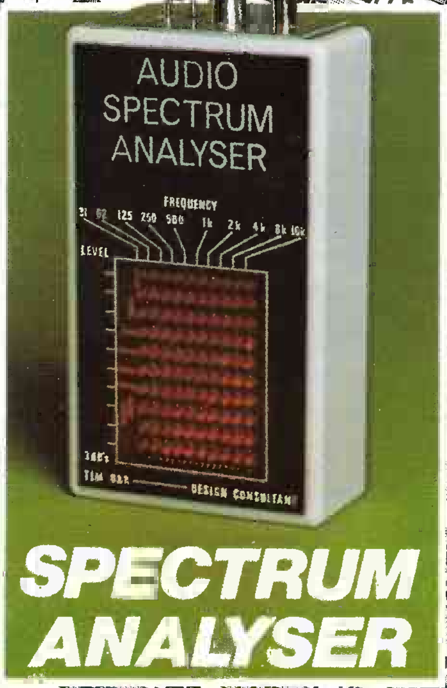
electronics today INTERNATIONAL

JUNE 1978

45p

CRASH COURSE ON
HOME COMPUTING
32 Page Supplement



SPECTRUM ANALYSER

Stars And Dots Game

Hitachi Mosfet Amp

Designing Amps

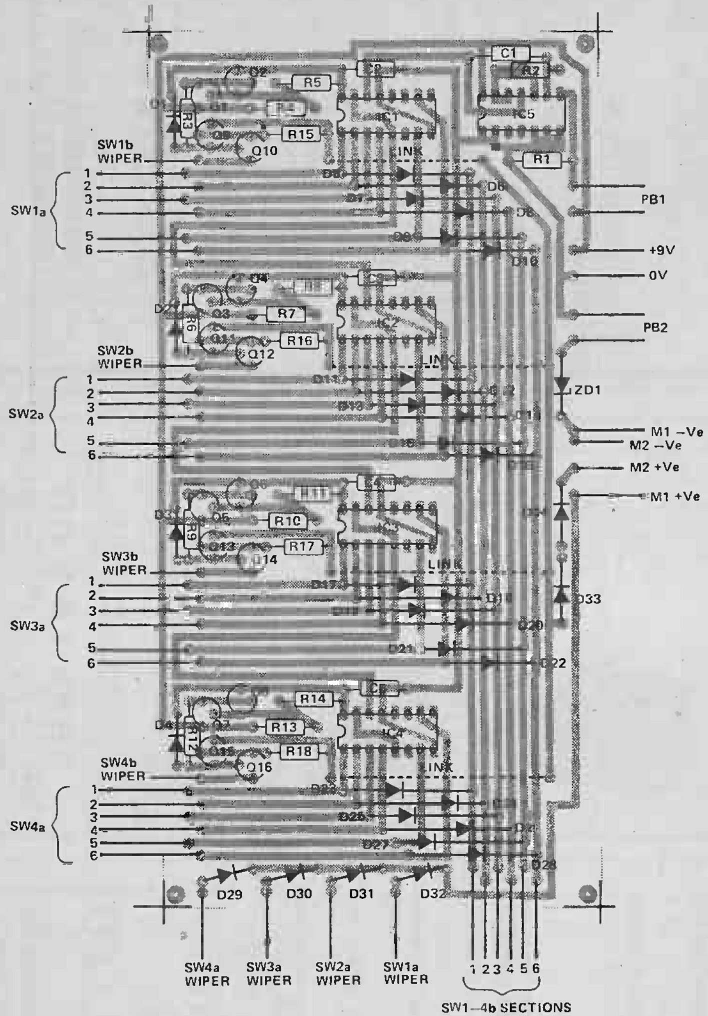
AM-FM Radio

TI 59 Review

Quarks

LCD WATCH OFFER £9·95

. . . NEWS . . . . PROJECTS . . . . MICROPROCESSORS . . . . AUDIO . . .



Above: Component overlay for the non-logic stars and dots game. Check IC orientation very carefully.
Right: Foil pattern for the game, shown full size at 175mm x 90mm.

BUYLINES

All the parts for this project should be available from most electronic outlets but shopping around can save money — so compare prices before buying.

