before the word begins, then follows a series of 1s and 0s (usually eight of them) before the data line returns high, signifiying the end of the word. The highs and lows of the word tend to flow together if there are more than one of either type together, so the clock is used to 'chop' them into something recognisable by the decoder. In this way, the data is said to be 'clocked in'. In Fig. 3 the clock is separate from the data for this purpose, but in some systems the clock is 'buried' in the data, to be separated in the unit being controlled. Other systems use the change from the rest state into the active state as a signal to the unit to start its own clock which then clocks the data into the receiver. The choice of method seems to depend on the manufacturer's individual preference.

So now that we all know how it works, let's take a look at the programming, without which the system is useless.

Programming

This job logically falls to a roadie. There is much said about roadies, but with the musicians all busy, who else is there? Joking aside, this isn't such a bad thing, as many of the 'gear humpers' are of exactly the right temperament to do an efficient job on the programming. It goes like this: the equipment having been set up for practice, the various controls need to be set. As the controls are now electronic (no, they're not pots driven by motors!), they have to be set up from the computer keyboard. First the address of the unit in question is given by the programmer, and then the various controls are set, one by one. During this process, the computer is programming itself in answer to the questions and, if the set-up is satisfactory, then this represents the repertoire-entry for that number. Alterations can be carried out by simply recalling a unit's address, and re-entering the set-up information. In a number of cases the levels used for practice are different from those used on a gig, so the programming may actually be done in silence, based on live experience. Normally a 'frame' program will be used, which has been prepared in advance, and which asks the appropriate questions, thus considerably speeding up the whole process. A typical example of questions (and answers) seen on a computer monitor is shown in Fig. 4.

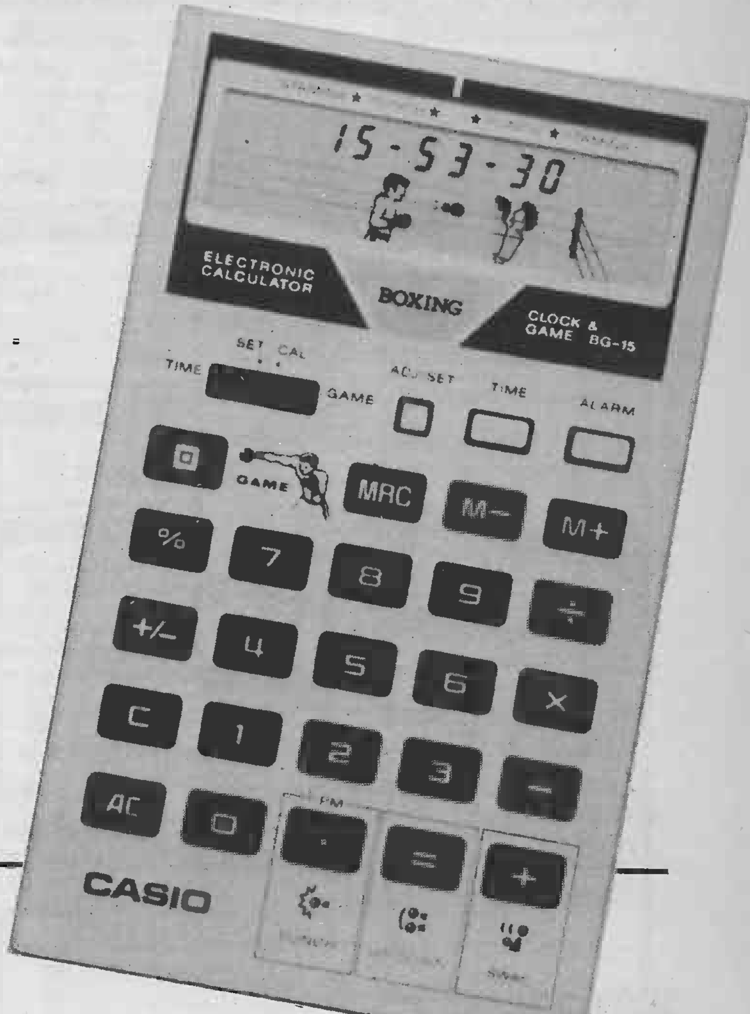
In this example, the system is capable of controlling 26 units (amplifiers, mixers, lights, etc), and these are given alphabetic addresses for convenience. In fact, the number of unique combinations in an eight-bit data word is 256, so there is

DIGEST

It's All A Game

If you have an uncontrollable urge to play electronic games, then Casio can furnish you with a cure — at least temporarily! They have added to their range of LCD watches and calculators two new models which incorporate games. The first of these is the GM10 watch; it is not only elegant with its anodised facia and off-the-shoulder black resin case with matching strap, but is also endowed with day/date facility, daily and hourly alarm function, chronograph and, of course, the game! This is of the space genre, the object being to intercept the UFO's laser beam with your three interceptors. Bonus points are awarded when numbers one to four are displayed in the hit zones. When three UFOs have been shot down without losing your three lives, you progress to the next round. There are 10 rounds in all, each becoming progressively faster. A 30-page instruction book is included, but only 12 of these will interest you, as they're the ones in English!

Dave 'boy' Green has nothing on this next offering. Casio's bid for the world title is the BG15, a four function, eight digit calculator with a constant time display, alarm and . . . the boxing game. This game has the right ingredients to keep you amused, incorporating advanced graphics which reveal the bulging muscles of the opponents and quite realistic knock-out scenarios. With this you can take out your aggression on your LCD partner without losing either your cool or your teeth! Blows can be delivered to the head or body and can be countered or blocked. You can also sway to avoid your attacker's punch and, in the case of a draw, the match will be awarded to you. The power of the punches depends on your boxer's weight class and stamina. Bouts can be won on a knock-out or on a points basis. There are up to 100 challengers of different weight classes from flyweight to heavyweight and the more you win, the higher the speed and technique the opponent uses. At the end of the bout the game will start over again — that's if you're still standing. The price for the GM10 is £22.95 and for the BG15, £18.95. Both are available from Tempus who can be found at The Beaumont Suite, 164-167 East Road, Cambridge CB1 1DB.



It Does Compute!

With the new Sabtronics model 2020 multimeter your computer can fly — well, almost! The combination of multimeter with a microprocessor interface provides a cost-effective solution to all your data aquisition problems as it is adaptable to home and personal computers. Just a few of the more impressive applications are:- monitoring and controlling temperature-related functions by using temperature transducer ICs. For use in water heating, greenhouse control and solar projects; testing the effects of stress using strain gauges; monitoring and controlling pressures of gases or liquids by using pressure transducers. Applications also include process control, lab analysis and school projects; as a lie detector by interpreting changes in skin resistance; plotting the effects of power supply drift versus time, temperature or input voltage; catching voltage or current problems by automatically monitoring the circuit and having the computer trigger an alarm when any adverse condition arises; testing components to certain tolerances for inspection or to generate parameter distribution; programming your plant-watering by measuring soil conductivity. Although these are merely suggestions, software drivers are provided for each computer system Sabtonics supports, obtainable by writing an application program.

The model 2020DMM boasts a 0.1% basic DCV accuracy, with 3½ digit large LED display. In addition, the new meter is capable of directly measuring AC and DC volts up to 1000 V, ohms to 20 M and AC and DC current to 10 A. The DMM is equipped with all cabling and I/O support necessary for the TRS-80, Apple, PET or Atari computers, with other models becoming available soon. The cost is £165 plus VAT. Further details from Black Star Ltd, 9a Crown Street, St. Ives, Huntingdon, Cambs PE17 4EB.

no reason why a system should not have numbered units — up to 256, if necessary! Ideas for multi-group outdoor concerts arise here! The numbers following the questions relate to the settings of the controls. Each control is capable of 256 positions (corresponding to a knob with 256 marked positions, if it were not electronic). It is obvious that the precision and degree of control setting is far superior to the best efforts of us mere mortals!

From a programming point of view, a mixer is considered to be a number of amplifiers, one for each channel, and something like an echo unit has its own questions on the screen, relating to individual controls such as reverb levels for example. The same applies to the special controls on a mixer (foldback, etc). The reader may now be wondering how the computer knows what controls are available on what equipment. It knows because the frame program has been specially written to suit the group's equipment. Software (programs) has to be prepared for any serious use of a computer, but given the ground rules, the software is usually produced by a member of the group, unless a 'package' has been bought from a company specialising in machine-control (as this branch of computing is called). This is, of course, expensive and while a dozen or so of the top name groups might invest to this extent, there will be hundreds who are prepared to try it themselves, with varying degrees of success.

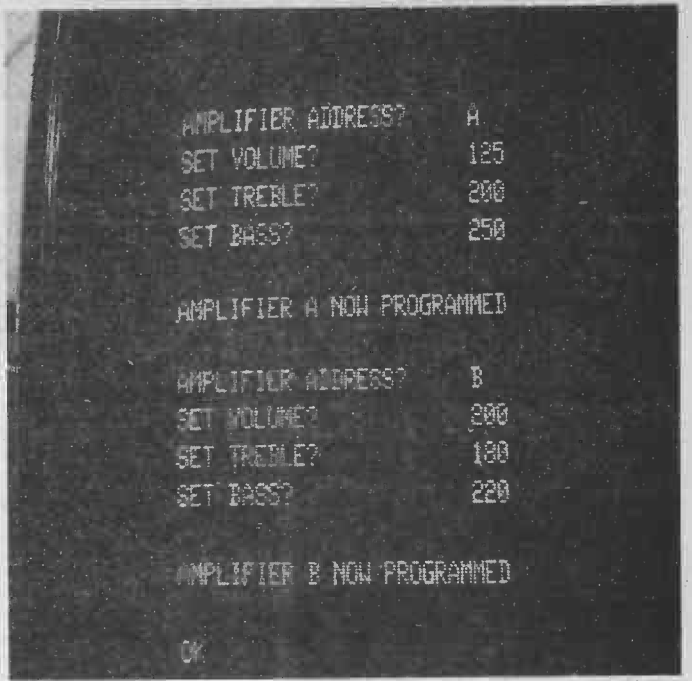


Fig. 4 Typical screen readout during programming.

Minding The Store

At rehearsal, as the gear is set up, the computer is busily programming itself. Therefore, when the group are satisfied with the performance of the equipment for that number, it is held permanently within the computer's memory, along with its title. As different numbers are practised, so a repertoire is built up. In fact this process, coupled with false starts and other problems, may take quite a long time to complete; but as with all such things, when the computer is switched off it promptly forgets everything — losing the lot! So there has to be some way of storing the information for later use.

On the smallest systems this can be done on cassette. For computers with only a small memory, a full repertoire may have to be stored in more than one section, each section being loaded during a gig — possibly in the interval, for the second section. Alternatively, the particular show may have to be assembled in advance, with just the required numbers being stored in the computer before the show starts. Each new set-up for all or just some of the equipment is sent out by pressing two keys on the computer keyboard, either between numbers, or during them, for special effects.

For larger systems, especially those being used for live recordings, a floppy disc unit is necessary, rather than a cassette recorder, as the great storage capacity and high speed of a disc unit means that a large repertoire may be stored on one disc and any part of it called up almost instantaneously.

Before the group begin to play the next number, the operator simply calls the list of titles on to the screen, and a question at the bottom of the list asks: WHICH NUMBER DO YOU WANT? In answer to this, the choice is keyed-in, and the system is immediately programmed for that item. The number chosen may require a change to some (or all) of the gear during the actual performance. The operator simply presses two keys, and the re-programming is done instantly while the group are still playing. Too simple? Well, he's got to press the keys at the right time, naturally!

Getting Started

At the present time there is very little commercial equipment available for the computerisation of sound equipment. There is a reasonable amount of studio gear around, but it is only applicable to live recordings, and does not help the group. There is, however, some equipment in the pipeline, and we are told that this will take three distinct forms:

- The 'bare board' type of computer-controllable preamplifier, which can drive slave amplifiers directly or can be fitted inside a normal amplifier.
- The above preamp inside a case, complete with power supplies and ready for use with slave amplifiers.
- The complete computerised amplifier in several output wattages: an almost featureless black box with no controls at all, except a power-on lamp and an instrument input socket in the front, with the two computer sockets on the back.

Other expected equipment may take longer to appear, such as echo units, effects generators, lighting control units, mixers, and a synthesiser with no controls. Needless to say, the latter item will not be cheap!

So do we sit back and wait? Well, if the preamplifier boards are going to be available reasonably soon, then we still need a computer, and some time to learn to program it, so perhaps that should be the first step. Computers such as PET, Apple, TRS-80 and so on, are reasonably priced and are potentially suitable, but even if there is a printer socket (there will certainly be a cassette socket) available, we still need to add a clock line. This is achieved by finding the point inside the computer where the system clock enters the printer or cassette circuitry, and using gates and a counter (three integrated circuits needed), the correct number of clock pulses can be sent along with the printer data (from a spare pin on the printer socket).

If the chosen computer has only a cassette socket, then the modification is still not too difficult. Having found the clock, we must now find the data where it enters the cassette circuit. This is brought out in the same way as the clock, or, ideally, to a new socket fitted for the purpose, to the computer. And we're ready for business! The foregoing is not as difficult as it may sound, but it certainly takes guts to stand by and watch a roadie disembowel a new microcomputer! For those who can't face that, take heart. There is at least one computer specially designed for the job already available, ready to plug in and run.

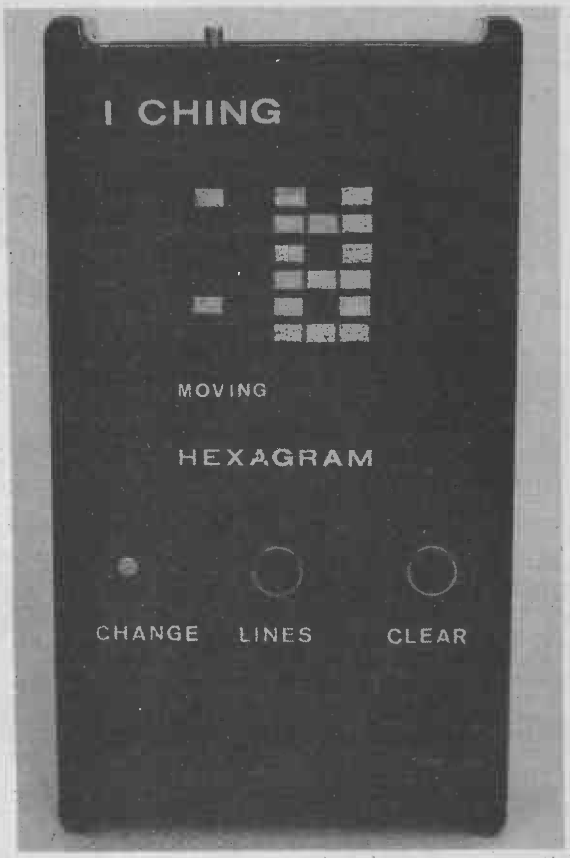
Conclusion

In the musical field as much as in any other, the microcomputer will make a powerful contribution toward evolution. Its applications are limited only by the imagination of the user. As the machines become lower in price (because everyone in Britain except you has already got three of the things), we should be seeing simple guitar tuning aids and so on, being run by computers almost powerful enough to take man to the moon, and with sufficient memory to catalogue the British Museum. **ETI**

I CHING COMPUTER

The I Ching, or Yi King (which means 'Book of Changes'), is an ancient Chinese text, one of the five classics of Confucianism. The book has been used for over 2000 years as a means of divination — an oracle for predicting events. The uniqueness of the I Ching lies in its presentation of 64 symbolic hexagrams which, if properly understood and interpreted, are said to contain profound meanings applicable to daily life. Throughout the ages, I Ching enthusiasts have claimed that the book is a means of understanding and even controlling future events.

The hexagrams are built up out of six separate lines, one above the other. Each line can take one of two fundamental forms: a full line, 'yang' (the male cosmic principle) or a broken line, 'yin' (the female cosmic principle); further, each type of line can be either a moving or a still line. Each hexagram will thus consist of a pair of the eight basic trigrams (pa kua) shown in Table 1.



Page The Oracle

The oracle is consulted by creating a hexagram while reflecting on the question that is being asked. The hexagram has traditionally been created by casting 50 yarrow stalks in a special procedure which provides a certain probability structure for turning up different types of hexagram. A more recent method is based on tossing coins, but this is considered inferior and does not provide the true probabilities.

Due to the mathematical and essentially binary nature of the I Ching, the medium of digital electronics is ideal for generating the random hexagram patterns with the authentic probability structure of the yarrow stalks. Also, by using solid state indicators, a visual display taking the form of the original Chinese hexagram can be produced.

Each line of the hexagram can be in one of four states (as described above); a moving (old) yin, a moving (old) yang, a young yang or a young yin. In terms of their probabilities, all six lines can be considered as totally independent of each other.

A moving yin has a probability of occurrence of 1/16, a moving yang 3/16, a young yang 5/16, and a young yin 7/16. Adding these probabilities in different ways we find a probability of ½ that a line will be yang (ie there is a 50-50 chance between yin and yang), and a probability of a ¼ that any line will be moving.

On the ETI I Ching generator the hexagram is displayed on an array of red rectangular LEDs; an additional column of green LEDs indicates any moving lines that are present. There is a push-button which must be pressed six times to create the hexagram; each press randomly throws up one line, with the probabilities described above being derived from a fast binary counter and logic decoding gates.

The lines remain invisible until the last one is complete; the display then illuminates the entire hexagram pattern. A 'clear' button is provided for removing the hexagram, enabling

TABLE 1

The Eight Trigrams with their commonly accepted equivalents

Chinese name	Chinese meaning	Natural element	Corresponding direction	Moral or Mental quality
Ch'ien	heaven	heaven	NW	strength
K'un	earth	earth	NE	weakness
chen	activity	thunder	E	being active
sun	bending	wind	SE	flexibility
K'an	pit	water	N	being in danger
li	brightness	fire	S	elegance
kên	to stop	mountain	SW	firmness
Tui	pleasure	collection of water	W	joyfulness

further hexagrams to be created, but it is not considered advisable to cast doubt on the oracle's answers by requestioning. A frivolous attitude to the I Ching will result in meaningless answers to your questions. The hexagram that is cast can be found in the Book of Changes, and the text that accompanies it should be carefully studied before interpretation. There are further descriptions for each line that may be moving; these should be studied also. Finally, if moving lines are present, the 'change' switch should be operated and any lines that are moving will change into their opposite, thus forming a new hexagram. This hexagram should also be looked up to complete the prediction.

