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electronics today INTERNATIONAL

JULY 1982 75p

ON-SCREEN EVERYTHING!

A project to use your TV as test gear

An ETI first!
Digital kitchen scales project
Easy-to-build: LCD display: Accurate

Keep an eye on the revs with our robot controller

Down to video basics with the first-ever circuit level description

..AUDIO....COMPUTING....MUSIC....RADIO....ROBOTICS..

KITCHEN SCALES

We now turn our attention to weighty matters. Surely it's time, in these days of digits with everything, that we got rid of the analogue scales readout? You bet it is. Design and development by Rory Holmes.

At last, the electronics enthusiast can make amends for the state of the kitchen table, sinking beneath an ever-growing pile of constructional debris. The ETI Digital Kitchen Scales offer a means of adding a digital readout to an ordinary mechanical pointer type of instrument.

The mechanics of weighing scales are particularly difficult for the DIY approach, requiring a frictionless movement with only one degree of freedom — vertical displacement. We decided to use the ready-built mechanics of a low cost spring movement scale and concentrated on the electronic problem of measuring displacement with high linearity, high resolution, and zero friction!

The resulting design consists of an

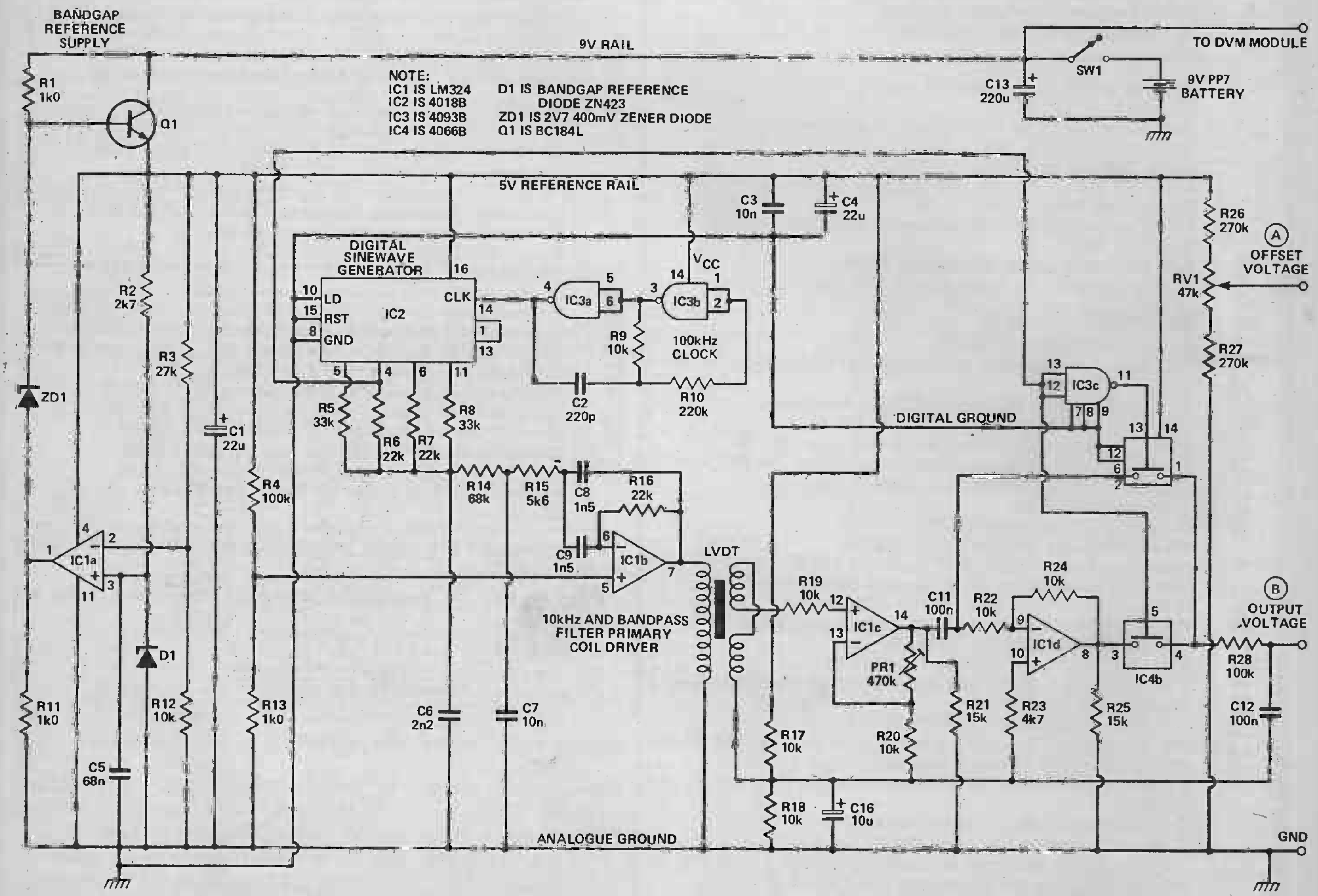


Fig. 1 Circuit diagram of the LVDT and associated circuitry.

HOW IT WORKS

The block diagram of Fig. 5 gives an overview of the circuit operation. Essentially, a Linear Variable Differential Transformer (LVDT) is used as a transducer, providing a voltage proportional to the displacement of its moveable core (a spring movement initially provides the linear displacement with weight). The circuitry generates the LVDT drive waveforms and uses a phase-locked detection technique to recover a stable voltage related to position (and thus weight). The voltage measurement obtained is displayed on a 3½ digit LCD DVM module to give a direct readout in kilograms.

Figure 3 illustrates the principle of the LVDT using an AC excitation signal. All the circuitry on the left of the LVDT shown in Fig. 1 is involved in supplying a stable 10 kHz sine wave to drive the primary coil. To achieve the required amplitude and frequency stability the sine wave is generated digitally using an even length walking ring counter based on IC2, the 4018 divide-by-(2 to 10) synchronous counter. IC2 is configured as a five stage divide-by-10 counter by feeding back the Q5 output on pin 13 to the input on pin 1. The Q1-Q4 outputs are summed with selected resistors R5-8, thus approximating the sine wave. The counter is clocked at pin 14 from a 100 kHz astable oscillator formed from IC3a,b. Since the counter divides by 10 the sine wave generated will always be one-tenth of the clock frequency, ie 10 kHz. The coil excitation frequency thus depends only on the C2/R9 astable time constant, and the amplitude only on the CMOS supply voltage.

The stability of the voltage levels is ensured by using a precision 5 V supply based on the bandgap reference diode D1. The op-amp used to regulate this supply (IC1a) actually powers itself from the 5 V output, thus stabilising its own power rails. A bias current of about 1.5 mA (also taken from the 5 V rail) is fed to the reference diode through R2, to produce an extremely stable voltage of 1V2 at the non-inverting input of the op-amp. The other (inverting) input of the op-amp is taken from the R3-R12 potential divider, the ratio of which sets the 5 V output due to negative feedback around the op-amp and series pass transistor Q1. ZD1, a 2V7 zener diode, allows the output of the op-amp to keep the base of Q1 at 5V6 while operating well below its own supply rail voltage.

The 5 V rail supplies all the circuitry but a separate digital ground is used for the logic ICs. This prevents digital noise from affecting the analogue signal measurement. C1 provides smoothing for the analogue supply rails, while C3 and C4 provide smoothing and decoupling for the digital circuitry.

Capacitor C6 filters the digital sine wave approximation from IC2, which is then attenuated to about 50 mV by the R14/C7 low-pass filter network. The resulting signal, a much better sine wave, is fed to the bandpass filter and coil driver amplifier based around IC1b. IC1b is configured as a standard 10 kHz active bandpass filter and gives a very pure sine wave on its output at pin 7 for driving the LVDT.

The LVDT primary coil has few turns and a correspondingly low resistance of about 4 ohms. Since IC1b (part of an LM324) can only supply about 25 mA of output current, the peak sine wave amplitude driving the coil should not be more than about 100 mV. Also, the output impedance of the op-amp should be very low. This is because the excitation voltage must remain constant as the primary coil inductance changes due to the core displacement. DC coupling is thus used between the coil and the op-amp output.

The sine wave swings ±50 mV about a reference level set at 50 mV above the analogue ground. This is only possible due to the ground sensing capability of the LM324 op-amp. Potential divider R13/R4 directly divides the precision 5 V supply by 100 to provide this reference level at the non-inverting input on pin 5.

The voltage output from the differential secondary of the LVDT (illustrated in Fig. 3) is amplified by IC1c. This op-amp is configured as a non-inverting DC amplifier with a high input impedance and a gain of around 20, the latter being determined by PR1. The 10 kHz sine wave signal is directly coupled from the coil and will be centred around the 2V5 reference rail provided by the potential divider R17, R18. The secondary is wired 'series opposing' such that there will be no signal when the ferrite core is centred.

The phase-locked detection is performed by multiplying the signal by +1 and −1 on alternate half-cycles of the sine wave to produce a bipolar signal centred about the reference level. IC1d, the last op-amp in the LM324 package, is configured as a straightforward inverter, AC-coupled to the sine wave signal. Two CMOS analogue switches, IC4a, 4b, switch the signal either directly (x 1) or through the inverter (x −1) on each separate half-cycle. They are switched alternately, using logic inverter IC3c, from pin 4 of IC2, a square wave output of the digital sine wave generator. This produces the waveforms shown in Fig. 4 since the square wave edges correspond to the zero-crossing points of the sine wave after detection.

The resulting phase-detected signal is low-pass filtered by R28 and C12 to produce a ±100 mV DC voltage, linearly proportional to the displacement of the LVDT. A further voltage is provided by the 10-turn potentiometer RV1 in conjunction with the potential dividers R26 and R27. A reference of ±300 mV (relative to the 2V5 rail) is available at the slider of RV1. The two voltages are fed to the differential input of the LCD panel meter. This allows the digital scale to be returned to zero readout, allowing further measurements when, say, 1 kg is already being registered. The diagram of Fig. 2 shows how the LCD voltmeter is wired up for our application to give a 200 mV full scale deflection (corresponding to 2 kg).

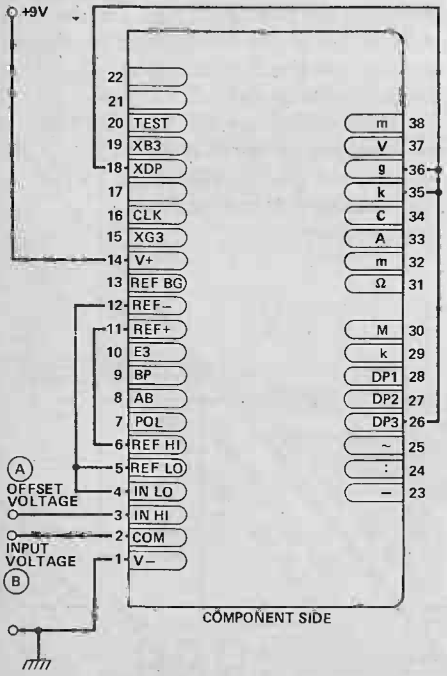
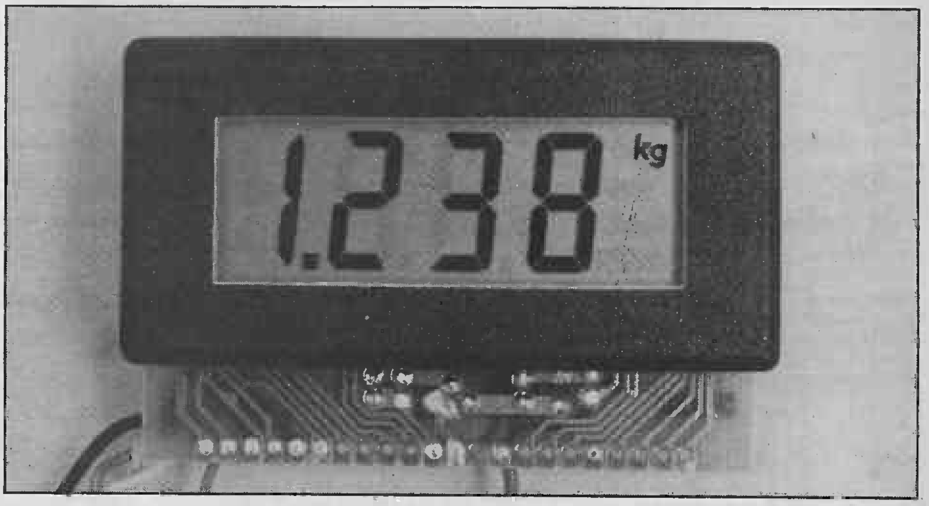


Fig. 2 Connection diagram for the DVM module.



easily wound inductive displacement transducer and the associated drive electronics on a small PCB, all supplied from a 9 V battery. An analogue voltage proportional to weight is obtained, which is then displayed on a 3½ digit LCD panel meter module. Up to 2 kg can be displayed on the scales, but a zero-offset control allows a given weight to be re-zeroed. This provides the useful facility of weighing and mixing ingredients simultaneously — when preparing cake mixture, for example.

The accuracy and resolution obviously depends a great deal on the initial accuracy of the spring and pivot

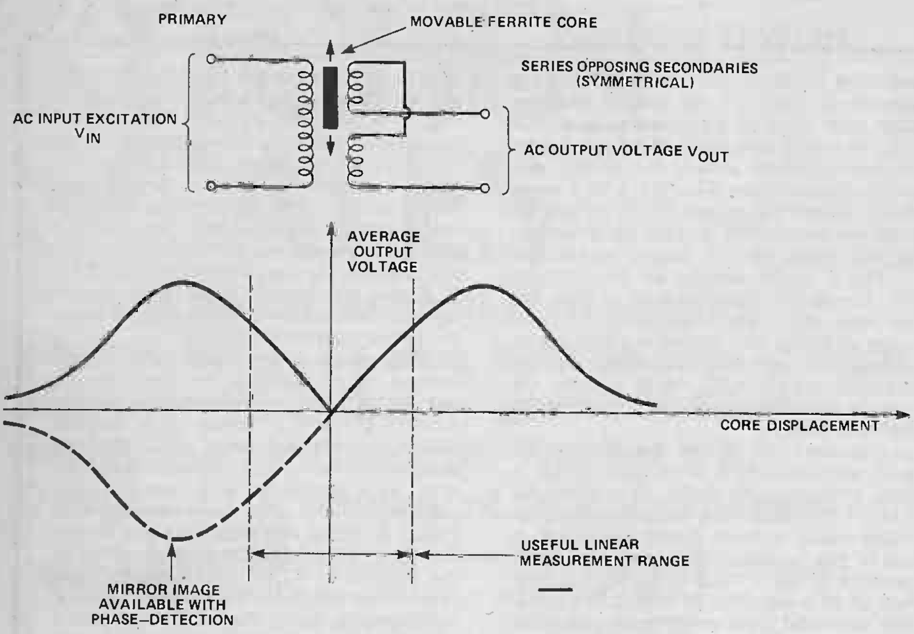


Fig. 3 The principle behind the LVDT, using an AC input waveform.

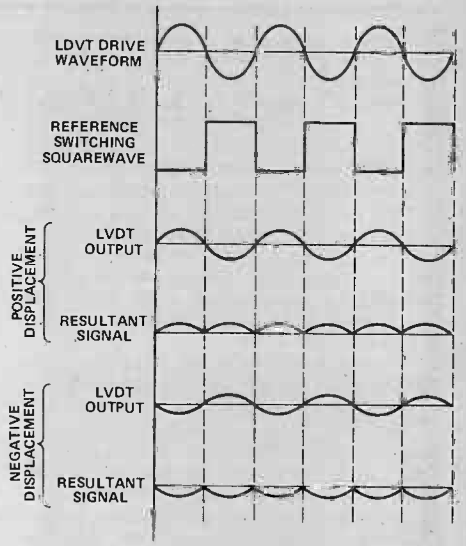
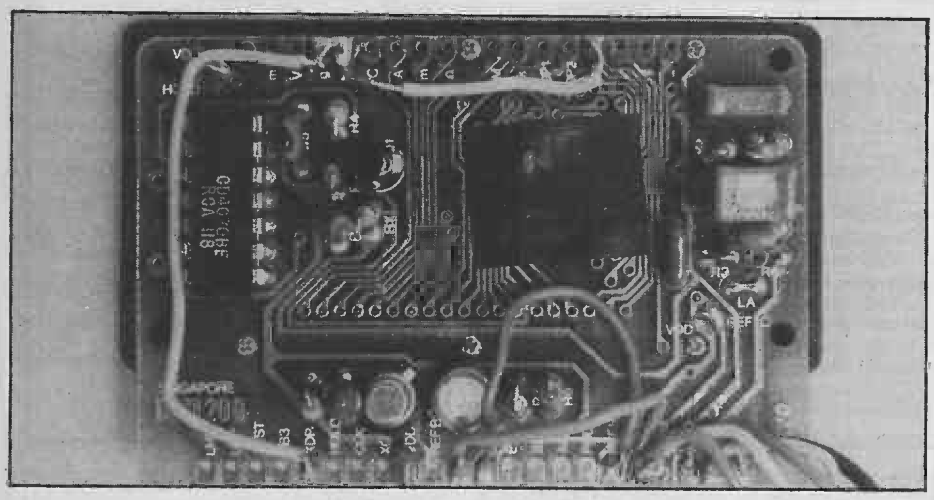


Fig. 4 The output signals generated by the circuit.



A view of the rear of the DVM module showing the wiring used in this application.

system used in the scales, but ¼% (5 grams in 2 kilograms) should be easily obtainable.

The inductive transformer we are using is known as a Linear Variable Differential Transformer, or LVDT for short. These are used extensively in industry for just such applications as this project — weighing machines, load cells, machine positioning and so on. The circuit features some novel techniques for allowing an LVDT of few turns to be used; specifically, a phase-lock detection system based on a digital sine wave generator, and a self-stabilising bandgap power supply for precision voltage levels. The block diagrams and boxed-off text give an explanation of the circuit operation and explain how the displacement measurement works.

Next month we will describe the construction of the PCB with an overlay, and give details of the LVDT coil winding and mounting arrangements.

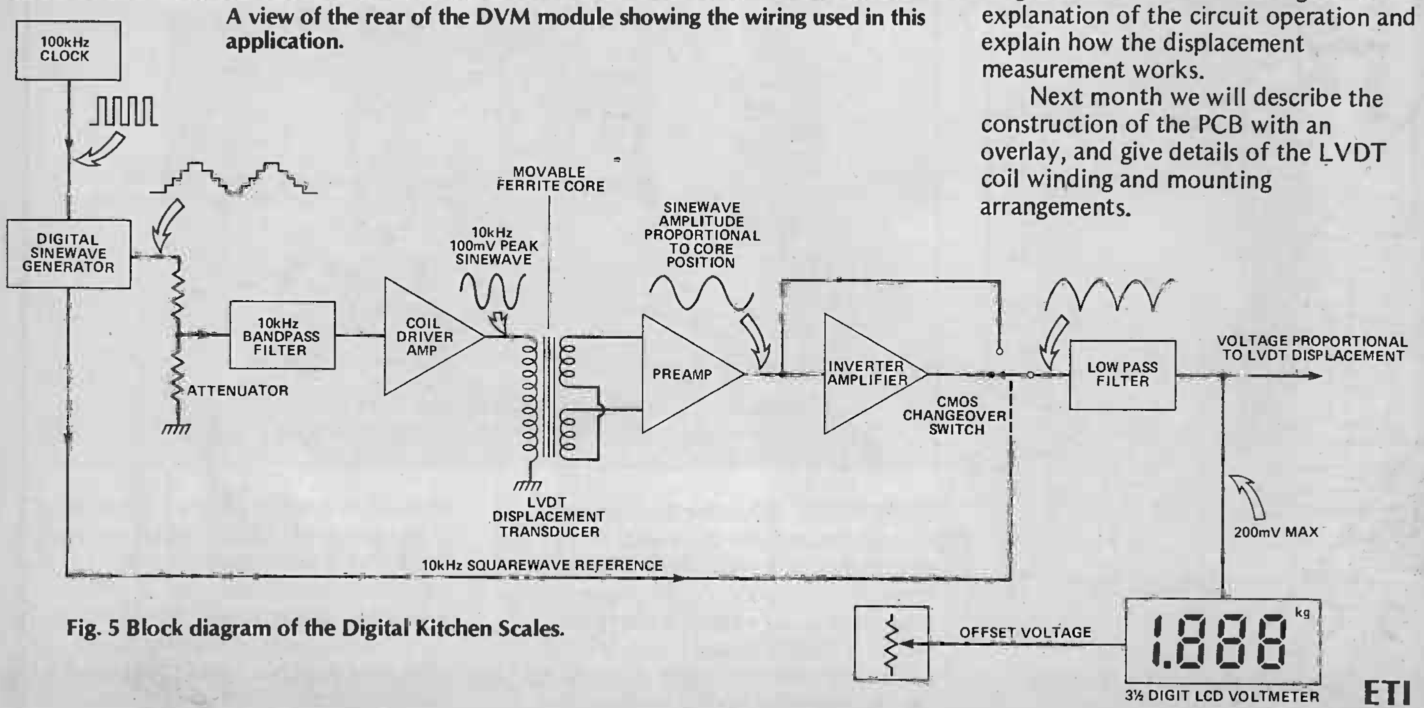


Fig. 5 Block diagram of the Digital Kitchen Scales.

ETI

LOGIC LOCK PART 2

This month we conclude this devilishly cunning device. Full constructional details are given for both the main board and our unique touch keypad. Design and development by Rory Holmes.

The PCB for the combination lock has been designed to fit into a general purpose ABS Verobox (120 x 80 mm); Fig. 1 shows the component overlay. The board should be assembled in the normal fashion, inserting Veropins at the five switch connection points. The five-way Molex PCB sockets and the screw ter4minal output sockets are obviously not essential items, but they simplify installation of the lock and we recommend their use. Orientation of transistors, diodes, electrolytics, and ICs should be carefully checked. IC6 is the opto-isolator; it has six pins, although the PCB allows for an eight pin DIL socket. It should be plugged into the six pins nearest the screw terminal connector. Figure 3 illustrates different options for output circuits that may be used for switching either AC or DC solenoids. An opto-thyristor is used for handling mains-operated solenoids, or lower AC voltages, while an opto-transistor isolator of either single or Darlington type is used for switching DC solenoids. For the DC arrangement a socket may be used and a BD139 transistor should be soldered in as shown on the overlay. If the mains output is used the opto-isolator should be soldered directly to the PCB, and a link is soldered in place of the BD139 across the base and emitter pads.

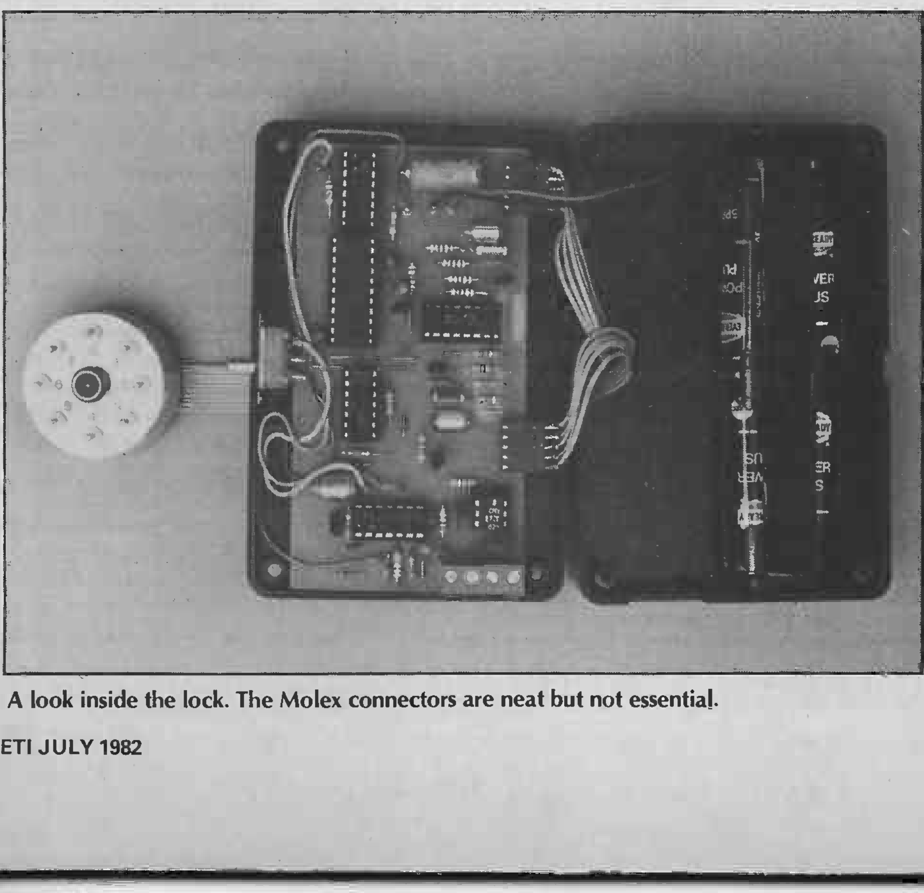
A PP3 battery connector clip should now be wired to the power supply points shown and the slide switch SW1 connected up to the five terminal pins using insulated wire. Before assembling the box, the keypad should be constructed to allow testing of the lock.

Keypad Construction

The construction method we describe here results in a very neat and attractive little keypad, no more obtrusive than the standard Yale lock fitting. The case for the keypad is nothing other than the familiar volume control knob! The hollow plastic type clad in a spun aluminium shell is used, with an external diameter of 38 mm. The internal plastic fixing bush is drilled away to allow mounting of the indicator bezel through a central hole. Ideally, the metal bezel should be mounted with an insulating washer or plastic bush, both to prevent contact with the aluminium casing and to provide the required height of 6 mm.

Figure 2 shows a cross-section of the keypad assembly. Eight holes of 5/32" diameter are now drilled evenly around the centre, for taking the touch contacts. These contact pins are made with standard PTFE-insulated lead-through terminals. The PTFE bush fits easily into the holes and the pins are then tapped home with a hammer, to produce a very firm and waterproof contact. It's a good idea to number the holes on the front panel before mounting the contacts, using Letraset or similar and a protective lacquer spray.

Once complete, a 10-way ribbon cable may be soldered to the contact pins on the inside of the knob. Since the ribbon cable is terminated with two five-way plugs it's important to observe the LED connecting wires. They are the first and last wires in the ribbon, the LED anode going to the +6 V rail on SK1. The anode should also be soldered to the LED bezel to form the central touch contact. The two plugs wired on the other end of the cable should be marked in some way to identify which way round they fit the sockets.



A look inside the lock. The Molex connectors are neat but not essential.

Time To Test It

Having assembled the keypad it can be plugged into the main PCB for testing. Set the switch SW1 into the normal 'operate' mode, and put a 6 V battery on the connector clip (four 1V5 AA cells, in the long flat type of plastic battery holder). Now, put the switch in 'program' mode and then back to normal. The LED will illuminate for about six seconds and then go out, ensuring that the lock has reset to its rest state. On connecting new batteries the memory should be zeroed — switch to 'program' again and enter 18 zeroes through the keypad, then return to 'operate' mode.

A combination sequence can now be entered by switching back to 'program' mode and touching both the required contact number and the central bezel with two fingers. (The skin resistance must bridge the contact pin to the +6 V rail on the LED bezel; it can be done with one finger, but it's easier with two). Any length of sequence up to 14 digits may

PARTS LIST

Resistors (all ¼ W, 5%)

R1-9	10M
R10,11	3M3
R12	47k
R14	100k
R15,18,19	22k
R16,20	220R
R17	2M2

Capacitors

C1-8,15	10n ceramic
C9	2u2 35 V tantalum
C10	33n ceramic
C11	1u5 35 V tantalum
C12,13,15	2u2 ceramic
C14	68n ceramic
C17	220u 16 V axial electrolytic

Semiconductors

IC1	4532B
IC2	4070B
IC3	5101
IC4	4029B
IC5	4093B
IC6	H11C4 (mains) or CNY17/1V (DC) or TL114 (DC)
Q1,2	BC214L
D1-7	1N4148
LED1	3 mm green LED with bezel

Miscellaneous

SW1	DPDT slide switch

PCB (see Buylines); two off five-way 0.1" PCB plug and sockets (Molex type KK); PCB-mounting screw terminals; short length of 10-way ribbon cable; knob for keypad (16 x 38 mm diameter, aluminium clad); eight off PTFE-insulated leadthrough terminals; four off 1V5 AA batteries, flat style holder and battery clip; case (ABS Verobox ref. 75-2860J, 120 x 80 x 35 mm); solenoid door bolt.