The box we used was a Vero box but the project could be attractively boxed in any of the many cases that appear in the shops today.

PARTS LIST

RESISTORS (all ¼w 5%)

R1, 5, 8, 11, 14	100k
R2	47k
R3, 6, 9, 12	680k
R4, 7, 10, 13, 15, 16, 17, 18	2k0

CAPACITORS

C1	470p Polystyrene
C2, 3, 4, 5	10n Polyester

SEMICONDUCTORS

Q1-8	BCY 71
Q9-16	BC 109
D1-34	IN914
ZDI	3V6 400mW

SWITCHES

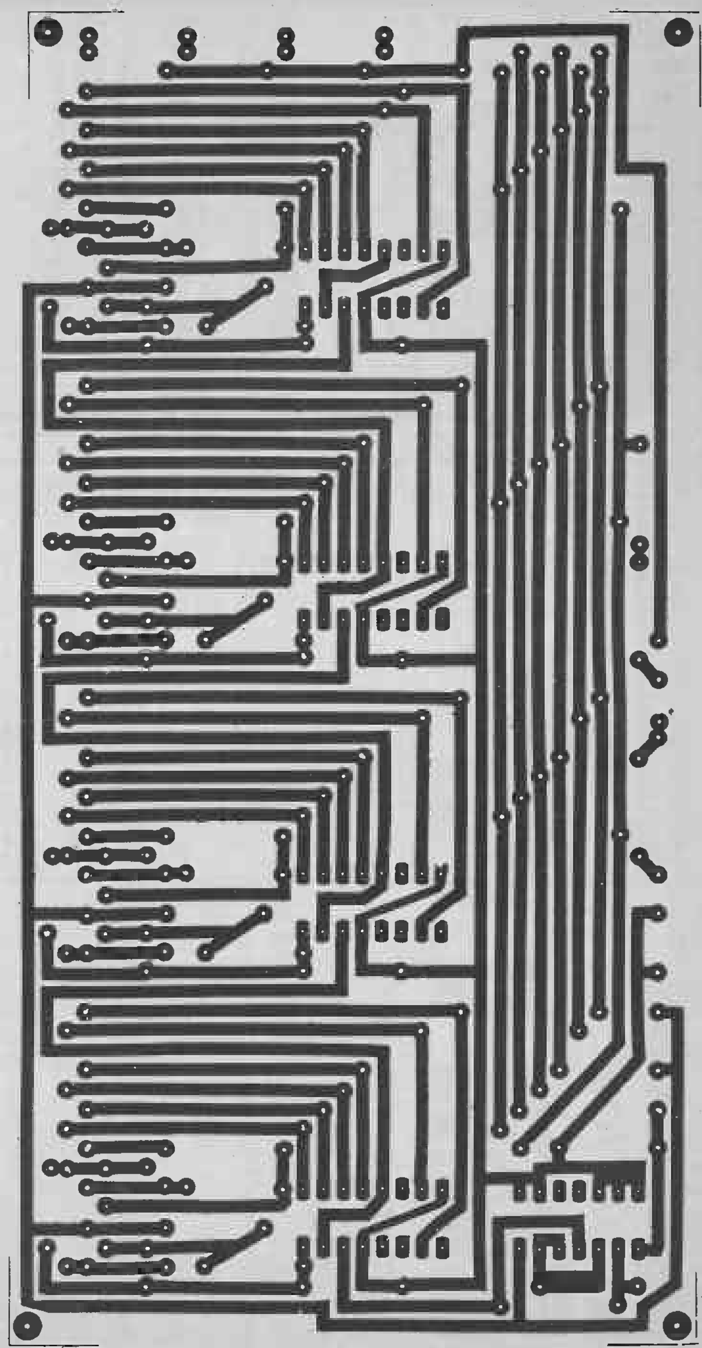
PB1, 2	Single Pole Push-to-make
SW1-4	2p-6w Rotary