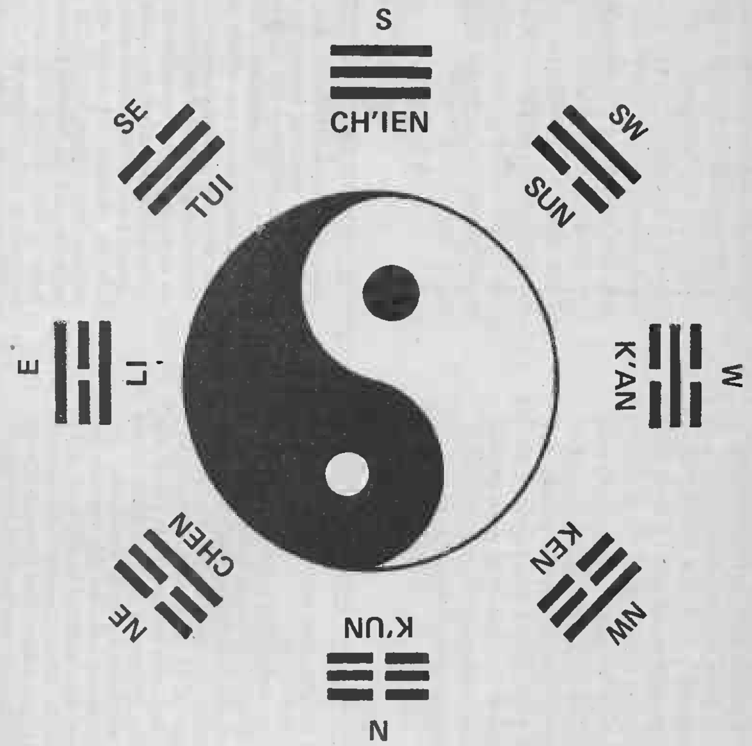


Fig. 1 The arrangement of the eight trigrams according to the legendary emperor Fu Hsi. The diagram is not upside down; south is traditionally shown at the top. Table 1 (opposite page) the commonly accepted equivalents and cardinal points according to King Wen.

Construction

By using two separate PCBs the machine can be made quite compact, most of the room then being taken up by the PP6 battery. A smaller battery cannot be used as the current consumption with all the LEDs on is about 40 mA. As these PCBs are not double-sided there are quite a few links which must be soldered in first: 11 on the logic board and six on the display board.

When soldering in the LEDs on the display board take a very careful look at the internal photo and the overlay diagram. This display produces the actual form of the hexagram; the LEDs should all rise to exactly the same height (1.5 cm from the board surface to the top of the LED). Observe also their polarity; all the cathodes should be on the right.

The rest of the display board is straightforward, but it is worthwhile double-checking the transistor pinouts before soldering them all in circuit (perhaps you possess an ETI component tester?). A length of ribbon cable can then be wired to the nine lead-out points as marked on the overlay.

When assembling the logic board don't forget to use IC sockets; there is adequate room on the board for these. One problem area may be the zener diode, ZD1; the holes for this diode are very close to the socket for IC7. On our board the end of the socket was filed

Make sure that both the socket and the diode will fit before soldering them in.

Twisted pairs of wires for connecting the switches should be soldered as indicated on the overlay diagram. The switches PB1, PB2 and SW1 should be temporarily connected up for testing purposes when assembly is complete. The leads of the ribbon cable from the display board should now be wired to the corresponding points marked on the logic PCB.

Test Patterns

The circuit is now ready for testing, and this is easier if it is done before mounting the boards in the case.

Connect a 9 V battery (PP6) to the supply leads and operate the 'clear' switch PB2. The display should be completely blank. Now press the 'lines' switch PB1 six times; on the sixth press the display should illuminate in a random pattern. The two outside columns of red LEDs should be fully illuminated, and the centre column will consist of any combination of on or off LEDs. Some of the green LEDs might be on; if so, then operating the 'change' switch will change the state of the centre red LED in the corresponding row (from on to off or vice versa). Pressing the clear switch again should blank the display ready for another pattern.

When testing the prototype machine it was discovered that the switching threshold of the Schmitt trigger gates (IC1) can vary considerably from one pack to another. This affects the frequency of the main clock built round IC1d, and the frequency of this clock will affect the brightness of the outside columns of red LEDs. Our clock had a frequency of 6 kHz with the first chip we tried, giving a well-lit display, but only 800 Hz with a different chip, causing rather dim LEDs. If the brightness of the outside LEDs does not match the centre column LEDs, then alter the value of C6 until they are of uniform brightness.

If all is well the circuit boards can now be assembled into the case. We used the West Hyde model given in the Parts List. This case has just enough room for a PP6 battery mounted at one end between the stand-off pillars (see the internal photo). A cutout must be made in the lid of the hexagram display, and a piece of polarizing filter sheet can be glued behind to sharpen the contrast.

Our case came with an internal battery compartment (it isn't required, though) and the display board was mounted on top of this with sticky pads, to provide the correct height. If there is no battery compartment, then insulated mounting pillars can be used.

There is plenty of room to mount the switches — we mounted the on-off switch at one end and the three control switches on the front lid.

Tao And Change

The electronic details of how this project works are boxed off as usual, but the philosophy behind the I Ching is as follows: all parts of the universe are harmonized in a rhythmic vibration where nothing is still; all beings undergo cyclic mutations and transformations. This is the Chinese view of creation. The static ideal gives way to a philosophy of continuous regeneration. Change itself is systematized and made intelligible in the 64 hexagrams of the I Ching, which are recurring patterns in the general flux. The 64 hexagrams symbolize all the possible situations and processes that can exist when yin and yang rise and fall in cycles. The Chinese call this unfathomable operation sken ('spirit').

The permanent Tao, an unchanging unity, underlies the manifold nature of existence, and by it imperceptible action it moulds the matrix of the universe from primordial chaos. The interaction of the complementary energies yin and yang are the source of the eternal transformations of the Universe and their alternating patterns unfold all the natural events of the world, such as night and day or winter and summer, manifesting the external aspect of the same Tao.

HOW IT WORKS

The overall circuit operates in two distinct modes. Initially, the user presses the 'lines' button PB1 six times and during this time two random binary bits are written into six sequential address spaces of a memory. On the sixth press, after recording the last two bits, the circuit switches to its other mode of operation. Here, a binary counter is clocked continuously and its outputs are used to scan the addresses to the memory. The previously recorded data is read out to a multiplexed LED display scanned by these same address lines. Essentially the two modes are; build up the hexagram pattern, then display it.

The line labelled 'control line' on the circuit diagram is used to switch the operating mode. This line is taken from the output of IC3a, a NAND gate whose inputs are wired to the B and C outputs of the slow binary counter IC2a. Thus the line is normally logic high and will go low when IC2a reaches a count of six. This control line is used to switch the address lines of the memory (IC7) and display decoder (IC6) from the slow counter IC2a to the fast counter IC2b. This is achieved with IC4 (a triple one-of-two CMOS switch) whose switch-select lines on pins 9, 10 and 11 are wired to the control line.

When power is first applied to the circuit, the slow counter IC2a is reset to all zeroes by the C4/R4 network. PB1 is the 'lines input' button. When pressed, it takes the input of IC1a (pin 12) high; this input is normally held low by R1, with C1 providing switch debouncing. The pin 13 input of IC1a will be held high by the control line; thus the output on pin 11 will go low while the input switch is pressed. This signal must initiate the following actions: first the random bit generator must be stopped, allowing the data to become stable, then this data must be written to the memory, and finally IC2a must be clocked to its next count position.

The random bit generator is implemented by driving the binary counter IC2b from a 6 kHz clock and decoding the four bit output with the logic gates IC3b, IC3c and IC5a. This provides two bits with the correct interdependent probabilities of occurrence; the truth table for these gates (Table 2) shows how these probabilities are derived. For example, when the counter is stopped, the D output has a 50-50 chance of being either high or low; this is the yin-yang indicator. The output of IC3c on pin 11 determines whether the line is moving or not — a logic low signifies a moving line. The counter is stopped by disabling the clock oscillator built around Schmitt trigger IC1d, C5 and R5. The pin 9 input to IC1d is wired to the push-button signal on pin 11 of IC1a. The oscillator is gated on when pin 11 is high and disabled when low (ie when PB1 is pressed).

The two random bits which are selected when the clock is stopped are fed directly to two data inputs of the memory IC7 (a CMOS 5101). The data will be stored when the write line on pin 20 is taken briefly low. The write line is wired to the output of IC1c where a 10 uS negative-going pulse arrives 10 uS after the clock has been stopped.

This pulse is again derived from the logic low push-button signal; the delay is provided by R2, C2, and Schmitt inverter IC1b, and the short negative-going pulse by C3, R3, and inverter IC1c. After the data has been stored, the positive-going edge of this same pulse is used to clock the slow counter IC2a which sets the next address for writing to memory.

This circuit action takes place on each operation of the switch, the memory location being incremented when the button is depressed and the free running clock generating random bits when it is released. On the sixth press of the button, when the last line of the hexagram is written to memory, IC2a then clocks to count 6, thus switching the control line from high to low.

The control line puts a low on the pin 13 input of IC1a, so inhibiting further operation of PB1, and allowing the fast clock to oscillate continuously. The control line also enables NAND gate IC3d via inverter IC5b; and by placing a logic low on the inhibit pin of IC6, the display decoder is also enabled. These controls allow the display to illuminate.

The address lines for the memory and display are now switched to the A, B and C outputs of the binary counter IC2b. For each three bit binary address that is routed through switch IC4 from the counter IC2b, one sequential output of the decoder IC6 is selected in conjunction with the corresponding memory word. (The 5101 chip is a 256 word by four bit memory, but in our application only six words of two bits are used; it's still the cheapest and simplest method though!)

The 1-of-8 line analogue decoder IC6 connects the base of the corresponding PNP transistor (Q1-6) via R8 to ground; this switches on the transistor which in turn takes the anode of the associated red and green LEDs to the positive supply rail.

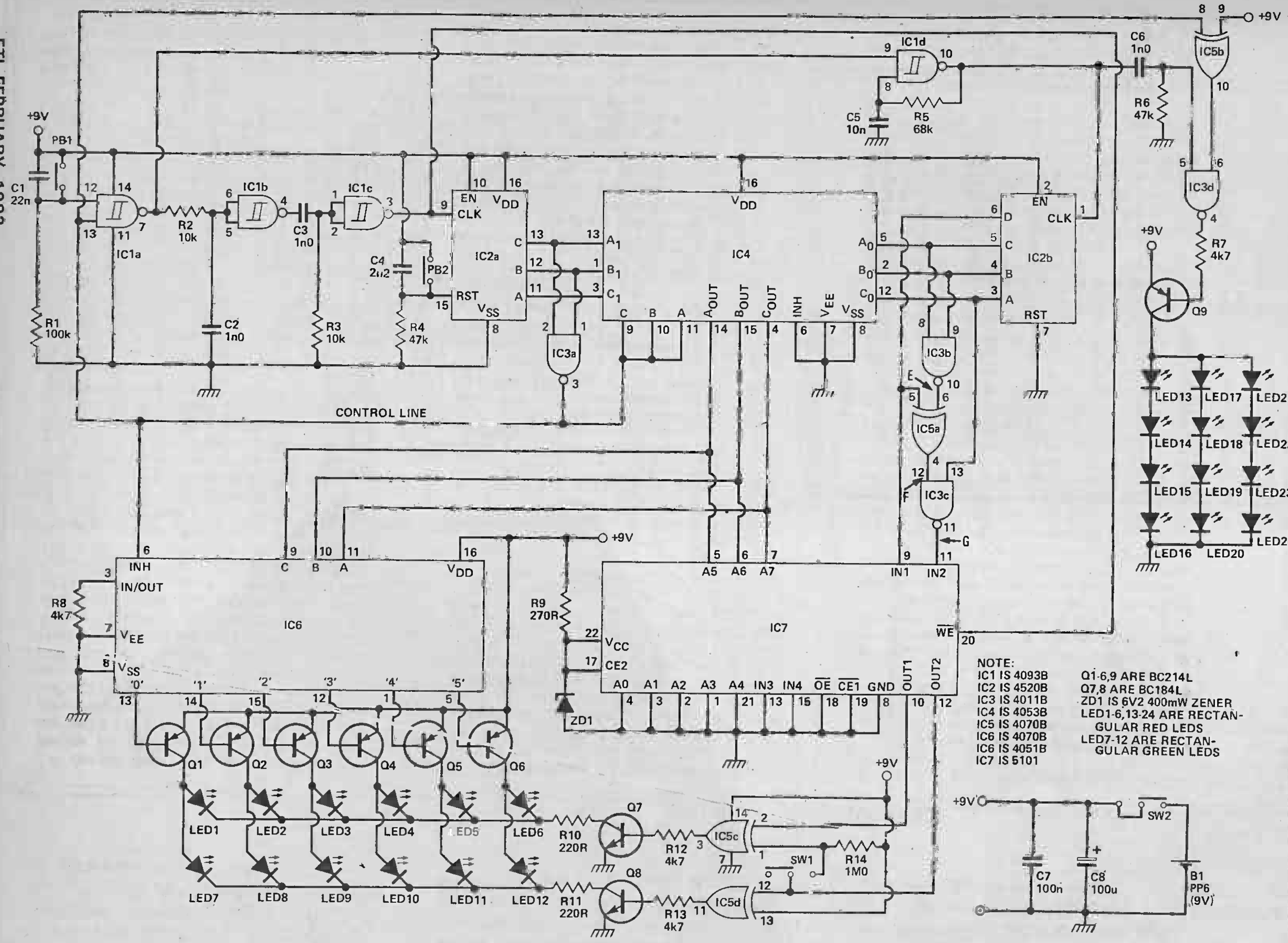
The two bit memory word selected by the address lines A5, 6 and 7 of IC7 is available on the data output pins 10 and 12.

The moving line information from pin 12 is fed via inverter IC5d and R13 to the base of NPN transistor Q8. The collector of Q8 is connected via R11 to all the cathodes of the green LEDs (LEDs 7-12). Thus when a moving line is present the output of IC5d will be high, turning on Q8 and providing a ground return for the selected LED.

A yang line is represented by a logic low on the data output pin 10; this provides one input to an EXOR gate IC5c. The other input is normally held high via R14, which makes the gate act as an inverter, thus driving Q8 and the red LEDs in the same manner as the moving line indicators. Since the address lines are driven by a continuous binary count, each line of the hexagram is repeatedly displayed in turn, giving the effect of a complete display. The outside columns of red LEDs must also be on continuously to complete the hexagram pattern. They are driven by a pulse waveform derived from the main clock using C6, R6 and IC3d to produce a train of negative-going pulses. These pulses turn on Q9 for a short period in each clock cycle, allowing a current burst to illuminate the series-parallel combination of LEDs (LEDs 13-24). Driving the LEDs in this way reduces the current consumption required for the same brightness.

SW1 is the 'Change' switch; when closed it has the effect of turning any moving lines into their opposite state; yin for yang and vice versa. The 'moving line' data output on pin 12 of IC7 will be connected to one input of EXOR gate IC5c, which then acts as a logic-level-controlled inverter, to produce the desired effect.

To create another hexagram the 'RESET' switch PB2 can be pressed. This resets the counter IC2a back to zero, which in turn switches the control line high again, blanking the display and returning the circuit to its first operating mode. The 'Lines' push-button now becomes operational again, allowing new hexagrams to be determined.



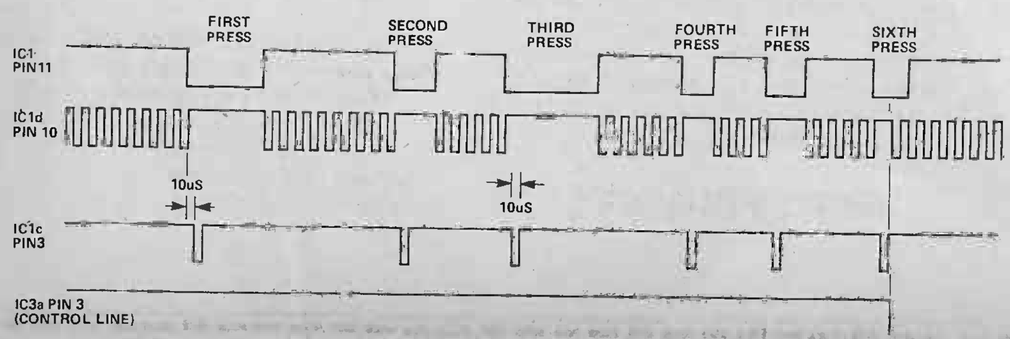


Fig. 2 (Above) Complete circuit diagram of the I Ching oracle.

Fig. 3 (Left) Timing diagram for the logic.

BUYLINES

The only out-of-the ordinary component in this project is the CMOS memory IC, the 5101. This is available from such mail order companies as Watford Electronics, Technomatic and Micro Times, all of whom are advertising in this issue. We can supply the PCBs for £3.98 a pair, as advertised on page 81. The case used was a BOC 708B from West Hyde Developments. Since this has a battery compartment which is unused (except to mount the display board) it might be advisable to by the cheaper BOC 708, without the compartment. English translations of the I Ching are available in both hardback and paperback. For example, we have been using the Richard Wilhelm and C.F. Baynes translation, a hardback published by Routledge and Kegan Paul, and the Mentor paperback editied by Raymond Van Over from James Legge's translation.