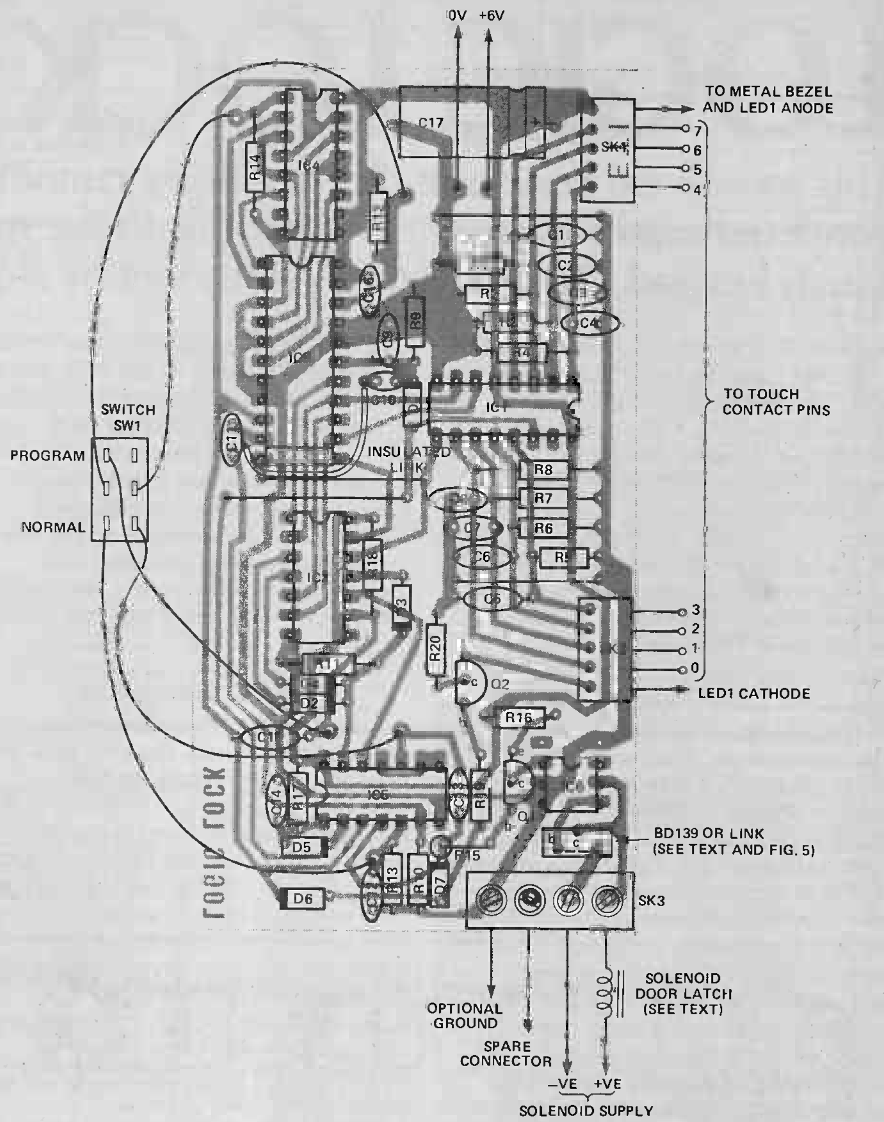
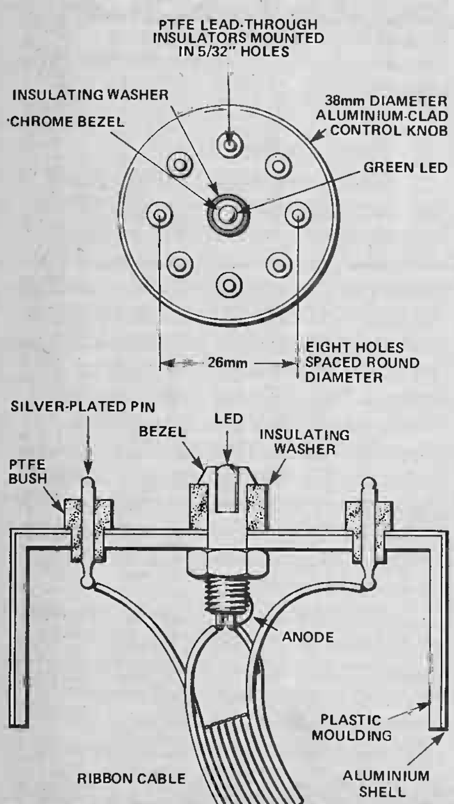
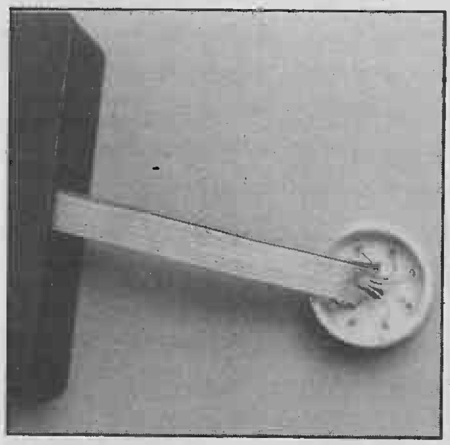


Fig. 1 Component overlay for the Logic Lock PCB.





A ribbon cable links the keypad to the main unit.

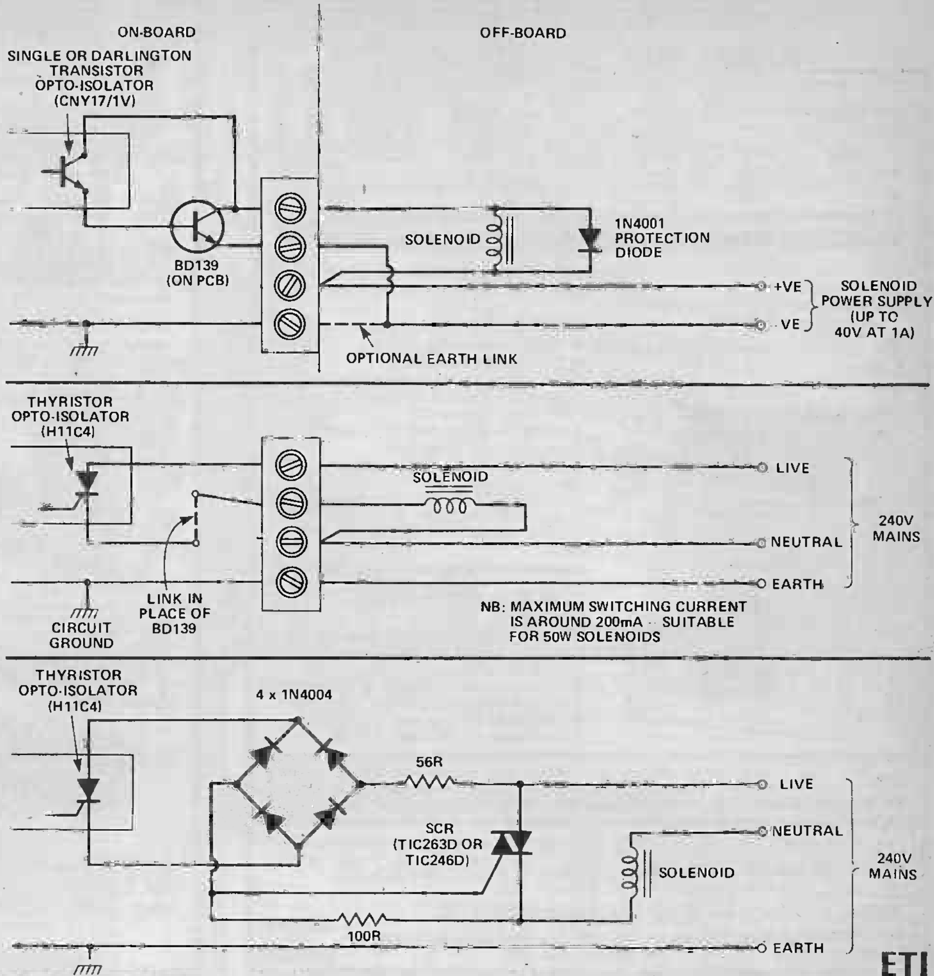
Fig. 2 (Left) Constructional details for the keypad. The LED indicates that an input has been accepted.

be programmed, with any number of repeated digits. After entering your last digit, switch back to 'operate'; the LED comes on again for six seconds to mark the end of sequence and reset the circuitry.

Now comes the moment of truth! Re-enter your combination sequence through the touch pins. As the last number is entered the LED should illuminate, signifying the solenoid activation period. For obvious reasons, errors are not indicated; the lock simply resets. If an error is made or the lock doesn't appear to open just tap in the code again from the start.

Once you are satisfied that the lock is working correctly, the main PCB should be secured in the lid of the Verobox using adhesive pads. It fits exactly between the corner pillars as the internal photograph illustrates. The switch is also secured on the lid alongside the PCB using Superglue. It

Fig. 3 (Right) Three possible output circuits that may be used to control mains or DC-driven solenoid bolts. Other possibilities are left to your ingenuity!



should be raised up on a paxolin spacer in order to clear the rim, with an appropriate slot cut in the main box for the toggle. A thin slot for the ribbon cable is also cut on one side near the base, allowing the five-way plugs to be pushed through one at a time. The specified battery holder fits into the bottom of the box, on the opposite side to the screw terminals. The batteries should be inserted first before securing with adhesive pads; make sure you allow room for the connector clip. The solenoid and power connecting wires pass through another slot cut out next to the screw terminals.

BUYLINES

All the hardware for this project (with the exception of the Molex connectors, which aren't essential) is available from Electrovalue. The board can be obtained from our PCB Service on page 71, while the solenoid door bolts are sold by BSG (Security) Ltd., 34/35 Dean Street, London W1V 5AP. Contact them for details.

AUDIOPHILE

Welcome to a geometrically perfect Audiophile. This month Ron Harris has been listening to the hyper-elliptical Shure V15V cartridge and the 'Carver Cube' amplifier. They went down a treat with his speaker cones.

This was to be the month of the great amp line-up, but the launch of a new V15 model is of sufficient interest — and rarity — to warrant immediate attention, is it not? Dissenting voices may pass (quietly) across to the Carver write-up also contained within these pages. See if I care that you choose to ignore half my erudite words of wisdom. Anyway, putting the amp comparison off a month gives me four more weeks to play with some of the best units on the market. . .

V15V Or Four And A Half?

Not a few people find the release of the V15V a little surprising — me included. For one thing it comes only a short time behind the MV30H — a version of the V15IV. Is that to fade away so soon, never really having had time to shine? For another, the Five will cost around £130 over the counter and that is quite a pile of green stuff more than the IV. That price also puts it firmly into competition with some excellent £100+ pickups — the Karat Ruby to name but one.

Herein lies the biggest question mark, for the V15V is *not* a moving-coil, despite the ever-circulating rumours that Shure were about to nail down their magnets and begin waving wire with the best of 'em!

With the hold that moving-coil pickups have on the top-end hi-fi market at present, it is a brave move to aim your new flagship straight into battle with them like this.

There are (inevitable?) similarities between the Five and its predecessor. It carries the stabiliser assembly, for one. Output level is similar, as is compliance. Once again this is a unit for low mass arms and low tracking weights; a combination which may be interpreted as a defiantly rude gesture in the direction of the massive arm and everything-must-be-heavy-to-be-rigid doctrine presently in vogue.

Five, Five, Five

The refinements are legion however. The greatest single advance lies in the cantilever system. In the V15V this is formed from a tube of pure beryllium. The tube wall thickness is only 0.005", or about 1/6 that of a human hair, as Shure would point out. (I love these silly comparisons! Ever tried playing a record with six hairs twisted together?)

The effective mass of this is reduced to the point where the resonance rises to beyond 33 kHz as opposed to the usual 19-24 kHz. This flattens top-end response and will tend to improve h.f. separation within the audible range.

Another improvement is the polish used to finish the stylus. Christened the 'Masar' polish, it is claimed to be orders of magnitude better than all that has gone before. Better finish means lower record wear and less surface noise.

As a final mechanical touch, the cantilever itself has been lengthened, thus allowing the vertical tracking angle to be altered. After some careful considerations, including a determination of exactly what cutting angles are used by the major record producing companies, Shure have gone for a VTA of 23°, higher than is considered conventionally accurate. They say this provides a better match to actual playing conditions and thus lower distortion in practise.

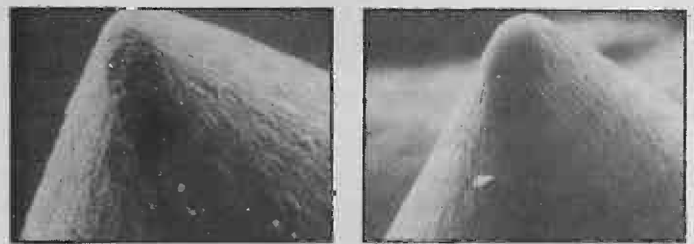
BASIC PROPERTIES OF MATERIALS

MATERIAL	MODULUS (STIFFNESS) dynes/cm² x10¹²	DENSITY grams/cm³	RATIO (MODULUS/DENSITY) cm²/sec² x10¹²
ALUMINUM	0.72	2.70	0.27
BERYLLIUM	2.9	1.85	1.58
BORON	5.5	2.53	2.18
SAPPHIRE	3.3-3.9	3.9-4.1	0.93
DIAMOND	7.4-10.5	3.15-3.5	2.88

	18 mil. O.D. 1/2 mil. wall	14 mil. O.D. 1 mil. wall	12 mil. O.D. 2 mil. wall	10 mil. dia. rod
RELATIVE LINEAR DENSITY (PROPORTIONAL TO EFFECTIVE MASS)	0.35	0.52	0.80	1.00
RELATIVE STIFFNESS	2.13	1.74	1.64	1.00
STIFFNESS/DENSITY RATIO (RELATES TO RESONANCE FREQUENCY)	6.25	3.44	2.06	1.00

COMPARISON OF SOME POSSIBLE BERYLLIUM GEOMETRIES

Left and above: a comparison of materials and cross-sections used for stylus production. Having decided to use beryllium, Shure claim a great advantage for their thin-walled tube approach over that of the 'solid rod' employed by some competitors.



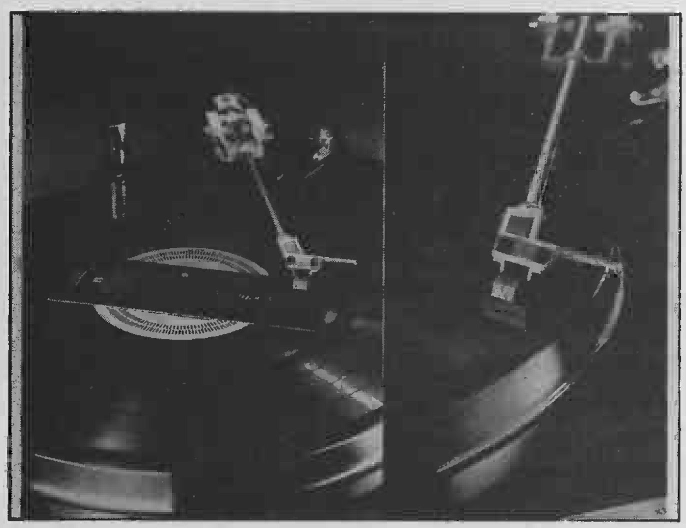
Above left: a conventionally polished stylus tip under extreme magnification. Above right: the V15V MASAR polish under the same power magnification. Note how much smoother the contact area appears. This could lead to lower surface noise and reduced record wear.

Electrickery

Electrically the new V15 exhibits less load dependence than previous models. The inductance has been lowered significantly such that interaction between the cartridge and capacitive elements of the amplifier input is reduced.

This filtering effect is one of the better theories advanced to explain the superiority of moving-coil cartridges in some parts of the audio spectrum. They have far less inductance as a rule, since the coils are kept low in turns to help keep down the moving mass. With moving-magnet designs there is less mechanical reason to limit the output voltage (dependent upon the coils) in this manner.

The earlier V15's, especially the III, were rather prone to interaction with the input capacitance. The cartridge required a large amount of capacitance to prevent a high frequency rise and a subsequent subjective 'hardness' upon which many commented, but few bothered to investigate.



A curious photo from the Shure publicity handouts — a V15V in a very strange record deck! It does show the alignment protractor in use though. (No doubt I shall receive irate epistles from said deck's producers now, berating my comments. . .)

Packaging Alignment

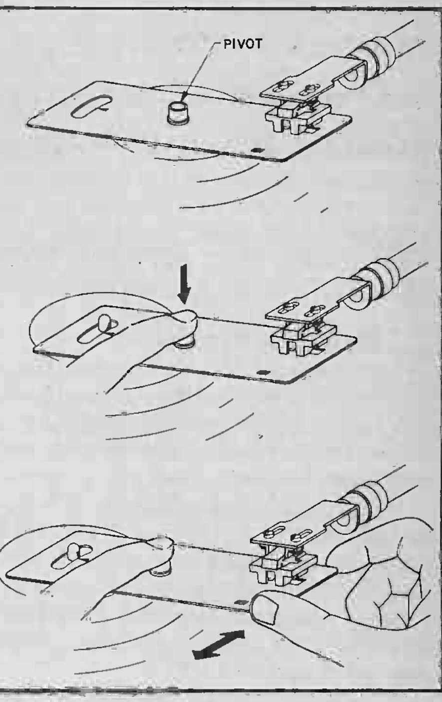
Part of the 'package' of the V15V is that the box itself forms an alignment protractor to simplify the fitting of the cartridge into the pickup arm. An "alignment-stylus" is also present, which greatly simplifies the levelling of the cartridge with respect to the record.

Shure's protractor is a 'two-point' alignment system which operates by holding the cartridge body in a precise location, whilst the arm is set-up for position. The system works well for arms which have slots in the headshell to allow alignment of the cartridge, but with such as the SME Series III, where the entire arm is moved back and forth and the headshell has no adjustment, I found the system a little awkward to use properly.

With the SME especially I found the usual alignment protractors more accurate and just as easy to use. I suspect, however, that this is as much due to my having set up an SME so many times that I could do it behind my back — wearing a blindfold whilst falling off a log in a storm downstream from a waterfall in the dead of night.

The Shure system is a great step forward inasmuch as it provides a universal method of easy alignment which requires no special tools or knowledge. As it comes free with the cartridge you've nothing to lose by trying it, anyway!

Using the new gauge has been designed to be as straightforward as possible. The cartridge body is left fixed into the packing it arrives in, and the headshell is adjusted until the protractor indicates alignment at two points on the record surface. Theoretically this will ensure good alignment across the whole record surface. Users of arms such as the SME, in which the whole arm is moved rather than the headshell, may find it a little tricky, however.



The "alignment-stylus" is simply a plastic straight-edge which fits into the body in place of the real thing and allows the user to set the headshell such that it and the cartridge are parallel to the record surfaces. It works well and is another of those clever little ideas that someone should have thought of a long time ago.

In addition V15V owners will receive a copy of the new Shure test record, upon return of the card packed with the cartridge.

Up In Arms

With all this technology and innovation going for it, the Five is clearly an expensive move for Shure and one that deserves serious consideration. As I said earlier, setting up the Five is straightforward, however you do it.

The compliance is high, around 30 c.u. and I could not recommend the use of this cartridge with other than a low mass arm, else the arm/compliance resonance could rise into unsuitable regions.

I tested the V15V in the inevitable SME Series III, to which it is perfectly matched. Allowing for the stabiliser the V15V tracked perfectly at 1 gram. (The stabiliser means that actual downforce is set at 1g5.)

Shure have invented a thing called "Total Trackability Index"; designed to show how much better the V15V is than anything else on the market. They don't actually say that themselves, of course, but I can see no other use for it! Americans do possess this in-built love of irrelevant numbers and statistics. Give them a subject — any subject — and they'll surround it with tables, percentages, averages, indices and other complications which add little to actual understanding. Serve us right for ever giving them independence, I suppose.

Resulting Results

I reproduce Shure's own test results herein, both to show how neatly presented they are and because I could not turn up any significant differences under test between my figures and these. Unusual that, as test methods will usually produce some variations, however minor.

As you can see the Shure acquits itself well technically with little or nothing to criticise. The lower inductance means that capacitance loading can vary from 100 pF to over 400 pF with no perceptible change in performance. The much improved rigidity of the cartridge body allows for better coupling to the headshell and seems to 'clean-up' results on a sweep test, measured against a V15IV.

Subjective testing revealed a character quite unlike any previous Shure cartridge I have heard! At first I had the unworthy thought that they were buying cartridges in and stamping 'Shure' all over them.

I hasten to add that I don't think for one second they are, it is just that it really *is* that different! Perhaps the easiest way to describe the change is to say that the difference between the V15IV and this new Five is very much that to be expected between otherwise matched units which are of the moving-magnet and moving-coil varieties respectively.

SHURE — SERIAL NUMBER 006359 — V15 Type V

22305

OUTPUT LEVEL @ 1 kHz, 5 cm/sec peak:	
LEFT CHANNEL	3.39 mV
RIGHT CHANNEL	3.61 mV
CHANNEL BALANCE @ 1 kHz:	0.6 dB
SEPARATION @ 1 kHz:	
LEFT CHANNEL	32.6 dB
RIGHT CHANNEL	30.2 dB
SEPARATION @ 10 kHz:	
LEFT CHANNEL	26.7 dB
RIGHT CHANNEL	19.6 dB
PHASE	CORRECT
400 Hz TRACKABILITY	>25 cm/sec peak @1 gram

FREQUENCY RESPONSE (Relative to 1 kHz)

LEFT CHANNEL		RIGHT CHANNEL	
1 kHz	0.0 dB	1 kHz	0.0 dB
2 kHz	-0.1 dB	2 kHz	-0.1 dB
4 kHz	-0.3 dB	4 kHz	-0.4 dB
6 kHz	-0.5 dB	6 kHz	-0.5 dB
8 kHz	-0.5 dB	8 kHz	-0.6 dB
10 kHz	-0.5 dB	10 kHz	-0.6 dB
12 kHz	-0.5 dB	12 kHz	-0.7 dB
14 kHz	-0.6 dB	14 kHz	-0.6 dB
16 kHz	-1.0 dB	16 kHz	-1.1 dB
18 kHz	-1.1 dB	18 kHz	-1.2 dB
20 kHz	-1.4 dB	20 kHz	-0.9 dB

QUALITY CONTROL APPROVAL ___HK___

Above: the V15V test results as supplied. Our lab found little to disagree with — and these are prettier than ours!

The Five has much of the mid-range quality of the very best moving-coils and a very good tight bass response. The treble is wide-open and clear and does seem to be less dogged by surface noise than other cartridges. (A more polished performance?...)

In short the Five is a very good design which should appeal equally to followers of both types of cartridge. I was able to A/B the V15V against the Dynavector Karat Ruby, which is probably one of its strongest competitors. Personally, I found the two indistinguishable in the mid-range, with the Shure tracking better and having a more extended and better controlled bass but with the Ruby showing a greater imaging capability and more extended treble.

Overall I would take the V15V, but personal preferences will dictate which performance parameters are more important to which listener.

Conclusions

Well, what can I really say? The Five is a highly refined design which offers a lot for its price and which looks as though it could be a serious challenger to pickups already in the £100 price range! It offers "moving-coil clarity" and "moving magnet" security!

A welcome addition to the hi-fi scene then, and one which is a radical improvement over previous models of this famous line.

●

Good things come in small packages?

Carver's Cube

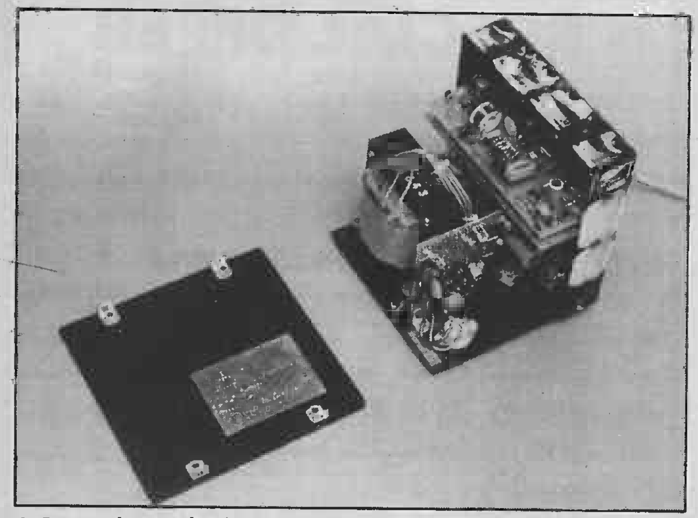
At long last I managed to lay hands upon Carver's M-400 amplifier, irreverantly known as the Cube. Many moons ago ETI ran a feature upon the internal mysteries of the beast, but none of us has had the chance to listen to a Cube in other than exhibition conditions, or to put the amp through some tests ourselves.

Finally, as part of the amplifier comparison we're working on, an M-400 was wrenched from the death-like grip of Carver's PR Company and rushed off to a test bench where they couldn't find us for awhile. Furtive this hi-fi game, sometimes.

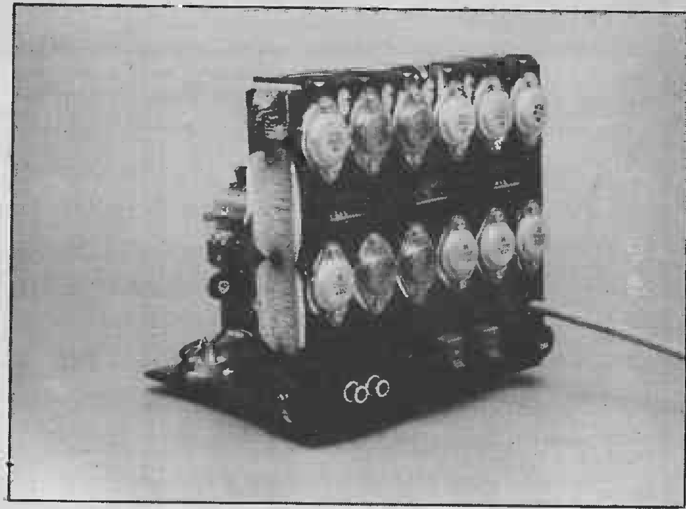
As you can see from the photos, the Cube is really tiny for all its 200 W per channel RMS abilities. Unusual, too, is the PSU arrangement, which is configured with the two channels connected in anti-phase to each other internally. Could cause havoc with some speaker switching arrangements, that could.

As it is it complicates testing a little. Carver have included some little hints in the manual as to how the M-400 should be bench tested.

Kind of them, that.



A Carver de-cased! Note the huge inductor (rear left) and the cleverly-mounted power meter on the back of the front panel.



From this angle you can see the ranks of output transistors lined up along the chassis. Ignore the white gunge, it's heatsink compound.

Room For Manoeuvre?

This time there was no way I was gonna hang around with the Cube in its box awaiting lab-time. Once it arrived at ETI, it was photographed hurriedly then whisked off into the mists of Kent to be listened to. (Would it were I could have taken the lady in the photos along with it. . .)

The system surrounding the M-400 was a pair of KEF 105 II's, a Denon PRA 2000 — rapidly establishing itself as the best preamp in the Universe — and a Thorens TD160S/SME III/V15V front end. There was no shortage of power-amps against which to measure the Cube — anything from a Denon 180 W Class A, to a Hitachi 100 W MOSFET design which can presently be found undergoing long-term test for our forthcoming amp comparisons.

In operation, Carver's miniscule monster proved to have a few little foibles. Rather like the late unlamented valve amps the Cube gets better after it's been running a while.

At first switch-on it can sound positively hard and rough on awkward signals, but leave it going 15 minutes or so and the change is remarkable! The upper mid-range smooths out and the transfer function becomes wholly more linear.

A decidedly odd little quirk this, one I have never encountered before in modern amplifiers and one which is damned difficult to pin down on the test bench, of which more later. The Carver does have a distinct personality of its own and it is perhaps accurate to dub this personality 'enthusiastic'! The Cube will deliver power into any load in prodigious amounts — over 520 W on my usual burst-test scales — without ever sounding strained. It projects the music forward as though eager to have you listen to it and rattles windows with surprising rapidity the first time any real bass appears on the recording.

Character Reference

Usually hi-fi components with this type of forward presentation are dubbed 'bright', but that implies a frequency balance tending to favour the top-end of the audio spectrum and that the Carver does not have. Once it has "warmed up", as it were, the Cube is as linear a device as you'll find anywhere. No part of signal sounds to be emphasised over any other — it's just that all of it gets thrown out of the speaker boxes and laid out before you, larger than life!

Whilst being initially unsettling, this presentation is very easy to become accustomed to, and makes other units sound flat and lifeless and lacking in dynamics thereafter. The M-400 would be an easy amplifier to become partisan about, I suspect.

Testing Failures

No, not the Cube, me. I tried in vain to get some meaningful reasons behind this 'warm up' syndrome down on my results sheets, but apart from some non-linearity in the output stage early on and that at a low level, I failed. The effect goes unproven, therefore, and you will have to listen for yourself to see if you agree as to the magnitude of its existence.

Other tests give the Cube a formidable specification. Burst power over 520 W, noise below the floor of my instruments with distortion barely above it, and a protection circuit which was unfoolable on the bench. It ignored 'music-type' peaks but rapidly shut-off anything remotely resembling trouble.

I would take Carver's claim of "an intelligent PSU" with a sizeable pinch of salt, but all the same the protection is excellent. (It was impossible to clip the M-400 under any real conditions anyway.)

Conclusions

Very entertaining indeed! An approach to hi-fi amplification which is both novel and effective. The Carver M-400 is not the most neutral amp I have ever heard but it is one of the most enjoyable to listen to. It possesses a clear character of its own and prior audition is vital to any intending purchaser.

The advantages of the design are enormous, tiny size, high power and cool running, low distortion and lower noise and a reasonable price tag. I hear Carver are launching another version, the M-1.5, which is rated at 750 W RMS per channel and is barely larger than the M-400.

Since, as I said earlier, it was well-nigh impossible to clip the Cube, the advent of this new powerhouse can only mean it's time to head for the bunkers, ear defenders in hand. **ETI**

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The TV display on the cover is an artist's impression. The project generates a black-and-white picture only.

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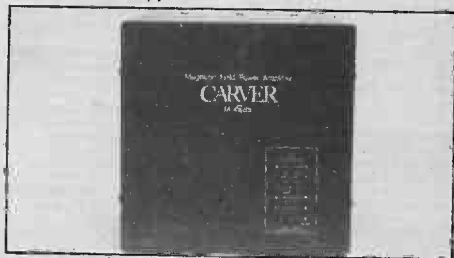
FEATURES



PROJECTS

INFORMATION

TV BARGRAPH

Lots of information to display? Make better use of the box in the corner; statistics, vital or otherwise, look good on the ETI TV Bargraph. Design and development by Phil Walker.

It's been some time since we did a really *unusual* piece of test gear. This month we are setting out to provide a solution to the problem of displaying many variables with good resolution. This sort of requirement can arise with spectrum analysis, statistical measurements, multi-point measurements of temperature, pressure, humidity, speed, current, voltage or indeed anything which can be converted to a proportional voltage.

It was realised some time ago by the electronic games and home computer manufacturers that most of the households who would be interested in their products would own at least one television set. This, we feel, is probably also true for you, our readers. So we have devised this instrument with that in mind.

The main problem in using a television set is that to obtain a high quality display on it the various synchronising signals normally produced by the gentlemen and ladies of the BBC and IBA must now be produced by our humble selves.

At this stage, we can go one of two ways. There are on the market several specially made integrated circuits for controlling VDU systems for home and commercial computers. However, this time we decided to steer clear of these and stay with gates and counters in the standard CMOS range.

The object of this project is to generate a display on a television screen consisting of a number of vertical columns. The height of each column is proportional to a specific input voltage. The columns may be upwards (positive) or downwards (negative) from a reference level. This reference may be changed if only positive-going signals are to be processed.

In order to generate the sync pulses we must first have some idea of their structure. Figure 1 shows the pattern of synchronising pulses aimed for in this design.

A normal TV picture consists of a total of 625 lines. Of these, 312½ are scanned each 20 mS such that each block covers the whole display screen, but the lines of one block fall neatly between those of the next. This reduces the impression of lines across the screen while also preventing objectionable flicker effects.

The apparent complexity of the sync pulse pattern is designed to ensure that a normal TV set can pick out accurately the right moment for line and frame flyback. The last thing necessary for a complete picture is tht the video signal shall appear at the correct times between the line sync pulses, and be blanked during the line flyback and frame flyback periods.