METERS

M1, 2	1mA panel meter

MISCELLANEOUS

PCB as pattern, PP3, box to suit



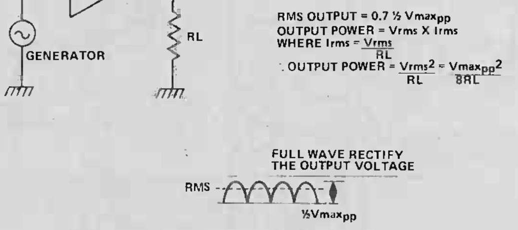
AUDIO AMPLIFIERS

Designing an amplifier is like re-inventing the wheel. There are thousands of published designs and possibly as many as a 100 different types of monolithic amplifiers as well as lots of off the shelf modules to choose from. If you design the amplifier yourself (or use someone elses design) you will probably encounter problems such as heat, noise, instability, distortion, power rating etc, etc. In this article Tim Orr sets out to help you cope.

Power Rating

The power rating for an amplifier is *generally considered* to be the maximum RMS power that a sine wave can deliver to a load (Fig. 1). The RMS power is given by:

$$P_{(RMS)} = \frac{V_{PP}^2}{8RL}$$



Therefore if RL=8R and Vpp is 1V, $pP_{(RMS)}$ = 15.6mW.

For V_{pp} = 10 V $P_{(RMS)}$ = 1.56 W.
For V_{pp} = 100 V $P_{(RMS)}$ = 156 W.

So the RMS power goes up as a square of the output voltage. However our hearing does not respond linearly to power, and so the difference between a 10 W and 100 W amplifier is always disappointing.

Heat

Not surprisingly, power amplifiers get hot. When they are delivering power to a load, the amplifier is also dissapating a considerable amount of heat itself. A reasonable rule of thumb is that both the amplifier and the load dissipate the same power, except when there is no output signal. Then the amplifier is the only thing that is getting hot. To get a very low crossover distortion it is usually necessary to run the output transistors in an amplifier in class A or AB. This means that the transistors are biased on (or partly on for AB operation). Thus they consume lots of current and get hot. Therefore designing power amplifiers is a compromise between heat production and distortion. IC power amplifiers, because of their small size, go for low heat generation and hence higher crossover distortion. Discrete component power amplifiers can use large heat sinks sometimes with forced air cooling and thus obtain THD figures from 0.1% to 0.01%.

Some IC power amplifiers get rid of their heat down the IC legs to suitably large areas of copper on the printed circuit board. There are also 'Stick On' heat sinks dor DIL packages. Also, when the going gets a bit hot some amplifiers employ a thermal shutdown mechanism. Generally though, high temperature operation means that the device life time is greatly shortened. Thus it is not surprising that the components that fail most regularly are the power transistors in amplifiers and power supplies.

Stability

The only difference between amplifiers and oscillators is the phase of the feedback and so it is hardly surprising that a problem exists. When the phase of the feedback becomes positive then oscillation can occur, if the gain of the amplifier is then greater than unity. The gap between a good amplifier and an oscillator is known as the phase margin. When the phase margin is reduced to zero, oscillations will occur.

More feedback when the phase shift is positive will increase the risk of instability. Less feedback when the phase shift is positive will make the amplifier more stable.

However, less negative feedback means more distortion. It is a compromise between stability and distortion. It is possible to increase the phase margin and thus stabilise the amplifier with a suitably placed capacitor. However, in the IC (monolithic) design this is not possible because this capacitor would probably occupy ▶

twice the area as the rest of the integrated circuit. So, the designers of IC power amplifiers usually make this stabilising capacitor small and set the amplifier gain high (less negative feedback).

You end up with a power amplifier that is only stable with high values of gain and which has a relatively high distortion. Even so, most monolithic designs need additional capacitors on their inputs and their outputs to maintain stable operation. Other stability problems are:

1) Amplifier gain and phase margin depend on power supply voltages. Thus, an amplifier may not be stable under varying conditions of supply voltage. During the power up, the amplifier may emit a squeak or a whoosh, due to high frequency instability.

2) Amplifier gain and phase margin depend on temperature. Thus as the amplifier warms up it may then become unstable, oscillate, the output transistors get very hot and the amplifier burn out.

Alternatively, the amplifier may be unstable only when cold. So you switch on and it squeaks (oscillates), warms up, stops oscillating, cools down, oscillates (squeaks), warms up, etc. etc. (Breaks the ice at parties!).