TABLE 2

	COUNTER OUTPUTS				GATE CONNECTIONS		
	D	C	B	A	E	F	G
	0	0	0	0	1	1	1
	0	0	0	1	1	1	0
	0	0	1	0	1	1	1
	0	0	1	1	1	1	0
YANG	0	1	0	0	1	1	1
	0	1	0	1	1	1	0
	0	1	1	0	0	0	1
	0	1	1	1	0	0	1
	1	0	0	0	1	0	1
	1	0	0	1	1	0	1
	1	0	1	0	1	0	1
	1	0	1	1	1	0	1
YIN	1	1	0	0	1	0	1
	1	1	0	1	1	0	1
	1	1	1	0	0	1	1
	1	1	1	1	0	1	0

D = 0 indicates yang; D = 1 indicates yin.
G = 0 indicates a moving line.

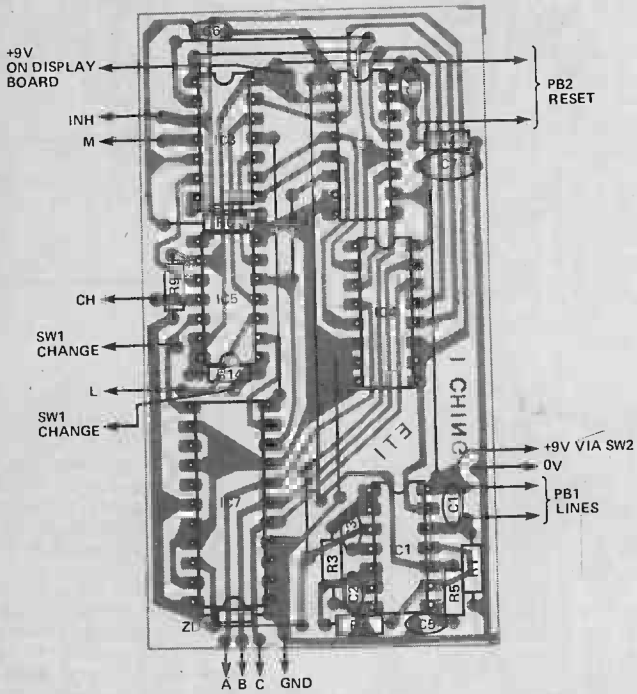


Fig. 4 (Above) Component overlay for the main board.

PARTS LIST

Resistors (all ¼W, 5%)	
R1	100k
R2,3	10k
R4,6	47k
R5	68k
R7,8,12,13	4k7
R9	270R
R10,11	220R
R14	1M0
Capacitors	
C1	22n ceramic
C2,3,6	1n0 ceramic
C4	2n2 ceramic
C5	10n cefamic
C7	100n ceramic
C8	100u 10 V tantalum
Semiconductors	
IC1	4093B
IC2	4520B
IC3	4011B
IC4	4053B
IC5	4070B
IC6	4051B
IC7	5101
Q1-6,9	BC214L
Q7,8	BC184L
ZD1	6V2 400 mW zener
LED1-6, 13-24	rectangular red LED
LED7-12	rectangular green LED
Miscellaneous	
PB1,2	push-to-make switch
SW1,2	miniature slide switch

PCBs (see Buylines); PP6 battery and clip; case ref. BOC 708B (see Buylines); I Ching (Book of Changes), see Buylines.

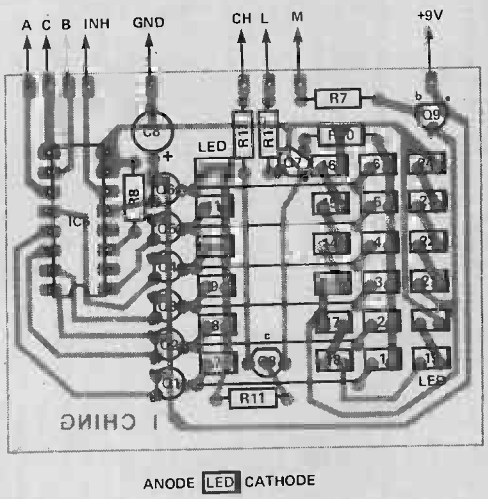


Fig. 5 Component overlay for the display board.

Inside the I Ching computer. Everything fits in neatly if the recommended case, battery and PCBs are used.

ETI

MILITARY ELECTRONICS

As the Soviet military might continues to outstrip that of the NATO forces, Kurt Fleischmann reports on how Western technology can redress the balance.

In the bow and arrow age of warfare, what mattered most was a strong arm, so that the arrow would reach the enemy; good eyesight so that the warrior could see the foe; and a good brain which could quickly discern the adversary's weaknesses and capitalise on them. In a manner of speaking, this is still what really matters today in the jet and nuclear age of warfare — except that man is now aided by sophisticated machines. Take, for example, what Marconi Space and Defence Systems Ltd (known to insiders as MSDS), call the 'Full-Solution'; a fully integrated Improved Fire Control System or IFCS.

This, created by MSDS to the specifications of the British MoD (these acronyms get everywhere), is a fire control system providing the following important qualities:

- A vastly increased hitting capacity on first or second try — at least in comparison to manually-operated Chieftain Battle Tanks.
- An increased firing power by speeding-up the process of engaging the enemy.
- The possibility, where necessary, of reaching further distances.

There is now less need for human judgement and decision-making and a greater likelihood of hitting the target as a result of greater speed and aiming ability, using the same British 120 mm and 105 mm tank guns.

These improvements are not idle boasts. They have been confirmed by experts watching Chieftains in action hitting two rather small targets (1 m x 1.2 m) three times within 53 seconds at ranges of up to 2900 m using only nine rounds.

Autorepair Circuits

When the arrow broke and/or when the primitive warrior ran out of bows and arrows, he had to look around for a suitable, pliable branch to make another bow and seek out and sharpen additional arrows. He was usually unable to finish this task, because his enemy would kill him while he was defenceless. In contrast, fault-finding and battle-time repair are made relatively easy for modern tank-users who also use IFCS. Checks are made by the computer automatically and constantly. In fact, fault-finding and repair is part of the basic set-up by so-called first line test equipment; faults are traced to a specific module, which is immediately replaced within the tank. The second line test equipment goes even further by finding any fault within the module, rectifying it by complete replacement, if necessary.

Unlike its predecessor 'Simple-Solution' which only took into consideration such factors as distance, types of ammunition, so-called trunnion tilt, the effects of cross winds and so on, 'Full-Solution', it is claimed, takes account automatically of *all* aspects of projectile flight. These include full simulation for ballistic computation as well as Autolay, the automatic laying and tracking at the proper elevation offset in relation to particular battle conditions. Other features include digital processors which can be reprogrammed and easy two-handed operation control.

The Think Tank

Not for nothing do Marconi refer to IFCS as 'intelligent'; it is almost able to think for itself. Not only can the system's digital computer (when correctly programmed) decide on the proper ammunition and the correct way of using it, but it is also 'self-correcting' — learning by its mistakes from sensors on the tank.

It does this using four main methods: through the Data-Handling Sub-System, or DHSS, the Tank Laser Sight (TLS), the Sighting Sub-System, the Sensors Sub-System and the Gun Control Equipment, or GCE. The DHSS has five specific purposes — ballistic prediction, lead-angle prediction, axes conversions, system control and data input and output — and is composed of four sections. The first is known as CIU (Computer and Interface Unit); it contains the GEC-Marconi digital computer (its source of power) and the interface module. The interface module's function is to adapt data from other sub-systems to the computer.

Section number two goes by the initials CCMU (Commander's Control and Monitor Unit) and is positioned so that it enables the commander to manually feed data into the computer as well as obtain this, when required, from the computer. Other uses of the CCMU include first line maintenance to diagnose and correct faults, make performance checks and help the gunner during long-range and indirect engagements.

The third and fourth sections, known as the Firing Handles, are used both by the commander and the gunner. Left-hand-operated, they activate the weapon and ammunition-type selectors as well as the laser and autolay demand switches and last, but not least, the firing switches.

Taking Aim

Next we come to the Sighting-Sub-System which has two eye-pieces for the gunner. The left eye-piece shows data displays for the status of laser and IFCS, the armament and ammunition chosen and the range as measured. The right eye-piece shows the Muzzle Bore Sight (MBS) mark. This is initially used as the aiming and lasing mark in the IFCS system; then the computer takes over by electronically creating an elliptical aiming mark in the view field, containing the target as well as the MBS.

This aiming mark is an ellipse just big enough to surround the tank target. It changes in size inversely with the distance. As a result, the visual relationship between the ellipse and the target size ensures the gunner's confidence in his ability to hit any target, irrespective of distance. Moreover, once the

relationship between target and ellipse is established, it is retained with the aid of the computer so that firing is successful; no matter what evasive action the enemy takes or at what speed or direction the tank moves.

A computer, ingenious as it is, is but a machine. So a great deal of information must be fed into it manually (or pre-programmed), such as gun jump or a computation for gun wear; through the Sensors Sub-System vital information for ballistic computation, computer turret control and gun position can be continuously and automatically adjusted for optimum results. All that is necessary is for the gunner and the commander to get the target into the view-finder and use two hand controls, one for activating the firing handle, the other the thumb controllers.

Artillery Aid

Another electronic aid the Army will make use of in any military engagement is known as BATES (Battlefield Artillery Target System). In the past, artillery control was voice-based — orders were transmitted by radio. But this was a time-consuming process, one which had been speeded up when FACE (Field Artillery Computing Equipment) was first used. FACE speeded up ballistic and weather calculations. It had little effect on order-passing. But speed is of the essence because the West in general, and Britain in particular, is hopelessly outnumbered in most conventional weapons (including artillery), by Warsaw Pact nations. So they have to make up in quality what they lack in quantity.

This is recognised by Britain's NATO partners and they are dealing with the problem in their own way. Where the British answer to the problem — BATES — differs, is that it will be a so-called distributed system. Artillery deployment will not be centralised in character, but will be called upon as and where and by whom it is needed; be it company, division, regiment or other level.

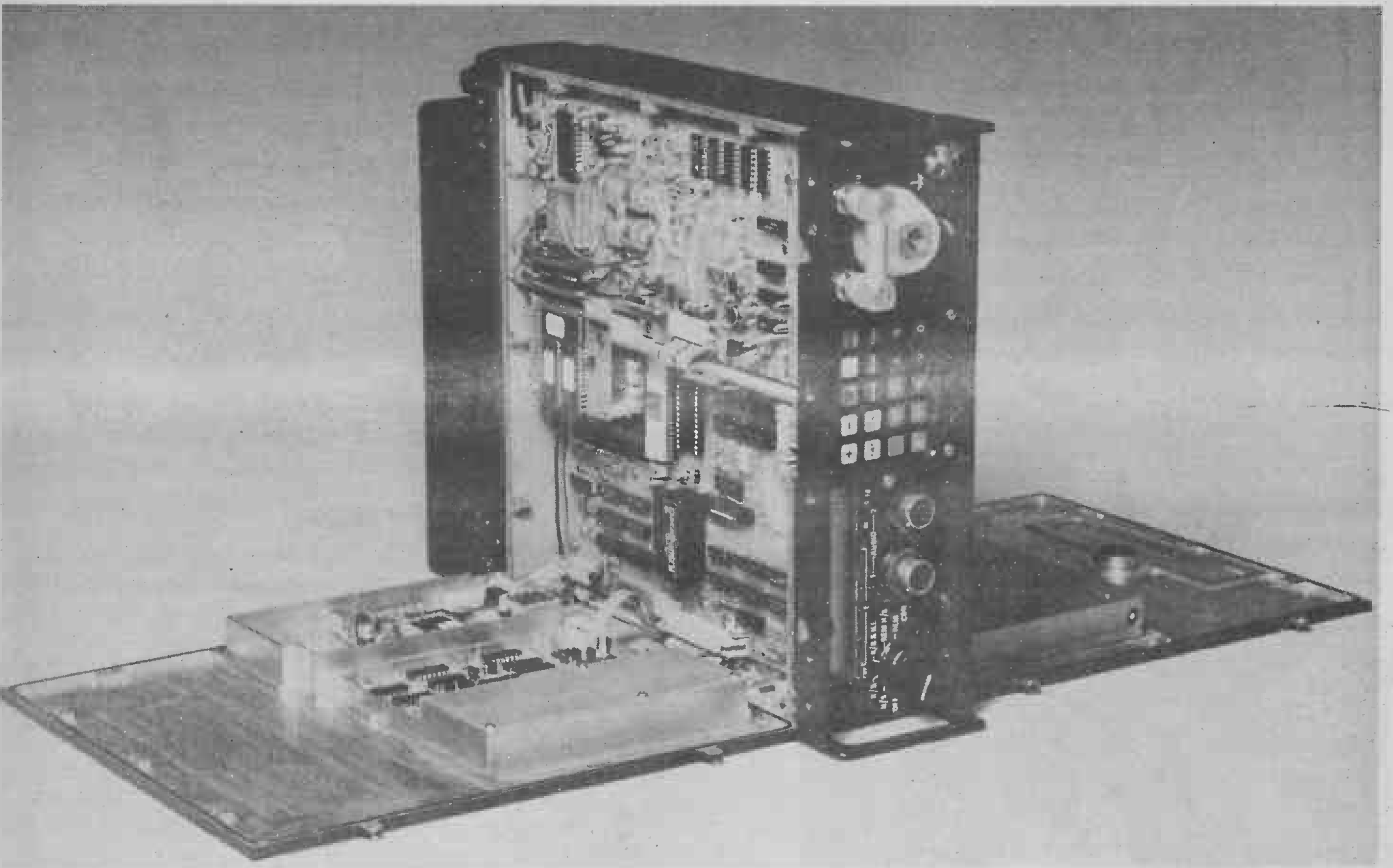
This means BATES can never be used by any unit in its entirety; although some (especially non-British) military men consider this a disadvantage. Moreover, according to British thinking, the greatest disaster is the complete destruction of the entire centralized unit, which can never happen with BATES.

A Tall Order

The BATES designers, Marconi and Scicon, had to overcome mutually antagonistic problems. BATES has to be tough, since wartime conditions are never a picnic. It has to be small enough to be easily transportable in Army trucks and operable in small fixed locations. It also has to be easy to use and repair because its de-centralised nature means that no specialists are likely to be around; it must also be capable of withstanding possible nuclear radiation as well as attempts by the enemy to attack the system electronically.

As with most machines for the military, BATES has one other problem which is of vital importance: that of making it usable by other systems, or in conjunction with them, on an international, NATO (or at least UK/USA) basis. Specifically, it can be used with the American TACFIRE and to some extent with Britain's own tactical computer aid, WAVELL, as well with some other artillery equipment either already in operation or shortly to be made available. Thus it has been decided, as a result of a study group's findings, that BATES should have a number of common modules, together with some special interface units.

The units will receive inputs from instruments measuring muzzle velocity and from weather radars. The former will be in touch with trunk systems, CNR and line systems. Two main types of modules will deal with so-called remote cells — observation posts or groups detached from the main body which need to put in, transmit, and receive information over the CNR or line interface units and units on the battlefield, where the gun



Edge hinges on each side of the case render the Philips Callpac radio exceptionally accessible for servicing and repair. When closed, the unit is nonetheless hermetically sealed and can withstand water immersion to a depth of one metre.

or missile is in actual operation. These units provide the data needed by those manning the weapons, and information from the front which may be needed back at the control post.

Of course, not only these remote cells but all the others in the system require processing ability. It has been decided that a multi-microprocessor is most suitable, as this is more flexible than a mini-computer.

Power Problems

Another problem is finding a source of power for equipment which must function in a variety of conditions, some of them rather primitive or chaotic. The only power supply available to a communications officer in the trench will be a manpack battery and even a tactical HQ in a remote part of the desert may not be much better off. In other situations mains electricity may be available, and BATES is designed so that it can work from a 24 V DC supply, with the aid of a float-charged battery, if for some reason the mains supply is interrupted. Of course, BATES cannot work directly off the mains supply but only through a transformer, which is part of the system.

Basically, then, BATES will be used to enhance control. Despite improvements in existing weapons and greater efficiency in communications generally, use must be maximised in the sense that no available target should be permitted to escape. This is where BATES should have a radical and revolutionary impact on the development of artillery in the British Army.

The Loneliness of . . .

Soldiering has at all times been a rather lonely occupation, despite attacks usually being mounted in groups and soldiers sticking it out together in trenches for a considerable time. The sense of loneliness is caused by the fact that in the last resort, the soldier fights and dies alone, being too busy keeping the enemy at bay to know what even his next trench neighbour is doing at any given moment, let alone the brigade a hundred yards away.

Some of this loneliness is reduced with the aid of modern military electronic communications equipment; by pressing a button it is now possible to know quickly what fellow-fighters are doing, where they are, and even where the enemy is. This is the main purpose and strength of communications systems, one of which (Philip's Callpac) will be discussed in some depth.

Callpac is described by its makers as a tactical radio device. It can be carried with ease by individual soldiers, or operated from an Army truck or from more permanent army installations.

What is new about Callpac? First it uses microprocessor technology, and second it is cheaply priced — at least according to the makers! It also combines, in an unusual manner, qualities not so far produced for military communications use. The radio spans the whole HF band from 1.6 MHz to 30 MHz and also has the facility of covering both short and long range communications. It is not stopped by unfriendly territory or climatic conditions and is designed so that one does not have to be an expert to use the equipment efficiently. Callpac may be used in conjunction with other communication systems (eg the Philips Unicom Series and the British Army Clansman). As a result of microprocessor control, features include rapid keyboard control, automatic antenna tuning, instant recall of nine pre-selected channels, and one-hand operation.

. . . The Long Distance Soldier

Callpac also boasts a groundwave range of more than 30 km; a power output that can be varied from 4 W PEP to 100 W

PEP (the last with the aid of an amplifier); protection against unauthorised programming and a full remote control and intercom facility through the use of a field telephone, as well as a local intercom facility between those that use the same transceiver.

Callpac receives its energy either from a NiCad rechargeable battery pack or from a DC source at the vehicle. Callpac can work at least 10 hours continuously (with a duty cycle of 1:9 as regards transmission and reception) unaided. After this the aid of the built-in battery charger is required. A Random Access memory retains any stored data if and when the power supply breaks down (which can easily happen under wartime conditions) or when the battery pack must be replaced.