The Circuit

The basic timing for the whole circuit is derived from a 2.5 MHz crystal oscillator. This was found to be essential as minute changes on

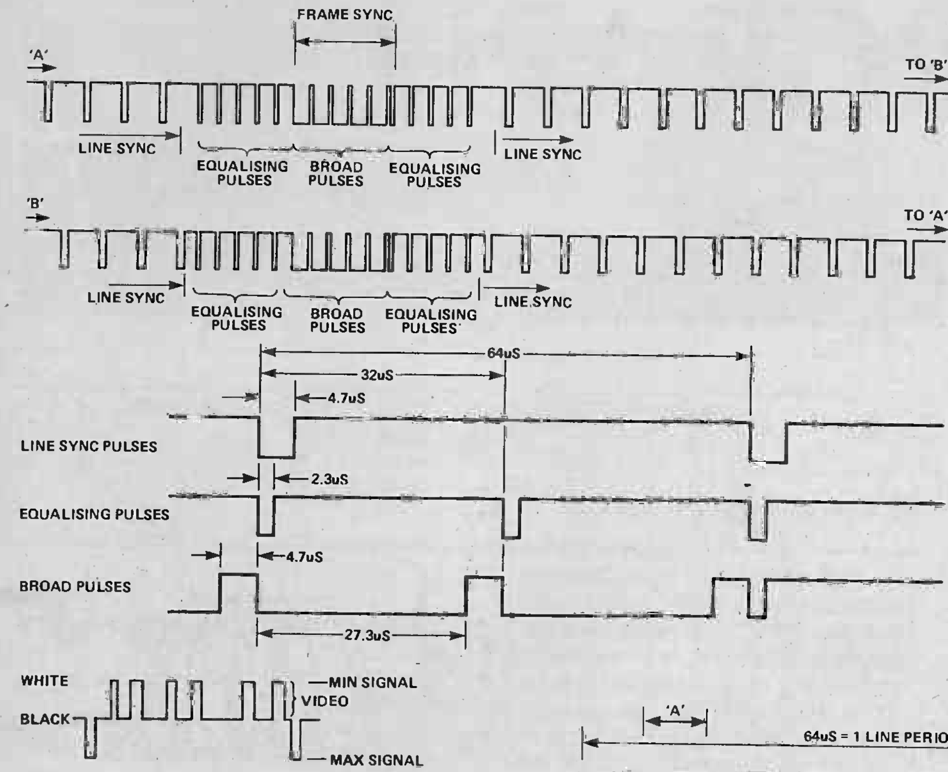


Fig. 1 Sync pulse structure.

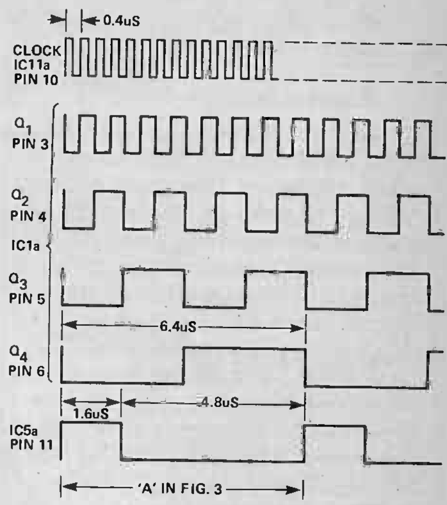


Fig. 2 Expanded section of part of Fig. 3.

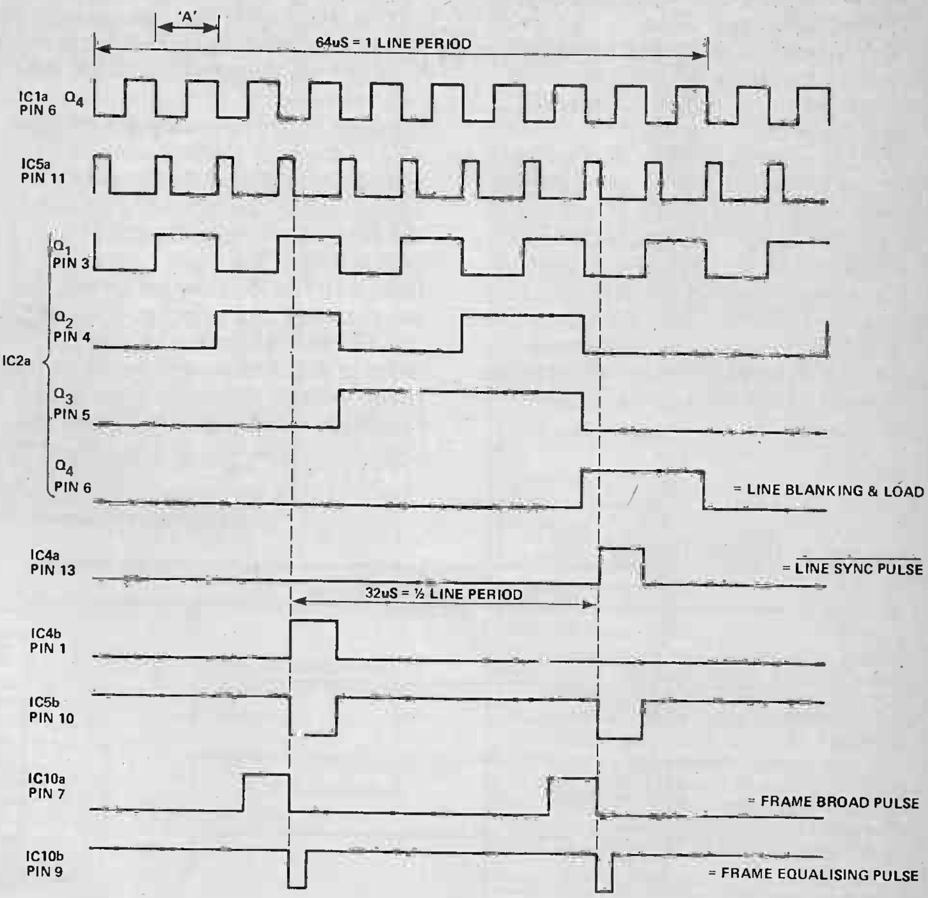


Fig. 3 Sync pulse generation.

the power supplies caused very visible display distortions when a free running oscillator was used. This 2.5 MHz signal is divided by 160 to give the basic line time of 64 uS. The logic associated with the line dividers gives two pulses per line period at IC5b which trigger the 'equalising' and 'broad' pulse monostables. The 4098/14528 dual monostable device (IC10) was used because derivation of these signals with normal logic would be very complicated (see Figs. 2 and 3).

The line sync pulse is generated by the logic (IC4a) and a choice between this and the other two pulses is effected in ICs 5c and 9.

The double line rate signal at IC5b drives a divide-by-5 prescalar (IC2b) and then a divide-by-125 device, IC3 (see Fig 4). This produces the 20 mS period for the frame scan. By virtue of the fact that the input frequency to these dividers is twice the line frequency and the total division ratio is 625, the number of lines scanned is 312½ in each vertical scan. This means that successive scans will be offset by half a line pitch vertically. The generation of the frame sync signal is accomplished by IC1b, IC7 and IC8.

When IC3's count reaches zero its output goes low for one input clock period. This causes IC1b to be reset to zeros and then to start counting and select the components of the frame sync at the appropriate times. When IC1b reaches its all 1s state, further counting is inhibited (Fig. 5).

The video generation is performed mainly on the channel cards. These consist of eight comparators and an eight-bit shift register on each card. The cards are controlled from the main board by a clock signal which determines the number of bars to be displayed. The load signal is the same as the line blank signal. This effectively causes all the channels to be sampled (in the line flyback period) and then displayed.

The other necessary signals provided are the reference and a ramp signal to which the inputs are compared (see Fig. 5).

Construction

In order to keep costs down the PCBs for this project are single-sided. However, this has meant a number of links on the boards. Most of these are used to complete the power supply lines to the devices. The actual construction of the boards is not difficult but we suggest that IC sockets would be a good idea. Most of the links on the board can be put in place before any other components although the one near IC11 and C1 should be left a little longer to allow C1 to be inserted. Insulated wire is recommended for all the links. The link from CLK to 8, 16, 32 or 64 should be made when the number of channels is known. It is easier in fact to put thick wire posts or PCB pins through these holes and link up afterwards.

The eight channel position is useful for setting up the complete system initially and making sure that it works.

It is most desirable that a fine-tipped soldering iron is used during construction and that a check is made for solder bridges, especially where tracks lie between IC pins. It is also most important that all polarised components are fitted the right way round. Especially note that the TL084's and the 4014 on the channel cards are not the same orientation.

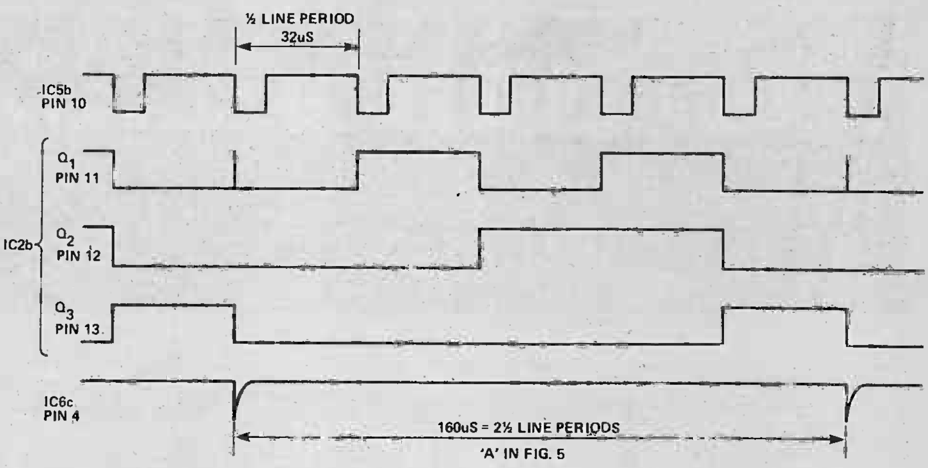


Fig. 4 The divide-by-5 effect of IC2b.

The completed boards in our prototype were fitted into a metal case made by Newrad; a small aluminium bracket was needed to support the rear of the main board as this overhangs the integral chassis member by about 1 cm. Support for the channel cards is by two pieces of PCB material with suitable holes drilled in them. One end of these members is bolted to the main board while the other is supported on pillars. When fitting the boards it is essential that no part of the 0 V supply line gets linked to the metalwork as this will short-circuit the reference supply.

On the channel cards, it is helpful if the channel inputs are fitted with PCB pins or similar to facilitate connection from the top of the board. The 2.5 mm jacks on the front panel used in our model could well be replaced with nine-way 'D' range connectors or anything else of that type, if desired.

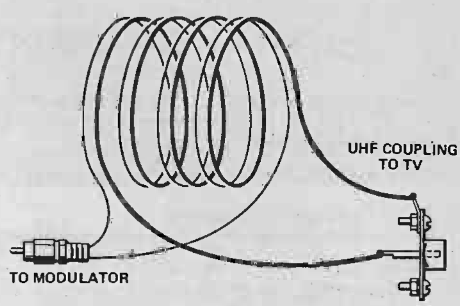


Fig. 6 The UHF coupling coil.

The output from the UHF modulator should be DC-isolated from the panel and our method, shown in Fig. 6, was to take about 20 cm of thin twin flexible wire and coil it about itself to a diameter of 1.5 mm (this makes a simple 1:1 transformer). Connect the ends of one wire to the coax socket pin and skirt, and the ends of the other wire to a phono plug pin and skirt. A possible improvement here would be to use a special two-hole ferrite core with two windings of four or five turns through the holes. Capacitance coupling of the UHF signal can cause stability problems in the voltage reference amplifier. Oscillation here may appear as broken or strange-shaped bars or possibly break-up of the picture.

Connection between the main board and channel cards and card to card is by 10-way jumper wire (or just link them direct). This is probably easiest done with the cards mounted on the support bars. Each channel card adds eight channels to the units' capability.

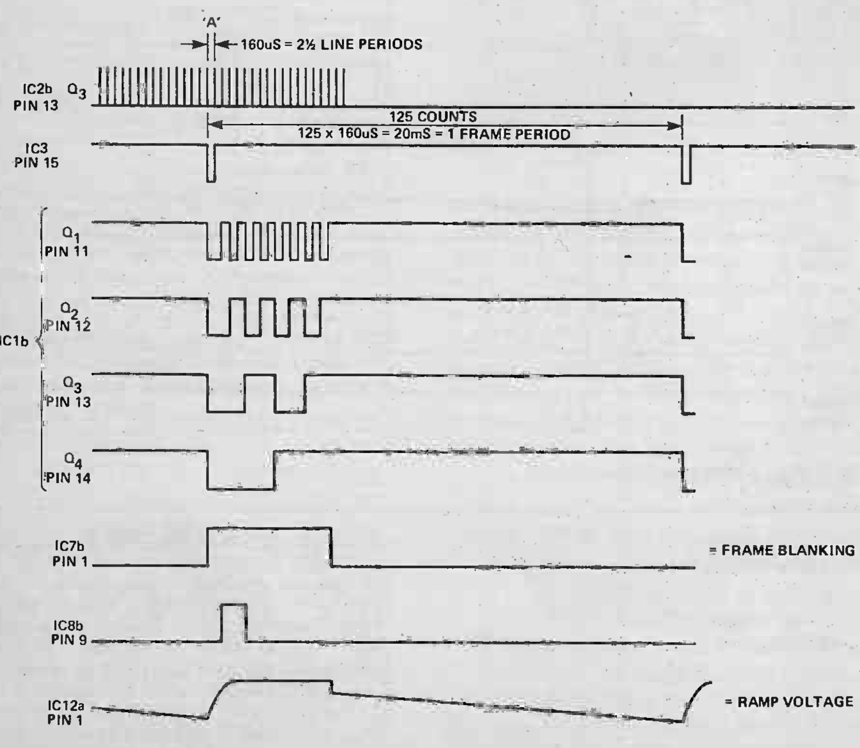
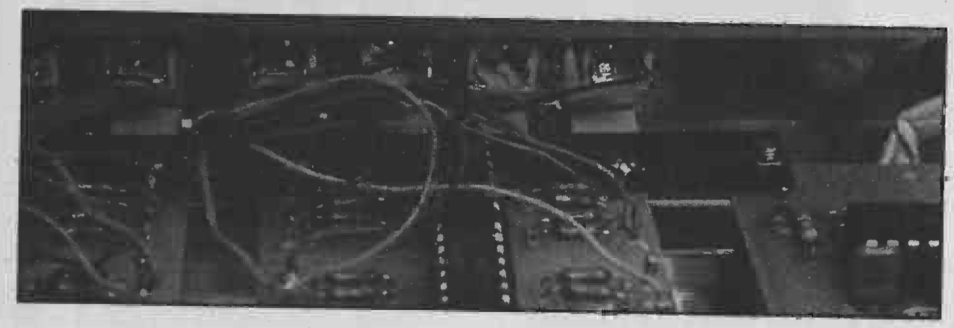
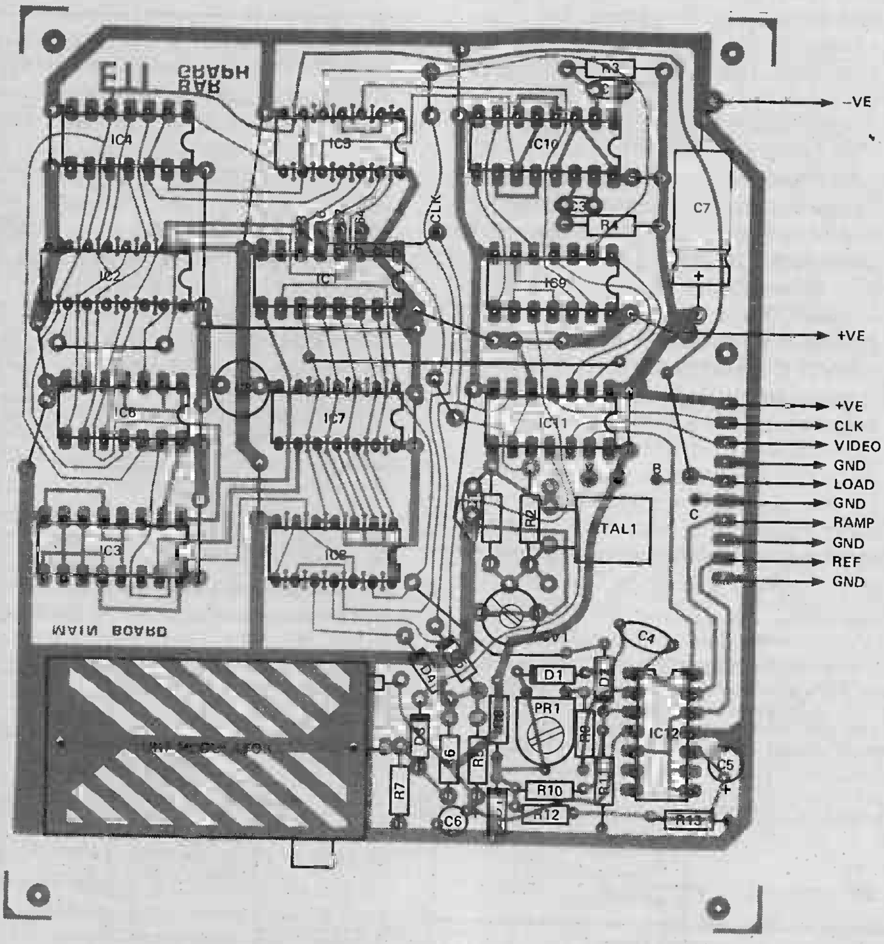


Fig. 5 Frame sync generation.





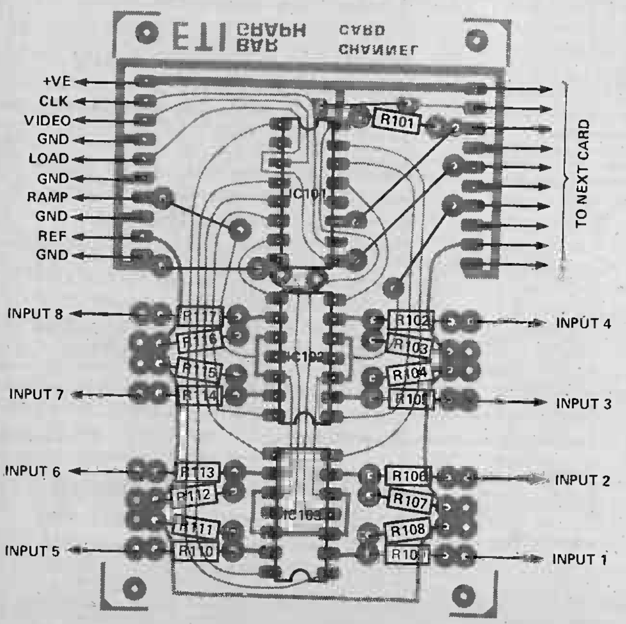


Fig. 7 Component overlay of the main board.

Fig. 8 Overlay for one of the channel cards.

PARTS LIST

MAIN BOARD

Resistors (all ¼ W 5%)

R1	1M0
R2	22k
R3,4,5	10k
R6,7,8	2k2
R9,10,12	10k
R11	100k
R13	Omit or select for reference voltage required

Potentiometer

PR1	10k miniature horizontal preset

Capacitors

C1	10p ceramic
C2	4n7 ceramic
C3	220p ceramic
C4	100n ceramic
C5,6	10u 35 V tantalum bead
C7	10u 25 V PCB aluminium electrolytic
C8	220u 25 V axial aluminium electrolytic
CV1	2-22p miniature trimmer

Semiconductors

IC1	4520B
IC2	4518B
IC3	40103B
IC4	4002B
IC5	4001B
IC6	4011B
IC7	4012B
IC8	4025B
IC9	4023B
IC10	MC14528 or CD4098
IC11	4070B
IC12	TL084
D1-D4	1N4148
ZD1	4V7 400 mW zener

Miscellaneous

UM1233 UHF modulator (Aztec), 8 MHz bandwidth; 2.5000 MHz crystal.

CHANNEL CARD

Resistors (all ¼W, 5%)

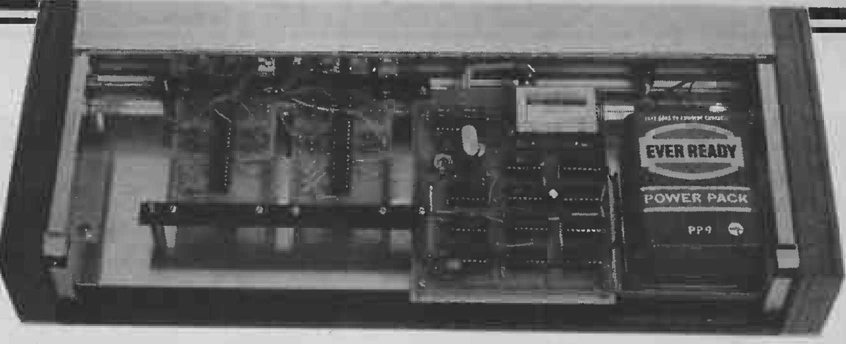
R101	56k
R102-117	27k
IC101	4014B
IC102,3	TL084

Miscellaneous

PCBs (see Buylines); 2.5mm jack sockets (eight per channel card); coaxial panel socket; miniature on/off toggle switch; PP9 battery and battery clips; phono plug; two off coaxial plugs and suitable length of coaxial cable; nuts, bolts, pillars, 12 x 12 mm aluminium angle or similar bracket; two off 12 x 100 mm (approx.) pieces of PCB laminate or similar (channel card support); wire, IC sockets etc; case (Newrad NP1426).

BUYLINES

Since we elected to use standard components rather than fancy chips for this project, there should be absolutely no supply problems. The modulator is from Technomatic or Watford, but make sure you get the wide-band (8 MHz) version. The case we used came from Watford. The PCB Service order form is on page 71.



The maximum number of channels sensibly usable is, we feel, 64, although 128 is possible on a good television set with the modulator specified.

When constructing the case for this project, it was found that the application of a drop of cycle oil to the self-tapping screws holding the extruded front pieces made it very much easier to screw them in. As we assembled and disassembled this part several times during construction, this was most helpful. Also, when assembling the chassis member to the side panels of the case, put the piece with the side flanges upwards and bolt it on underneath the lugs on the side panels. This is necessary to allow sufficient room for the battery inside the case.

Setting Up

There is very little in the way of setting up to be done. The basic unit should give some sort of display when powered up although no bars (or very few) will be present. To see anything more, it is necessary to feed some voltage into the channel inputs. As specified, full-scale should be about 6 V. This sensitivity can be altered by changing the input resistors as required. The reference voltage can be adjusted slightly using R13 or overridden by driving IC12 pin 10. The ramp rate is controlled by PR1 and should provide a fair control range for many purposes enabling the reference voltage crossing to be positioned near mid-screen.

When first trying the board, set CV1 to about half-capacitance (plates half-meshed) and adjust it only if the picture is unstable or cannot be pulled in by the television line and frame hold controls. (Don't forget to tune it in as accurately as possible).

Use

When fully operational the device should give the required number of bars along the screen, with a vertical height proportional to the input voltage for each one. The vertical resolution is about 270 steps, corresponding to a pair of interlaced lines. In the display mode built into the board, bars are not visible until the input is greater or less than 0 V. They then appear above or below the centre line; if the input sensitivity is too low then extra conditioning amplifiers will be required.

To change the display mode,

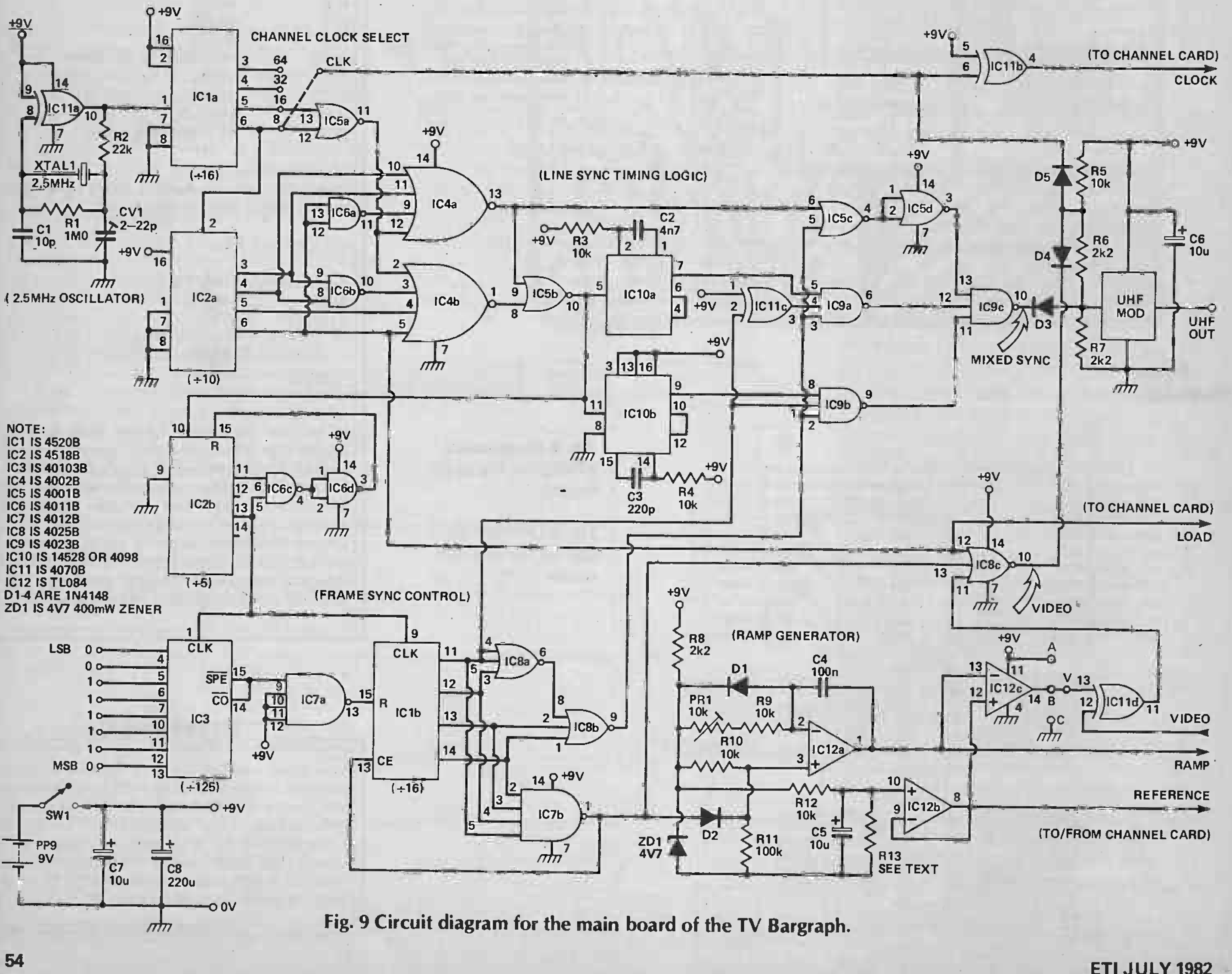


Fig. 9 Circuit diagram for the main board of the TV Bargraph.

break the track linking points V and B on the overlay. Linking V to A gives bars which start at the top of the screen and move down for decreasing input, while linking V to C gives bars which start at the bottom and move up with increasing input. Leaving out D5 will give double-width bars which merge with those on either side.

It is possible when using large numbers of channels that some will be off the edges of the display screen; this can only be cured by using a higher quality TV or monitor as the line flyback time in cheaper sets is often longer than that specified (12.8 uS). A way round this is to miss out the channels on the first card and start with the next. The device is ideal where a semi-graphical display of multiple inputs is required and has the advantages of easy expansion and no requirement for computing facilities.

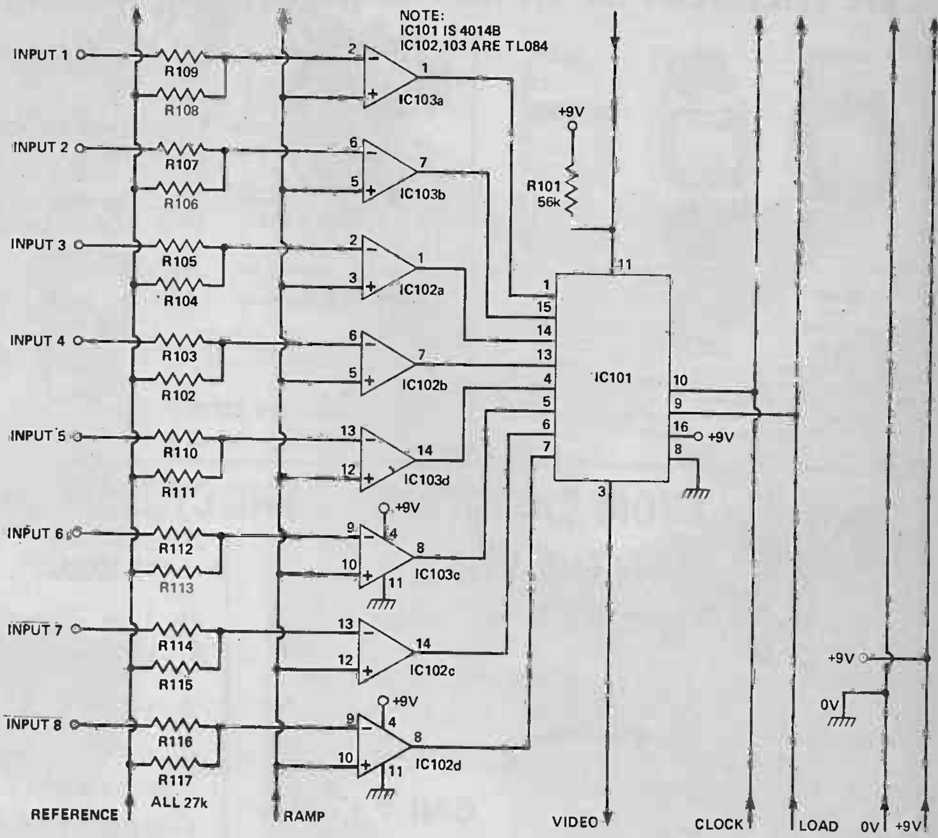


Fig. 10 Circuit diagram for one channel card; this provides eight input channels.