3) The load put on an amplifier will affect the phase margin. Designing an amplifier that will drive any load is difficult. Often a power amplifier will have a capacitor resistor network from its output to ground. This network is used to increase the phase margin.

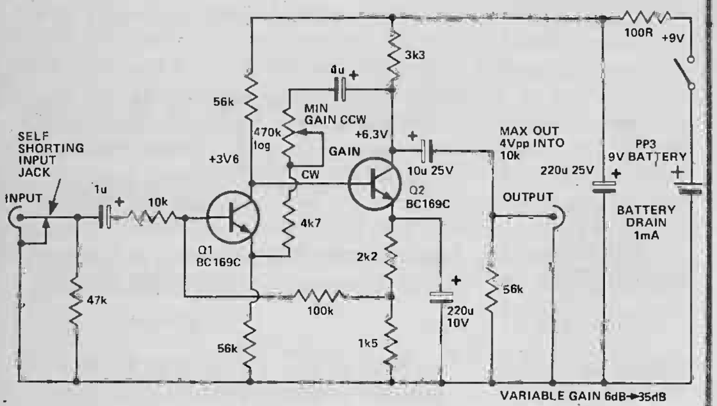
Distortion

If you put a pure sinewave into an amplifier and you get out of it the same sinewave plus some harmonics, then you have got distortion. Any other spurious signals are not distortion products and are not included in the THD calculations.

Crossover distortion is usually generated by the output transistor pair (Fig. 2). This is caused by one of the transistors switching off before the other one can switch on. The result is a 'lump' in the output waveform which gives the sound a 'buzzy' quality. The distortion can be reduced by turning the output transistors on a bit more, by biasing their bases further apart. This increases the quiescent current and thus more power is dissipated. Also, overall negative feedback can be used to iron out the kinks, but this will increase the chance of instability.

Another type of distortion is harmonic distortion. An amplifier, used in open loop is usually fairly non linear. This non-linearity will cause any signal passing through the amplifier to be distorted. Negative feedback is used to iron out the non-linearities and so reduce this source of harmonic distortion.

It is interesting to note that the hi-fi market wants low THD figures of 0.1% to 0.01% but the music market actually prefers (in some cases) higher figures of about 2%.

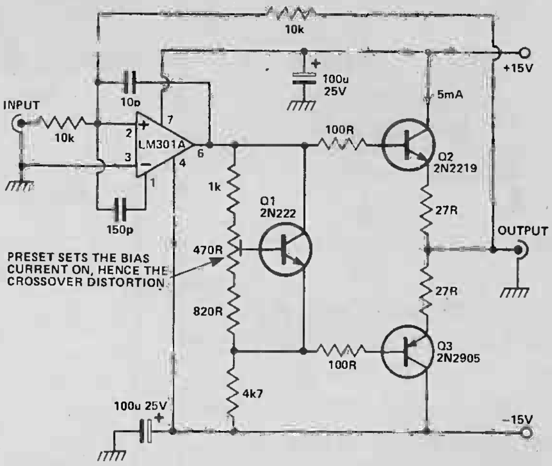


Mains hum is easily picked up with high impedance microphones, particularly if the microphone cable is long. Also, a treble cut occurs when using long cables. The output impedance of the microphone and the capacitance of the cable produces a low pass filter which cuts off the high frequencies, so that a high impedance microphone should only be used on a short cable.

For low impedance types, a low-noise high gain amplifier is needed, as output is much lower, and the circuit above is such an amplifier. The noise generated by transistors is a function of collector current. The current through Q1 has been optimised to give low noise operation.

The amplifier has an open loop gain of more than 60 dB. Negative feedback is applied, via a variable 470k pot, so that the closed loop gain is controllable from 6 dB to 35 dB. This allows the gain to be tailored to suit different types of microphone and hence get the best overload and S/N ratio conditions. A maximum signal output of 4 V into a 10k load is obtained and the current drain is 1 mA making it possible to run the amplifier from a PP3 9 V battery.