Callpac is reliable due to the use of the latest microminiature techniques and its tough plug-in modular design. The mean time between failures is estimated at 5000 hours, with an average repair time of 30 minutes. This is made possible by a system of exchanging faulty modules for working ones, virtually on the spot. It can be put to many uses as a result of being microprocessor-based and it can be worked in conjunction with Philip's PRC/VRC 4600 series of VHF radios.

Operator using the Callpac electronic communication system.

Day Of The Triffid

Another electronic aid to Army communications is known as Triffid. This is a tactical relay system which makes contact possible between Army command posts and field units, wherever they may be located. The Triffid equipment was first conceived when it became necessary to think of modernising radios which were part of the Bruin trunk communication system. However, Bruin was to make way itself at the beginning of this decade for the Ptarmigan automated network, so designers had to construct Triffid in such a way that it could be used in Ptarmigan. This meant a seven-module construction with three frequency bands from 225 to 1850 MHz, using one of the RF heads.

Triffid, a joint effort of Marconi Communication Systems Ltd, AEG Telefunken, and Siemens AG, has the following advantages. Modules and antennas can be changed easily whenever frequency band changes are necessary; the modules are light and small enough to need no more than one operator and they are sufficiently self-contained to be exchangeable for each other. Triffid can operate in extremes of temperature and relative humidity, near other radios or radars, in fast vehicles on bad roads and even on the backs of lone soldiers. Moreover it is designed to withstand RDI (radio frequency interference) and even the effects of some nuclear attacks, since it is shielded against blast and flash by the vehicle in which it is carried.

A Signal Success

Over 200 Triffid sets are now used by the Signal Regiments in BAOR with more than 450 actual modules in the hands of the ultimate users. It has been found that non-specialists are easily able to get the best out of Triffid and can keep the equipment in top shape using the in-built test-diagnosis devices, even when subjected to the electromagnetic interference which can be expected when friend and foe are trying to keep in touch with their own units.

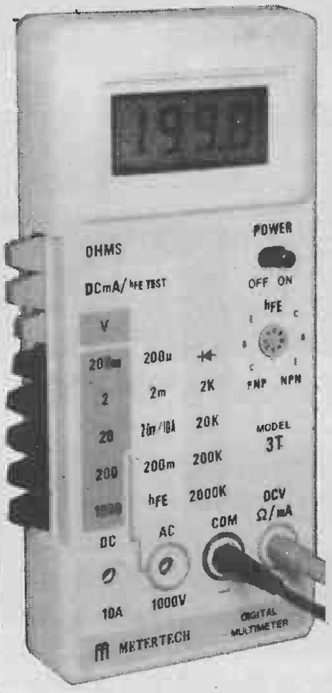
Triffid obtains its power in battle field conditions from a 24 V lead acid battery, charged by a petrol generator in conjunction with a power-conditioning unit. DC voltages are regulated by a power-supply module in conjunction with a switching inverter, making sure that large voltage variations can be utilized and that overload can cause no damage.

Frequency can be varied by means of a frequency synthesiser, set by four front panel switches; there are three frequency bands, 225-400 MHz, 610-960 MHz and 1350-1850 MHz. The equipment at once informs the operator if anything is wrong by an in-built alarm and monitoring system.

All told, the machines described above afford the modern Army the assistance which is essential if a numerically inferior NATO fighting force is to have any chance of resisting, let alone deterring, the armies of the Warsaw Pact Nations. **ETI**

Science For Britain

Edmund Scientific Products of New Jersey, USA, have appointed Rheinburgs Sciences Ltd of Tonbridge to be their exclusive UK distributor for their extensive range of astronomical, scientific and associated equipment. The range consists of over 4,000 items of equipment for the astronomer, hobbyist, education and industry and the majority of these are now available for sale in the UK. To promote these, Rheinburgs have a newly equipped showroom at Tonbridge and two comprehensive free catalogues. The General Scientific Edition comprises 48 colour pages covering such subjects as Optics, Holography, Tools, Motors, Microscopes, Laboratory equipment and much more; whereas the Astronomical Edition contains details of products for all astronomical needs. To obtain your free copy of either or both of these catalogues, simply write to Rheinburgs Sciences Ltd, Dept. PR1, Sovereign Way, Tonbridge, Kent TN9 1RW.

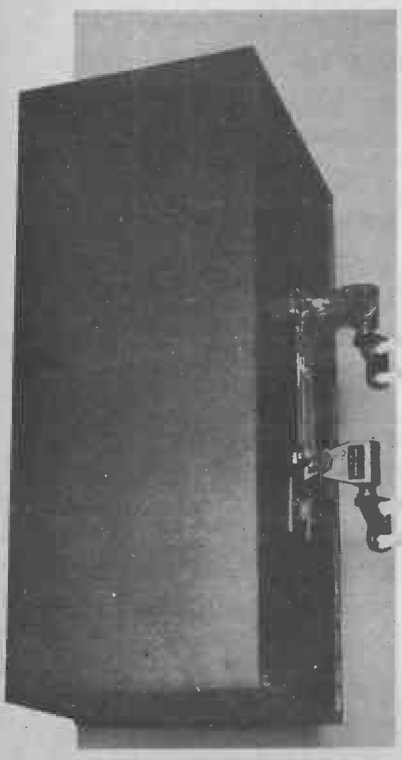


Miser's Meter

One of the items we managed to get our mucky little paws on at the Breadboard Exhibition was an incredibly cheap multimeter from Marco Trading. The U4324 (actually it's not a U but our typesetting machines can't handle Russian characters) has a high sensitivity taut band suspension movement and full coverage of AC and DC voltage and current ranges, as well as resistance. The sensitivity is 20000 ohms per volt on DC and 4000 ohms per volt on AC. There are six DC and five AC current ranges and a decibel range. The meter can handle signals with frequencies from 45 Hz to 20 kHz. The U4324 costs a mere £10.50 plus £1.50 postage and packing and can be obtained from Marco Trading, The Old School, Edstaston, Wem, Shropshire, SY4 5RJ. Their catalogue is also available free of charge, just send them a stamped, addressed envelope and you'll get one!

Speaker Mounting

Fidelity Fastenings are announcing the introduction of a versatile new product — their Multi-Angle Wall Mounting Speaker Hinges (Model FF4). These lightweight fixtures, manufactured from a robust, glass-filled nylon material, are very strong — each is capable of supporting a speaker of up to 50 lb (22 kg) weight. They feature a patented device which offers the facility of tilting and swivelling to virtually any angle desired, allowing optimum positioning for stereo speakers. Each pair is supplied with screws, wall plugs and easy-to-follow installation instructions. They cost £8.99 a pair (including VAT) from hi-fi, record and many other outlets. For further information contact: Fidelity Fastenings, The Ridgeway, Iver, Bucks.



Playing For Time?

There are plenty of alarm clock calculators around, but Casio's latest addition can perform calculations *with* time! The HQ25 is an eight-digit LCD calculator with fully independent memory, an internal clock showing time in hours, minutes and seconds plus am/pm and a variable daily alarm. It can also handle sums involving addition and subtraction of time, or multiplication and division by integer or decimal figures. Time data is entered as hours, minutes and seconds, and answers similarly indicated for periods not exceeding 100 hours (or negative limit of 10 hours). Within the range, touching one key gives instant conversion between sexagesimal and decimal expressions — a feature that could be useful in problems involving angular measure. Power is supplied by two AA size batteries and the HQ25 has a recommended retail price of £11.95. Sales enquiries to Casio Electronics Co Ltd, 28 Scrutton Street, London EC2A 4TY.

COMPUTER EXPANSION SYSTEM

This month we look at the Ins and Outs of home computing with this versatile and simple expansion board. Treat your ZX81 to more peripherals, too. Design by Watford Electronics.

This month we cover the wonderful world of I/O by presenting an input/output card with numerous applications. The card is based on two 6520 peripheral input/output chips. One chip is used mainly as a Centronics parallel printer driver. This is located at the third 1K section of the card's 8K block. Assuming the user to have a sound card in slot 4, ie at 8000 Hex (Group 1 users), then this parallel card will sit in slot 5 with the parallel printer driver at 8400 Hex. It is possible to move this around by simply linking the relevant 1K block select through to $\overline{CS2}$ (pin 23) on the 6520. Be careful not to select anything at 8000 ($\overline{CS0}$) if you wish to use the sound card at this location. See Table 1 for details.

The PCB has the second 6520 (IC2) located at 8600 Hex ($\overline{CS3}$) and the digital-to-analogue converter at 8800 Hex ($\overline{CS4}$). These can easily be moved around as the user desires in the same way as the Centronics interface I/O chip as described above. Also on the PCB is DIL reed relay energized by PB_7 of IC2 for tape control or other small current (<500 mA) switching applications. Pads for five transistors are provided for other driving applications. IC5 is a 75450 and again provides a different means of output driving.

The circuit diagram and PCB have all these differing devices on to illustrate the vast range of outputs that can be achieved — it is expected that many users will add more of these type of devices to suit his/her needs.

Provision is made on the PCB for all of the various output devices shown in the circuit diagram, but it is assumed that readers will have their own ideas on uses for this card. Several sets of general purpose outputs are provided for and may be linked to your own homebrew circuits. The kit of parts supplied by Watford Electronics for this board does not include the components for our suggested applications (see Parts List).

The outputs for a Centronics printer interface are taken from pads down the left of the PCB, as shown on the overlay. The remaining port connections of IC1 have take-off pads on the other side of the chip, while connections to IC2 can be made via SK1 on the overlay, using a header plug and ribbon cable. Output driving transistors may be soldered to the pads on the PCB, or wired off-board. Almost any type is suitable—BC108, 2N3053, TIP series etc—but remember you may have to change the value of the base resistor (R4-8) to suit the drive current requirements.

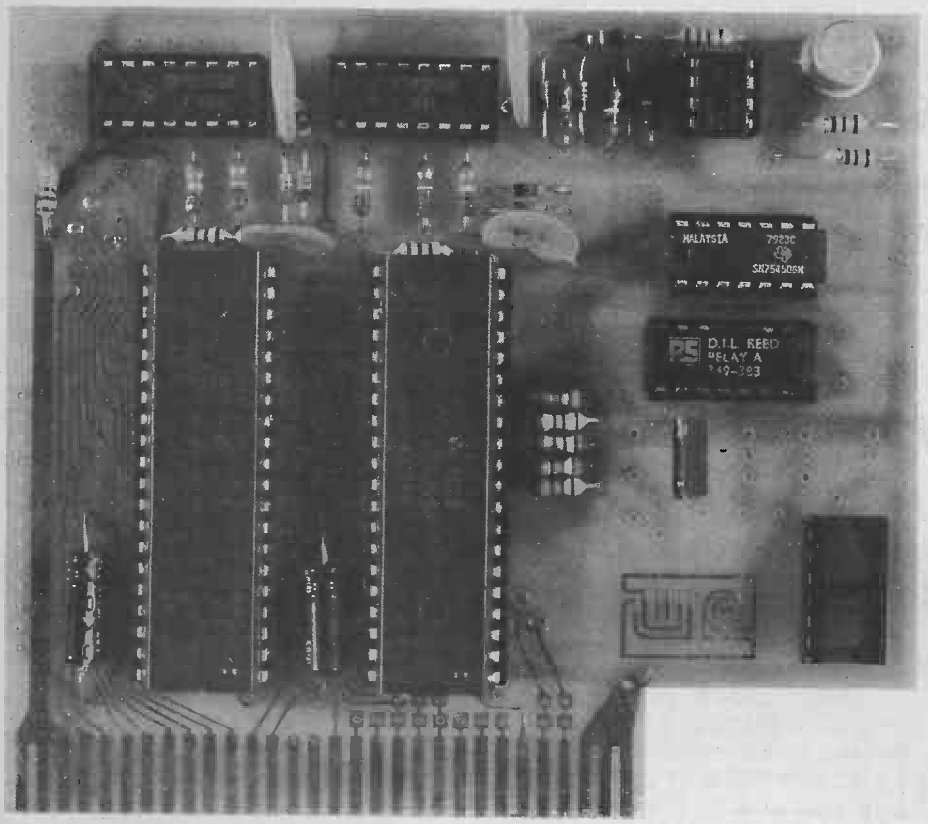
Construction And Use

Construction for this board is quite straightforward, but if you're planning to move things around in the memory map it would be wise to make these modifications before soldering any of the components in place. On both sides of the board you will find small square pads, often linked in pairs by thin tracks. These correspond to the link points shown as small circles in the circuit diagram connections and are provided for ease of re-location. Simply break the track between the pad(s) you wish to modify and connect the new chip select signal by soldering flying leads between the relevant pads. The circuit diagram shows which select line is which on the edge connector and should be used in conjunction with Table 1.

TABLE 1

SELECT LINE	START ADDRESS
$\overline{CS0}$	8000
$\overline{CS1}$	8200
$\overline{CS2}$	8400
$\overline{CS3}$	8600
$\overline{CS4}$	8800
$\overline{CS5}$	8A00
$\overline{CS6}$	8C00
$\overline{CS7}$	8E00

These square pads are also provided for PB5, PB6 and PB7 of IC2, should you wish to take them elsewhere for applications other than those illustrated, and on the spare inputs to IC4. These are normally tied high by R3, since the D-to-A application only uses eight bits (six for IC3 and two for IC4); if you want to latch extra bits, just break the links to R3 and connect the required inputs to the right pads. For example, if you wanted 12 bit D-to-A (bear in mind you'll need very high tolerance resistors on the outputs for this), then you could connect D0-D3 to the unused inputs of IC4 with flying leads from the PCB tracks left of IC1, using the through-plated holes as soldering pads. The data will now have to be written separately to each latch (ie two bytes needed), so IC3 and IC4 will need different chip selects. Modify these as described above, wire up an extra four diode/resistor pairs off-board and you're there.



PARTS LIST

I/O BOARD

Resistors (all ¼W, 5%)

R1,2,3,9	1k0
*R4-8,18	10k (or other value to suit application)
R10-17,19	100k
*R20,21	1k0 (or other value to suit application)

Capacitors

C1,2,5,6	100n disc ceramic
C3,4	4u7 25 V axial electrolytic

Semiconductors

IC1,2	6520
IC3,4	74LS174
*IC5	75450
*IC6	LM308
D1-8	1N4148

Miscellaneous

*RLA	DIL reed relay

PCB (see Buylines); two off 40 pin DIL sockets; three off 16 pin DIL sockets; two off 14 pin DIL sockets; one off eight pin DIL socket.

LIGHT PEN

Resistors (all ¼W, 5%)

R1	20k
R2	100k

Capacitor

C1	100n ceramic

Semiconductor

Q1	TIL81 phototransistor

Miscellaneous

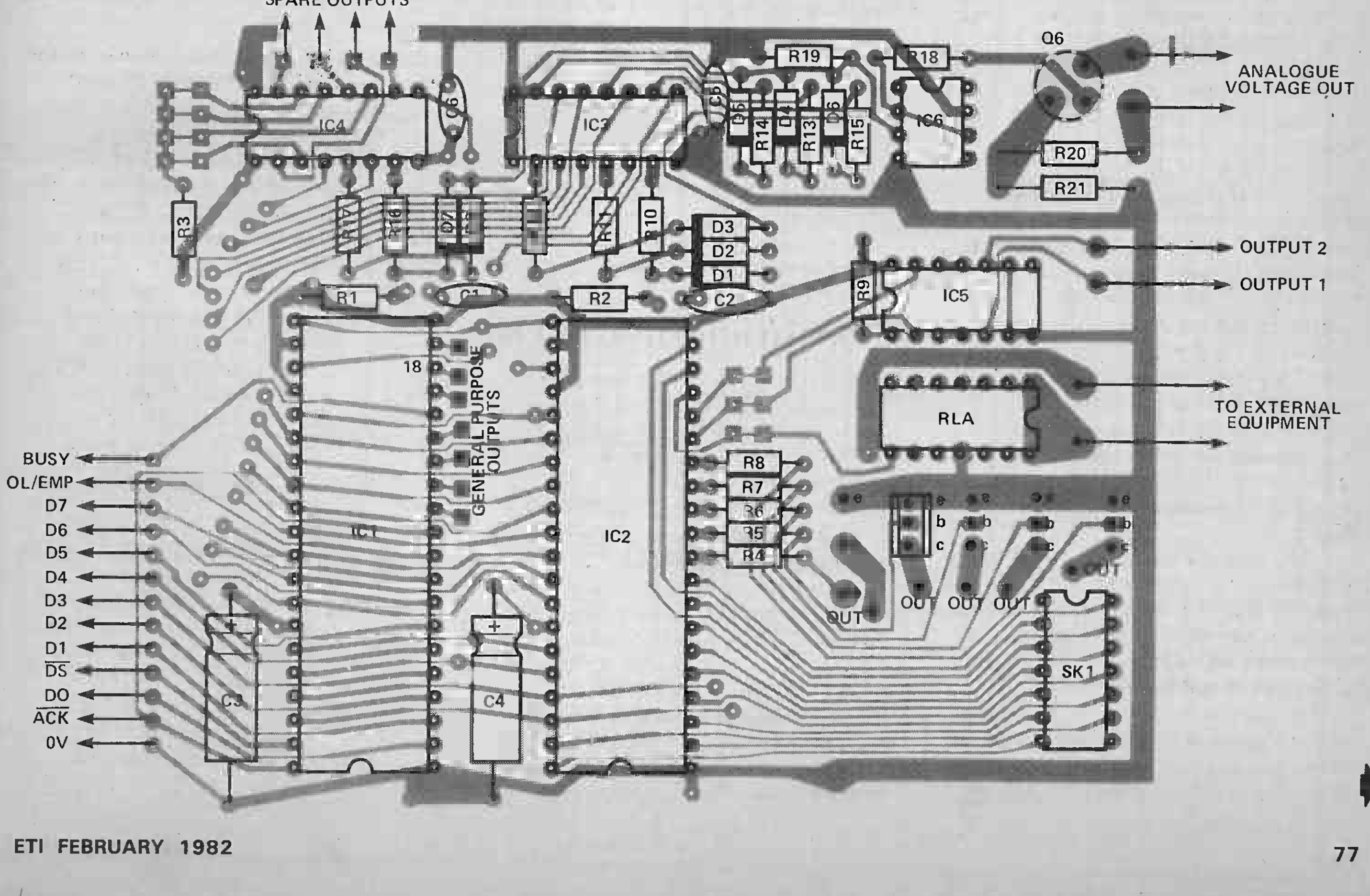
PB1	push-to-make switch

BUYLINES

A kit of parts for this expansion card is available from Watford Electronics for £19.95 plus 50p p & p plus VAT. Please note that as user requirements will vary, the components marked with an asterisk in the Parts List will not be supplied with the kit.