HOW IT WORKS

SYNC GENERATOR AND TIMING

IC11a is connected as an inverter and with XTAL1 and associated components forms a crystal-controlled oscillator working at 2.5 MHz. This is divided by 16 in IC1a and 10 in IC2a to give the 15,625 Hz for the line sync generator. The outputs from IC2a are decoded by IC6a, 4a and 4b to give two outputs. One is during count 1000 for the line sync, the other is at count 0011. The outputs from IC4a and 4b are disabled for about 1.6 uS after the active clock edge by IC5. This allow the counters and logic to settle and prevents glitches due to propogation delays. It also starts the line sync pulses at the right time relative to the line flyback blanking pulse taken from IC2a pin 6.

As IC2 is a dual decade counter, counts 1000 and 0011 occur at equal time intervals after each other. These outputs from IC4a and 4b are combined in IC5b to form a pulse chain at exactly double the frequency of the line sync pulses (Fig. 3). The regularity of this signal is important to get correct interlacing of the final picture. This signal triggers the two sections of IC10 and also IC2b. IC2b, IC6c and IC6d divide the input by 5 before driving the clock inputs of IC3 and IC1b. IC3 is a presettable eight-bit down counter which is configured to divide by 125, giving an output equal to the input clock period each 125 input cycles. Thus we get a low pulse 2½ line periods long (five double frequency pulses) every 312½ line periods (5 x 125 = 625 double frequency pulses). The low pulse from IC3 is inverted by IC7a and resets IC1b to all zeroes; it then holds it there for 2½ line periods. At this point IC1b will be incremented by the output from IC2b (ie every 2½ line periods) until it reaches the all 1s state. IC7b detects this condition and its output inhibits further counting. The output from IC7b is also used to blank the video signal during the frame flyback period. The actual frame sync signal is generated during counts 0001, 0010 and 0011 of IC1b. The logic for this is provided by IC8a and 8b. The output from IC8b determines whether normal line sync or frame sync is required and IC1b pin 11 decides which type of frame sync signal is to be sent. The signals are actually switched and combined in IC9, IC11c, IC5c, d.

IC10a and 10b are monostables triggered by the double line frequency pulse chain and provide the 2.3 uS 'equalising' pulses and 27.6 uS 'broad' pulses reqired for proper synchronisation in the 625 line system.

VIDEO AND CHANNEL SAMPLING

Having generated all the sync pulses to stabilize the display format, the video information must be generated. IC12 generates a negative-going ramp signal, synchronised with the frame blanking signal via D2. The frame blanking signal forces the non-inverting input of IC12 high, so the output of IC12 goes high to try and reduce the differential voltage between its inputs. However, the inverting input cannot go more than about 0V7 more positive than the cathode of ZD1. This means that C4 will charge very rapidly. When the frame blanking period is over, D2 is effectively out of circuit and the voltage on the non-inverting input to IC12 will be about 10/11 of V_{REF}. By normal op-amp operation the inverting input to IC12 will also be at this voltage, causing a current of 1/11 V_{REF} / (R10 + PR1) to flow through C4. The result of this is that the output voltage from IC12 will now fall linearly until it approaches the 0 V rail or another frame blanking pulse occurs.

Another section of IC12 provide a buffered reference voltage while a third acts as the comparator — switching as the ramp voltage passes the reference.

The ramp and reference voltages pass to the channel cards where the video signal is generated. Each input signal is compared with the ramp voltage in a section of IC102 or IC103. The more positive the input signal, the sooner its comparator will switch and the higher up the screen its bar will start.

The outputs of all the eight comparators on each card are fed to the parallel inputs of a 4014 eight-bit shift register (IC101). During the line flyback blanking pulse the data is loaded into the 4014; during the rest of the line time it is clocked out under the control of the channel clock. If more than one channel card is in use the output from each additional card is fed to the serial input of the next card along. This effectively extends the length of the shift register. For best results 8, 16, 32 or 64 channels should be used. Four or 128 channels are possible with modifications while 24, 40, 48, 56 etc will give poor display formats.

The video signal from the channel card(s) returns to the main board and, via IC11b, is mixed with the sync and blanking signals in D3,4,5 and R5,6,7 before going to the UHF modulator.

The line blank signal mentioned above is derived from IC2a pin 6 (Q_4) while the frame blank signal comes from IC7b. The output of this last device is high for a total of 40 lines in each half frame (16 x 2½) leaving 272½ lines for the actual display. This is still probably more than a normal portable TV will display vertically.

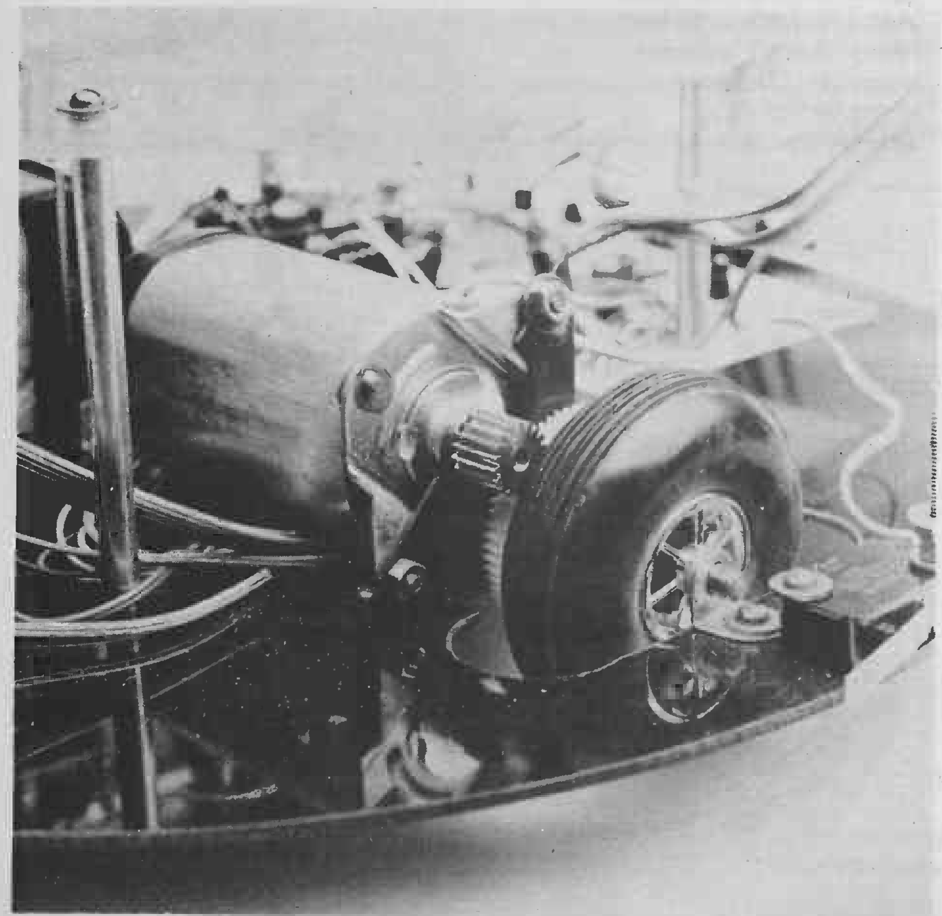
The feeding of the channel clock signal via D5 into the video mixer causes the bars to be separated and half width. Omitting D5 allows the bars to broaden and merge with each other. The jumper points V, A, B, C are pre-linked to give a video inversion at $V_{RAMP} = V_{REF}$. Alternative effects can be obtained by linking V to one of the other points, when bars starting from the top or bottom of the screen will be obtained instead of starting from the centre line.

MOTOR SPEED CONTROL FOR ROBOTS

With our Mobile Robot moving towards completion, reader C. Fisher of Bristol sends us this discrete circuit, derived from a Motorola design, which offers followers of our Robot a 'cheap and cheerful' alternative method of getting things moving in the living room.

In most DC motor speed control circuits, a voltage proportional to actual motor speed is compared with a voltage proportional to the desired speed and an error, or correction, signal is obtained. The error signal is amplified and used to adjust the speed of the motor in such a way as to reduce the error signal to a near-zero value. Provided that the circuit has been properly designed, the motor will run at a speed close to that desired.

The voltage supplied to the motor will determine the output power or torque, as well as the speed, so that at low speeds very little torque is produced. The normal method of overcoming this problem is to supply the motor with constant voltage pulses with a duty cycle proportional to the error voltage so that the full torque is produced at low speeds.



Light Work?

There are a number of ways that can be used to derive a voltage proportional to motor speed. One method which has been favoured in the past, involves measuring the back EMF of the motor. This can lead to problems, as both the input and the output are obtained from the same point, namely the motor.

It is possible to obtain a voltage proportional to motor speed using an opto-electronic tachometer system. A circuit employing this technique is shown in Fig. 1.

The motor armature, or output shaft, is painted with twenty stripes, ten black and ten white. A fibre optics Y-guide is focused on to the pattern and one branch is used to provide illumination from a small DC-driven lamp. The other branch of the Y-guide feeds light reflected from the pattern to a photo-transistor, Q1. The output of Q1 will be a signal with a frequency proportional to motor speed as ten pulses will be produced for each complete rotation of the armature.

The transistor Q2 is a pulse shaper which feeds a tachometer circuit giving an output directly proportional to input frequency and therefore motor speed.

Comparative Difference

The FET, Q4, acts as a buffer to minimise the loading on the tachometer circuit and provide a fairly low output impedance, which is appropriate to the differential comparator which follows it. In addition, Q4 acts as a level shifter to ensure that there is sufficient output to bias the comparator when the tachometer output is zero. The diode, D3, provides a measure of temperature compensation.

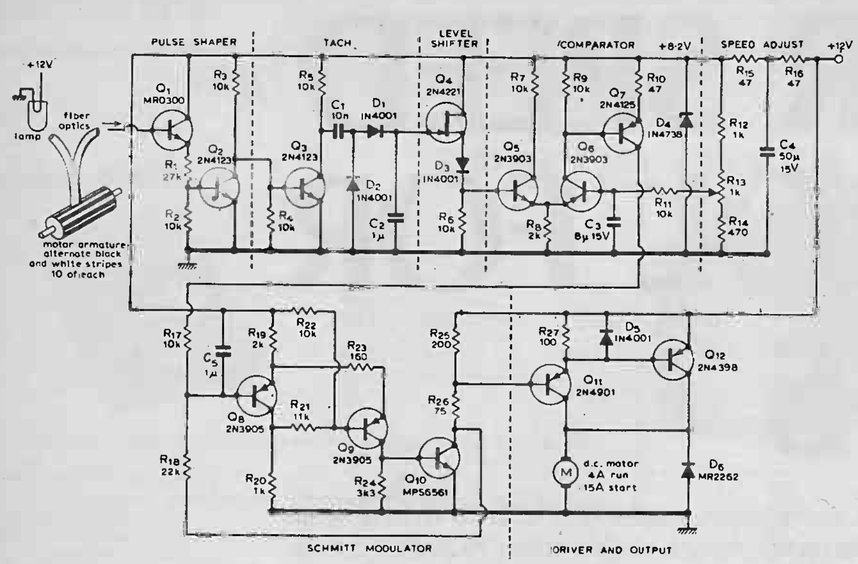


Fig. 1 Complete circuit diagram of the robot motor speed controller.

A Small Step?

The above method uses a photo-tachometer to obtain its reference for positional control. In our Mobile Robot, we decided to employ an infra-red system.

There are several reasons behind this choice, not least of which is the fact that our Robot employs a combination of on-board and distributed intelligence.

The arm carried by our Mobile is controlled in an entirely different fashion. . .

But then if you want full details of the ETI Mobile Robot, you'll have to buy our August issue, wherein all will be revealed!

The comparator compares the tachometer output with the voltage at the wiper of the speed adjustment potentiometer R13 and produces an error signal if a difference exists. The capacitor, C3, prevents motor speed overshoot if the setting of R13 is changed rapidly.

The network R15, R16, C4 and D4 forms a voltage-stabilising supply circuit for these circuits.

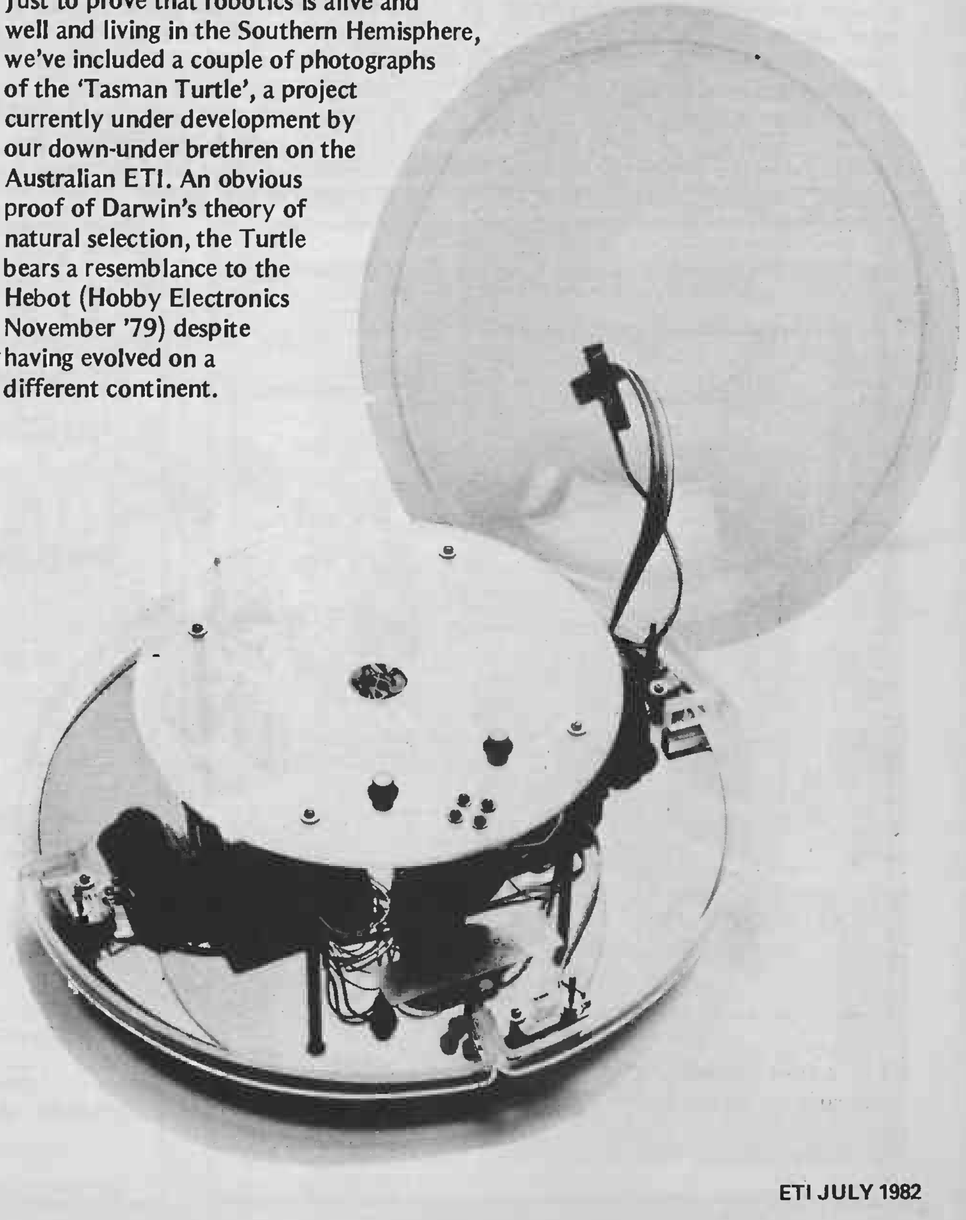
Errors Eliminated

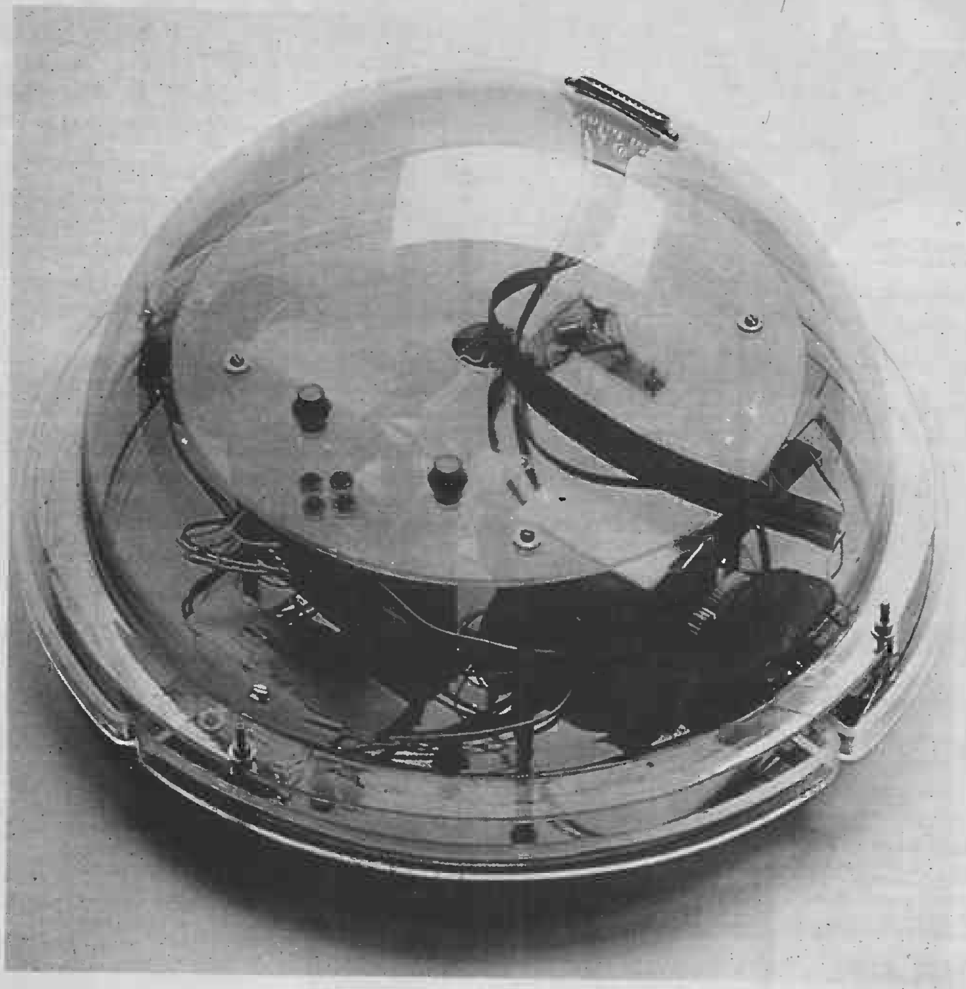
If an error signal exists because the potential at the wiper of R13 is higher than the output of the tachometer, it means that the motor is rotating too slowly. The pulse width modulator formed by a Schmitt trigger circuit will trip and full power will be supplied to the motor. As the motor speed increases the error signal will fall until it is almost zero. At this point the Schmitt trigger will remove power from the motor. The process is continuous and the motor is supplied with a train of pulses, the width of which will be proportional to the error in the motor speed, or the load on the motor.

Feedback for the circuit is obtained by directly measuring the motor speed, and results in accurate speed control over a wide range of output powers, as can be seen in Fig. 2. This drawing also shows the effect of power supply voltage variations. Figure 3 indicates the effect of temperature variations.

Temperature compensation may be improved by removing diode D2 and connecting one or more diodes in series with R9.

Just to prove that robotics is alive and well and living in the Southern Hemisphere, we've included a couple of photographs of the 'Tasman Turtle', a project currently under development by our down-under brethren on the Australian ETI. An obvious proof of Darwin's theory of natural selection, the Turtle bears a resemblance to the Hebot (Hobby Electronics November '79) despite having evolved on a different continent.





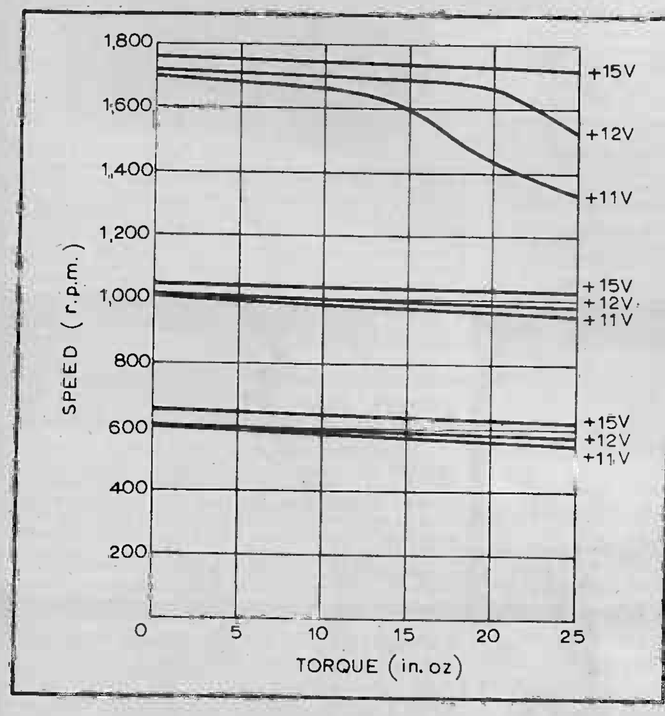


Fig. 2 How motor speed varies with changes in load and supply voltage for the circuit described.

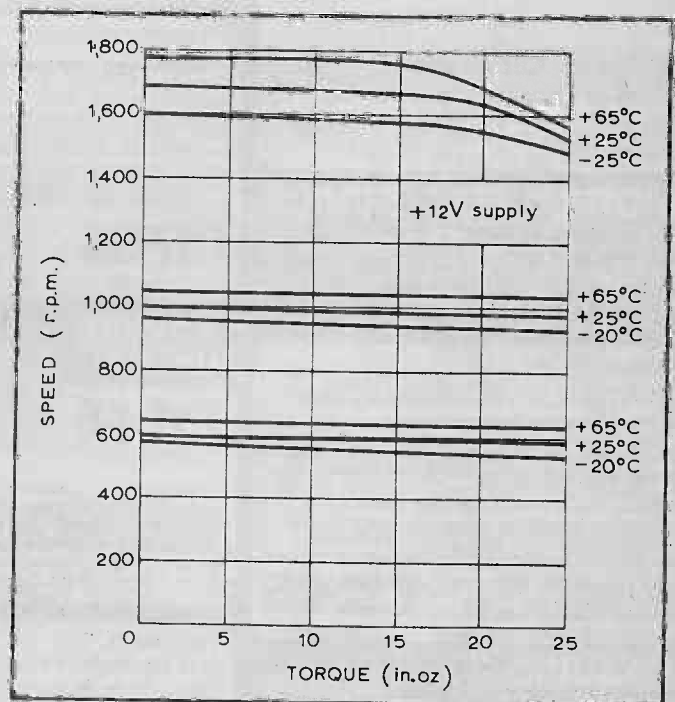
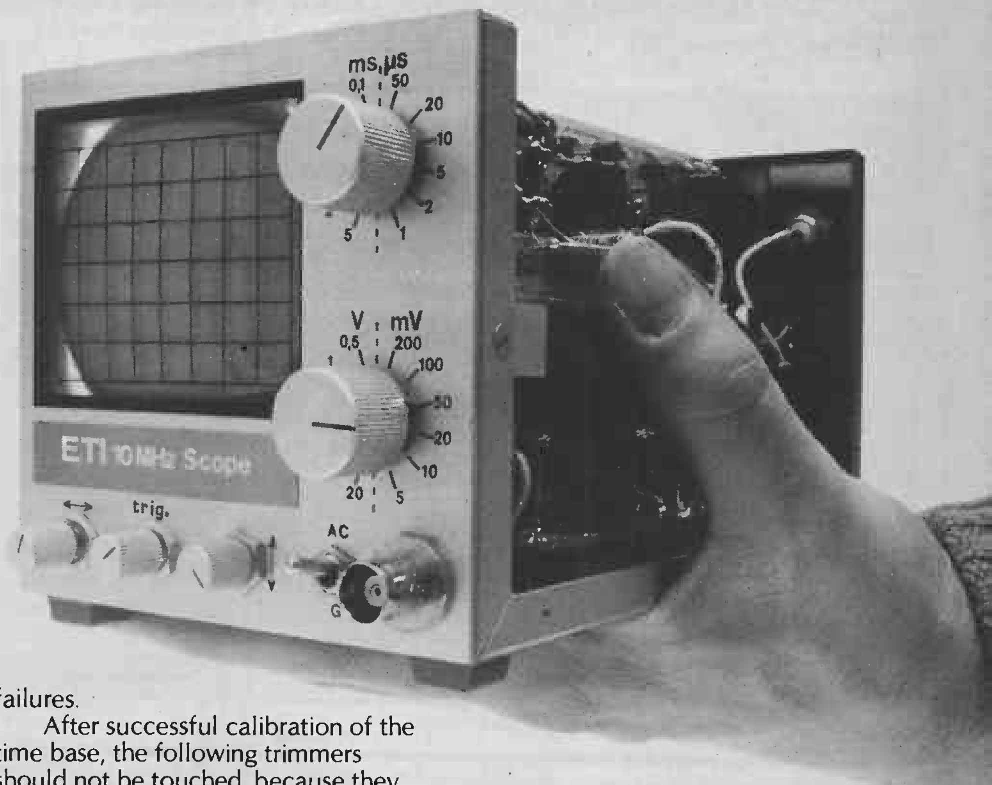


Fig. 3 How temperature affects the motor speed. The supply is assumed constant at +12 V.

ETI

OSCILLOSCOPE PART 3

In the final part of this project we give the all-important procedures for calibrating the scope timebase, Y-amplifier and frequency response. Once everything's set up, you'll have a scope to rival any other at the price. Design by K. W. Dugge.



To calibrate the timebase, apply a 50 Hz AC voltage to the input; the time selector switch should be on 5 mS/div. Adjust PR4 so that one cycle is exactly four scale divisions long. Apply a 5 kHz signal to the input (the frequency should if possible be checked with a counter). The time base switch should be on 50 uS/div. Again, one cycle should be four divisions long. If one cycle is longer than four divisions (sweep too fast) an additional capacitor must be placed in parallel with C20 (2n2). A place is provided for this on the circuit board. If, for example, the sweep time needs to be 10% longer, then the additional capacitor should be 10% of C20, ie 220pF.

If the length of one cycle is too short, then an additional capacitor must be placed in parallel with C44 (220nF). If the period of the 5 kHz signal (which should be four div x 50 uS = 200 uS) appears as only 180 uS ($\Delta = -10\%$) then a value of 10% of 220nF (22nF) should be fitted in parallel with C19 and the setting of PR4 repeated.

The remaining positions of the time base selector switch do not require adjustment, as they should automatically be correct — apart from unavoidable tolerances — thanks to the fixed resistors associated with the switch. All that is required is that the calibration be checked in each position of the time base selector using single, exactly-known frequencies, in order to track down possible component failures.

After successful calibration of the time base, the following trimmers should not be touched, because they will influence the calibration; PR10 (time base calibration), PR11 (trace length), PR15 (brightness), PR16 (focus), PR17 (astigmatism) and PR14 (10 V supply voltage).

Y-Calibration

Set the sensitivity on 1 V/div and the input mode switch on 'DC'. Set the trace, using the Y-shift control, to the centre of the screen. Apply 3 V DC — checked with an accurate multimeter — to the input. With PR8 (Y-output stage gain) set the trace to the top line of the scale. Switch the input mode selector to 'G': check whether the zero line has shifted. If necessary, reposition the zero line with the Y-shift control and re-adjust PR8 to bring the 3 V trace back to the top line of the scale. For the fixed linearity check, set the input to 0 V and the trace to the bottom scale line. Increase the input voltage; for each successive 1 V increase in the input voltage, the trace should move up by one division.

SPECIFICATION

Bandwidth: 0-7.5 MHz (−3 dB) for six divisions (one div = 7 mm); 0-10 MHz (−3 dB) for four divisions.

Input: BNC connector, switchable AC/DC/ground.

Sensitivity: 5 mV/div to 20 V/div in 12 calibrated 1/2/5 steps.

Case Size: approximately 175 x 105 x 100 mm.

Weight: approximately 1 kg.

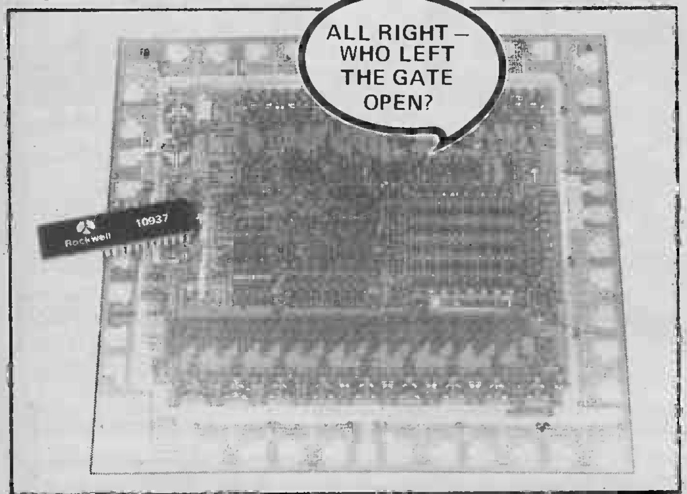
TABLE 1

SW1 on 5 mV, test probe 1:1, adjust CV1.
10 mV, test probe 1:1, adjust CV9.
20 mV, test probe 1:1, adjust CV5.
SW1 on 5 mV, test probe 10:1, adjust trimmer in probe. This trimmer must not now be altered! Also, the other trimmers, once set, must not be changed!
SW1 on 10 mV, test probe 10:1, adjust CV8.
20 mV, test probe 10:1, adjust CV4.
SW1 on 50 mV, test probe 1:1, adjust CV7.
100 mV, test probe 1:1, check that the adjustment is still correct.
200 mV, test probe 1:1, check that the adjustment is still correct.
500 mV, test probe 1:1, adjust CV3.
1 V, test probe 1:1, check that adjustment is still correct.
2 V, test probe 1:1, check that adjustment is still correct.
SW1 on 50 mV, test probe 10:1, adjust CV6.
100 mV, test probe 10:1, check.
200 mV, test probe 10:1, check.
500 mV, test probe 10:1, adjust CV2.
1 V-20 V, test probe 10:1, check.
Input square wave 50 kHz approx., SW1 on 5 mV/div, test probe 1:1, adjust CV10 on the main circuit board) for best waveform.