Unbalanced Line Driver



The high open loop gain of an op amp is combined with the power handling capabilities of descrete transistors to produce a line driver amplifier. The output driver stage (Q1, 2, 3) is included in the overall feedback, and acts as a power booster on the output of the op amp. Transistor Q1 is used as a VB_{BE} multiplier. That is, it sets up a voltage of about 1V5 between its collector and emitter. The actual voltage can be set by the preset connected to its base. Thus the bases of Q2 and Q3 can be biased apart by a set amount, just sufficient to make them work in class B operation.

If there are any ambient temperature changes, Q1 automatically adjusts the bias voltages to Q2,3 to maintain a constant bias current. There is overall negative feedback from the output, providing a voltage gain of 0 dB (x1). The output is partly short circuit protected by the 27 ohm emitter resistors. This amplifier can deliver high level, low distortion signals into low impedance loads. It could be used as an output driver in an unbalanced audio mixer.

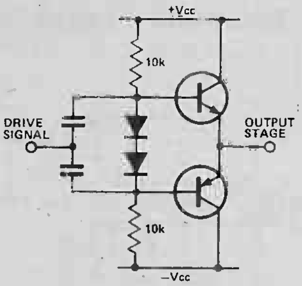
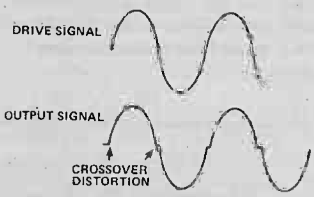


Fig 1 (above) is the classical output pair that produces the equally classical crossover distortion illustrated below. Careful biasing of the output pair can reduce the effect but it is usually present in most amplifiers of this type.



Noise

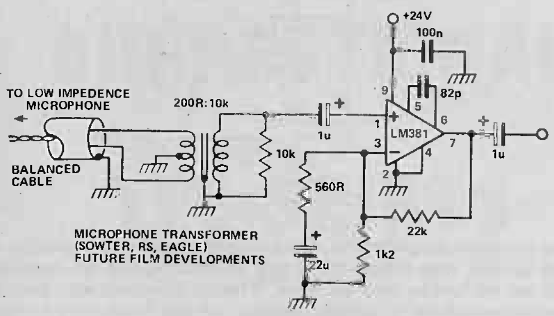
Noise is generally not a problem in power amplifiers but it is in the pre-amplifier stages of an audio system. An overall system signal to noise ratio of 70 dB (3 000 to 1), is quite good and not very difficult to achieve. Better than this is studio or professional quality. When amplifiers are used to reproduce stored signals, such as from a disc, radio or tape recorder, then an overall S/N ratio of 70dB is quite adequate. This is because the S/N ratio for these storage or transistor systems is quite low.

For example the best disc technology will only give us a 60 dB S/N ratio. The best studio quality tape recorder (unprocessed), will give 65 dB. Radio transmissions are about 50 dB on FM, and cheap cassette players only clock up 30 dB's.

As tapes and discs are used then their S/N ratio deteriorates. Also, most listening environments have a high background noise level (air conditioning, street noise, jets etc.).

The most demanding situations where the noise of a preamplifier will be important are in amplifying the signals from low impedance microphones, magnetic cartridges for record players and tape recorder pickup heads. In the following sections there are several examples of low noise pre-amplifier designs.

Balanced Microphone Preamplifier

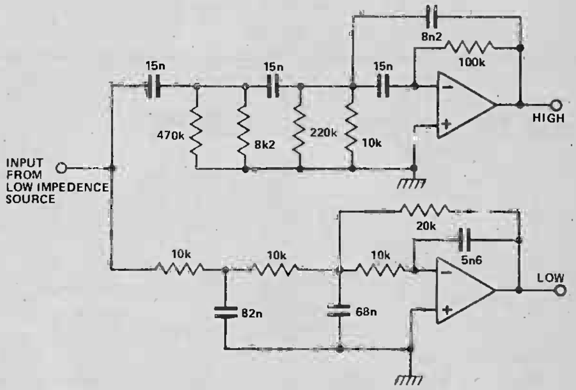


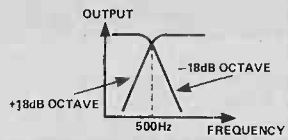
Professional audio equipment generally uses balanced inputs and outputs. This means that the inputs and outputs are differential, which is usually obtained by having balancing input and output transformers.