If you wish to interface your ZX81 to the system, the single-ended jumper lead will cost £4.99 plus VAT for a 24" length, while the 2 x 23 edge connector is £1.75 plus VAT.

Fig. 1 (Below) The component overlay for the I/O board of the ETI Universal Computer Expansion project.



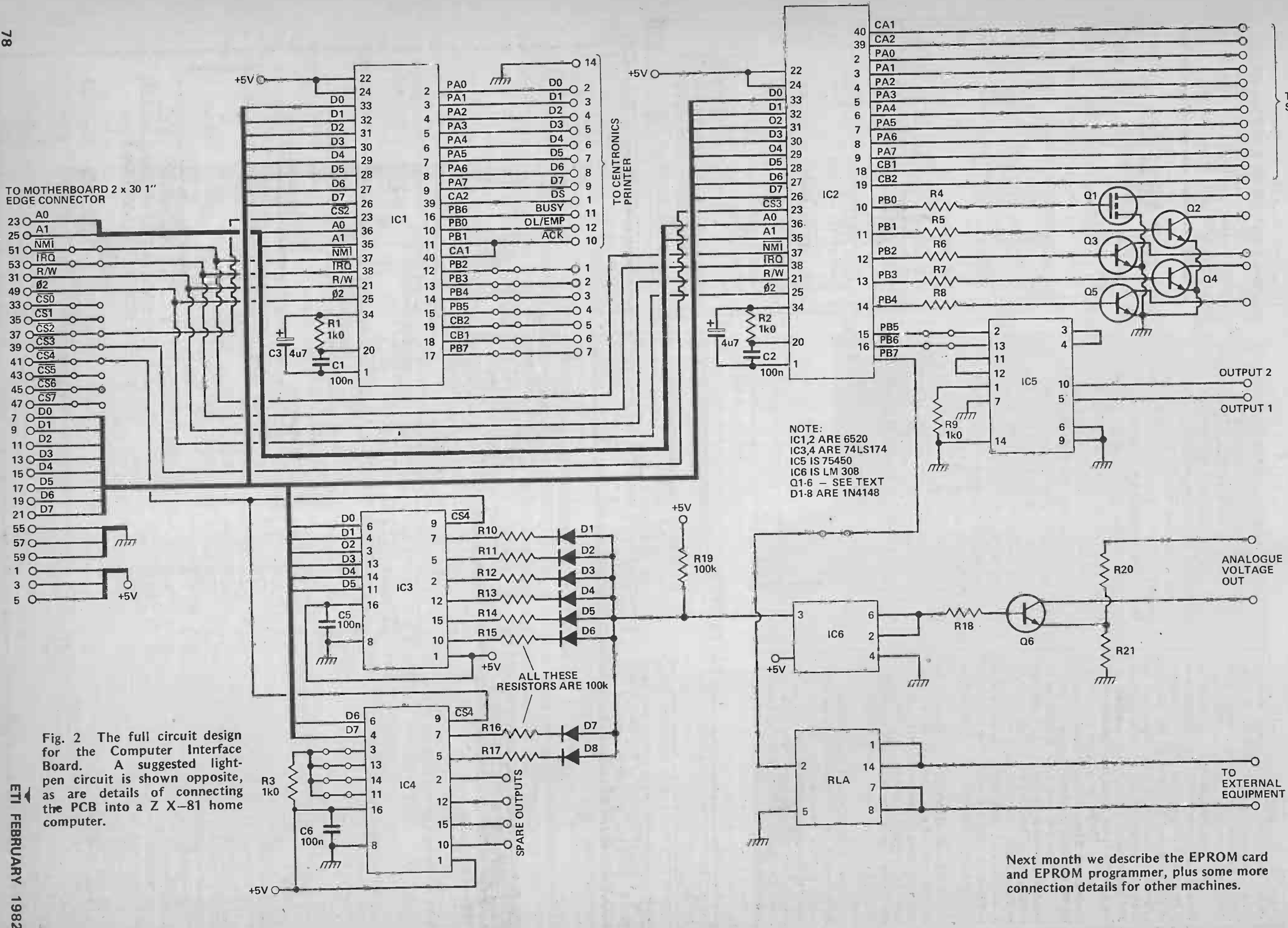


Fig. 2 The full circuit design for the Computer Interface Board. A suggested light-pen circuit is shown opposite, as are details of connecting the PCB into a Z X–81 home computer.

Next month we describe the EPROM card and EPROM programmer, plus some more connection details for other machines.

HOW IT WORKS

The heart of the output card is its two 6520 PIA chips, which are very powerful devices. Where they reside in their particular 8K block is determined by which 1K block select, CS0-CS7, is connectedd to pin 23, the CS2 input.

The digital-to-analogue converter is formed using two 74LS174 Hex quadruple D-type flip-flops with clear. Data is clocked into the latch by $\overline{CS4}$ (although this select is variable as mentioned). This means that an analogue yoltage output, which is buffered and amplified by IC6 and Q6, is available using a simple POKE to the relevant address.

IC5 (a 75450) is a dual peripheral positive-end driver. Typical current handling for the transistors in this devices is 200 mA.

A 14 pin DIL reed relay is included to make the driving of a peripheral such as a tape recorder easy. The relay has one set of single pole single throw contacts rated at 0.5 A and as such can be used in the 'remote' inpout common to most recorders. The cassette is then easily switched on or off by toggling PB_7 of IC2.

A very cheap light pen can be made up for this card using a TIL81 phototransistor, which will conduct when light is incident on its lens. The collector is tied to +5 V through a 20k resistor, while the emitter is connected directly to 0 V. No connection is made to the base.

When light falls on the phototransistor the voltage at the collector drops; if the light source is then removed, the collector will return to the upper potential. This low-going pulse is transmitted through a 100n capacitor to the CA1 line of a PIA. On the light pen a push-button switch is provided so the user can request an input at the current location of the light pen. This switch is normally held at +5 V via a 100k resistor, but when it is depressed the line goes low. This line is connected to PA0 of the PIA.

The many owners of ZX81s will doubtless be overjoyed to know that the expansion board can now be interfaced to their system. The changes involved are simple; first, remove IC6 from its socket on the motherboard. Connect pin 10 of the now empty socket to pin 40 ($\overline{MREQ}$) on the motherboard input socket and connect 0 V to pins 9 and 27 of the input socket. No modifications are required to the Sinclair. Using a 2 x 23 edge connector (see Buylines), a 40 pin header and a jumper lead, connect the computer to the motherboard as follows, using Table 2 for the Sinclair connector and Table 3 for the header. Connect A0-14 and D0-7 directly across; A15 cannot be used. Connect 0 V on the Sinclair to pins 9 and 27 of the header, connect CLK (Sinclair) to Ø2 (header), and MREQ (Sinclair) to pin 40 (header).

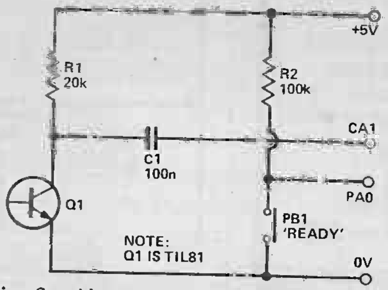


Fig. 3 Above: A suggested circuit for a light-pen for use with the expansion.

If you are using the RAM card then $\overline{RAMCS}$ must be taken to +5 V on the Sinclair to disable the internal memory—this is not necessary if the sound or I/O cards are being used.

TABLE 2

	Side A	Side B
1	D7	5 V
2	$\overline{RAMCS}$	9 V
3	POLARISING KEY	
4	DO	0 V
5	D1	0 V
6	D2	CLK
7	D6	A0
8	D5	A1
9	D3	A2
10	D4	A3
11	$\overline{RINT}$	A15
12	$\overline{NMI}$	A14
13	$\overline{HALT}$	A13
14	$\overline{MREQ}$	A12
15	$\overline{IORQ}$	A11
16	$\overline{RD}$	A10
17	$\overline{WR}$	A9
18	$\overline{BUSAK}$	A8
19	$\overline{WAIT}$	A7
20	$\overline{BUSRQ}$	A6
21	$\overline{RESET}$	A5
22	$\overline{M1}$	A4
23	RFSH	$\overline{ROMCS}$

Socket pinout on rear of ZX81.
Row A is component side of PCB.
Row B is underside of PCB.

ETI

TECH TIPS

Simple Intercom

J.P. Macaulay, Crawley

The heart of this simple intercom is the low power audio amp IC — the LM386. This usually has a gain of 20, but for this application the gain is increased to 200 by adding C1 between pins 1 and 8. The loudspeakers used are miniature transistor radio types and double as microphones, depending upon the position of SW1. The input signals from the speakers are fed into the transformer T1, an LT700 which is used in reverse in a step-up mode.

The output of the transformer feeds into the LM386 input via the 10k volume pot. This is required because of the differences in output of small speakers when used as microphones, often a variation of 4 to 1.

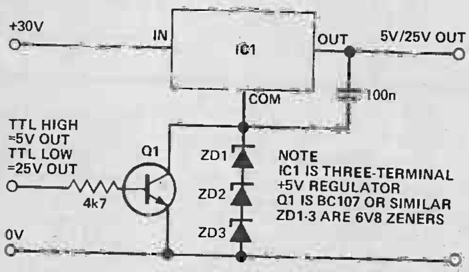
One of the main reasons for using this particular IC is that its current consumption is only 3 mA and the battery used to power it can therefore be expected to last for almost its normal shelf life. Since the amp won't give up the ghost until the battery voltage falls below 4 V, this is several months.

The output power of the circuit is limited to some 250 mW, although in this application that is more than enough. The LT700 is available from Electrovalue.

Switched Supply For EPROMs

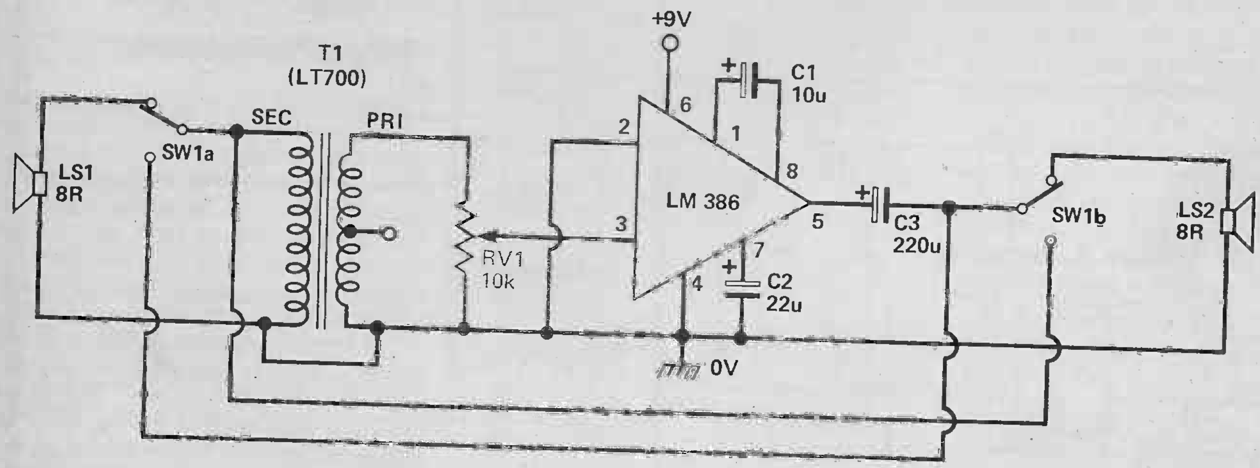
A.G. Blewett, Ramsey

While developing an EPROM programmer for my computer, the need arose for a circuit which would supply +5 V or +25 V stabilised to the programming pin



on a 2716/2732 EPROM, selected by a TTL level signal on an output port. After trying various sorts of transistor switches (all of which suffered from an unacceptable voltage loss), the following simple circuit was evolved. The transistor shorts out the zener diodes when it is turned on by a high level (greater than 2V4) signal at the input. Under these conditions, the regulator has its common lead virtually at ground, so it produces +5 V at its output.

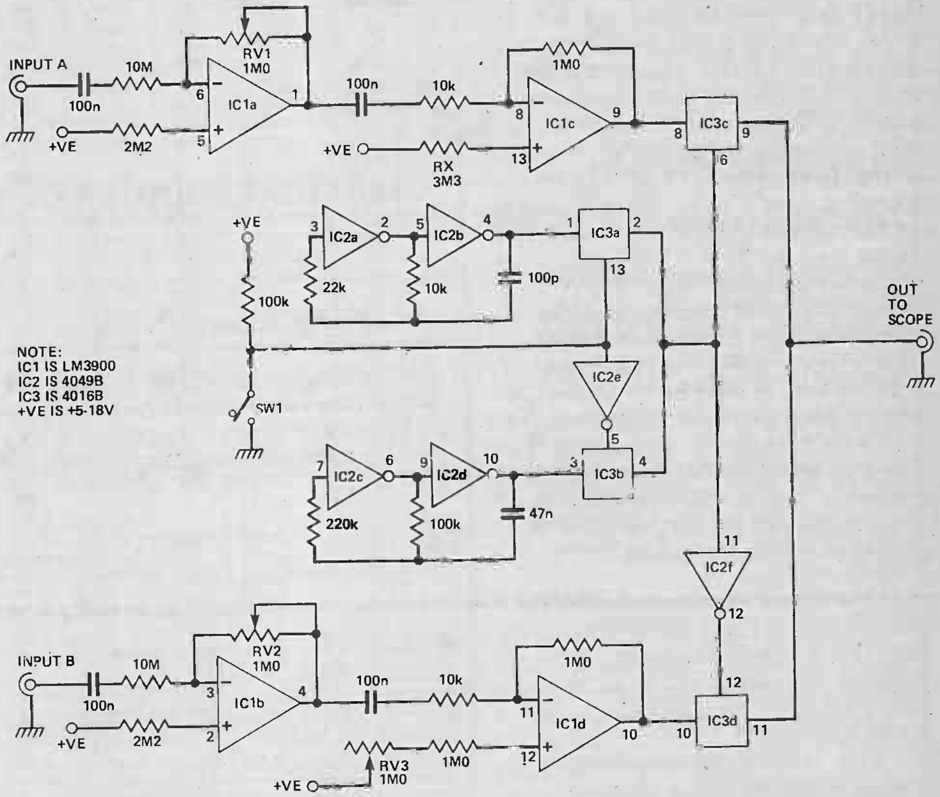
With a TTL low signal at the input (less than 0V7), the transistor turns off, the zeners conduct, and about 20 V appears at the regulator common terminal, producing +25V at its output. As the programming current to an EPROM is only a few milliamps, dissipation in the regulator is not a problem when it's producing 5 V; if other uses are envisaged, however, this point should be considered. It would be possible with suitable choice of zeners to use a number of transistors to digitally switch more than two voltages, in other applications.



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 at competitive rates. Drawings should be as clear as possible and the text should be typed. Text and drawings must be on separate sheets. 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.

Dual Trace On Single Beam Scope

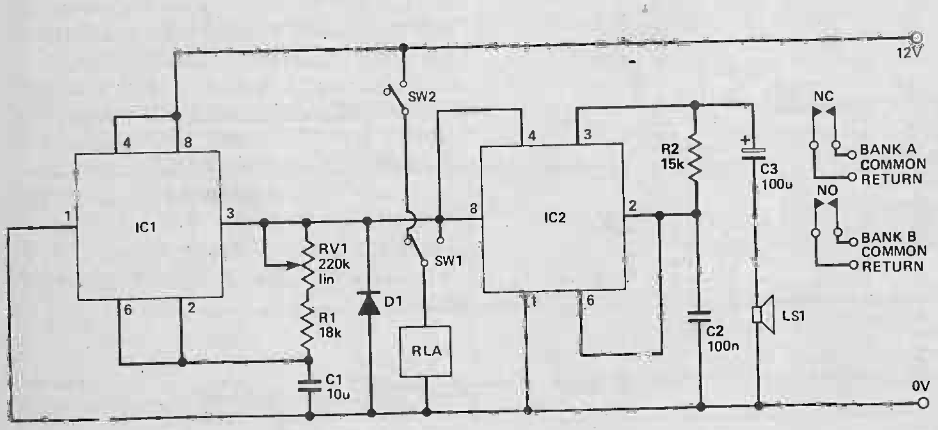
J.C. Harris, Kent



Three inexpensive ICs form the basis of this electronic switch to give a dual trace on a single beam scope. Operation of SW1 gives either a chopped display or alternate display. RV1 and RV2 are gain controls and RV3 varies the separation of the two displays. IC1a and IC1b are impedance converters and have input impedance of 10M. IC1c and IC1d (with gains of 100) combined with IC1a and b, control the display amplitude. Also the DC operating points of IC1c and d, controlled by RX and RV3, determine the separation.

CMOS switches IC3a,b (controlled by SW1) pass either high or low frequency square waves, generated by the two astables built around CMOS inverters IC2a-d. These square waves control switches IC3c,d, passing chopped (HF switching) or alternate (LF switching) signals to the scope (which must have an input impedance greater than 10k). The remaining two inverters invert the control signals to one of each pair of switches so that one is on and the other off at any instant. Power supply requirements are simplified by the use of the quad op-amp LM3900.

The lead to the scope should be short to avoid trouble at the higher switching rate.