DIGEST

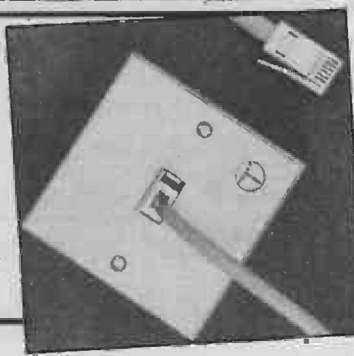
Precisely Stereo

After all the nasty things we've said about TV sound, it seems the industry has finally got the message. Regular stereo TV transmissions have begun successfully in West Germany, and Grundig has launched its first stereo TV in Britain. The sets will give genuine stereo from stereo video recorders and video disc players, and processes mono broadcasts to give a 'spatial' sound. Despite the four speakers (two forward-facing tweeters, two side-facing woofers), the sets are slightly narrower than conventional mono televisions, and a plug-in board will give stereo broadcast sound when this becomes available. Are you listening, BBC and ITV?



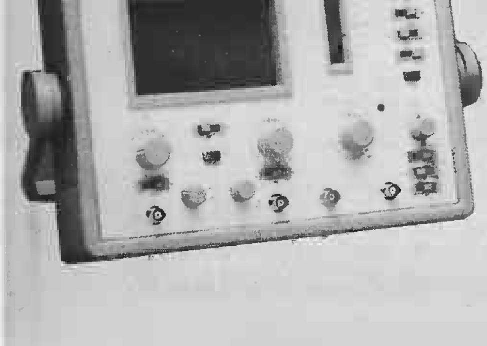
Plugging Telecom

In case you were wondering, this is what British Telecom's new four and six way modular plugs look like. Manufactured by Pressac of Nottingham, the cable connections follow the current trend of being IDC, the blade contacts giving high pressure, gas tight connections.



Scopex Scoop

Scopex are feeling rather pleased with themselves — they've developed the world's first tubeless, dual trace, digital storage, battery portable scope. The Scopex Voyager uses a 128 x 256 LCD matrix display to replace the normal cathode ray tube; this makes the device only 330 x 260 x 98 mm! Compare *that* with your normal digital storage scope. The 64 x 102 mm display does not use the conventional twisted nematic LCD material, but dye-phase-change material which gives a very wide viewing angle, high contrast and excellent visibility in a range of lighting conditions. Input waveforms can be stored with an equivalent bandwidth of 150 kHz and it's possible to keep the waveform in memory, even with Voyager switched off, for at least 100 days. Thus a waveform may be stored in the lab and compared with one 'on-site' miles away. The Y-amp sensitivity is 10 mV/cm to 5 V/cm, with a sweep speed from 20 uS/cm to 50 S/cm. The weight of the Voyager is 2.5 kg, and the price — well, that's £2,500 plus VAT, slightly out of reach of the hobbyist but likely to take industry by storm. Nice to see a British company leaving the Americans and Japanese standing. For more comprehensive details on the Voyager facilities, contact Scopex at Pixmore House, Pixmore Avenue, Letchworth, Herts SG6 1HZ.



Osborne On Offer

If there's anyone left who hasn't heard about the Osborne I portable microcomputer (we've seen it on two magazine covers already this month), here's your chance to get hold of one at a special launch price, valid until July 31st this year. For £1250 plus VAT, Adda Computers will supply the total package including free delivery in London and the Home Counties, 10 free diskettes and a one year parts and labour warranty. The Osborne I weighs only 24 lbs and can fit under an airline seat. It includes 64K of memory, a business keyboard with graphics and 10-key numeric pad, a monitor screen, outputs for separate printers and other peripherals, two disc drives, connections for battery pack Mailmerge, SuperCalc, CP/M operating system and two BASIC languages. Comparable packages cost two to three times as much. Adda Computers live at Mercury House, Hanger Green, Ealing, London W5 3BA.

Musical Boxes

L & B Electronics, designers and manufacturers of modular lighting control and amplifier systems, have been made distributors for the Paia range of kits which are being introduced into the UK. The Paia range is extensive, to say the least — the catalogue has 16 pages crammed with synthesisers (from the tiny to the huge), organs, sequencers, drum synths, string synths, all the special effects you've heard of and some you haven't, plus a selection of books. One of the sequencers is called the 'Orgasmatronic' and there's a kit for an 'Encephalo-Gratification Generator', but I'm sure the designer has seen his psychiatrist and is OK now. The catalogue contains endorsements by Larry Fast and Peter Gabriel, which can't be bad, and for more information contact L & B Electronics, 45 Wortley Road, West Croydon, Surrey CR0 3EB (telephone 01-689 4138).

Sharp Practice

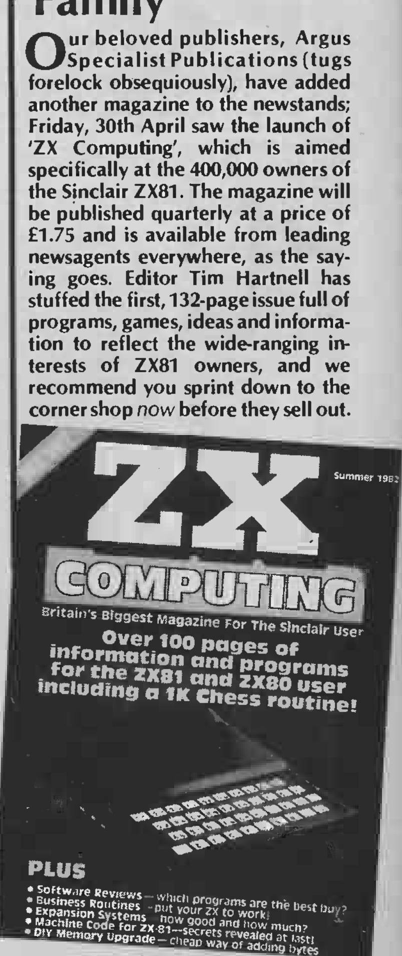
Pocket computer freaks will be interested to know that the Sharp PC-1500 can now be purchased from Tempus. "Suddenly a pocket computer approaches the personal computer in ability" says the brochure, and it's certainly a powerful tool for use in business, management, engineering and hobbies. Its small size makes it ideal for portable use (only 195 x 25.5 x 86 mm), and it's packed full of goodies. The memory contains 16K of ROM and 3.5K of RAM, expandable to 7.5K of RAM with an optional memory module; programs and data are retained when the computer is switched off.

Almost any symbol can be displayed on the 7 x 156 dot display area (would you believe circuit diagram symbols?), or a 26-character line when using alphanumerics. For the first time ever in a pocket computer there is a QWERTY keyboard layout; with the add-on colour graphic printer/cassette interface (!!!), the PC-1500 can be used as a small personal typewriter. Six user-definable keys are provided, plus a MODE lock key so you can only RUN the program, avoiding inadvertent erasure. The BASIC features variables, two-dimensional arrays, variable strings and many other advanced features.

Combining the built-in clock and 'beep' functions allows the PC-1500 to act as an alarm clock — with on-screen messages! — to remind you of your schedule. Forthcoming extras include an RS-232C interface. At the moment we don't even have a reproduceable photograph of this wonder device, but hopefully we'll be able to get one for review before too long. Meanwhile, the PC-1500 is available for £169.95 including VAT, from Tempus, 38 Burleigh Street, Cambridge CB1 1DG.

A Birth In The Family

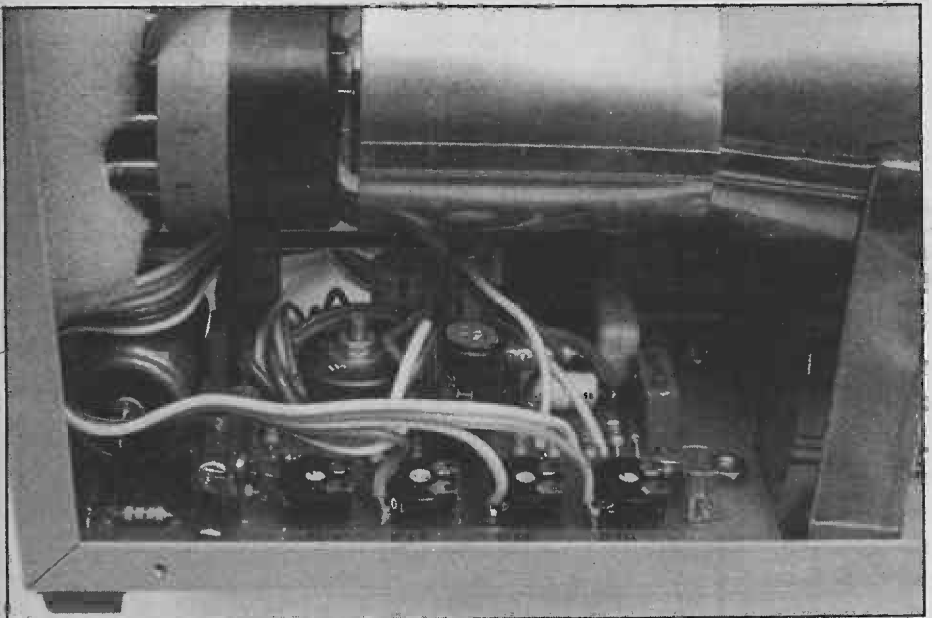
Our beloved publishers, Argus Specialist Publications (tugs forelock obsequiously), have added another magazine to the newstands; Friday, 30th April saw the launch of 'ZX Computing', which is aimed specifically at the 400,000 owners of the Sinclair ZX81. The magazine will be published quarterly at a price of £1.75 and is available from leading newsagents everywhere, as the saying goes. Editor Tim Hartnell has stuffed the first, 132-page issue full of programs, games, ideas and information to reflect the wide-ranging interests of ZX81 owners, and we recommend you sprint down to the corner shop *now* before they sell out.



Frequency Response Adjustment

Close the case (screw the cover on completely, as the additional capacitance of the cover will affect the calibration). Connect a 500 Hz (approximately) square wave to the input using a 1:1 test probe; set the input mode switch on 'DC'. Adjust the amplitude of the signal generator during each of the following steps so that the picture size is about three or four divisions. The adjustments in Table 1 should give the optimum square wave shape (no rounded corners, no overshoot on the edges).

This completes the setting up of the instrument.



An internal view of the power supply board.

BUYLINES

Since we started publishing the Oscilloscope project we've been informed by AEG-Telefunken that they can supply the cathode ray tubes, type DG7-32, at the very reasonable price of £28.75, excluding VAT but including post and packing charges. The tubes are normally available ex-stock. The full address is:

AEG-Telefunken (UK) Ltd,
Electronic Components Division,
217 Bath Road,
Slough,
Berkshire SL1 4AW.

The telephone number, should you need it, is Slough 872164. All four boards for the scope can be obtained using the order form on our PCB Service page (page 71).

ETI

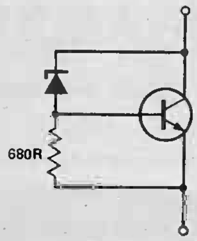
DESIGNER'S NOTEBOOK

This month we're dipping into the Notebook of Phil Walker, one of ETI's project editors, and showing you some of the unusual techniques he's collected.

We depart from the usual style of Notebook this month to bring you a pot-pourri of small circuits and techniques that you may not have come across before. Ever needed a two-bit DAC? An awkward low-current supply rail? Logic level shifting? Look no further, these are just three of the nine design ideas presented here.

Uprated Zener Diode

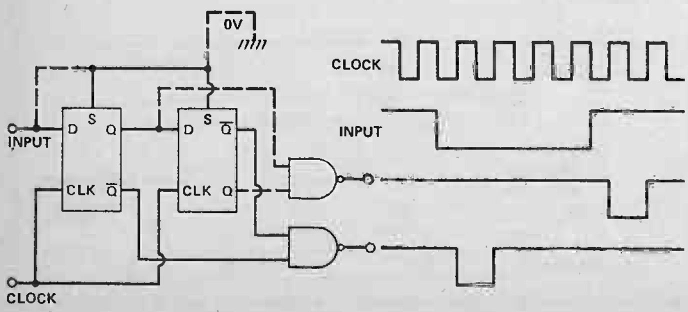
This circuit can be used to simulate a high power zener diode where the correct component is not available or too expensive. The configuration increases the allowable dissipation in the circuit up to the limit of the transistor or the diode rating times the transistor current gain. The stabilised voltage is about 0V6 to 1 V greater than the nominal zener voltage. The variation of output voltage with load current may not be quite as good as a normal diode but this may well be offset by convenience or cost considerations.



Single Output Pulse From An Input Level Change

When dealing with asynchronous inputs to a digital system it is often necessary to signal that a change of input has occurred. This circuit is mainly concerned with producing a single pulse synchronous with the system clock when the input changes state. The output is a pulse, one clock period wide, after the input goes from high to low or vice-versa (depending upon which output is used).

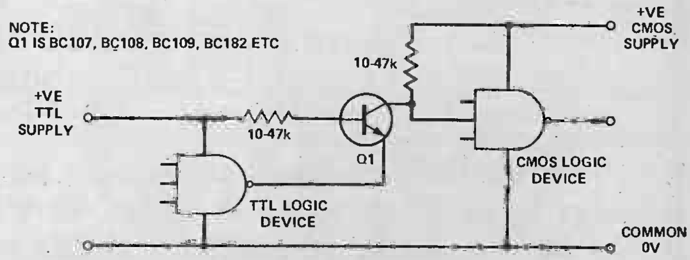
If only the falling edge of the input is of interest, then the input signal may be taken to the set inputs of the latches. This will enable the circuit to respond more quickly and reliably to successive pulses. In general the clock frequency should be at least four times the input frequency.



TTL to CMOS Logic Interface

When using mixed logic families it is necessary to transfer the signal from one set of logic levels to another. If all the devices are operated from the same supply rails this is easy, but if the rails are different then some form of interface circuit is needed.

For a TTL to CMOS interface this can be most simply a TTL gate with an open collector output and pull-up resistor, but if this is not available then the following circuit may be used. The circuit operates quite well for low to medium frequencies.

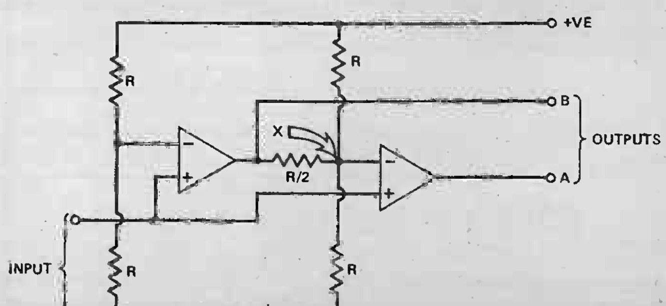


Two Bit A/D Converter

This is a very simple circuit which gives an approximate conversion of an input voltage level to a two bit binary code. Its accuracy is limited by the output circuitry of the op-amps and for best results CMOS types could be used.

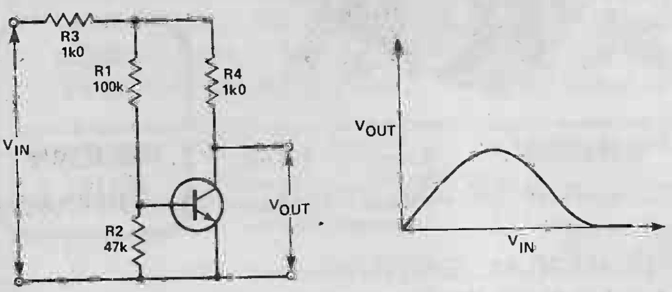
As the input voltage rises from 0 V, at first both the A and B outputs are low. This makes the voltage at point X about one-quarter the supply voltage. As the input voltage reaches this level, output A will go high. Later, when the input voltage reaches half the supply voltage, output B will go high. This then makes the voltage at point X go to three-quarters of the supply, forcing output A to go low. Still later, as the input voltage continues to rise it will reach this last value and output A will again go high.

The reference for this circuit is the supply rail. If the op-amps or comparators used cannot drive to very near the supply rails then adjustments may be made to the resistor values to compensate.



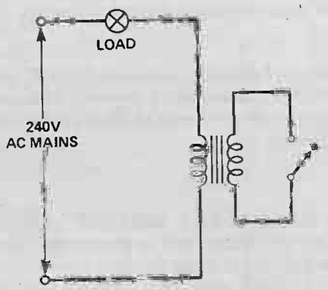
Transistor Function Generator

A one transistor circuit in which the output voltage follows the input up to a threshold set by R1 and R2. The output then falls at a rate determined by R3 and R4 until it is virtually zero. By varying the resistance values many different transfer functions can be obtained.



Secondary Mains Switching

A novel way of switching a mains load without having mains voltages on the switch itself is to put the switch on the secondary of a suitable step-down transformer with the primary connected in series with the load. The transformer primary presents a high impedance when the switch is open and a low impedance when it is closed. The main disadvantages of the arrangement are that the switch must carry a larger current than normal and the transformer must be rated for about the same power as the switched load.

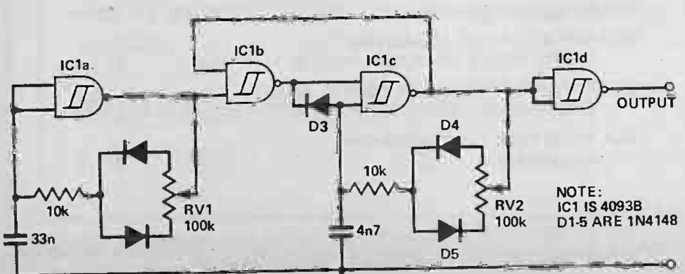


Simple Pulse Burst Generator

Using a 4093 CMOS quad NAND gate package it is very easy to make a circuit which produces bursts of pulses. These bursts have the property that they are composed of complete pulses, all of which have the same duration. The circuit shown here is configured to produce a variable number of pulses in each burst while the repetition rate of the bursts remains roughly constant.

The first IC section produces the variable mark/space ratio burst control signal, the next two sections are the gated oscillator while the last section acts as a buffer and gives the output as positive-going pulses.

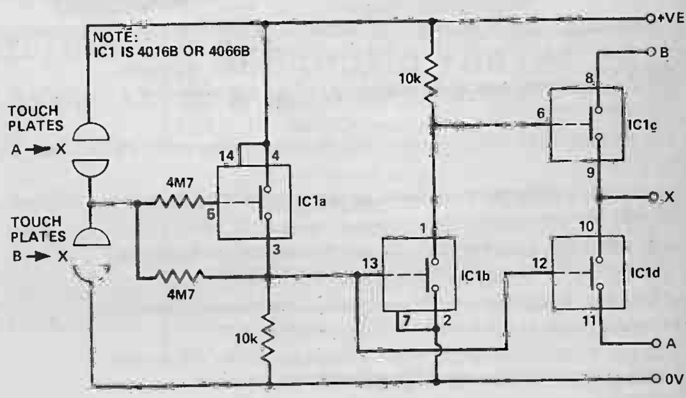
The frequency of operation for both sections is determined by the product of the capacitance and the fixed plus variable resistance. If a 50% fixed duty cycle is desired then the resistor/diode combination can be replaced by a fixed resistor in series with a variable resistor.

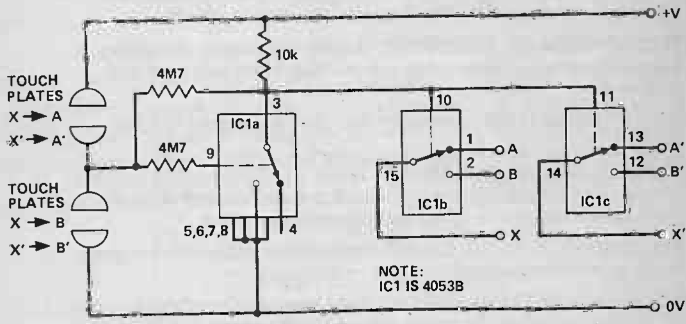


Bistable Touch Switch For Analogue Signals

This uses two sections of a 4016 or 4066 CMOS switch IC. One section of the device is used as a latch, while the others can be used as a changeover switch or as three make or break switches.

A similar switch can also be made using a 4053 triple 1 of 2 selector. In this case we get two analogue change-over switches with a bistable action. Either of these circuits could be used where audio control or signal selection is required but the hi-est of fi is not essential.





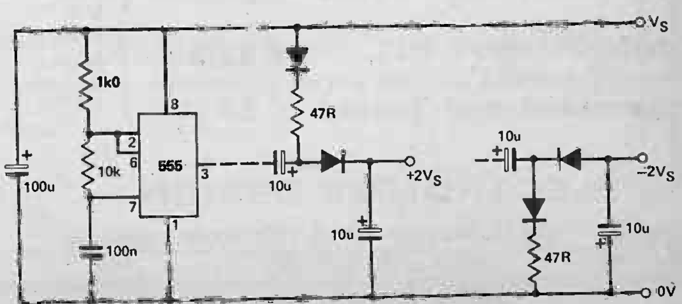
Extra Supply Rails

A 555-type timer IC can be used to provide that awkward low current supply rail when an extra battery would be inconvenient. The device is connected as a free-running astable oscillator and drives a simple charge pump. The polarity of the diodes and capacitors in the output circuit determines whether the output is positive or negative. Output impedence of this circuit is usually quite high, being determined by the capacitor values. The capacitor values should not be too high as this will overload the output circuit of the IC.

If the standard type of 555 IC is used the main supply rail should be decoupled at the IC pins with an electrolytic capacitor to prevent the well-known switching spike of the device affecting the rest of the circuitry.

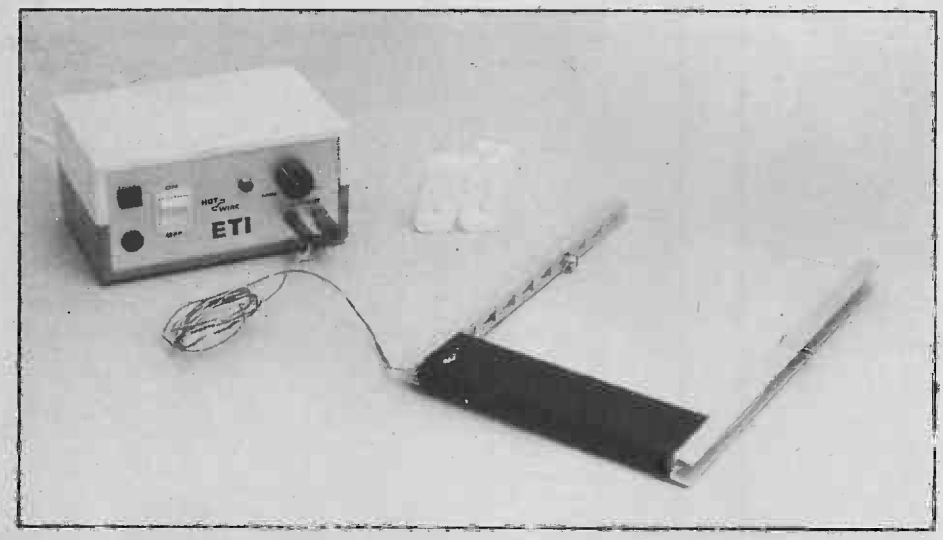
The output voltage from this arrangement will range up to about equal to the input voltage, superimposed on to the relevant supply rail.

ETI



POLYSTYRENE CUTTER

The ETI Hotwire is just the thing to get you going. No, it's not for stealing cars, it's for modelling. Turn that waste polystyrene packing into beautiful models with the Hotwire and some imagination. Design and development by Phil Walker.



This easy-to-construct project is a controller for a hot-wire polystyrene cutter. This method of cutting foam polystyrene is probably better than most others as it does not create any rough edges or crumbs; it actually works by melting the material as it comes into contact with the hot wire.

The object of the controller is to maintain the wire at a fairly constant temperature sufficient to melt the polystyrene foam quickly but without charring. This is accomplished by using a simple type of phase controller to regulate the power applied to the wire. The circuit employs a 747 dual op-amp, both parts of which are used as comparators. Speed of operation is not critical here as the circuit is operating at mains frequency (50 Hz).

Taking A Pulse

The first part of the circuit produces a 100 Hz pulse signal which synchronises the rest of the circuit to the output from the bridge rectifier. The second part generates a variable time delay which is used to regulate the amount of power developed in the cutting wire. The longer the time delay, the less power is developed and vice versa.

The control element used in this project is a thyristor as this will withstand the high peak currents in the circuit without the necessity for large drive currents.

Construction

This is fairly simple since most of the components are mounted on the PCB. Make sure that the diodes and IC are the right way round. Bolt the small heatsink to the rectifier bridge using some heatsink compound before mounting it on the board. Allow it to stand about 6 mm away from the board to avoid thermal stress effects. The thyristor is mounted on top of the larger heatsink, both being held by the same screw. Heat conductive paste should be used here as well. R9 will get quite hot in operation and should be stood away from the board if possible to allow air flow around it.

When mounting the PCB in the case, it is advisable to do so with the capacitor C1 at the bottom so that it is not heated by the other components.

Fairly thick wire should be used for connecting to the transformer and output sockets as they will be carrying several amps. RV1 is wired so that *minimum* resistance occurs at *clockwise* rotation.

Some Cutting Remarks

In our prototype the cutting head was made from two short pieces of slotted aluminium extrusion of the type sold for shelving systems. These were screwed to a piece of wood to form a handle while also insulating them from each other. The steel wire was clamped with some large nuts and bolts so that it was under some tension. The wires to the control unit were also clamped to the large bolts and held in place along the arms of the head with sticky tape.

It is recommended that the ceramic insulators sold by good electrical shops be used for the ends of the cutting wires in order to keep the metalwork isolated. Plastic connector block could be used but may melt under extreme circumstances.

Once everything is working correctly you can begin to exercise your creative talents on the nearest piece of polystyrene. Apart from a modelling tool, a gadget for 3-D doodling and something to keep the kids quiet during the summer holidays, you could use the Hotwire for cutting out large letters — ideal for advertising displays or exhibition stands.

BUYLINES

All of the parts for this project should be readily available from the usual outlets. The thyristor, SCR1, can be either a 2N4443 or a 2N4444 — the latter has a higher voltage rating and a higher price. The PCB can be obtained using the order form on page 71.

HOW IT WORKS

The 15 V AC from the transformer is rectified by BR1 to give a raw 100 Hz pulsating DC supply. C1 is charged to the peak voltage of this supply via D1 and provides the power for the circuitry. The raw DC supply is taken via R2 to IC1a where it is compared with the voltage across ZD1. The output from IC1a consists of a train of negative-going pulses which occur around the zero crossings of the AC input. These pulses are used to synchronise the variable time delay circuit by discharging C2 at the zero crossing of the AC input. The capacitor then charges at a rate set by R4 and RV1 until its voltage reaches the level set by R5 and R6. At this point the output of IC1b changes from its low to high state and switches SCR1 into conduction.

Once SCR1 has been switched on it causes the raw DC supply to be applied across R9 and the cutting wire until the voltage falls to zero at the end of the half cycle. At this time the thyristor turns off, the variable time delay circuit is reset and starts again. The proportion of the total time for which the output is on is determined by the time delay set by RV1; hence this controls the amount of power dissipated in R9 and the cutting wire. The main function of R9 is to reduce the peak surge current which would flow in the circuit, but it will also give some protection against inadvertent short circuit (the wire itself has a resistance of a couple of ohms). LED1 is incorporated to indicate when the output is operating and gives a visual indication of the power setting.

PARTS LIST

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

R1	6k8
R2	3k3
R3,7	1k0
R4	5k6
R5	2k2
R6	15k
R8	1k8
R9	1R0 10 W wirewound
R10	180R 2 W wirewound

Potentiometer

RV1	100k linear

Capacitors

C1	1000u 25 V axial electrolytic
C2	68n ceramic

Semiconductors

IC1	747
D1	1N4002
D2	1N4148
BR1	6 A bridge rectifier, square package, 50 V or greater
ZD1	3V6 400 mW zener
SCR1	2N4443 (see Buylines)
LED1	5 mm red LED

Miscellaneous

FS1	20 mm 1A6 slow-blow fuse and holder
SW1	Double pole rectangular mains rocker switch
LP1	Mains panel-mounting neon indicator with integral resistor
T1	15 V 60 VA mains transformer

Heatsinks (finger-style TV21 for thyristor, TV4 for rectifier); PCB (see Buylines); case (Verobox 21039, 180 x 120 x 90 mm); panel mounting socket for LED1; two off 4 mm banana sockets, grommet, wire, nuts, bolts, brackets etc; 0.010" steel wire (guitar top 'E').

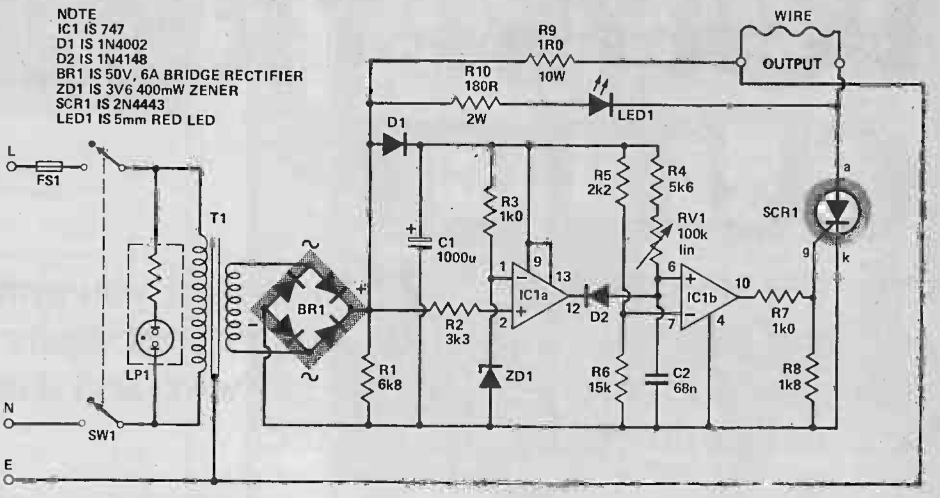
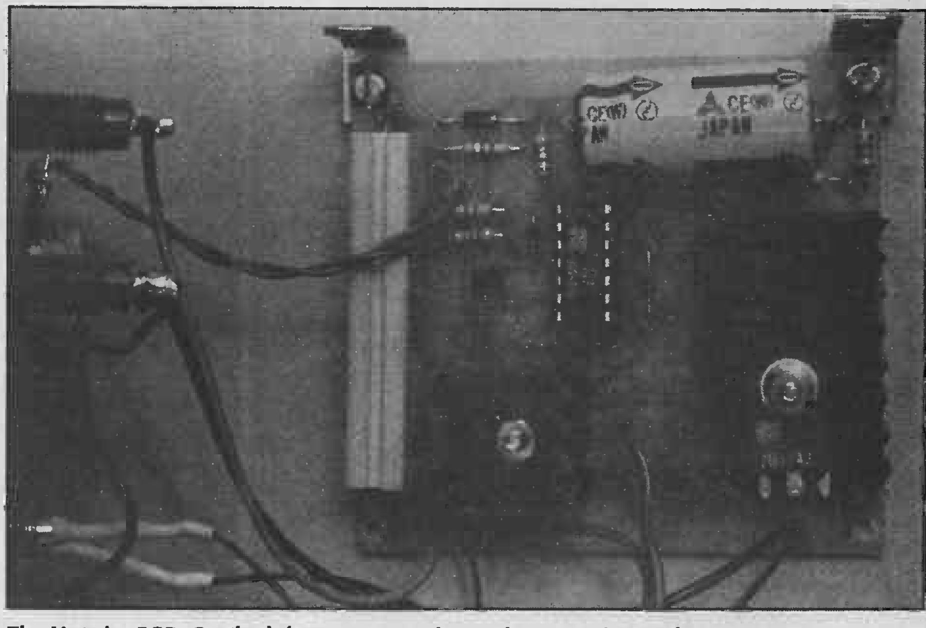


Fig. 1 Circuit diagram of the ETI Hotwire.