The advantage of using a balanced system is that any unit can be connected to any other unit without any ground loop problems. A balanced system eliminates these problems. Also, mains hum pick up is reduced. A balanced audio cable has an outer screen and a twisted pair of wires in the centre. Any mains hum (or other signal) which is picked up on the twisted pair will have the same amplitude on each of these central wires. This is a common mode signal. The microphone signal applied to these two wires is a differential signal. Thus, when the microphone signal plus mains hum is connected to the transformer, the differential signal appears at the output windings and the common mode signal is rejected. Thus the mains hum is suppressed.

The transformer also provides a voltage gain, and the LM 381 provides a low noise amplification of about 32 dB (x40).

Active Crossover Unit

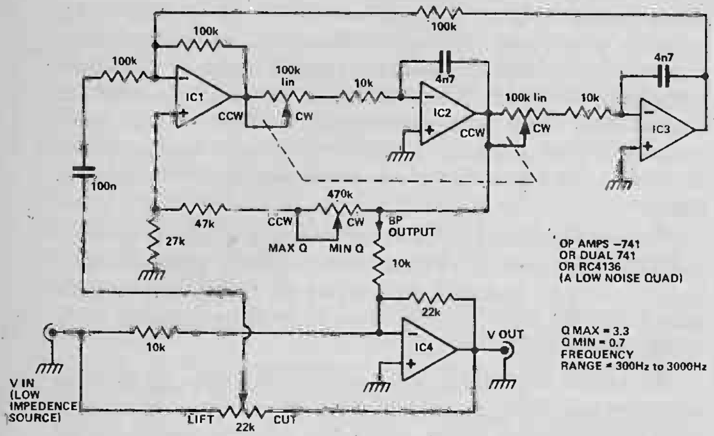




The circuit shown is for a two speaker system having a crossover frequency of 500 Hz. The filter structures are third order Butterworth multiple feedback, low pass and high pass. (Third order implies that roll off slopes of ±18 dB/octave are obtained.)

▶

Parametric Equaliser

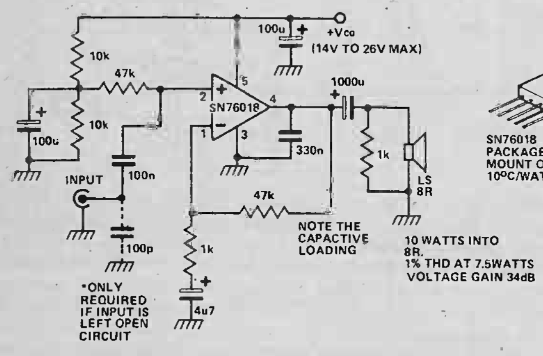


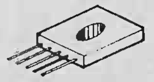
This is possibly the equaliser for the amplifier system that has everything. The parametric equaliser has got three controls. It is a bandpass filter which can have variable cut or lift, so that a particular frequency band can be enhanced or rejected. The resonance can also be controlled so that area of frequency affected can be broad or narrow. Also the centre frequency of the bandpass filter can be varied so that it can be tuned to operate at a particular frequency. The circuit operation is quite simple.

Op amps IC 1, 2, 3 form a state variable filter, the Q and centre frequency of which can be varied. Op amp IC4 is a virtual earth amplifier. When the equaliser is in the lift position, the signal is fed into the state variable filter. It then comes out of the bandpass output and into IC4. In this feed forward position the equaliser has got a peak (lift) in its response. When the equaliser is in its cut position, the bandpass filter is in the feedback loop of IC4 and so there is a notch in the frequency response.

Care must be taken not to cause overloading and clipping when using high Q lifts.

10 Watt Power Amplifier. (SN6018)





SN76018 PACKAGE MOUNT ON HEATSINK 10°C/WATT

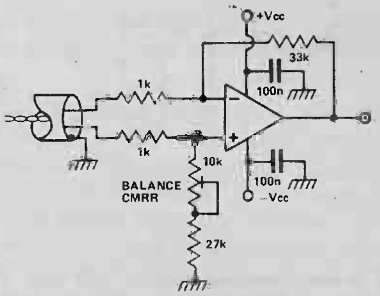
This is a very simple and inexpensive monolithic power amplifier made by Texas Instruments. It comes in a package that looks like a plastic power transistor with five legs.