Automatic Stage Lighting Bank Changeover Unit

I. Hodgson, Middlesbrough

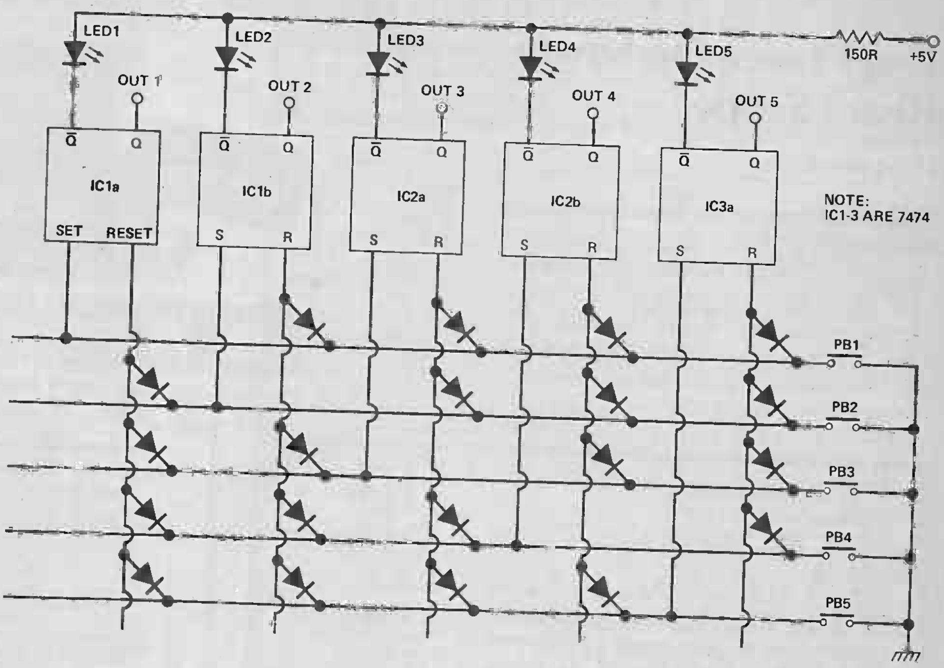
The main problem with concert lighting is to constantly change the colours on the stage, sometimes in sync with the music, while still allowing the operator to empty several cans of his favourite ale. This unit enables the bank switch-over rate to be preset (with RV1) and patched in with SW1, bypassing a foot or hand-operated switch SW2.

IC1 forms the main oscillator with a period which can be varied by RV1 between 0.6 and 8 seconds. The main oscillator switches first RLA, a 430 0hm relay which effects the switching of the banks, and second another osicllator, IC2, which acts as a bleeper to indicate the changeover rate during operation. The tone produced by IC2 is used to power a high impedance speaker or earpiece, the total resistance being made up to about 100R with a series resistor.

Electronic Switch

M. Harrison, London

This circuit is the electronic equivalent of the interlocking push-button switch bank commonly used as an input selector in amplifiers, and in many other applications. This circuit has the advantage that it only requires small SPST switches on the front panel, thus simplifying mounting, and it is cheaper than mechanical switch banks. The outputs can be used to drive CMOS analogue switches (after buffering), or reed relays. The circuit is quite simple, consisting of set-reset flip-flops arranged so that when one is set (by pressing one of buttons 1-5) all the others are reset by the diode OR gates on the reset inputs. LEDs 1-5 are to show which input or channel is activated, and are best mounted above the appropriate button. The circuit can easily be extended or reduced for any number of channels required.

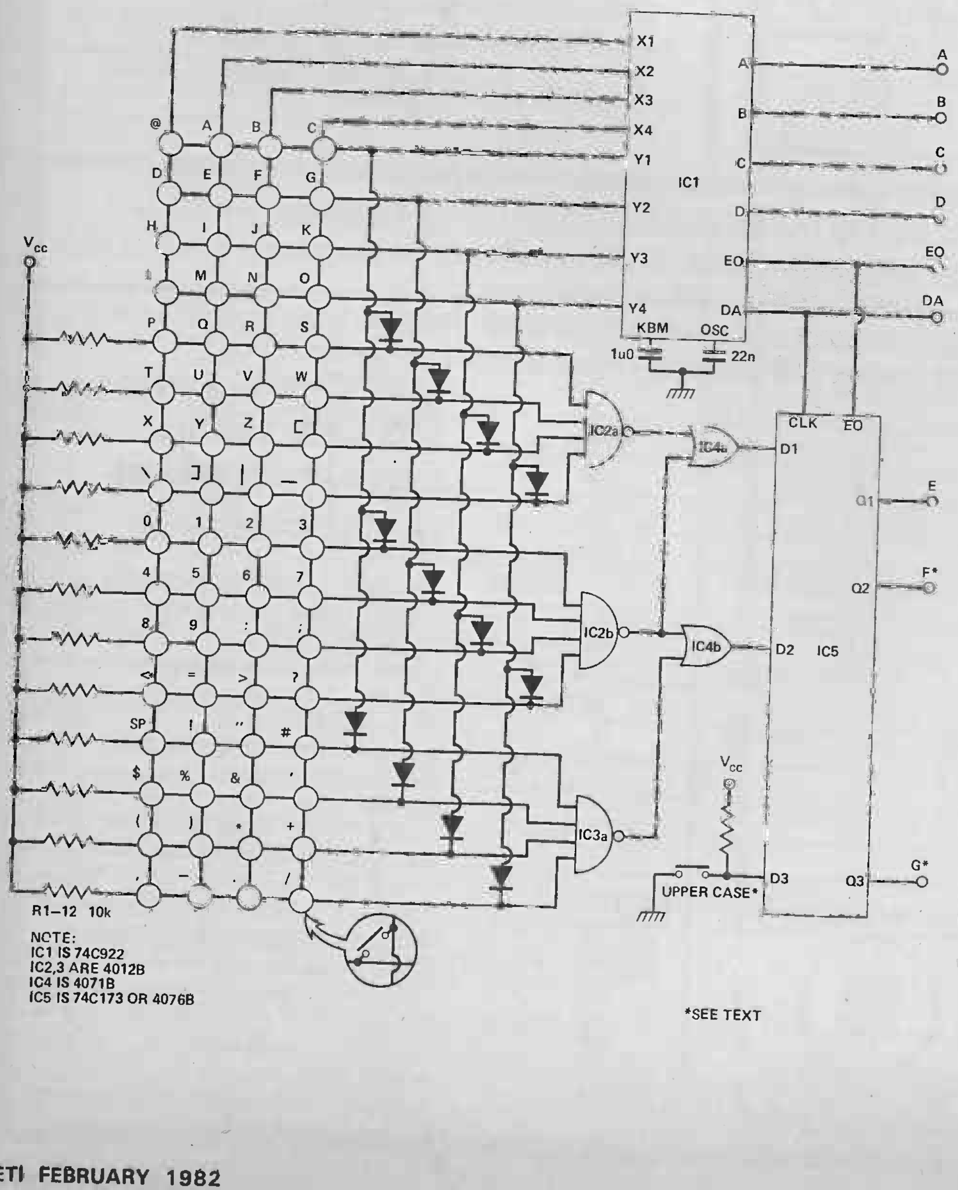


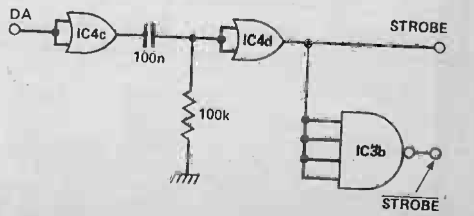
Low Cost ASCII Encoder

Quentin Rice, Mitcham

This encoder is based on the 74C922 16-key keypad encoder. This is a self-scanning encoder with debounce and latched outputs. In this case it has been expanded to four groups of 16, comming the active-low X outputs and OR wiring the Y lines through the diodes. The NAND gates detect which group of 16 is being keyed; this is encoded through the OR gates, and latched into the flip-flops. Both the flip-flops and the 74C922 have tri-state outputs for bus operation. A data available output goes high when a key is pressed. A strobe is available, if required, using the remaining gates.

The circuit is shown with a six bit condensed code, ie with some upper and some lower case characters found on 64 character generators. A full upper case operation can be obtained by reversing outputs F and G. With a 64-key keyboard, it is possible to obtain all 128 ASCII instructions with only one shift key.





PEST CONTROL

If I could talk to the animals. . . I'd tell 'em to go away. Get the message across with this harmless little gadget. Design and development by Phil Walker.

Is your garden the main attraction for all the local neighbourhood cats, dogs, hedgehogs, rats, voles, aardvarks and other small furry creatures? If so, this device should protect your seedlings, kiddies' sandpit or dustbin from their attentions until you can make a more permanent arrangement. It is harmless to the animals themselves, being merely annoying rather than painful.

Our first design involved frequency-shifting a tape of Barry Manilow up into the ultrasonic range so that only animals could hear it, but it was felt that this would lay us open to prosecution by the RSPCA. Consequently a second approach was adopted.

Many people have noticed that, when they operate the ultrasonic remote controls on their TVs or hi-fis, their household pets will protest loudly and leave the vicinity rapidly. We have designed this little device to capitalise on the phenomenon, in reponse to our readers requests for a gadget capable of protecting precious plants without the need for violence!

The 'Allez-Cat' consists of two basic parts: the first is an oscillator tuned to 40 kHz while the second is a voltage doubler and pulse generator. The pulses are about 10 milliseconds long and occur two to three times a second. This is done to reduce battery drain and increase the annoyance factor (for a cat, dog, hedgehog . . .). The voltage doubling action increases the available output power for a given battery voltage.

The whole device (apart from the battery) fits into a small plastic box and can use 6 V or 9 V batteries such as a PP1 or PP9, as convenient. If it is to be used out of doors, some form of protection from rain or other water should be provided as the transducer isn't waterproof.

Construction

The device is constructed in a small plastic box which is modified to clip onto a PP9 or similar battery using the battery connectors. The assembly of the PCB should offer no problems provided that care is taken with the polarity of capacitors and diodes. Do not fit the transducer to the board at this stage.

Mount the switch on the lid of the box using two short bolts. The large grommet should fit fairly loosely into the large hole in the lid and the transducer may be pushed into its centre using reasonable care. (Push in from the rear and avoid pressing the mesh front). Attach some thin single strand wire to its pins, and put the lid on one side for the time being.

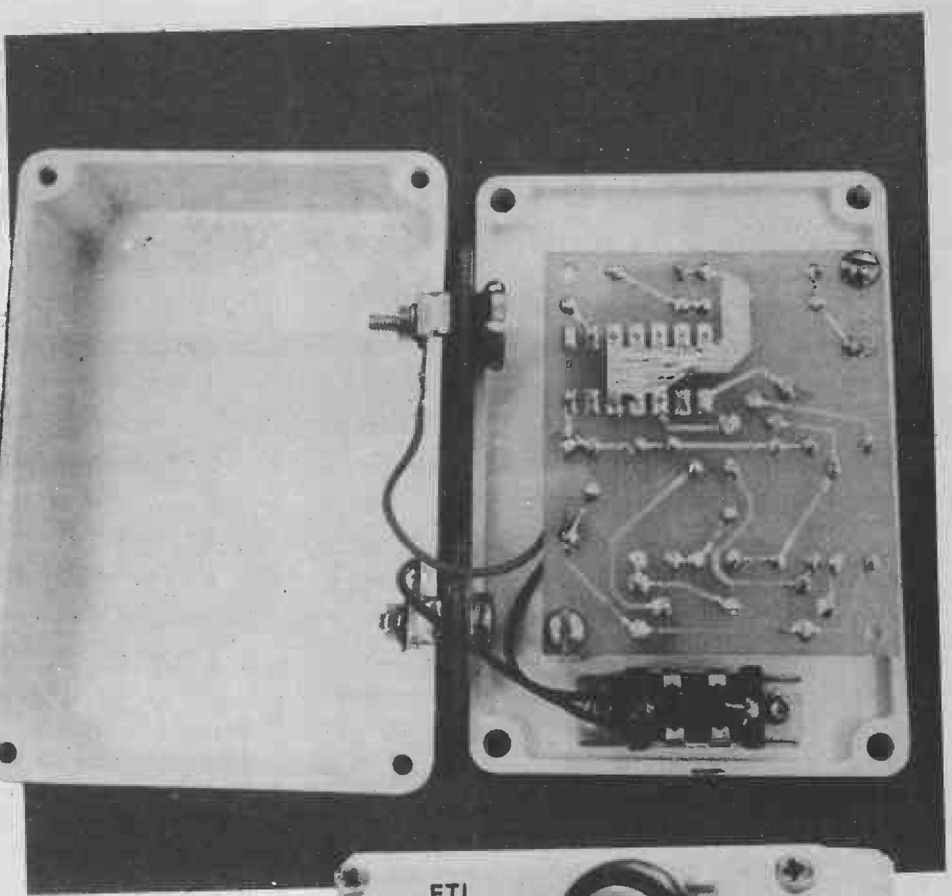
Assemble the battery connectors to the side of the box as shown in Fig. 1, remembering to attach wires about 2" long to the solder tags first. The small grommet is cut in half (ie into two discs), each half being sandwiched between the battery connector and the case. This gives a steadier mounting, as the rear of the connectors is bowed slightly. Attach the wire from one connector to its proper hole on the

PCB. Then attach another short length of wire to the other supply hole on the PCB.

Bolt the PCB to the lid of the box using the spacers and ensure that the wires to the transducer are soldered to the proper holes on the board. Connect the two remaining wires to the switch and assemble the two halves of the box with self-tapping screws (these are provided with the box). You are now ready to control the migratory habits of the animal kingdom.

HOW IT WORKS

IC1-R6-R7-R8-C4 together with TX1 form a power oscillator which drives TX1 at its resonant frequency of 40 kHz for maximum efficiency. Q1,Q2 and their associated components form a standard multivibrator oscillator with a low natural frequency. R5 and Q3 form an electronic switch which applies the negative spikes occurring at the junction of C3, R3 and D2 to IC1. This means that the power supply to IC1 consists of short spikes of up to twice the battery voltage. The duration of the spikes depends on the current drawn by IC1, since this current flow alters the mark/space ratio of the multivibrator from a square wave to a series of widely spaced pulses. The repetition rate of these pulses is determined by C2 and its associated timing resistors.



The Allez-cat PCB easily fits into a small Vero potting box, with the battery connectors bolted through the side of the case (above). The whole project then clips directly onto a PP9 battery (left).

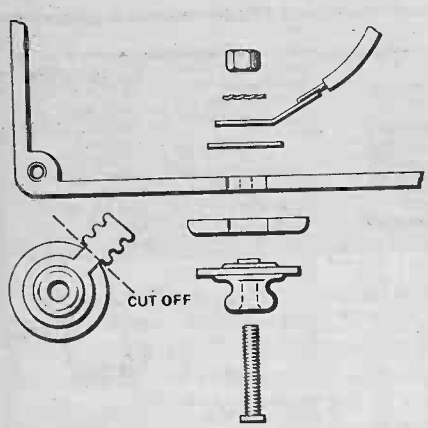


Fig.1. (right) Constructional details of the battery terminals.

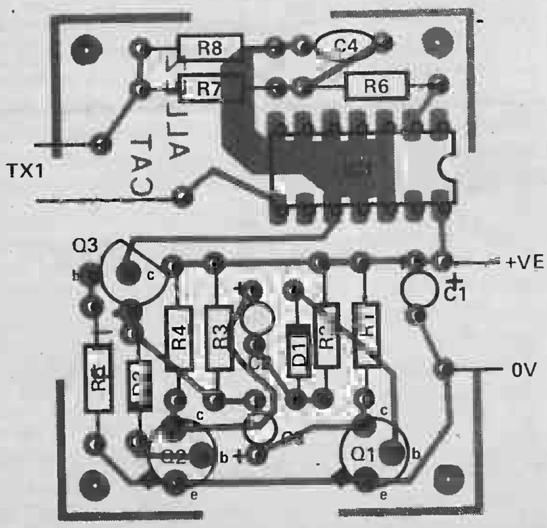


Fig.2. (below) Component overlay for the Allez-cat pest control project.

PARTS LIST

Resistors (all ¼W, 5%)

R1,6	1k0
R2,3,4,5	10k
R7	100R
R8	10R

Capacitors

C1,2,3	100u tantalum
C4	10n polycarbonate

Semiconductors

IC1	LM380
Q1,2	BC107
Q3	BC182L
D1,2	1N4148

Miscellaneous

SW1	miniature rocker or slide switch
TX1	40 kHz ultrasonic transmitter

PCB (see Buylines); PP9 battery; pair of PP9 battery connectors; two solder tags; fixing hardware; grommets; case (Vero 75-1469L).

BUYLINES

No problems with any of the parts for this project, with the possible exception of the ultrasonic transmitter. The catch is that most mail order companies sell these in matching pairs, but for this application we don't need the receiver. A quick chat with Watford Electronics established that they are prepared to oblige with single transducers, so direct your enquires thence. PCBs, as ever, are available from us at the price listed on page 81.

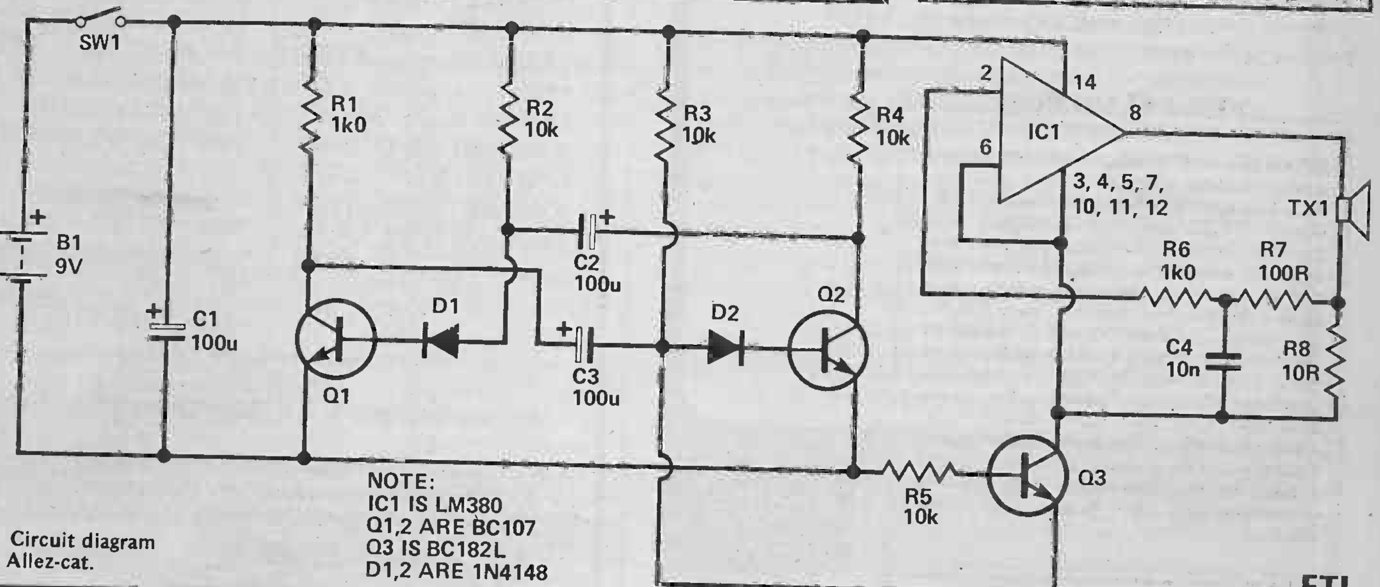


Fig.3. Circuit diagram of the Allez-cat.