The Hotwire PCB. On the left you can see the earth connection to the pot case.

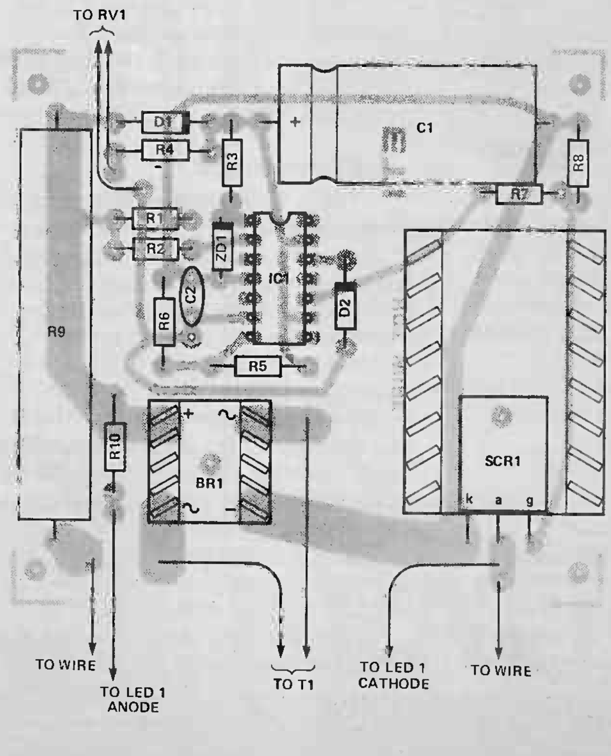


Fig. 2 Component overlay for the polystyrene cutter.

ETI

VIDEO SYSTEMS

It's probably the most advanced piece of engineering you'll ever have in your home — but it isn't that difficult to understand. Stan Curtis makes the hardware look easy.

In a matter of just a few years home video has become big business; second only to home computers as a means of taking away our hard earned money in exchange for boxes of wonder electronics. The pace of development continues with four video recorder formats now in use, three video disc formats imminent in the shops, and new camera/recorder/television technologies just around the corner. This rate of product change (Grundig released then replaced three recorders in 15 months!) coupled with a puzzling reluctance on the part of the manufacturers to release anything resembling technical information has left the electronics enthusiast a little in the dark. Many have just thrown in the towel and work on the basis of "an input socket and an output socket and what's in between is none of my business". Others have made innocent enough enquiries of the so-called technical departments of some of the importing companies. The standard responses vary from a shovel-load of pseudo-scientific mumbo-jumbo to downright suspicion of the "why do you want to know; you're not going to tamper with one of our machines, are you?" kind.

So the time has obviously come for ETI to present a basic primer on the state of today's video technology together with some background on basic video principles. This should, hopefully, deflect the Editor from any more mutterings about a do-it-yourself ETI video recorder! (Don't you bloody believe it! — Ed)

The Basic Principles

Before you can start to understand how video equipment works it is useful to learn a few of the principles and a few of the key words. For example, just how do we get a picture on the television screen? Each complete picture is termed a frame and lasts for 1/25th of a second; in other words synchronised to the 50 Hz mains supply with 25 frames per second. Each frame consists of 625 horizontal lines (in the UK) which are written across the screen during the frame time. Unfortunately the picture rate of 25 per second causes a flickering effect which is most annoying, so a way had to be found to increase the effective picture rate to 50 per second without increasing the video bandwidth. The answer was interlaced scanning, where the picture is scanned at 50 frames per second rate (to avoid flickering) but on each scan only half the lines are traced out, leaving a gap between each pair for the missing lines. Each scan is called a field and during the second field all the missing lines are scanned. The picture is made up of odd lines, even lines, odd lines, etc so that in every second exactly the same amount of data is transferred (hence the same signal bandwidth) but without the flicker.

That Syncing Feeling

In order that the picture be accurately reconstituted on the screen it is necessary that there be some sort of synchonisation between the signal source and the receiver. The synchronisation is achieved by the use of pulses. There is a sync pulse at the start of each line and a series of sync pulses at the start of each field. These field sync pulses are repeated at half line spacings so that the line sync is not lost and a series of equalising pulses (of opposite mark/space ratio) are also added to maintain the average signal level. This arrangement is probably best understood by studying the figures given in the TV Bargraph project this month.

Sync pulses and picture signals are kept separate by keeping the former below and the latter above the black level. Thus it is quite simple to separate out the sync pulses at a later time. The combined signal of both picture information and the sync pulses is usually referred to as composite video.

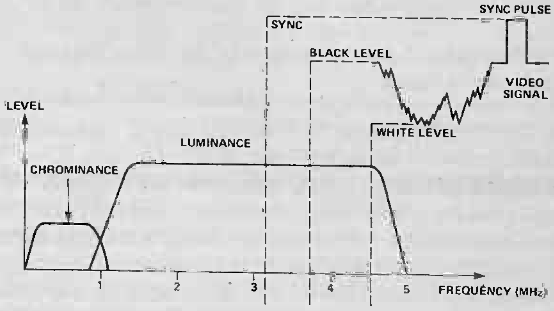


Fig. 1 The bandwidth of signals during video FM recording.

A very wide bandwidth is required to handle this signal which extends down to DC. The DC component must be accurately maintained because any change in its value will affect the average brightness of the signal. The high frequency bandwidth can be calculated by considering the picture resolution. Each of the 625 lines has a duration of 64 microseconds of which 13 microseconds is used for a black margin at either side of the picture. Thus for horizontal resolution of 575 picture elements there will be a need for a bandwidth of about 5.6 MHz. Similarly we can see that if a domestic video cassette recorder has a bandwidth of 3 MHz the horizontal resolution will drop to below 300 picture elements.

The video picture signal varies in DC level at any instant, the voltage determining the grey tone of the picture. The highest DC level represents white while the lowest DC level is black, the greyness varying linearly between these limits. The brightness signal is termed the luminance signal to distinguish it from the colour or chrominance signal.

Hue And Y

Once colour is considered the video theory becomes steadily more complex. Colour has two characteristics; hue which describes its colour (red, yellow, etc) and saturation which describes the percentage depth of the colour. Thus a 10% red will be a faint pink while 100% will be a deep strong red.

The colour camera converts the colours of the subject into three outputs, red, green and blue, from which any of the original hues can be reconstituted. They can also be mixed in the ratio 30%-59%-11% to produce the luminance signal (Y):

$$Y = 0.3R + 0.59G + 0.11B$$

The percentages are chosen to follow the sensitivity of the eye. The chrominance signals are then derived by subtracting Y from each to give the three difference signals R—Y, G—Y, and B—Y.

Although it is possible to send each of these signals separately it is obviously more convenient to combine them as a single colour signal. The first operational system to do this was the NTSC developed in the early 1950s in the USA. Later came the SECAM system in France and the PAL system developed by Telefunken in Germany. The three systems are incompatible with each other as many people have learned to their cost when they have imported NTSC equipment from the USA.

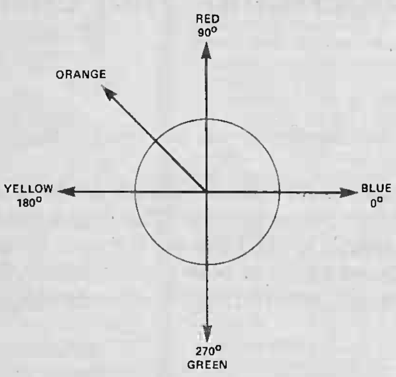


Fig. 2 How colours are determined by the phase angle of the sub-carrier signal.

The colour signal is encoded using suppressed carrier quadrature modulation. This means that the R—Y signal is modulated on the 4.43 MHz subcarrier whilst the B—Y signal is modulated on the same subcarrier 90° out of phase. When the two subcarriers are combined the result is a single signal whose phase angle varies in relation to the two components (see Fig. 2).

Thus the hue of the colour is defined by the phase angle and the saturation by its amplitude. To do this we have only used the B—Y and R—Y components since the G—Y component can always be derived from the other two.

Suppressing The Truth

Now we come to the suppressed sub-carrier bit. The chosen frequency of 4.43 MHz sits right inside the 5 MHz luminance bandwidth and its presence would therefore cause a visible pattern on the screen. The solution is to suppress the carrier frequency leaving just the sidebands.

Again some sort of synchronising signal is needed to enable the colour signal to be reconstituted accurately. So for colour a 10 cycle burst of 4.43 MHz carrier is inserted ahead of the video picture signal. This gives an accurate reference frequency to enable the suppressed sub-carrier to be reformed by a local oscillator in the TV which is 'kicked' into sync by this colour burst. The phase of this burst also acts as a reference in decoding the difference signals.

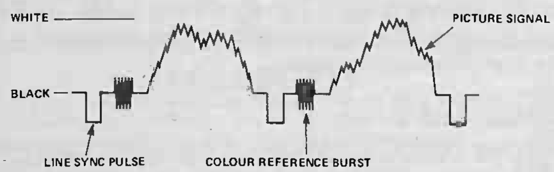


Fig. 3 The composite video signal with line sync pulses.

The foregoing applies to both the NTSC and the PAL systems but in the latter the phase of the R—Y signal is reversed on alternate scan-lines and so the reference colour burst changes phase through 90° on alternate scans. This allows phase errors to be averaged over adjacent lines, avoiding the colour shift which has earned NTSC the nickname 'Never The Same Colour'.

Video Cassette Recorders

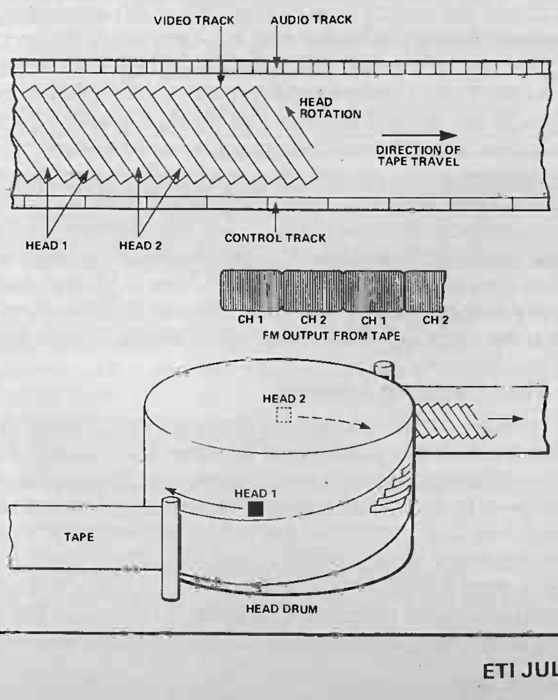
The first video cassette recorder appeared in the early 70s with Sony's ¾" tape U-matic being introduced in 1970 and the Philips NR-1500 system a year or so later. Both systems used the helical scan technique (see the box) and although both aroused some interest in the domestic market the great majority were sold to educational and industrial users. In time the U-matic recorders became the standard format for industrial users while Philips went on to the 1700 series and, to all intents and purposes, had the domestic market entirely to themselves.

However, in late 1975 Sony introduced the first examples of their Betamax home VCRs, whose technology was broadly based upon the U-matics although scaled down to use half-inch tape. Not long afterwards JVC (despite making U-matics under license from Sony) jumped in with their competing system, VHS. Initially this system offered longer recording times than Betamax (two hours) and for a few years a war was waged in the main market (the USA) with each format trying to offer longer play times. Indeed half-speed VCRs went on sale in the USA, and although they offer frugal use of tape the picture quality is truly awful. The broadcast video bandwidth is about 5 MHz and the average VHS recorder can manage about 2.8 MHz. Halve the recording speed and the video bandwidth drops to 1.4 MHz while the video noise level rises. The result is a fuzzy, grainy picture which is almost unviewable. Once VHS reached a playback time of six hours (half-speed) the competition became pointless — after all how many six hour movies do you want to watch?

HELICAL SCANNING

A conventional audio tape recorder uses linear scanning — the tape moves across the recording heads horizontally with the audio signal being recorded along the length of the tape. This system works well at the tape widths and speeds used because of the limited audio bandwidth (only 20 kHz or less). However, a video signal has a much greater bandwidth, as described in the main text; to record the TV pictures requires about 200 times as much information per second, yet the video tape is only four times wider than audio tape and travels about the same speed. How can the machine pack all the extra information on?

The trick is to make the recording head move as well; a speed of approximately 1500 RPM is used. Instead of passing the tape horizontally across the rotating head drum, the tape guides position it at an angle as shown in the diagram. Two tiny recording heads are positioned half-way up the drum and on opposite sides, so that one is always in contact with the tape. The rotation of the drum means that the heads sweep across the tape at about 5 metres per second; 200 times faster than an audio recorder. As the first head passes across the tape it writes a diagonal stripe of information; the slow movement of the tape across the rapidly spinning drum ensures that the second head will write its stripe adjacent to the first, and so on. This technique is called helical scanning and is used by all video recorders of all formats at present.



EEC VCC

Meanwhile back in Europe Philips, working at what seemed to be a leisurely pace, conceived their 2000 system which gradually became known as the VCC (Video Compact Cassette). The system was launched in partnership with Grundig and at once a major blunder was revealed. Somewhere along the line both companies arrived at a different understanding of the same drawing and positioned their audio heads at different points. The result was instant incompatibility between the two compatible models. There were red faces and dark mutterings all round, after which it would appear that Grundig dug their heels in and Philips did some quick mods! In its final form the VCC format offers a turn-over cassette offering four hours recording-time per side — so on the basis of playing time they have really socked it to the Japanese. The picture quality is very good and this is due in part to the clever use of a technique called Dynamic Track Following (DTF) which is explained in the second boxed section.

A more recent format intended for portable recorders is the Funai which also appears under the Technicolour and Grundig brand names. These recorders use an audio-sized cassette filled with ¼" metal tape. Although the quality is very good the high writing speed has limited the playing time to 30 minutes. The tape transport mechanism is very small and light with the result that the weight of a typical ¼" video recorder is now not much above 3 kg; hence the Japanese are now designing combined camera/recorders.

DYNAMIC TRACK FOLLOWING

The Philips and Grundig VCC video cassette recorders use an ingenious control system called Dynamic Track Following (DTF). Unlike the VHS and Beta recorders the VCC machines do not have a linear control track recorded on the tape; instead they have an arrangement based around the use of two video heads whose height can be adjusted by the means of a piezo-ceramic element.

During the recording process one head is held in a fixed position and the other is capable of being moved by a special error correcting signal. When the vertical blanking period occurs (and hence no visible picture) Head One is switched to playback and it sweeps the track just recorded by Head Two. One of the recorded signals is of 233 kHz and the detected signal causes Head Two to be moved until this signal reaches its maximum amplitude. When this is achieved the two heads are in their correct relative positions.

During playback the control is maintained by detecting pilot signals recorded along with the video signal. If the playback head reads only one signal then it is tracking correctly. If, however, it is mistracking it will sense two frequencies and an interference (or beat) frequency will occur. Thus if Head One is too high the error signal will be 47 kHz, and too low, 15 kHz. For Head Two too high the error signal will be 15 kHz and too low 47 kHz.

With this system the video heads will always be positioned correctly even with a still frame playback — in consequence a feature which VCC recorders excel at producing.

There is, though, a possibility that as the tape speed drops both heads will be lowered until they run out of their range. This is corrected by the Automatic Tracking Control (ATC) which, when it senses both heads mistracking, feeds a signal to the tape servo system to increase the linear tape speed.

Transports Of Delight

Electronically all these video cassette recorders are basically the same, their main differences being in the design of the tape transport; each format has adopted its own tape path and arguments continue about which is the best arrangement. For example, on Betamax recorders the tape remains wound around the video head drum at all times and the picture can still be viewed when the tape is being wound or rewound. The normal VHS deck has to unthread the tape for fast wind, rewind, and stop operations and this causes a tedious operating delay if you want to wind, check the picture, wind etc while looking for a particular portion. The latest generation of VHS machines can keep the tape against the head drum for cuing back and forth so the differences between these two formats are gradually becoming fewer.

The linear tape speeds are lowest on the Betamax (1.87 cm/sec), 2.34 cm/sec for VHS, and 2.44 cm/sec for the Philips VCC. Similarly there are differences in the writing speed, Betamax being 6.6 m/sec, VCC 5 m/sec, and VHS 4.85 m/sec. The linear speed is important to the fidelity of the soundtrack because the audio signal is recorded conventionally along a narrow track at one edge of the tape. As all three of these VCR formats have a linear speed of about half that of an ordinary audio cassette deck, the audio quality is for the most part pretty indifferent. A typical video recorder can have its audio performance compared to a low-cost cassette deck and still come out badly. Some video recorders now fit Dolby B, which is worth a 10 dB improvement on the signal-to-noise ratio, and Toshiba have a similar noise reduction system.

How Do VCRs Work?

The drawings show the block diagram arrangement of the video record and playback circuits. First of all it must be remembered that the recording process can only handle the wide bandwidth of the video luminance signal (DC to over 3 MHz) by using frequency modulation. The carrier signal frequency will vary with the amplitude of the video signal. Thus the peak white level may shift the carrier to 4 MHz, a black level to 3.3 MHz, and the sync pulses down to 3 MHz. This change in carrier frequency is referred to as 'deviation' and the total modulation is called 'modulation index' (M). Then

$$M = \frac{\text{deviation}}{\text{centre frequency}}$$

and for video recording will typically be 0.5. The FM will be passed through a low-pass filter to remove all components above the maximum deviation to give a band response as shown in Fig. 1. The process of frequency modulation is achieved by letting the luminance signal control the frequency of an oscillator whose output drives the recording head.

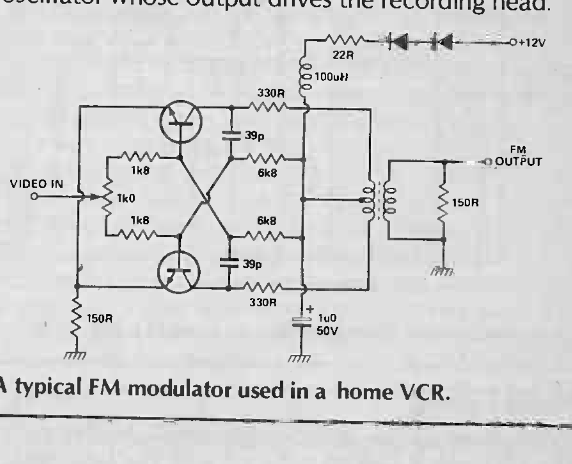


Fig. 4 A typical FM modulator used in a home VCR.

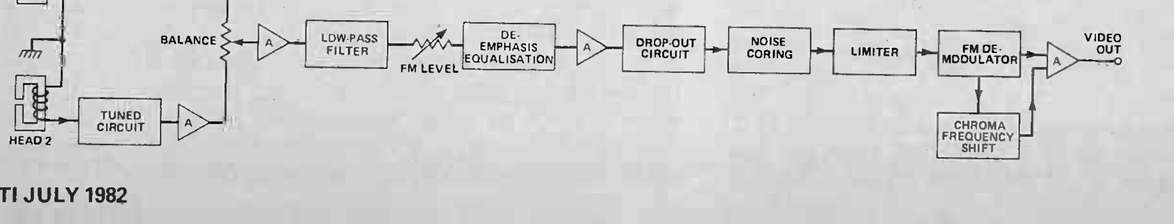


Fig. 5 Block diagram of a VCR playback system.

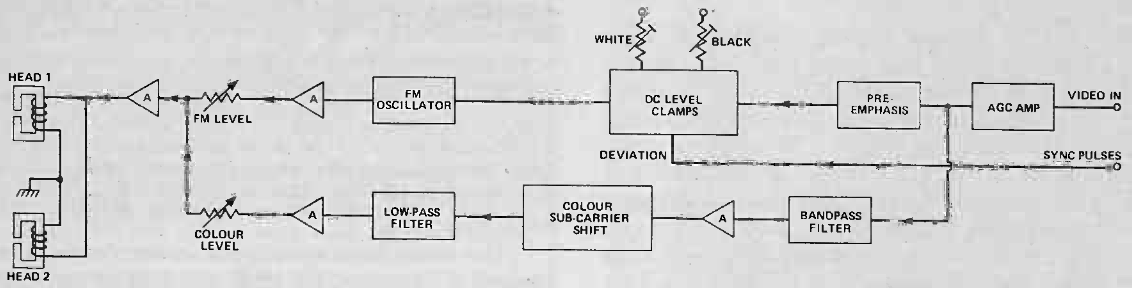


Fig. 6 Block diagram of a VCR recording system.

The chroma or colour signal doesn't modulate the same oscillator, even though on television signals it's modulated on a high-frequency sub-carrier (4.43 MHz). Instead the chroma signal modulates a low-frequency subcarrier of 750 kHz.

Two Heads Are Better Than One

If we now look at the block diagram we can see that on replay the output from the two video heads (alternately as they sweep across the tape) is first fed to a tuned circuit which resonates at about 5 MHz to peak up the frequency response which is falling off rapidly above 3 MHz. The output from the two heads is then balanced and passed through a low-pass filter to remove out-of-band noise, etc. De-emphasis follows to equalise the HF boost applied during recording. There then follow drop-out compensation and noise-reduction circuits which we will look at separately as they are of some interest. Then, after limiting to remove any AM components the signal can be demodulated and the chroma component frequency shifted to restore it. The chroma and luminance signals can then be mixed in a video amplifier to give a composite video output.

The recording process is almost the reverse with a slight variation. The chroma signal is frequency shifted but the luminance signal must have its maximum and minimum amplitudes defined by DC clamps before modulation in order to establish the maximum deviation. The two FM carriers are mixed at the output and fed to both of the video heads.

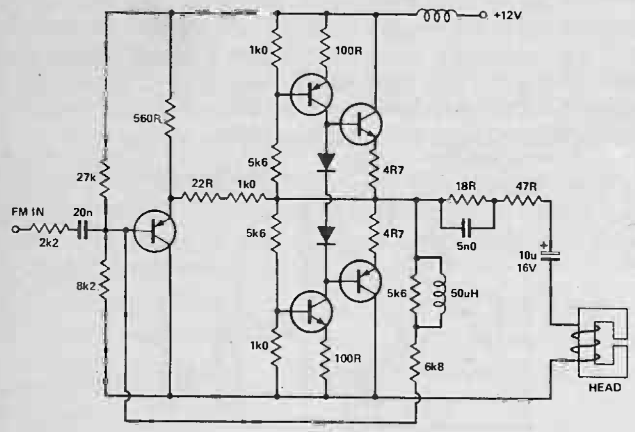


Fig. 7 A typical record drive amplifier as used in a JVC VCR.

Noise Coring

Video noise is an unavoidable result of the recording process, although it can be reduced with the best designs of video recorder and high-performance video tape. The effect of the noise is to make the picture grainy and hence lose the sharpness of lines and edges. To improve the subjective appearance of a picture, VCR manufacturers use a video-noise reduction circuit technique which is called noise coring. This works in the same way as a replay-only noise gate in audio. First, the high-frequency video signal is separated from the low freqencies. The HF signal is put through a clipper or limiter which removes the noise energy located on its average axis. The HF and LF signals are then recombined.

As with so much video circuitry there is virtually no value in drawing a circuit diagram of such a system, because it would consist of just three integrated circuits and a few resistors. For example, JVC use the 9V107 Filter/Amplifier, the SN7667 Limiter, and the VC2011 Mixing Amplifier/Buffer. The latest models use even fewer ICs!

If excessive coring is applied the picture will appear sharp but will also seem very unnatural, because much of the fine picture detail will be lost along with the noise.

Drop-out Compensator

The term drop-out is almost self-explanatory. When a segment of the tape has shed its oxide or has an embedded impurity then the recording will be interrupted and the signal will drop out. On the television screen these gaps are visible as random white lines that appear fleetingly on the screen. Get enough of them and they'll certainly ruin your viewing, so again the manufacturers have sought ways to minimise their effect. One technique is to substitute a picture line for the missing one but without an expensive memory this seems, at first glance, more than a little difficult.

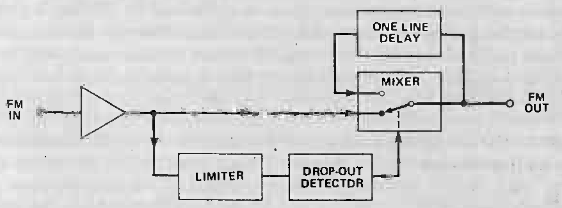


Fig. 8 Drop-out compensation.

However, as the drawing shows the usual circuit is quite simple. The FM signal is played back through a limiter (to maintain constant amplitude) and then fed to a drop-out detector which senses gaps in the signal. If a gap is found a DC control pulse is fed to the switch/mixer to disconnect the direct signal path and to connect instead the output of a 'one line' delay line. Thus the previous signal is substituted for the missing one. In this way the worst drop-outs remain unseen although a long term drop-out cannot be accommodated.

The Video Disc

Quite staggering sums have been spent by the electronics industry in developing and launching three competing video disc systems. It is seen as the Great White Hope for making billions of dollars of profits in the coming decade, although many industry observers feel there will be strong consumer resistance to a playback-only system. My favourite quote was from an RCA spokesman; "What's £200 million to a company like RCA"! The RCA system is called Selectavision and is made in the USA. From Philips (Holland)/Magnavox (USA)/ Pioneer (Japan) there is the LaserVision system and from JVC (Japan) there is VHD; so called because they haven't thought up a punchy trade name yet. It's no surprise that all three systems are totally incompatible and use completely different approaches.

Just Lasing Around

The Philips system is called LaserVision and is an optical system reading the signal encoded as pits on a reflective disc by means of a laser beam. This system was first launched under the Magnavox label (a subsidary of Philips) in America in late 1978. The system works well and the players have sold steadily but there have been continual problems with disc quality with (according to some observers) a 90% reject rate sometimes occurring. The current models use an expensive gas (neon) laser, although the design originally conceived the use of solid state lasers which will become available at a far lower cost — eventually, that is, so Philips are keeping an unusually low profile in their marketing, at least until they can make a worthwhile profit on the players. The difficulty is in manufacturing a solid state laser which has a wavelength short enough to focus on the very narrow signal track on the disc. Because the video signal is recorded in an analogue (not digital) form, it is not easy to correct the errors and ghosting which occur if parts of two adjacent tracks are simultaneously illuminated by the laser beam.

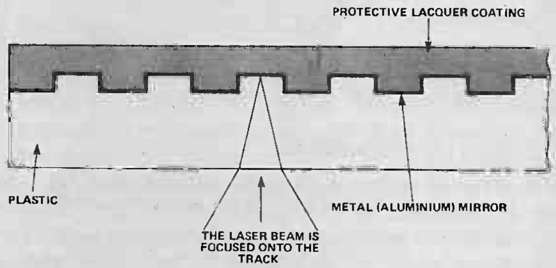


Fig. 9 A cross-section through the LaserVision disc.

Two types of disc can be viewed on the LaserVision player. These are CLV (Long Play) which gives a playback time of one hour per side; and CAV (Active Play), giving about 36 minutes per side. The long playing CLV disc keeps each video field the same length but increases the rotational speed as the 'groove lines' get nearer to the centre of the record. Thus the speed at the outer edge is about 500 RPM but by the centre of the disc thia has risen to 1500 RPM. The CAV disc is played with a constant motor speed but the length of the fields decreases closer to the centre as a consequence of the reducing diameter. This is the type of disc which makes LaserVision more interesting. The laser can move across the disc in 24 seconds, passing over some 54,000 separate television frames. Thus with the correct control mechanism the frames can be read one at a time (rather like a massive card index library), watched in fast or slow motion in either direction or be held in a 'still frame' mode — for months if necessary because in the absence of physical contact there is no wear on the disc.

The CLV discs will normally be used for recordings of entertainment because these 'special effects' are not possible. The rotational speed of the disc varies so that there is never a fixed number of picture frames in each rotation of the disc.

The Versatile VHD

The late-runner in the video disc contest is VHD (Video High Density or Video Home Disc) system developed by JVC in Japan and backed in the UK by the Thorn-EMI group. This system is very similar to both LaserVision and Selectavision but uses a 10" diameter disc instead of one of 12". The disc (which plays for one hour each side) is pressed from conductive plastic, with the signal pressed in as raised and lowered patterns but no groove as such. The signal is read by a capacitive pickup which follows the spiral track by using a sort of parallel tracking servo-controlled arm. The VHD disc suffers some wear in use because the tracking stylus slides over its surface and so the repetitive play of, say, a still frame could, as with Selectavision, shorten the disc's life.

As it stands the VHD system is not ideal for the playback of still-frame pictures. Each rotation of the disc holds in its signal track two full television frames ie four interlaced fields. Thus if the same track is scanned repeatedly there will be some visible 'judder' of the picture as the two different frames alternate. Two solutions exist. The first is to feed the frame into a digital frame store where it is converted from analogue to digital form and loaded into a memory. The 'frozen' picture can then by continually readout from the memory, converted back to an analogue video signal and fed to the television to give a perfect still frame.

This is the approach the television companies take and it's an ideal approach except for one thing — the video frame stores cost £17,000 and upwards, hardly suitable for fitting inside a £350 video disc player. So JVC have adopted the somewhat more pragmatic approach of recording each TV frame twice, so that for every rotation of the disc only one picture is seen. Now such a doubling-up will mean that the programme will be viewed in slow-motion. Solution? Easy — just double the rotation speed and accept that these discs (Type II) will only run for 30 minutes a side. If we put a three hour blockbuster movie on to these discs we will need three, viewing all six sides, so the early years of video disc may resemble the days when a complete opera could only be heard by playing a stack of 78 RPM records! Already there is talk in Japan of an 'autochanger' video disc player — the juke-box of the future.