Thus it can be screwed down to a heat sink without any problems. The THD specifications for this device are:

10 W at 10% THD (R1 = 8 ohm)
7.5 W at 1% THD (R1 = 8ohm)
0.05 W to 6.5 Watt at 0.2% THD (RL = 8ohm)

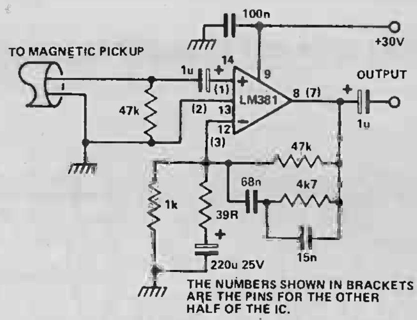
No isolation from the heat sink is required. It should be used in applications where high fidelity is not required. Note that it requires two stabilising capacitors.

Electronic Balanced Input Microphone Amplifier.



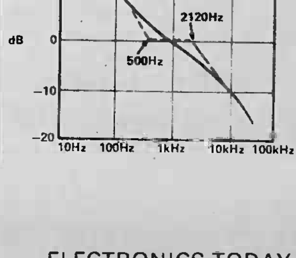
It is possible to simulate the balanced performance of a transformer electronically with a differential amplifier. By adjusting the presets the resistor ratior can be balanced so that the best CMRR is obtained. It is possible to get a better CMRR than the one you would obtain from a transformer. Also, a transformer can itself pick up mains hum, it is expensive and heavy. So, electronic balancing can be quite competitive. One problem is obtaining a truly differential low noise amplifier. I would suggest a RC4136 which is a quad low noise op amp.

Record Player — Magnetic Pickup



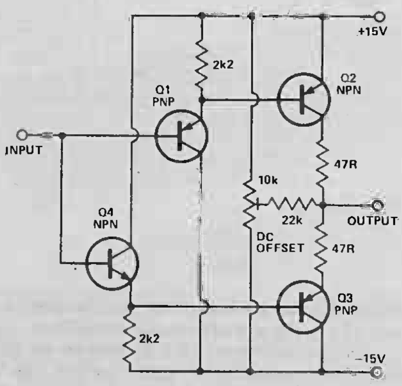
If you were to amplify the signal from a magnetic pickup on a record player and listen to it the sound would be terrible. It would be all treble and no bass. This is because the pickup is magnetic and gives an output voltage which is velocity sensitive. That is the faster the needle wiggles in the record groove, the larger the output voltage, or rather the output voltage (for the same amplitude of excursion) is proportional to frequency. To restore the natural sound, the signal must be equalised with a frequency response as specified by the RIAA.

This play back equalisation gives 20 dB lift at low frequencies and 20 dB attenuation at high frequencies and is 0 dB at 1 kHz. No equalisation is required if you use one of the cheaper ceramic pickups, which have a flat response.



Graph illustrating the non-ideal approximation to the ideal RIAA equalisation curve, the response flows smoothly unlike the 'defined' RIAA response.

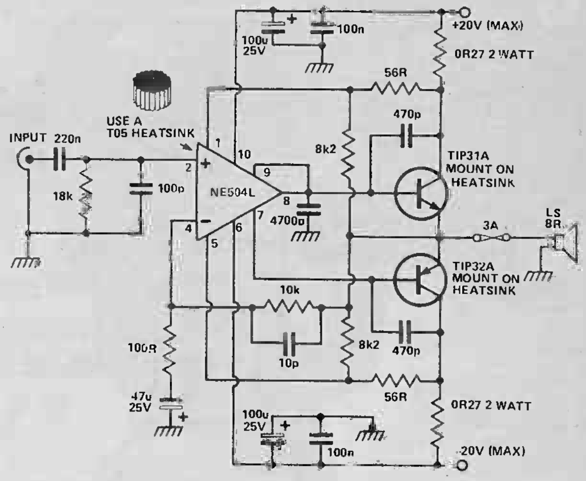
50 Ohm Driver



When you want to buffer a test generator to the outside world it is often very difficult to get an amplifier with sufficient bandwidth and power handling to do the job. The circuit is a very simple unity gain buffer. It has a fairly high input impedance, a 50 ohm output impedance, a wide bandwidth and high slew rate.