ETI

Breadboard Report

Breadboard '81 took place as usual this year at the Royal Horticultural New Hall in Westminster. We are pleased to report that it was once again a roaring success with more stands than ever before to keep our electronically minded customers satisfied.

The latest sophistication in miniaturised circuitry and the booming personal computer market were strongly in evidence this year with stands covering such myriad aspects of the subject as sound, light, music, radio, test gear, CB, games and logic. Among the exhibits were Marshall's Closed Circuit Television Camera — designed to be built by amateurs; the Kansas City series of arcade games software for Tandy TRS-80 and Video Genie personal computers; the Acorn Atom personal computer — manufactured by the company which has designed the computer for the BBC's computer literacy programmes; Sinclair's now famous ZX81 personal computer with printer and add-on memory; hand-held, mobile and base CB radio units; plus much, much more.

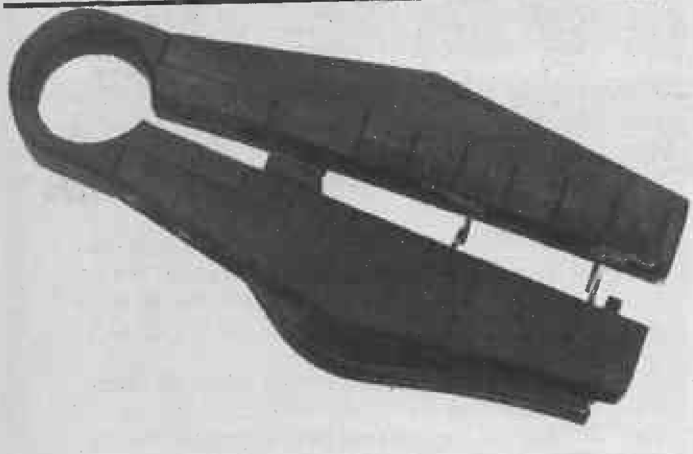
Of the stands which had more general appeal, there were plenty supplying components, games and often just general information and helpful advice. Not to be missed was, of course, our own stand! Yes, the ASP stand proved to be very popular; we were not only selling magazines and special publications to visitors, we were also displaying some of our most popular projects — including our Armdroid which attracted lots of interest when it tried to grasp items of clothing, the odd ear and anything that happened to get in its way when it was demonstrating its prowess.

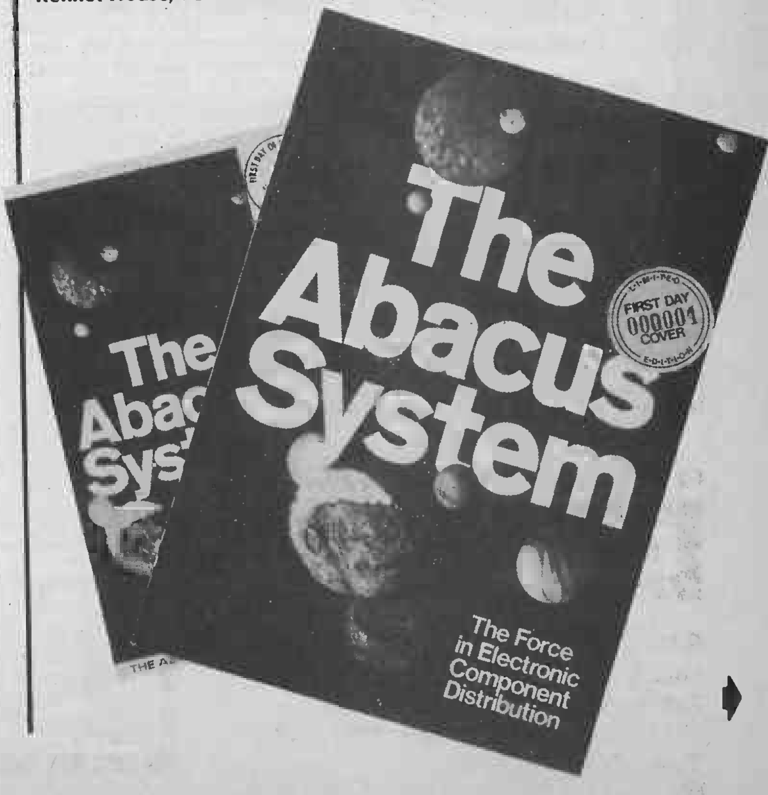
Also on show were the ETI V3 loudspeakers coupled with the Hobby Electronics Capricorn amp; this combination supplied our staff with music throughout the five-day exhibition. New ideas were also there for your perusal; a PET computer was demonstrating a soon-to-be-published game for Computing Today called The Valley; there was a CB rig to play with and plenty of our smaller projects. (We hope that whoever stole our Alcohometer has many happy hours with it, and would he or she mind not bringing his wire cutters with him next year?).

On a lighter note, there were plenty of ideas for Christmas presents to be found this year, so if you missed it, come next year and solve all those problems about what to buy Uncle Fred and Cousin Wilber!

The Force Be With You

New 152-page catalogue from Abacus Electronics PLC has just been published. It is called 'The ABACUS System — The Force in Electronic Component Distribution' and contains details of all stock components held at the company's new Franchise Division at Newbury. Technical details, manufacturers' current list prices and precise ordering instructions accompany the list of 2500 to 3000 items which include: semiconductors, microprocessors and computer boards, connectors of all descriptions including IDSs and a wide range of instruments, such as oscilloscopes, multimeters, frequency generators and capacitive meters. National Semiconductor, SGS Ltd, 3M's Scotchflex, TEEE of France, Ferranti (Dundee) Connectors, Reliability Inc, Ampliversa, Thandar, Elpac and Assman are just a few of the manufacturers whose products are listed in this catalogue. Copies are obtainable from: Abacus Electronics PLC, Kennet House, Pembroke Road, Newbury, Berks RG13 1BX.





Simple Stripping

The growing use of wire wrapping techniques as an alternative to soldering has led AB Engineering Company to produce a new Strip-Wrap tool. As its name implies, the tool is designed to cut the wire end and strip the required length of insulation from the wire before making wire-wrap connections. Designed to operate in a similar manner to an office stapler, the Strip-Wrap is light in weight — only 90 gms — and is small enough to fit the pocket. It is manufactured in glass-fibre reinforced polyamide with hardened, tempered and precision ground tool steel cutting blades. It has a capacity from 32-18 awg and the insulation can be stripped to leave between 5 and 60 mm of bare wire for wrapping requirements. This length is adjustable against a graduated scale engraved on the unit. The stripping blades are also adjustable and self-centre the wire for ease and speed in use. Both cropping and stripping blades are replaceable. In use, an operating lever folds out of the body of the unit. This is intended to reduce the pressure required to operate the Strip-Wrap. A light thumb pressure on this lever neatly crops the end of the wire and makes an incision in the wire insulation at the pre-set distance from the cropped end. A light pull on the wire then removes the insulation. The tool is suitable for use with PTFE, Kaynar and other heat-resisting insulation materials. In addition to stripping the wire insulation for wire-wrapping, the tool can also be used as a conventional wire stripper and cutter. Further information on the Strip-Wrap is available from AB Engineering Company, Timber Lane, Woburn, Beds.

THE ZX PRINTER

With a tiny TV and a compact computer to his name, Uncle Clive now gives us the ZX Printer. Peter Freebrey gets into print to LLIST the virtues of the latest beast from the Sinclair stable.

Whereas the ZX81 was available for review very quickly after its existence was announced, the ZX Printer has been hovering in the wings for many months. This has meant that many of us have either seen one at exhibitions or demonstrations or have, at least, read many press releases or advertisements about it. I think that because of this, a little of the wonder at Sinclair's latest achievement has been lost. The ZX Printer is quite a 'wondrous machine' for, although it is simple in operation and as yet unproved for long term reliability, it offers the ZX user the option of adding a printer to his system for the remarkably small outlay of £49.95, including PSU and VAT!

On opening the box you will find that, like many pieces of equipment these days, the ZX Printer is well protected for travel. It is packaged together with the uprated 1.2 amp power supply in what must be the disposal expert's nightmare — expanded polystyrene. The 1.2 amp power supply *must* be used when using the ZX Printer with either the ZX81, or the ZX80 with replacement ROM. Needless to say, you may use this with or without the printer attached, so this means you now have a spare PSU for that new project that needs a 9 V supply. The ZX Printer is very compact and has that weighty feeling normally associated with functional and robust equipment. The printer is supplied with one roll of metallised printing paper, although other metallised papers may well work — if you can find one that will fit. It comes as no surprise that Sinclair Research suggest that their paper will give superior results. Replacement rolls are available from Sinclair Research at £11.95 per five rolls. Each roll is 65 ft long and you can print about 100 lines per foot. That gives you the option of about 6500 lines or 260 screenfuls of information or any mixture thereof!

Hard Copy Made Easy

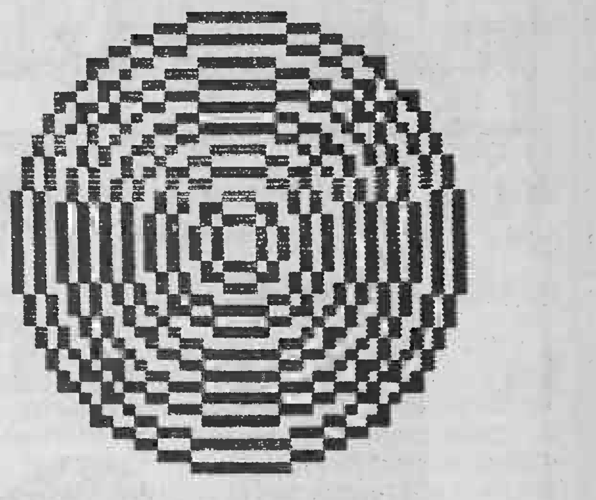
The ZX Printer is extremely simple and quick to put into operation. It takes longer to put a mains plug on the power supply lead than it takes to plug in the printer and learn the use of the one control available — that being paper feed. The plug on the printer lead has an integral socket to accept the add-on 16K RAM pack. The 17-page instruction book gives a clear description of how to load a fresh roll of paper, use of the feed button, how to tear the paper neatly off the roll (!), cleaning the printer

and the general principle of operation. There is also a selection of programs aimed at giving you the pleasure of seeing your new acquisition work, at the same time showing the use of those BASIC statements associated with the ZX Printer. Strange as it may seem, these BASIC statements are nowhere explained or described in the text; instead we are directed to read Chapter 20 of the ZX81 instruction book.

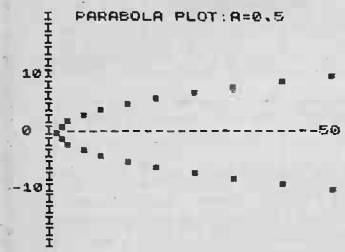
It would amuse me to report that the ZX81 instruction book referred you to the ZX Printer instructions but, although it does tell you that the printer will have its own instructions, it then continues with a short chapter which quite clearly explains the function of the three relevant BASIC statements. These are LLIST, LPRINT and COPY. The first two are just like LIST and PRINT except that they direct the display to the printer instead of the television screen or monitor. The third statement, COPY, enables you to print out a copy of whatever is at that time displayed on the screen.

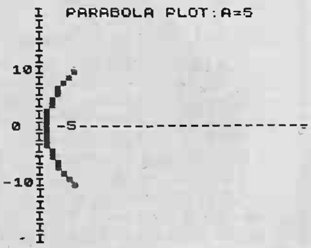
Having only three major print statements to consider, the ZX Printer is very easy to use. Formatting the display to the printer may be carried out using the TAB and AT statements, albeit that when using the AT statement, line commands are ignored and only column commands in the range −21 to +21 are actioned. Although unlikely to cause much confusion, it must be remembered that the output from LPRINT is not printed immediately but stored in a buffer store as a display one line long which the computer will print only:

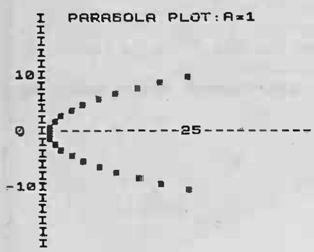
1) when the buffer is full
2) after an LPRINT statement that does not end in a comma or semicolon
3) when a comma or TAB item requires a new line
4) at the end of a program if there is anything left unprinted



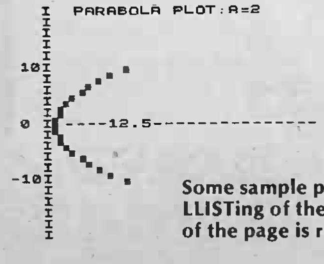
```
10 FOR J=1 TO 10
20 FOR K=1 TO J*12
30 PLOT 32+J*2*SIN (K/(J*6)*PI
),22+J*2*COS (K/(J*6)*PI)
40 NEXT K
50 NEXT J
```








```
  5 INPUT A
  6 PRINT AT 1,21;"          "
  8 PRINT AT 1,7;"PARABOLA PLOT
:A=";A
 10 FOR Y=-10 TO 10
 15 LET X=Y*Y/(4*A)
 20 PLOT X+10,Y+20
 30 NEXT Y
 40 FOR J=1 TO 21
 50 PRINT AT J,4;"I"
 60 NEXT J
 70 FOR K=6 TO 28
 80 PRINT AT  11,K;"-"
 90 NEXT K
100 PRINT AT 11,2;"0"
110 PRINT AT 6,2;"10"
120 PRINT AT 16,-1;"-10"
140 PRINT AT 11,X/2+4;X
150 GOTO 5
```



Some sample printouts using the COPY statement, together with a LLISTing of the program that generated them. The example at the top of the page is reproduced life-size.

How It Prints

As already mentioned, the ZX Printer uses a metallised printing paper. This consists of a black paper base coated on each side with a very thin layer of aluminium. To 'print' on this paper, current is passed through the aluminium, thus evaporating the metal and revealing the black paper base underneath. To achieve reasonable resolution, a fine pointed conductive stylus is used as the source of a spark discharge. This is rapidly drawn across the paper from left to right, much as a television picture is scanned across the screen. Although the speed of printing may vary, the voltage pulses to the stylus are synchronised with the stylus hitting the paper so that the printing should always be vertical. To avoid the need to return the stylus quickly to the left-hand edge of the paper there are, in fact, two styli mounted on a moving belt which follow each other in quick succession. The belt and the paper feed roller are both driven continuously while printing; so that when the next stylus comes round, the paper has been moved up ready for the next line.

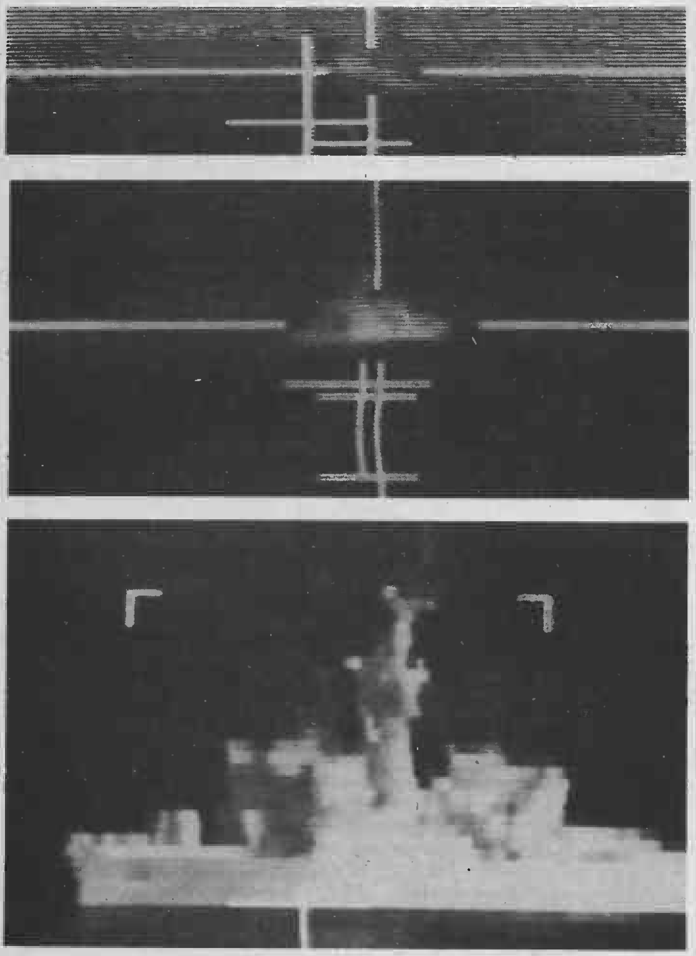
The printout presented by the ZX Printer was clear and readable with none of the disturbing fuzziness that has been seen with some systems. Graphic symbols can 'join up' from line to line giving a clear continous picture. Keeping the printer clean is probably very important and although the review model has been tested to some extent, long term reliability can only be commented upon at a later date. However, having seen printers in almost continuous use at both the PCW Show and at Breadboard '81 it would seem that this little machine has plenty of stamina! As with the ZX81 and the ZX80 before it, the ZX Printer offers something that, for the initial outlay, is quite remarkable and can only add to the effectiveness of your ZX system.