VIDEO DISC MANUFACTURE

Of all the video disc types, probably the hardest to manufacture is the optical disc used in the LaserVision system.

The original video programme is recorded onto a professional video tape recorder using 1" or 2" tape. It is played back on to the disc 'cutting lathe' where a high-power laser beam tracks a modulated light beam over an ultra-smooth glass disc coated with a photo-resist material. The exposed disc is chemically developed, etched, and washed to leave a visible spiral of pits etched into the glass surface. The glass is then coated with a fine coating of silver to form a conductive layer, which also allows the disc to be played as a quality check without damage (there is no physical contact with the disc's treated surface).

The disc is then plated with nickel, followed by a layer of aluminium. The glass master is then removed (and damaged beyond further use) to leave a negative which is referred to as the 'father'. More nickel is electro-deposited onto the record side of the father to produce a positive 'mother', from which a number of nickel negative stampers can be grown again by electroplating.

The video discs proper start off as a blank sheet of 1.3 mm acrylic (Perspex) which are thoroughly washed and then coated with a 30 micron thick layer of photosensitive lacquer. The disc is then gently pressed against one of the negative stampers to give the spiral track of indentations; they can then be exposed to the ultraviolet light which hardens the photo-resist. The discs are then loaded into large vacuum chambers where they are immersed in an aluminium vapour for about 30 minutes. This vapour causes a very fine reflective coating to be deposited on the disc; a coating which is then protected by a layer of clear lacquer.

So far one disc side has been produced, so it is glued to another side and the final two-sided disc is balanced electronically to ensure stable rotation in the player.

The entire process is semi-automatic up to the last important stage — final inspection. At present the discs are checked by actually playing them with an operator watching the programme on a television screen. The inspector checks four discs at a time (four screens) in what must be one of the most boring jobs of all time. However it has not yet proved possible to automate this process or to rely upon fast playback during inspection.

The RCA Selectavision Video Disc System

This system uses a flat circular disc of 12" diameter which has the television sound and picture signals recorded on a spiral groove rather like an audio disc. However, there are 10,000 grooves per inch (compared to 250 on a audio disc) and the information is recorded as frequency-modulated vertical undulations of the V-shaped groove. The plastic disc contains a fine carbon dust to make it conductive and is covered with a film of oil to lubricate the playback stylus and so increase the

Bosom Buddies

Just when we were wondering where all the photos of lovely ladies had disappeared to, this dropped through the letterbox. At 3½" x 1½", this Standard Telephone and Cables radiopager is probably the smallest in the world. It has four separate tone patterns to indicate messages from different sources and can be muted during meetings to record calls silently for playback later. The picture shows how the unit can be hidden in your cleavage; people with differently-shaped chests can clip it to their clothing.

TK Transmissions

Remote control is the forte of TK Electronics and their latest kit departs from their usual wares in that it uses mains control, not IR or ultrasonics. Any electronic appliance which plugs into the mains can be controlled using a hand-held transmitter, also connected to the mains. Up to a total of 16 receivers can be operated anywhere in the house and switched on and off by more than one transmitter. The receiver may be coded so that you don't interfere with your neighbours and vice versa. The transmitter has the advantage that it can be activated by logic signals — the computer-controlled home becomes a reality. The kit contains a transmitter and two receiver units for £42 plus VAT, although transmitters and receivers are available separately. TK Electronics are at 11 Boston Road, London W7 3SJ. Enquiries on 01-579 9794; tell 'em ETI sent you.

Credit card customers with poor memories have been accommodated by TK, who've got a new phone number, 567 8910 (geddit?). Dialling this number and quoting your Access/Barclaycard number will get you same day despatch of components. A 6" x 9" SAE to the above address will secure the free 1982 shortform catalogue.

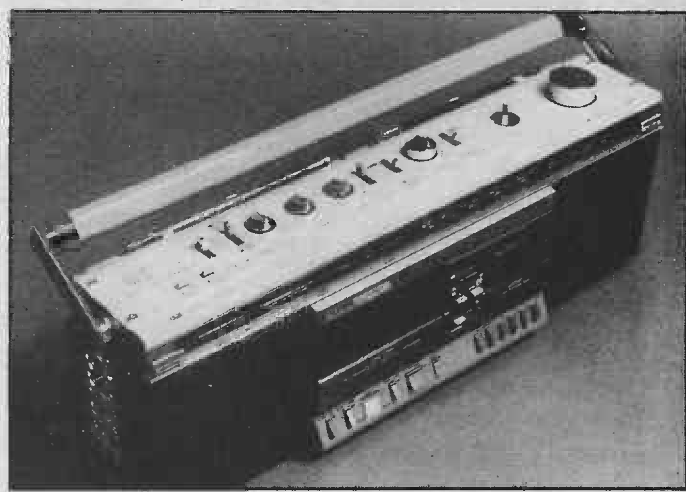
Shorts

- Triangle Digital Services of 23 Campus Road, London E17 8PG, are now supplying a free handbook on their industrial speech synthesis system. Various products are described included the low cost custom vocabulary service.
- We get some amazing stuff sent to us. Would you believe that the Edinburgh District Council is fitting microprocessor-controlled weighing systems in their new abattoir. Obviously a case of steak and chips — oh dear, these offal puns. . .
- Two new high-speed analogue multiplexers from Burr-Brown, the MPC 800 and MPC 801; they provide up to 16 single-ended (eight differential) channels or eight single (four differential) channels respectively. Each chip is self-contained with onboard address decoding for channel selection.
- Crow of Reading have introduced the Barcovision video projector, which can give a maximum picture size of 6 metres diagonal on any flat screen, with viewing angles only restricted by perspective distortion. I want one. . . and a videotape of Debbie Harry.
- Greenweld have sent us their latest catalogue, and you can have one too for the sum of 50p plus 25p postage. Also included are 60 pence worth of discount vouchers, bargain list, wholesale list and a reply paid envelope.
- Texas' new CCD image sensing chips use a patented new technology to avoid the need for a two-phase clock. The chips are sensitive to light across the visible spectrum, and feature 1728 x 1 and 128 x 1 pixel resolution.
- Verospeed are adding more than 100 CMOS devices to their catalogue. Industrial users will now be able to get same day despatch at competitive prices for semiconductors they require urgently. Speed indeed. . .
- Kentec of Sevenoaks have introduced a one-board Z-80 based computer with 2-4K RAM, 2-8K ROM, 36-key keyboard, I/O port and 300 page manual. It's intended to be a low cost development tool; the price is £65 including power supply.
- Lander Microsystems, 32 Clockhouse Lane, Collier Row, Romford, Essex are selling the LM124 EPROM programmer for use with the TRS-80 Model I level II (16K). Most single-supply EPROMs can be blown, without personality modules, and commands include BURN, COPY, EXIT, FILL, LOAD, MEMORY, NEW, PAGE, READ, SAVE, TEST and ZERO.

Amazing Aiwa

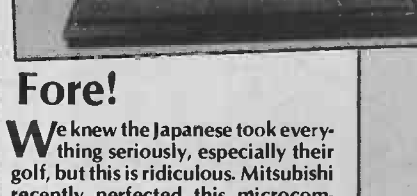
New from Aiwa (or I-eee-waaaaah, as their ads on commercial radio would have us believe) is the CS-W7 compact/micro stereo radio cassette recorder. This incredibly versatile unit incorporates two tape formats, micro and compact, with a versatile dubbing facility which works from either one to the other. For example, you can record from one tape onto the other in either direction while listening to a completely different programme on the radio. You can make two simultaneous recordings from the radio or an external source. You can combine voice and music from separate recordings. Wow!

The audio output of the CS-W7 is 5 W per channel; the radio is four-band with a sleep timer. The microcassette has two playing speeds so you can choose between accurate speech recording or a longer playing time; both tape units offer metal tape compatibility and editing functions. The CS-W7 retails at around £190 from authorised Aiwa dealers.



Wrist Radio

Come back Dick Tracey, all is forgiven. Trafalgar's Radio Watch '82, believe it or not, is the same size as an ordinary digital watch with all the usual functions, but also contains an AM radio with hi-lo volume control and high-quality earpiece. Damned if I can see where you plug it in, though. Definitely one of the smallest portable radios, so over to you, Sony. Trafalgar Watch Co Ltd., Trafalgar House, Grenville Place, Hale Lane, London NW7 3SA.



Fore!

We knew the Japanese took everything seriously, especially their golf, but this is ridiculous. Mitsubishi recently perfected this microcomputerised golfing aid so you can brush up on your strokes at home. Working with Namio Takasu, a Japanese professional golfer, Mitsubishi have designed the GL-500 to display such data as head speed, face angle, hitting area, swing arc, ball direction and other esoteric information every time you swing at the ball. The unit contains four sensors in the mat base, a built-in microcomputer, a charger and carrying case, so the Japanese can practice almost anywhere. Brits, however, will have to stick to knocking balls across the office into the wastepaper basket — there are no plans as yet to market the GL-500 in Europe.

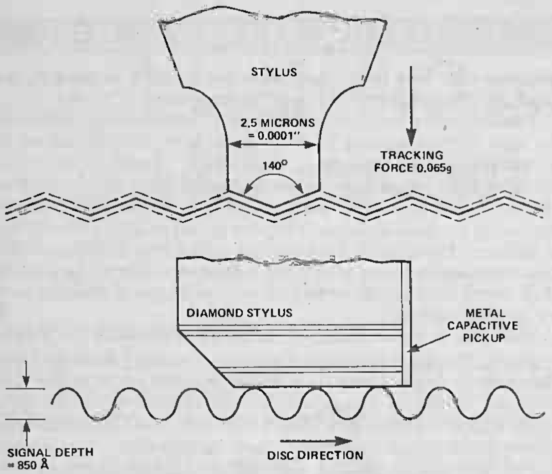


Fig. 10 RCA Selectavision groove geometry.

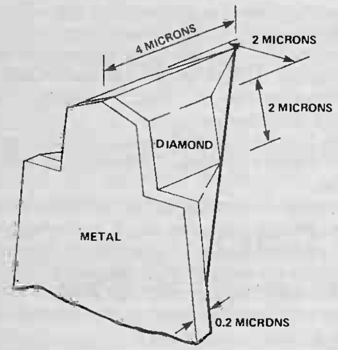


Fig. 11 The RCA stylus.

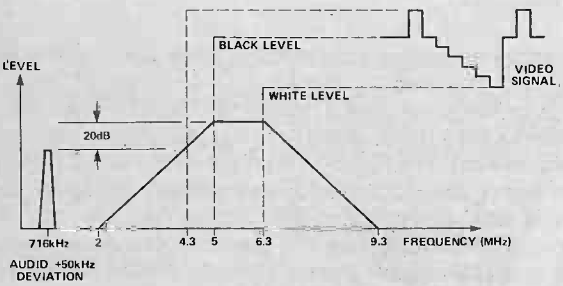


Fig. 12 The RCA video disc signals.

life of the disc. The stylus runs in the groove but actually senses the signal through the use of capacitive coupling between the stylus and the disc — each being one plate of a capacitor. The stylus is long enough to ride over the signal peaks pressed into the groove, so the disc surfaces rises and falls under the stylus electrode giving a capacitance variation of about 1×10^{-4} pF peak-to-peak. This almost insignificant change in capacitance is what constitutes the output signal.

The capacitance is made part of a 910 MHz resonant circuit which is fed with a signal from a 915 MHz oscillator; a frequency which falls at the half-amplitude point on the resonant curve. As the disc-stylus capacitance varies, so does the resonant frequency and the amplitude of the 915 MHz oscillator output signal. Thus over a period of time the 915 MHz signal is amplitude-modulated by the disc signal which can be simply recovered using a diode detector. The output signal is, in fact, two frequency-modulated carriers, these being 716 kHz (audio) and 5 MHz video. These carriers are fed through limiter amplifiers to take care of the 20 dB or so of level variations that occur, and the constant amplitude signal is fed to phase-locked loops for demodulation. The remaining circuitry contains quite complex arrangements for reconstituting the composite video signal and others that detect and compensate for playback errors.

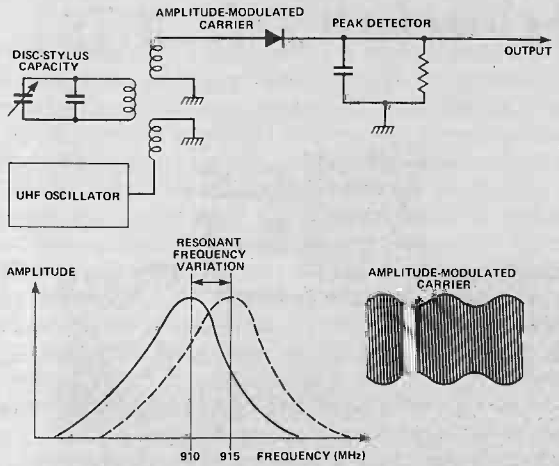


Fig. 13 The Selectavision playback demodulation system.

Are You Being Servoed?

Early prototypes of the CED disc system (as it was known) used an ultra-lightweight tone-arm which supported the capacitive pickup in a conventional record player fashion. However the current design of pickup is quite heavy, incorporating the high frequency resonators and amplifiers; and so it cannot be guided by the extremely small side forces generated by the microscopic groove walls. For this reason the pickup is mounted in a servo-controlled arm which tracks across the disc in response to the stylus motion (see Fig. 14). Obviously no servo or gear-train could follow every small movement of the stylus so the pickup is allowed some 2 mils of free motion.

As the drawing shows, a conducting 'flylead' is positioned in-line with the stylus and its position is detected by two sensors, one to each side. The two sensors are varactors whose capacitance is modulated by a 260 kHz oscillator. Each is of opposite polarity, so with the flylead absolutely central the capacitively coupled 260 kHz variations cancel out. An offset in position will cause a 260 kHz component to be detected by the flylead and result in an error signal being sent to the arm motor which will reposition the arm. The use of these opposing sensors largely cancels out most of the temperature variations and provides a very stable electrical centre.

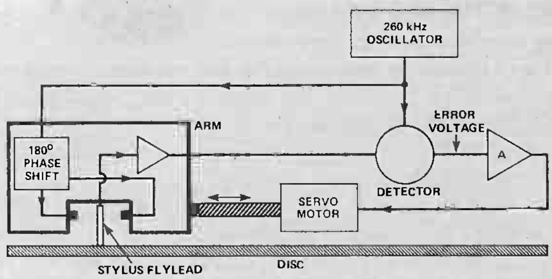


Fig. 14 The stylus position servo system.

Looking Ahead

Finally, the future. Matsushita and Pioneer (with some work by RCA) have produced optical video disc systems which can record programmes (or data) and subsequently replay them. During recording the laser works at full power and burns into the surface of a blank disc; when switched to low power it reads it back LaserVision-fashion. The Matsushita system puts 15,000 still pictures on to an 8" disc and although that represents only five minutes or so of a video programme, it will only be a short step to a complete optical record/playback disc. Meanwhile Sharp and Matsushita (Panasonic) are working on a magnetic disc recorder which also uses a laser, but in this case it alters the magnetic characteristics of the disc coating. **ETI**

SERIES 5000 BRIDGING ADAPTOR

Some like it loud. Here's how to operate the two ETI-5000 MOSFET power amp modules in the Series 5000 power amp in bridge configuration with the addition of a simple, inexpensive module.

This project consists of a unity gain phase inverter that can be installed within the Series 5000 power amp. The input to one of the power amps is disconnected from the input socket and is wired to the output of the bridging adaptor. The input of the bridging adaptor is connected in parallel with the input of the other channel. This leaves one of the input sockets unused, although it could be connected to the other input socket if required.

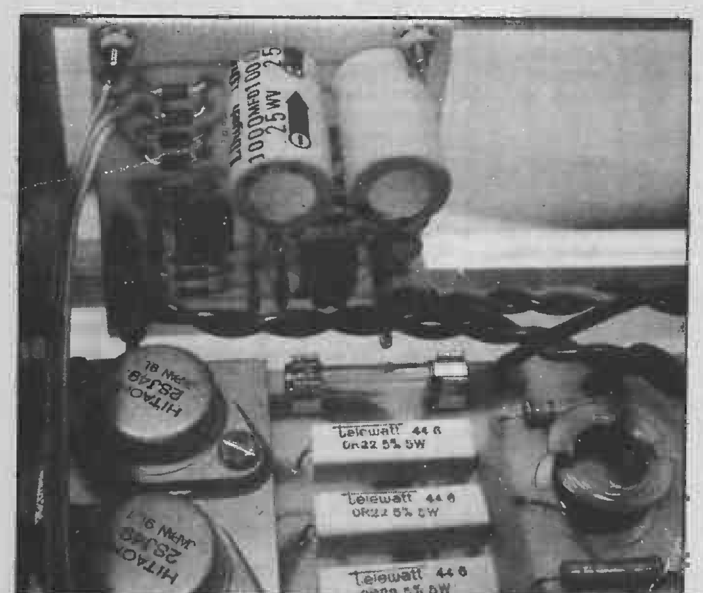
The bridging adaptor must not degrade the distortion figures of the amplifier to which it is connected. Similarly good noise figures and freedom from slew-induced distortions must be ensured through careful design of the unity gain amplifier stages. Unfortunately, amplifiers with a gain of one tend to be the most difficult to stabilise because of the relatively high amounts of negative feedback. To overcome this problem and to maintain good noise figures, NE5534N op-amps were used in the design.

Noise Problems

The conventional way to achieve an inverting amplifier is to ground the non-inverting input and insert the input signal into the inverting input via a resistor. In this configuration the inverting input is also connected to the output of the op-amp through another resistor and forms a virtual earth point. The input resistor therefore forms the input resistance of the stage. Since this is connected to the output of the preamplifier the value of this resistor must be high, ie around 10k-100k. Unfortunately, this would seriously degrade the noise performance. To overcome this problem the bridging adaptor has been broken into two stages. The first is simply a unity gain buffer. This stage has low noise figures and an output impedance low enough to drive the following inverter stage. Since the input resistor has been kept to a small value in the second stage a good noise figure results.

PARTS LIST

Resistors (all ¼ W, 5% except where stated)	
R1,2	100k
R3-6	1k0
R7,10,11	100R
R8,9	1k5 2 W
Capacitors	
C1	220n polyester
C2	1n0 ceramic
C3,4	10p ceramic
C5	10n ceramic
C6	100u 25 V PCB electrolytic
C7,8	1000u 63 V PCB electrolytic
C9,10	10u 25 V tantalum
Semiconductors	
IC1,2	NE5534N
D1-4	1N4001 or equivalent
ZD1,2	12 V 400 mW zener
Miscellaneous	
PCB (see Buylines); mounting hardware; hookup wire.	



The board installed in the Series 5000 amp at the left hand end of the chassis.

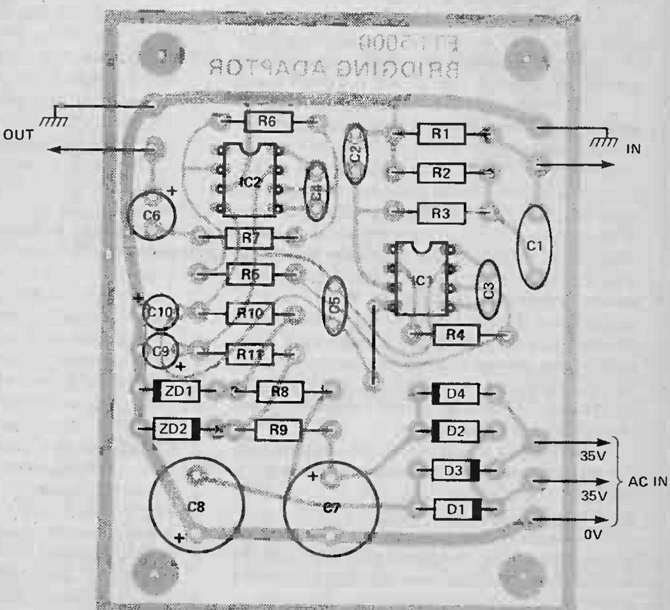


Fig. 1 Component overlay for the bridging adaptor.

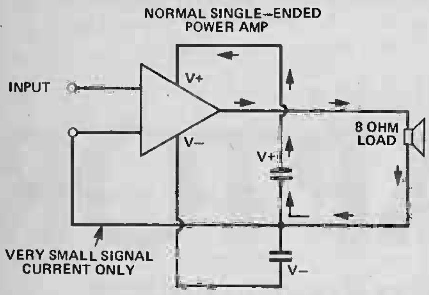


Fig. 2 Single-ended power amp showing how current flows in the power supply and the load.

Construction

Construction of the bridging adaptor is not difficult since all components are mounted on the PCB. The components can be mounted on the board in any order, although it is probably best to leave the two large electrolytic capacitors until last. As usual, be careful of the orientation of all polarised components such as the electrolytic capacitors, ICs and diodes.

Solder input and output leads to the board and bolt to the side bars on the left hand side of the power amp (viewed from the front), as shown in the accompanying photograph. Use twisted pairs of 32 x 0.2 mm plastic-covered hookup wire, as with the existing input wiring. Solder the output directly to the input of the power amp closest to the bridging adaptor. Solder the input leads of the bridging adaptor to the input socket of the other power amp. Included here is a block diagram of the Series 5000 power amplifier showing suitable modifications to incorporate the bridging adaptor.

Performance

The prototype bridged Series 5000

HOW BRIDGING WORKS

The amount of power an amplifier can deliver into a certain load is determined by the simple equation:

$$P = V^2/R$$

where V is the supply voltage and R is the resistance of the load. To achieve more power we must either decrease the resistance of the load or increase the supply voltage. Either of these will cause an increase in the amount of current to flow, and this must be catered for in the design. Unfortunately, power transistors are limited by the maximum voltage they can withstand so the supply voltage cannot be increased indefinitely. An amplifier with a supply voltage around 50 V is probably capable of supplying around 40 V peak to the load, the remaining 10 V being dropped by the output transistors, driver transistors and the power supply. This corresponds to a power level of around 100 W RMS into an 8 ohm load.

In order to increase this the load could be decreased to 4 ohms, for example. The simple equation above predicts a power level twice that of the 8 ohm case. In practice this ideal is never met since the increased current causes increased voltage drops. In the case of a MOSFET output stage such as the ETI-5000, the relatively high on resistance will cause quite a high voltage drop, decreasing the maximum output power to around 150 W for a 4 ohm load.

In order to increase the power of audio amplifiers it would seem we must increase the supply voltage and design the amplifier so that it is capable of withstanding higher signal currents. A closer inspection of Fig. 2, however, reveals another alternative. The conventional power amplifier consists of the amplifier itself and a power supply, as shown in the digram. The power supply is represented by the pair of capacitors. These correspond to the main storage capacitors in the power amp. The rest of the power supply has been omitted since its purpose is simply to maintain the necessary DC voltage differential between the ends of the capacitors.

In a class B output stage only one of the output capacitors is supplying energy to the load at any given time. The arrows in the diagram indicate the direction of the current flow when the power amp is delivering a positive-going output signal. As can be seen, the large signal current flows from the positive supply capacitor to the power amplifier, through the load and via an earth return path to the electrolytic capacitors. Every wire in this current path has resistance, so voltage drops occur at all points in the circuit. These voltage drops can be extremely significant in the performance of the power amplifier.

The distortion figure for the ETI-5000 module, usually around 0.001%, can be degraded to worse than 0.3% if the resistance in the power supply leads exceeds a small fraction of an ohm. If extremely low distortion figures are required the entire heavy current path and earth leads should be wired with one of the very *low resistance* speaker cables available.

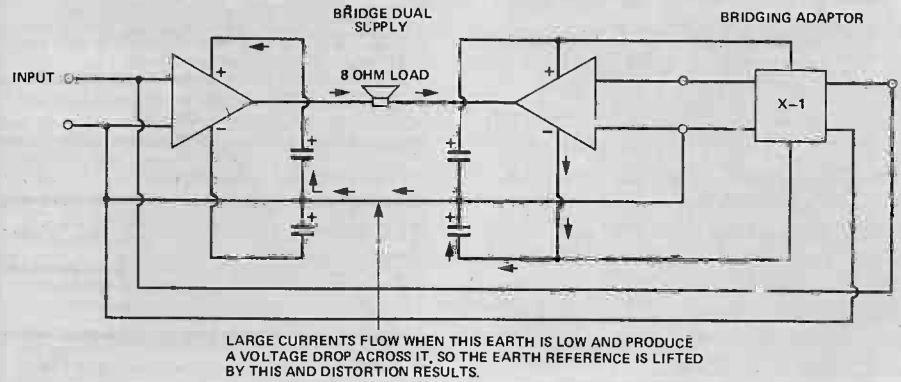
We have seen above that at any given time in a class B power amp only one of the capacitors is supplying power to the load. So the load has access to only one of the supply rails. If both supply rails could be used at the same time the voltage available to the load would be doubled without having to redesign the amplifier, so long as the resulting current were within its capabilities. This is the purpose of the bridge configuration with power amps, sometimes referred to as 'bridging'. The principle is shown in Fig. 3. Two identical power amplifiers have been used here, the output of each going to opposite ends of the load. The input signal is fed to the input of the first amp in exactly the same way as in the more conventional approach. The arrows indicate the direction of current flow for a positive-going signal voltage. At the same time, the input signal is fed to the second power amp via a unity gain phase inverter. A positive-going input signal voltage becomes a negative-going signal at the input of the second amp. While the output of the first power amp is swinging positive the output of the second amp is swinging negative, so the load experiences double the supply voltage (neglecting for a moment the increased voltage drop due to increased signal current).

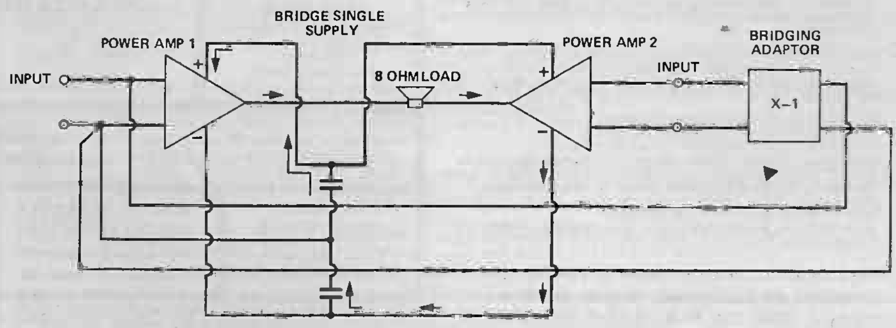
In the 4 ohm case discussed earlier the signal current is doubled, while the supply voltage remains much the same; the maximum power is therefore doubled. In the bridge case, however, the maximum signal voltage is doubled, thus also doubling the current. Since power is given by the product of voltage and current the power increases by a factor of *four*. In a real amplifier, of course, this power is never achieved. Once again the voltage drops across the output transistors, etc will decrease the power considerably, and this is especially true when using MOSFET output devices. To make a closer estimate of the power that can be expected of an amplifier when connected in bridge, determine the power delivered into a load of half that used in the bridge and double this value. If the bridge is to be used with an 8 ohm load, for example, determine the power delivered by one amplifier into a 4 ohm load and double this figure. In the case of the ETI-5000 module the power into 4 ohms is around 150 W RMS, so the power achieved by two 5000s in bridge should be around 300 W RMS. Measurements carried out with the bridging adaptor gave power figures between 280 and 300 W RMS, in good agreement with the estimate.

There are also limitations, however, which must be considered for successful operation of a bridge amplifier. First, since each amp is effectively driving a load half that of the real load, the load resistance connected to a bridge amplifier must be twice the minimum load specified for individual power amps. Since the minimum load recommended for the ETI-5000 module is 4 ohms the minimum load used in bridge should be 8 ohms.

Another problem associated with bridging is that both power amps used should share the same power supply to ensure the integrity of the earthing system. If this condition is not met, the distortion figure and stability margin of the amp will almost certainly be degraded. In Fig. 3, two independent power amplifiers are connected in bridge. This is done by joining their earth reference points together and driving the loudspeaker with out-of-phase signal voltages. Current resulting from a positive-going signal voltage flows from the positive supply through the first power amp and through the loudspeaker to the second power amp, and then to the negative supply rail of the second power amp. The circuit is completed by the connection between the two earth points. The problem is that, since this connection has a finite resistance, a voltage drop will occur across it, varying with the signal voltage and modulating the earth current for the second power amp.

The solution is to operate both power amps from a single power supply. Figure 4 shows a pair of amps connected in bridge and using a common supply. Once again, the arrows show the direction of current resulting from a positive-going signal voltage. Notice that in this case the connection between earth reference points has been eliminated and both power amps have access to the same single reference point. This is one of the reasons the Series 5000 power amplifier was configured with a single supply even though two power transformers and a total of four electrolytics were used; the two channels in a stereo power amp could be bridged, forming a mono power amp. For stereo operation two such amplifiers are required.





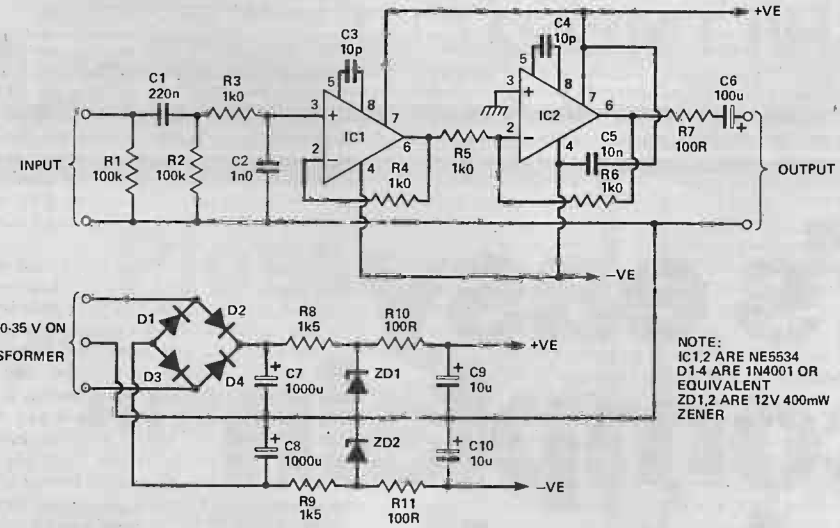


Fig. 3 (Top left) Two separate bridged power amps showing individual power supply and load currents.

Fig. 4 (Centre left) Bridged power amp and single supply showing load and supply current flow.

Fig. 5 (Bottom left) Circuit diagram of the bridging adaptor.

amp performed favourably and gave distortion figures around the resolution of our THD analyser (approx. 0.003%). Similarly, noise figures were not degraded and the adaptor tested was free of slew-induced distortion. The power output achieved was around 300 W RMS when connected to an 8 ohm load. Connection to a 4 ohm load is *not* recommended for the reasons given in the accompanying box.

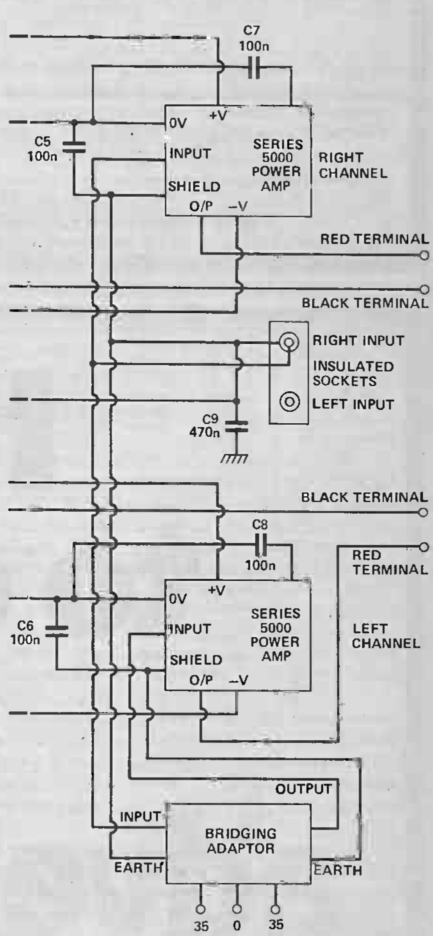


Fig. 6 How to wire the bridging adaptor into the Series 5000 amplifier for bridged operation.

HOW IT WORKS

The Bridging Adaptor is a unity gain (gain of x1) inverting stage that has its input in parallel with one power amplifier module and its output driving the other power amplifier module. Thus the power amp module it drives operates out of phase with the other power amp module.

The bridging adaptor has two stages — a non-inverting input buffer stage and an inverting output stage. The active device in each stage is an NE5534 high performance op-amp. A on-board rectifier provides dual supply rails regulated by two zeners.

Input is coupled to the non-inverting input of IC1 via an RC network consisting of C1, R2, R3, and C2. Resistor R1 provides a DC return for the input line. Resistor R3 is a low value to ensure good noise performance for IC1, and together with C2, a lowpass filter is established to limit the slew rate of incoming signals to prevent slew-induced distortions. Feedback for IC1 is provided by R4, connected between the output and the inverting input. The output of IC1 drives the inverting input of IC2 via R5. Feedback around IC2 is provided by R6. The feedback constants for both IC1 and IC2 are arranged so that each stage has a gain of one.

The output from IC2 is coupled via R7 and C6, which provide a low frequency rolloff, C6 also providing DC blocking.

The bridging adaptor is powered from the Series 5000 amplifier power supply transformers. Diodes D1 to D4 form a bridge rectifier providing about ±52 V DC with respect to the winding centre tap. Capacitors C7 and C8 provide smoothing. Two zener diodes, ZD1 and ZD2, are used to provide regulated positive and negative 12 V DC supply rails for the two ICs. Resistors R8 and R9 provide current dropping for the two zeners and R10/C9, R11/C10 provide further filtering. Capacitor C5 provides a high frequency bypass for the supply rails. Capacitors C3 and C4 provide frequency compensation for IC1 and IC2 respectively.

BUYLINES

As usual, we can supply you with the PCBs; the order form is on page 71. Nothing else should cause any problems; the NE5534 is available from Watford Electronics, or as an alternative you could use the TDA1034 from Technomatic.

MICROPHONE SWITCHING UNIT

Muddled by multiple mike inputs on your mixer? Don't let your brain take the strain; our Voice-Switch unit will do all the work for you. Design by Vivian Capel. Development by Peter Green.

This unit was specifically designed as an addition to public-address microphone mixers, although other applications are possible. With many public-address systems there is a need to operate a number of microphones at the same time, such as when covering debates, multiple interviews, and similar items. If all the microphone channels are left open and faded up, extraneous noises such as foot shuffling, note rustling and even whispered asides can be caught and relayed to the audience, as well as the possibility of acoustic feedback being increased.

To avoid this the operator keeps open only the channel actually in use, and fades up each in turn as required. The snag is that when the debate gets lively, someone often starts speaking before the operator can fade up his microphone and the first few words are lost to the audience.

An additional difficulty can be to identify the microphones if they are passed from one participant to another or there are unexpected changes in seating positions. Thus it sometimes happens that the wrong microphone is faded up and a minor panic ensues until the correct control is found and turned on. Operating the mixing console in such situations can be a harrowing experience!

Vox Popular

Much if not all of this hassle can be avoided by the use of voice-switched microphone channels. Some expensive professional mixers have this facility, and there is also an American-made add-on unit of which one is required for each channel controlled. This makes the set-up rather bulky as well as costly.

With the module here described, although one is required for each channel, it is small enough to be easily accommodated inside most mixers alongside the appropriate input socket or fader. A switch and sensitivity control must be fitted to the control panel, but connection to the mixer circuitry is simple and needs little disturbance to the existing wiring.

So what exactly does it do? It mutes the channel to which it is connected, opening it only when the microphone picks up sound of a predetermined level. Thus all the channels can be faded up but none will be live until the particular microphone is actually used. Switching is fast, during the first cycle of received sound, so there is no audible loss of starting sound when the sensitivity control is correctly set.

Open Channel

The channel remains open as long as the microphone is being used and for five seconds afterward, after which it mutes. Thus it stays on during pauses in speech or when the voice is dropped providing the break is no longer than five seconds. If the pause is longer, the microphone will come on again as soon as speech is resumed, but in practice five seconds has been found to be about right. Longer or shorter times can be obtained by changing the value of a capacitor.

In addition, a very useful extra facility is the provision of an LED indicator mounted on the control panel near each fader control. This is switched on when the channel is open and so informs the operator immediately which microphone is being used. Hence he can make instant adjustments to the volume as required, even if his view is blocked and he would otherwise be unsure of which control to operate. This does away with the need for colour-coded cables and other devices previously used to identify microphones on stage.

The supply voltage can be from 9-12 V, and so can be taken from the mixer supply; current required is not large and well within the ability of most mixer mains supply circuits. For battery operated mixers certain steps can be taken to reduce the current needed even further, and these will be described later.

Mounting And Wiring In

The small size will enable the module to be fitted without much trouble. One is needed for each

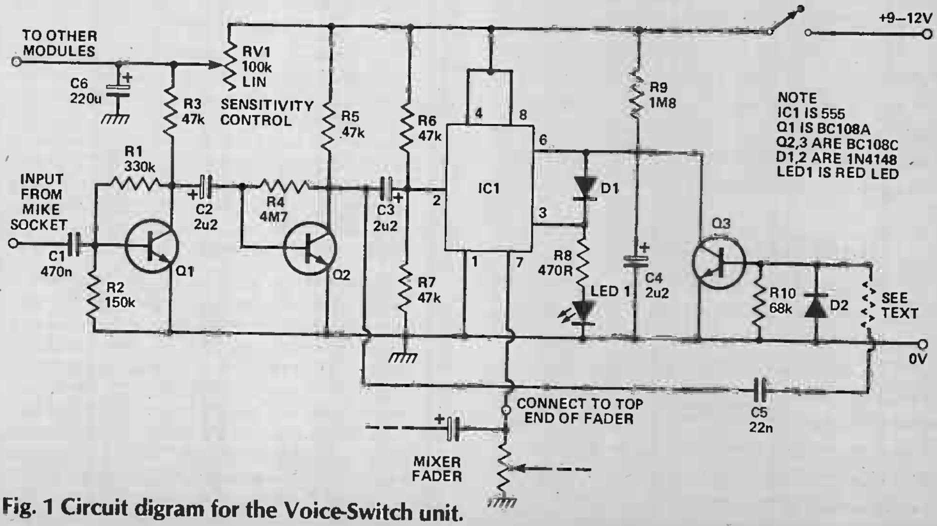


Fig. 1 Circuit digram for the Voice-Switch unit.

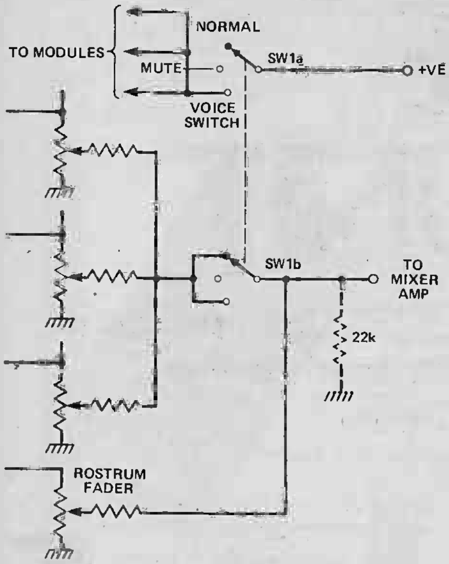


Fig. 2 A double-pole double-throw switch with centre off position can be used to include manual muting of all but the main channel. This is how to connect it up.

channel to be controlled. Generally, it has been found best to fit them to all microphone channels except the one used for the chairman or main rostrum, as this one is usually in continual use.

The best point for connecting the muting circuit is to the top end of the channel fader. With battery operated mixers it is not necessary to use screened wire for this, but it should be used for mains powered units. Earth the screen at one end only to avoid hum loops. The input terminal on the PCB is wired to the live terminal of the microphone input socket; existing wiring to the socket is not disturbed. In the case of a balanced input, it can be taken to either of the two signal terminals.

Positive supply wires should be taken individually (not looped) to the Voice-Switch control switch. This can be an ordinary miniature on/off toggle switch, the other terminal of which is taken to the positive supply. When in the 'off' position, there is no muting action by the 555, and the mixer works normally. An additional refinement which has been found useful, though not essential, is to include manual muting of all channels (except the main rostrum channel) in this switch.

Doubling Up

In this case, a double-pole double-throw switch with a centre 'off' position is required. One pole is wired to switch on the supply to the modules in the down position as with the simple on/off switch. The other pole is made to break the circuit between the isolating resistors that go to the wipers of the faders, and the input to the mixer amplifier, when in the centre 'off' position. It should be easy to locate the position on the mixer PCB where the resistors from the fader wipers are connected together. The track is cut here and a wire taken from each of the two sides to the switch. The resistor from the main channel will have to be taken to the other side of the cut so that it will not be switched with the others.

One of the two wires coming from the modified PCB is taken to the moving contact of the switch, while the other is connected to *both* the fixed contacts. Thus the circuit is broken only in the centre 'off' position. The action of the switch is therefore: UP, mixer operating normally; CENTRE, all channels except rostrum muted; DOWN, all channels except rostrum controlled by Voice-Switch. The purpose of this facility is that when microphones are being handled and passed around, handling noise can switch on the Voice-Switch. It is therefore useful to be able to manually mute them until the participants are settled, when the Voice-Switch can take over. It can be seen now why the main rostrum or chairman's channel needs to be separate and on all the time, as he will undoubtedly be introducing the programme or participants while they are coming on and taking their places.

Wiring from the positive supply to the switch should be fairly substantial, 13/02 or 16/02, as it is necessary to avoid common-impedance coupling between the units. Likewise the earth wires should be taken individually to an earth point and not connected from one to the other.

Sensitivity Control

Individual sensitivity controls could be fitted if required, but where all microphones are of the same type or the same sensitivity, as will generally be the case, it is sufficient to have a single control for all channels. This means only one control has to be fitted to the control panel in addition to the switch. If desired, the control could be a preset mounted inside the mixer and set up by trial and error. This may be preferable with a permanent installation that is to be operated by unskilled persons. For temporary set-ups though an external control is best as the acoustics of the hall play a part when establishing the correct setting.

HOW IT WORKS

The heart of the module is a 555 timer (IC1) which performs the switching when triggered by the input signal. Coupling to Q1 is via C1; capacitance of this component must be kept low (820nF is a maximum). The first transistor is a BC109C or BC108C. It must have the suffix C as this denotes the highest gain obtainable for the type. Considerably amplified, the signal is applied to Q2 which is also a high-gain C-type transistor. Component values here are chosen to deliberately overload the transistor and produce a squared-off wave of high amplitude at its collector.

IC1 needs a negative-going square wave which takes the voltage at the trigger pin from half to less than one-third of the supply rail. As the voltage is held at the half level by R5 and R6, the first negative-going excursion triggers the device.

Normally the output of a 555 is taken from pin 3, but this is of no use for our purpose because the output alternates between positive rail and earth. To mute the channel we need to short the audio signal to earth at some suitable point, then release the short when the voice switch operates. Having the full rail voltage applied in one position of the switch is inconvenient and could cause problems in the mixer circuits.

Instead, the output is taken from pin 7. This is normally used to discharge the timing capacitor after the timing cycle is complete in readiness for the next cycle. It therefore floats during timing, but shorts to negative or earth outside of the timing cycle. It thus can be used for audio muting.

Discharge of the timing capacitor is now carried out by pin 3 when it goes to negative at the end of timing, and diode D1 prevents the capacitor charging from pin 3 when it is positive during timing. In addition the LED indicator is supplied from pin 3 via R8.

The timing network is R7 and C4, which gives the five second delay. It is not necessary for timing to be exact so these need not be close tolerance components. If a longer time is desired, the value of the capacitor should be increased or the resistor decreased, and vice-versa for a shorter time.

If the device timed out every five seconds, although it would switch on again almost immediately there could be perceptible breaks, especially if the event occurred in the middle of a word. So it is necessary to prevent the completion of the timing cycle as long as sound is being picked up by the microphone. This is done by Q3 which is connected across the timing capacitor. Pulses from Q2 are applied to its base via C5 and rectified by D2. The resulting positive-going signal turns Q3 on and thereby keeps discharging the capacitor. It is only when the signals cease that the capacitor is allowed to charge fully and so time out.

Q3 must be of a low-gain variety, that is having the suffix A, otherwise it will turn on with low-level sounds and remain conductive due to ambient noise; hence the unit would fail to switch off. If only higher gain transistors are available, a 33k resistor (shown dotted) should be included to reduce the input, but this is not required for low-gain transistors.

It is also necessary for C5, the input coupling capacitor, to be of low leakage such as a polycarbonate or mylar type. Any leakage will result in a permanent positive voltage on Q3 base and a failure to switch off.

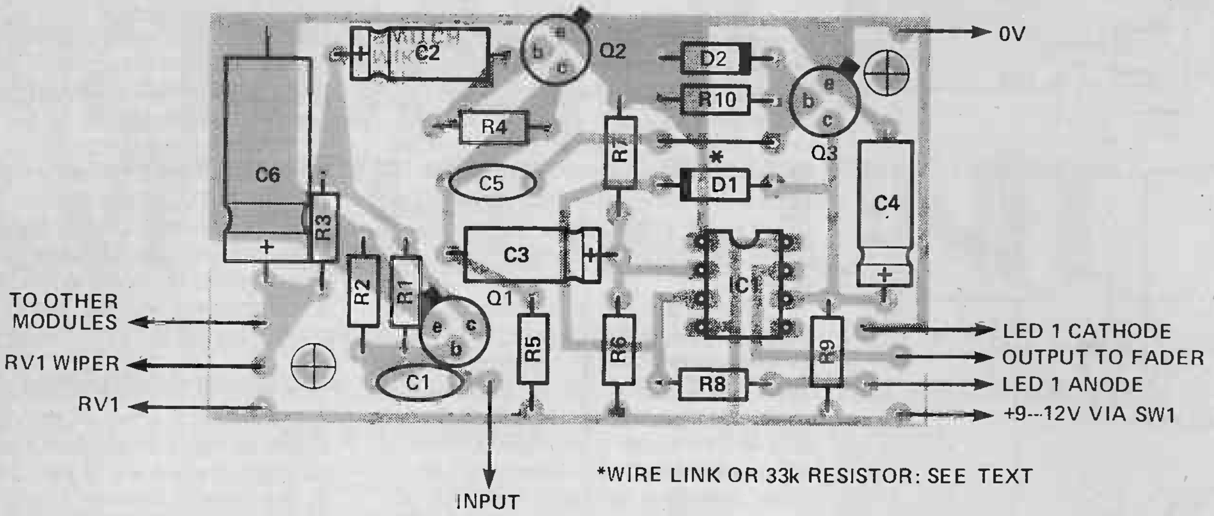


Fig. 3 Component overlay for the Voice-Switch. If the modules are all using a common sensitivity control, then the decoupling capacitor (C6) is only needed on the first module.

The design allows for either individual controls or a single one. The control varies the potential applied to the first stage and hence its gain. Wiring from the modules to the control can be looped if this is more convenient. If this is done the decoupling capacitor should be mounted on the first module so that the run from the control is decoupled.

The modules will operate with gain to spare using 200 ohm ribbon microphones which have a low signal-voltage output, so should have plenty of gain for most other available microphones.

To set the sensitivity control, turn it to an advanced position, whereupon the channels will be randomly switched on by sound coming back from the loudspeakers. Turn it down, a little at a time, until each channel only switches on immediately the microphone is used. If the first syllable of speech is lost, the control has been turned down too far.

With a good PA system the loudspeakers are of such a type and angled in such a way as to reduce sound coming back to the platform to a minimum. If poorly installed, returning sound will be excessive and this will not only predispose the system to acoustic feedback, but may make the setting of the Voice-Switch sensitivity control very critical.

LEDs and Battery Operation

The LEDs can be mounted above or below each fader so that a clear indication is given as to which channel is in use. This has been found to be a valuable feature.

For battery mixers the current requirement can be reduced. First a higher value resistor can be placed in series with the LED (R8); say 1k0 instead of 470R. Light output is reduced but not too much; a 1k2 or even 1k5 can be used but at further and noticeable sacrifice of illumination. This is assuming a 9 V battery is fitted. A red LED gives more light per milliamp than the other colours, so one should be used here.

Another current saver is to use a CMOS 555 instead of the bipolar variety. These are more expensive but well worth it in terms of battery saving. The complete module (including LED with R8 equal to 1k0) takes 10 mA on and 5 mA off with a bipolar 555. Using a CMOS 555, these figures are reduced to 5 mA on and 0.3 mA off; this is a total saving of some 50%. The CMOS version has the same packaging and pin connections and can be plugged into this circuit as a direct substitute. (This does not apply to all circuits.)

Operation

When first switched on, all channels come on and the LEDs light up, thus giving a visual check that all are working. To avoid random noise at this stage, the faders should be down. After five seconds the LEDs will go out whereupon the faders can be advanced to the normal operating level. Then, each will come on only when the microphone is used, and a fine adjustment of the fader can be made. There is no need for visual identification of the microphone in use as the LED indicates this.

Should there be spurious triggering, ease the sensitivity control back a shade. If the manual muting facility has been included, the faders can be left up when there are microphone movements or changes of participants, and the channels manually muted by switch. Any such operations will not affect the main microphone.

PARTS LIST

Resistors (all ¼ W, 5%)	
R1	330k
R2	150k
R3,5-7	47k
R4	4M7
R8	470R
R9	1M8
R10	68k
Potentiometers	
RV1	100k linear
Capacitors	
C1	470n polycarbonate
C2-4	2u2 16 V axial electrolytic
C5	22n polycarbonate
C6	220u 16 V axial electrolytic
Semiconductors	
IC1	555
Q1,2	BC108C or BC109C
Q3	BC108A
D1,2	1N4148
LED1	red LED
Miscellaneous	
SW1	SPST or DPDT with centre off (see text)
PCB	(see Buylines)

BUYLINES

We refuse to believe that you'll have any difficulty in obtaining the parts for this project — everything is so common you'll probably be able to pick them up in Woolworth's! Bi-Pak, Rapid Electronics and Cricklewood are among the mail order companies you may care to try. The addresses are given elsewhere in the issue. People unable to etch their own PCB can obtain one using the order form on page 71.

ETI

TECH TIPS

Mains Failure Emergency Light

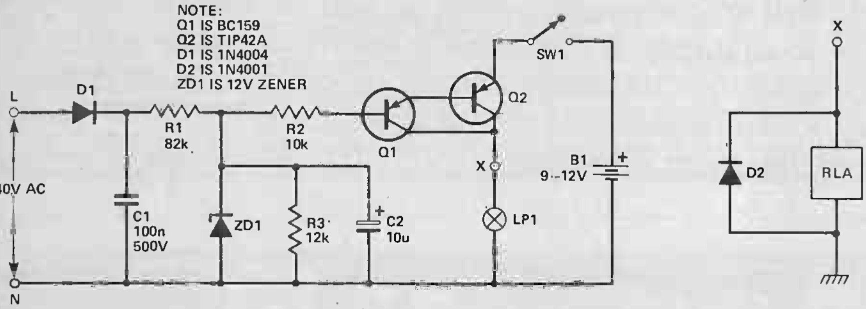
J. P. Macaulay, Crawley

This simple circuit has been found very useful in situations where a coin-slot electricity meter is used. It also would be useful no doubt in the event of another power strike.

The mains is half-wave rectified by D1 and smoothed by C1. The resultant DC is fed through a current limiting resistor R1 to produce a reference voltage of 12 V across ZD1. R3 and C2 form a simple leakage and smoothing circuit.

If B1 is of the voltage specified the Darlington pair Q1, Q2 are biased hard off. If mains failure occurs the base of Q1 is pulled to ground; the base current flows through R2 and R3. The bias is sufficient to take the Darlington pair into saturation, providing current to the lamp which then lights up. Since the circuit is not isolated from the mains, care must be taken to earth the box in which it is mounted. Connection to the mains can be via a normal mains plug.

If required the alternative load, a relay, can be driven instead of the lamp. This can then be used to switch in an auxiliary power supply. When the mains is reapplied the circuit returns to its original state. Current consumption is zero when the mains is present.



Penalty Kicks

G. Durant and D. Hall, Selby

We designed this following hand held game to be simple for construction, cheap and most of all fun for the operator. The idea of the game is to put yourself in the position of the goal keeper and to guess which way the striker is going to kick the ball, by turning the rotary switch to the marked positions. The shoot button is then pressed; a noise will indicate whether the operator has guessed right.

IC1, a 555, is wired up as a astable multivibrator running at about 100 Hz feeding the clock input of the 4017. If the 'shoot' button is not operated, the Q output of the latch formed by IC5a,b is low, allowing the 4017 to count. Three LEDs are driven by the 4017; in the reset state these are blanked by the latch.

Switch SW1 is turned into the position the 'goalie' thinks the ball will come towards the goal. The 'shoot' button is pressed. IC2 stops counting and one of the LEDs lights.

If you guessed correctly the 'saved' LED is lit via IC3a,b or c and a 1 kHz sound generator built around IC8 is sounded for just over a second, being controlled by IC7, a one-shot monostable.

If you 'let the ball into the goal' a buzzer is triggered, formed by IC6.

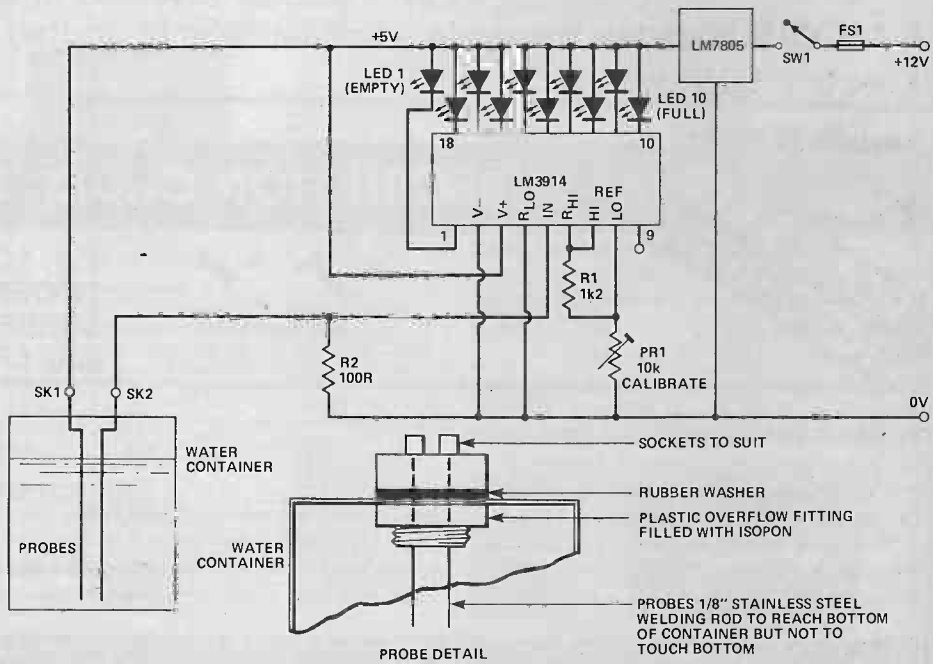
Caravan Water Supply Monitor

P. A. J. Thomas, Cowes

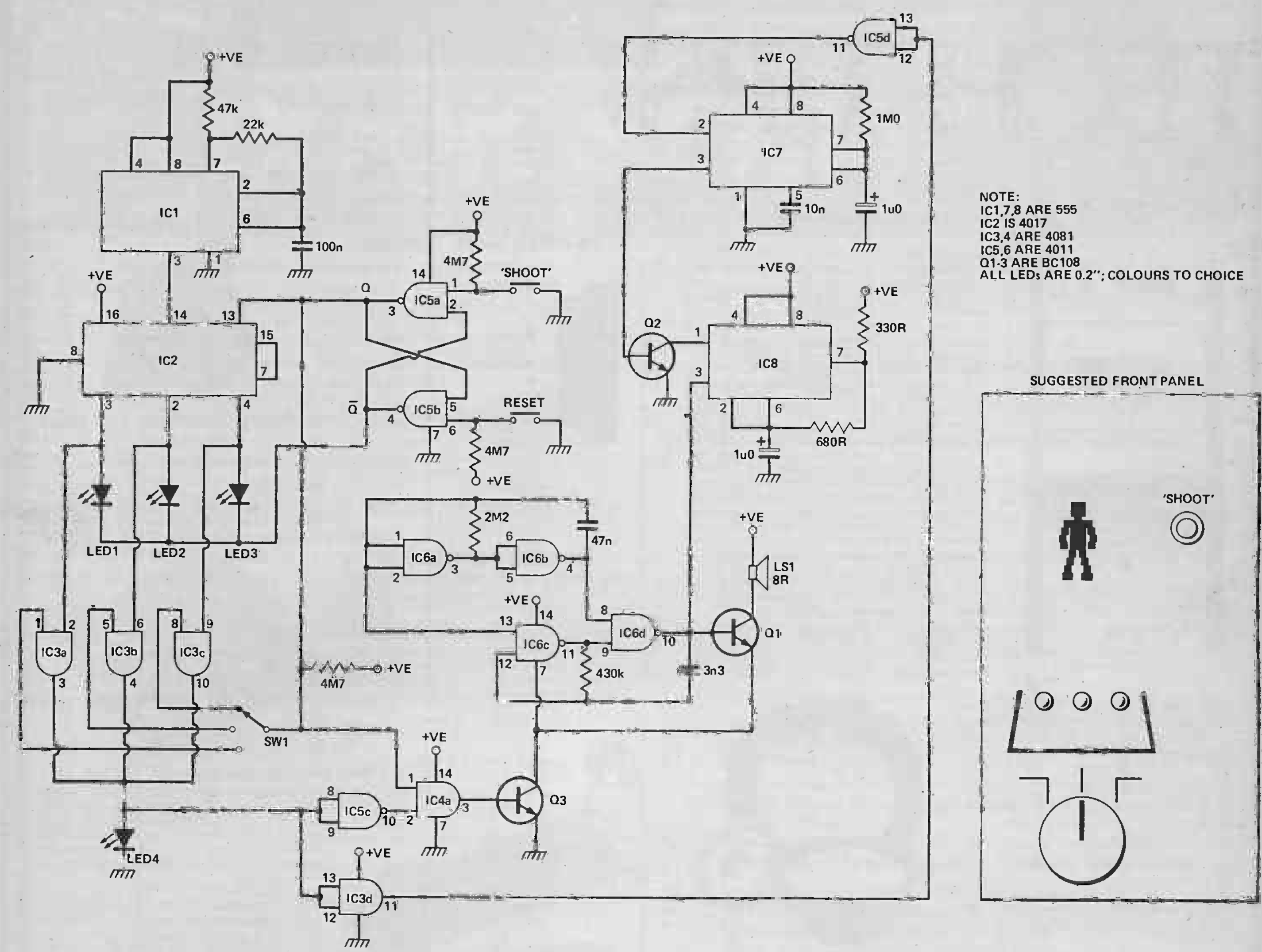
Being a keen caravanner I found that knowing how full the water containers were at any time was a problem, since they are opaque (being either plastic or aluminium). I designed the following circuit to give a visual display over the sink inside the caravan. The circuit is duplicated for drinking water and waste water levels, only the colour sequence of the LEDs being reversed. For drinking water, it is; full to half-full, green; half to one-quarter full, amber; quarter-full to empty, red.

The circuit is based on the LM3914 dot/bar driver using ⅛" stainless steel welding rods as probes. Electroplating is overcome by reversing the polarity through the probes each time the containers are filled (or emptied in the case of the waste container), by using a two pin plug and socket. The probes are spaced ½" apart but this isn't critical.

The probes are held in a plastic overflow joint connector, used for water tanks, and this is filled with Isopon; the surface nearest the water is smeared with Araldite to prevent the fibreglass tainting the drinking water.



The display, dot or bar mode, gives a linear display of the contents very accurately, thus preventing overflows of waste water or running out of drinking water.



CMOS Sustainer for Electric Guitar

S. P. Giles, Edmonton

I believe this must be one of the simplest and cheapest sustainers for electric guitars around. IC1a and IC1b are both CMOS inverters, wired to act as op-amps. Any inverter will do the trick, such as 4009, 4049, 4069 or 4007.

The gain of IC1a is determined by the collector-emitter resistance of Q1 plus R2. If the output level is to remain constant while the guitar note decays away, the gain of IC1a must be increased by a corresponding amount. This is achieved by rectifying the output of IC1a through IC1b and D1 and passing the resultant DC voltage, which is smoothed by C4 and R7, to the base of Q1. This forces the collector-emitter resistance of Q1 to increase in proportion to the input level from the guitar. RV2 can be set to any desired level and when set high can easily overdrive the input stage of the guitar amplifier giving a valve-type of distortion.