ETI

Defence Digest

In Sight

This series of three photographs shows the images produced on a cockpit-type display by an infra-red seeker for the AGM-65F Maverick missile during captive flight tests. In the top segment, the seeker 'locks on' to the guided missile destroyer USS Bagley at a longer range than the flight crew's visual range. In the middle, the image is shown at the time the ship was sighted by the flight crew. At the bottom, the Bagley is shown in what would have been the terminal stage of the missile's flight. Hughes Aircraft Company is developing this version of the Maverick missile family for the US Navy.

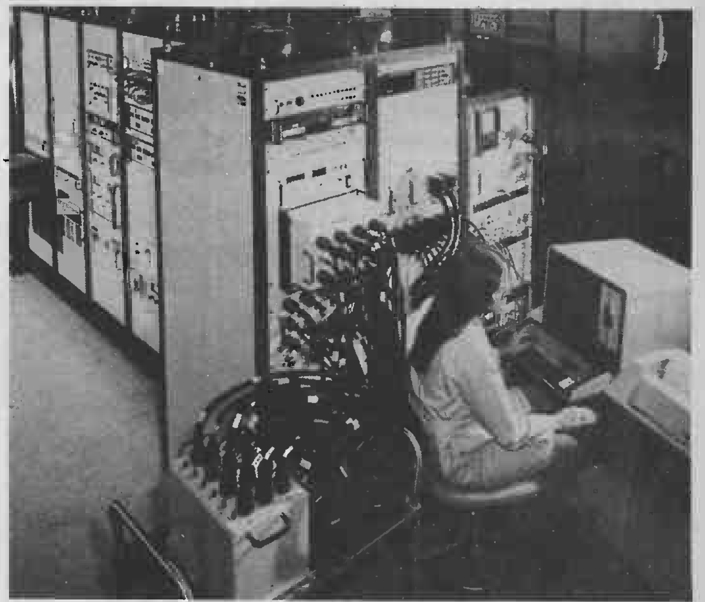


Training For Undergraduates

The US Navy has selected the McDonnell Douglas/British Aerospace/ Sperry team to continue programme development work on the Navy's VTXTS undergraduate jet flight training programme. The contract has been awarded to the team, headed by McDonnell Douglas (who will be the prime contractor) with British Aerospace as the principal sub-contractor for the airframe and Sperry as principal sub-contractor for the simulators. The Navy selected a modified version of the British Aerospace Hawk as the training aircraft. The Hawk, powered by a Rolls Royce Adour engine, has accumulated more than 100,000 operational flight hours as a trainer with the Royal Air Force since its introduction into service in 1976. No Hawk aircraft has ever been lost due to engine failure. The modified Hawk planned for the Navy will have a gross take-off weight of 12,129 pounds and an empty weight of 8,723 pounds. The Rolls Royce Adour engine produces 5,340 pounds of thrust and uses 1,250 pounds of fuel an hour, less than half the fuel burn rate of the existing Navy jet fighter. Modifications to the Hawk for the Navy training programme call for strengthened landing gear, an arresting hook and catapult launch fittings. Manufacture of a significant portion of the fuselage and major components, plus final assembly, will take place at the McDonnell Douglas plant in Long Beach, California.

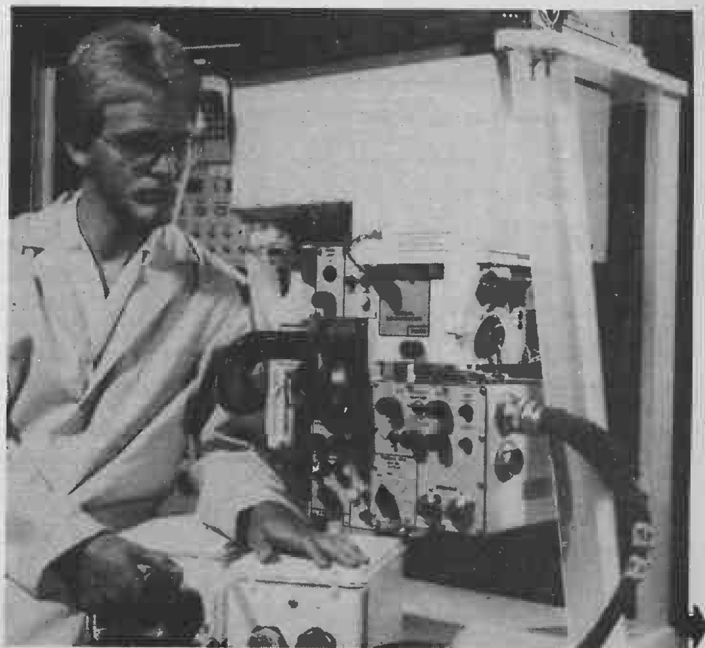
Versatile Tester

German troops will use computer-controlled test stations like this one to run fast and accurate field checks of the electronics of major weapon systems. The stations, installed in trailers for mobility, will be used to test and isolate faults in the electronic subsystems of weapons such as the Roland air defence system and the Gepard and Leopard 2 tanks. Four of five production test stations ordered by West Germany have been delivered by Hughes Aircraft Company's Electro-Optical and Data Systems Group. Hughes designed and built four versions of the test stations, called REMUS, under a $8.7 million subcontract. Each REMUS station has the capability to test electronic equipment within a specific frequency range.



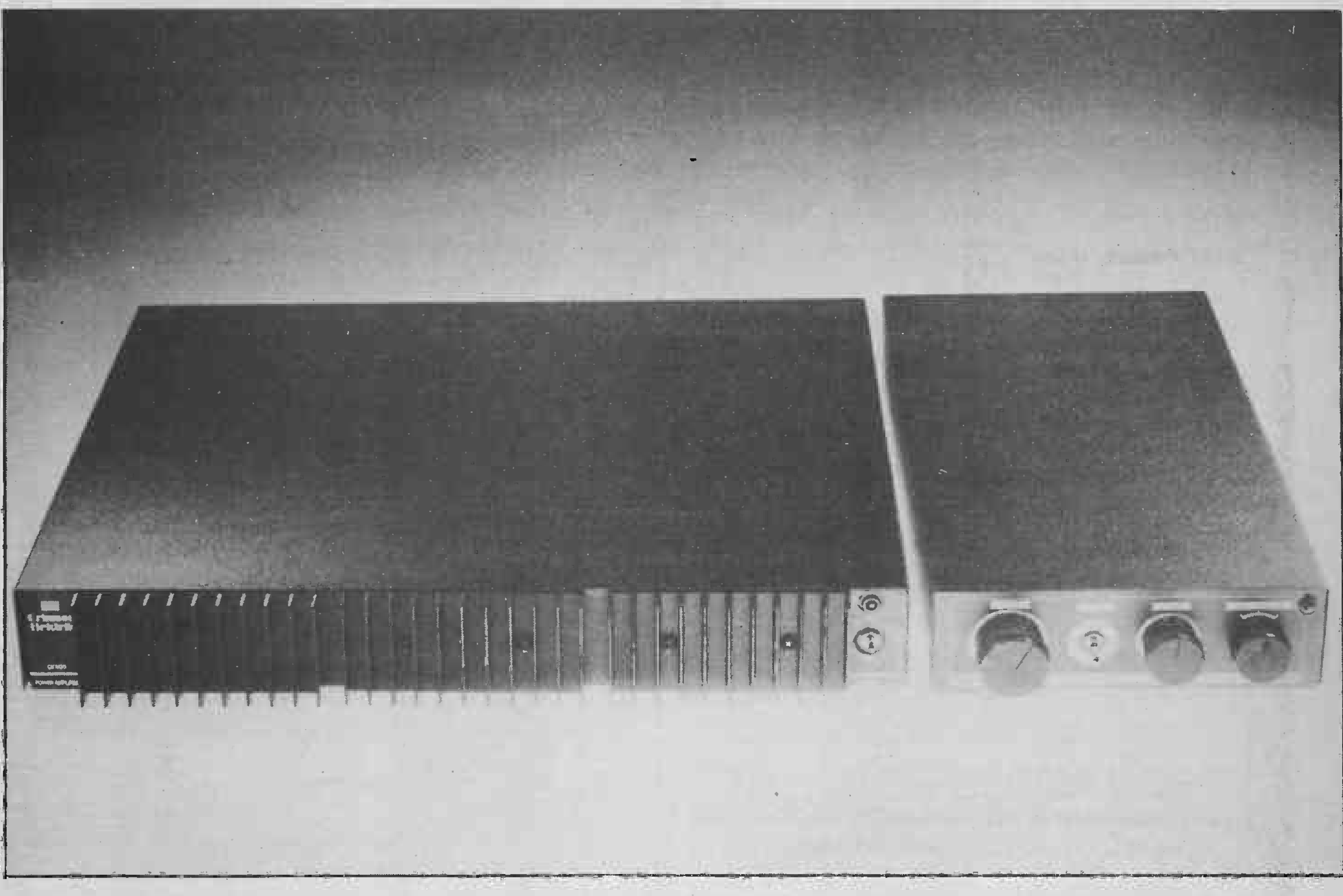
Tank Sight

Hughes Aircraft Company technician is shown displaying the sight unit that will enable gunners of the US Army's new M1 Abrams main battle tank to pinpoint targets through darkness, smoke or haze. For the first time, M1 tanks have been deployed to Europe where they are currently being used for training US Forces. Two M1s are also in Switzerland for evaluation by the Swiss Army. Hughes' Electro-Optical and Data Systems Group at El Segundo, California, is producing the thermal imaging night vision system and the laser rangefinder for the M1 at a rate of about 30 per month. Production is scheduled to increase to about 60 systems per month next year. These systems are integrated into the gunner's primary sight unit (shown here) by M1 prime contractor Chrysler Defense Incorporated.



AUDIOPHILE

It's King Crimson time, as Ron Harris gets an exclusive first look at the new pre/power amplifier kits from our friends Elektrik.



Hello, good evening and welcome. Well, I've started Audiophile just about every other possible way known to mankind *except* the obvious, so why not? What do you expect for 75p anyway? Hemingway?

Right, having gotten the always awkward opening out of the way, let's have at it, good readers. This month we have an EXCLUSIVE. Note the capitals! Crimson Elektrik have long been regarded as the doyens of kit-build amplifiers and a new model from them is an EVENT (note the capitals again. Connects the two together you see, psychologically speaking. Cunning things these psychs.).

This is the first review of the CK1010/CK1100 combination — the brand new Crimson Elektrik 100 W pre/power amplifier kit. These two are part of a range which also includes:

- CK1040, a 40 W per channel power amplifier, costing £119
- MC2K, a moving-coil preamp add-on to the CK1010 preamp, costing £25
- PSK, a PSU module — needed if you use the 1010 preamp without a Crimson power amp, at £20. All prices include VAT.

Sit down at the back there please, I haven't forgotten. The CK1010 costs £90 and the CK1100 power amp will set you back £149. Also including VAT.

Ranging In

The obvious question is — why a new amp at all, if the 'old' one is so good and doing so well? Crimson say this model is spawned from the comments of users of the present range, inasmuch as it contains all the improvements they have been asked for that were sensible. So anyone who wanted a Paisley pattern case to match the wallpaper has been politely ignored. However, those who noticed how easily satin finish steel picks up — and holds — finger marks have not cast their words into a uncaring void. The new case is crinkle finished and easily cleaned!

In among the circuitry — which is new anyway — mods such as a switched sensitivity volume control have been added, in this case to facilitate use at low-level with a good range of control on the pot.

A worthy addition then, and not simply a new box and old ideas. Would that all companies restrained themselves in such a manner. Coincidentally, I think it is noticeable how the classier firms are the last to rush toward a new release, just to add a control or two. How many new Quad units have there been? Or SME?

Might be a rule in there somewhere, you know; 'Ron's Law of Inverse Quality Marketing' I could call it. Always fancied being in the history books . . .

CK1010 Preamp

As with all Crimson kits, the PCB arrives pre-assembled, with no soldering required on the board itself. The buyer is asked only to mount the PCB in the case, wire it up to the input and output hardware and fit the controls. Easy for such as are reading this, is it not?

All the necessary bits and pieces are provided, right down to feet for the case and an aluminium facia for that professional look. The chassis itself is steel and has all the metal work done to it already. It is finished in a matt finish known as passivation — which everyone has seen and no-one knows the name of. Makes a good earth though. As I have mentioned previously, the case lid is crinkle finished both for looks and for avoidance of wear and tear.

Inputs catered for are disc, tape and tuner, with moving coil available through the MC2K option. Personally I think they should have fitted two tape inputs, but that's a personal point of view, most people would not need it — until someone turns up to record their LPs . . .

Controls offered are volume, balance, stereo/mono, input select, tape monitor and 10 dB attenuator (coupled to the volume control). No filters and no tone controls. (Do you *really* need them, or is your present amp simply bass-deficient or top-end prominent?)

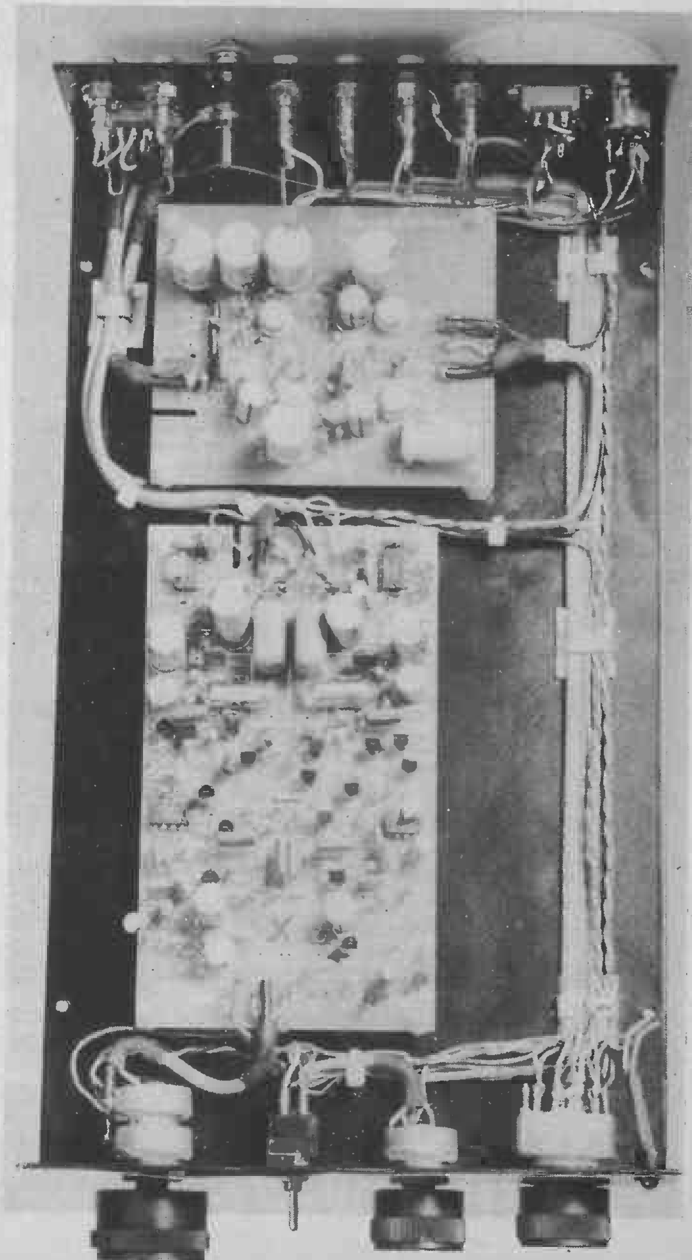
All outputs, except power supply, are phono socketed and quite right too, in my opinion. The only use for the dreaded DIN is to carry the output to the Crimson CK1040/CK1100 and bring back the low voltage supply.

Design Noted

The preamp falls basically into four gain 'blocks' which use both ICs and discrete transistors where applicable, to reduce noise effects in the earlier stages of the design: for example, the magnetic input is a cascode circuit employed to match the high impedance source of the cartridge.

Switching thumps are eliminated by JFET switches set at the output. All in all the 1010 is an improved version of the present Crimson 'Super' preamp board, using more or less the same high stability component positions designated therein. The stereo/mono switch, the attenuator and the MC/MM select switch for the phono input are all situated on the back panel of the unit, next to a sensibly huge earth terminal!

The unit I had for trial was fitted with the moving-coil board already, to allow me to exercise some favourite cantilevers.



CK1100 Power Amp

Power amp PSUs are something we've been going on about at some length recently, simply because they are an often overlooked area in which great improvements can be effected. (Read the article on page 26 of this very issue, for example.) Crimson have not employed two complete supplies but have adopted a very acceptable approach to the same end. A large single toroid, with two secondary windings, feeds two separate rectifier and reservoir networks, such that each channel has its own energy-store to draw upon in case of dire (peak) need.

Dynamic crosstalk will be reduced with this configuration, when compared to the more usual single supply and reservoir set-up all too common in commercial amplifiers.

Protection for the output is provided by a shut-off circuit rather than fuses in the output. A mains fuse *is* provided — and not even the most extreme of the 'super-ear-and-no-instruments' review brigade has suggested *that* makes a difference to final sound (yet?)

Once again all the hardware and heatsinks etc. are provided in the kit and there is nothing more to spend out on, short of a mains plug. The required DIN-to-DIN lead for connections to the preamp is included with the 1010 kit. At first glance the case may seem a little large, but the heatsinks occupy a fair amount of panel length, and distancing the modules from the PSU is no bad thing. Keep your wiring neat and it will repay you with added dBs of noise-ratio.

Overall the constructional standard of the pre-assembled boards is very high, maintaining Crimson's reputation. Assuming the builder wires them up correctly, the amps are guaranteed for two years, so be careful with the iron — overheat in haste and repent at leisure.

I would estimate that an experienced constructor could assemble a 1010/1100 set-up in about four hours, if he makes no serious errors. Take your time and get it right the first time. Probably best to spread it over two evenings, taking one unit at a time.