The circuit is simply two pairs of emitter followers. The base emitter voltages of Q1 and Q2 cancel out, and so do those of Q3 and Q4. The preset is used to zero out any small DC offsets due to mismatching in the transistors.

20 Watt Amplifier



An audio power amplifier can be constructed from a power driver op amp plus a pair of transistors. The power driver is a NE540 made by Signetics. It generates quite a bit of internal heat and so a TO5 heat sink is required. Note that this design uses five stabilising capacitors.

The amplifier works quite well once any stability problems have been sorted out and the power output is quite adequate for a domestic amplifier system.

ETI

SPECTRUM ANALYSER

A ten channel unit designed for ETI by Tim Orr, who knows a thing or two about Active Filters and Op Amps.

AUDIO SPECTRUM ANALYSERS can be a valuable tool used in the setting up of a room acoustically, with a graphic equalizer such as the ETI design published in September 77, to monitor programme material or just as a gimmick to please yourself and friends.

When setting up rooms pink noise is pumped into the room using an amplifier. A microphone is then used to monitor the sound and its output is the input to the analyser. Now by adjusting the graphic equalizer a flat response can (hopefully) be obtained.

Design Features

Spectrum analysis can be done by two main methods. The first is to have a tuneable filter which is swept across the band of interest. The output of the filter when displayed on an oscilloscope, will be a frequency/amplitude graph of the

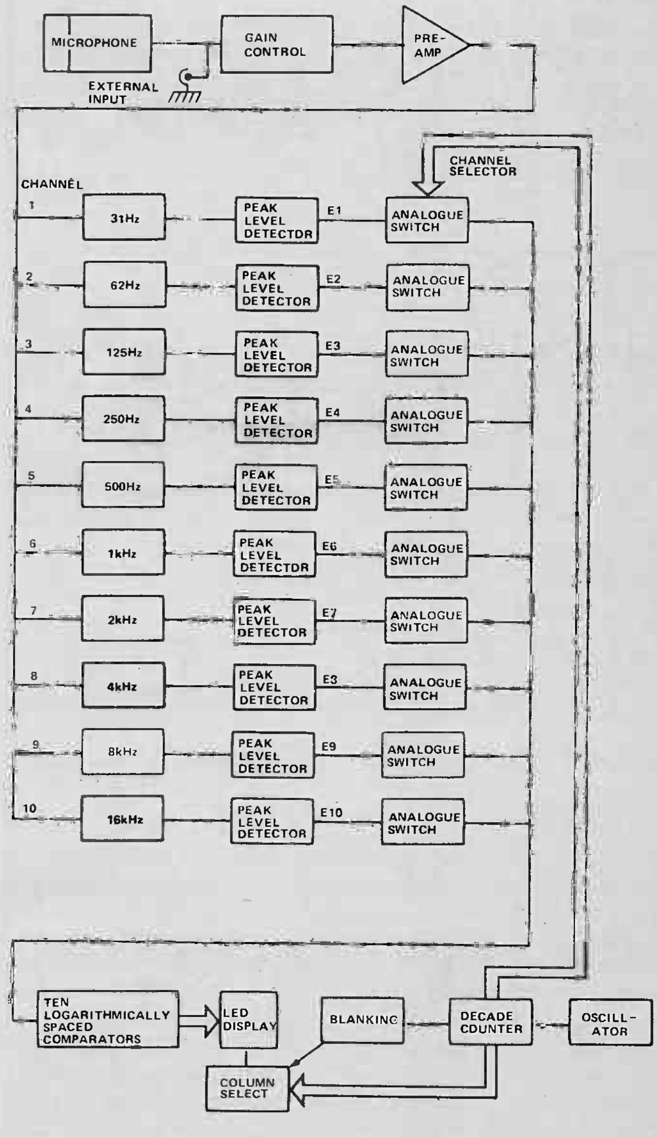


Fig. 1. Block diagram of the Spectrum Analyser.

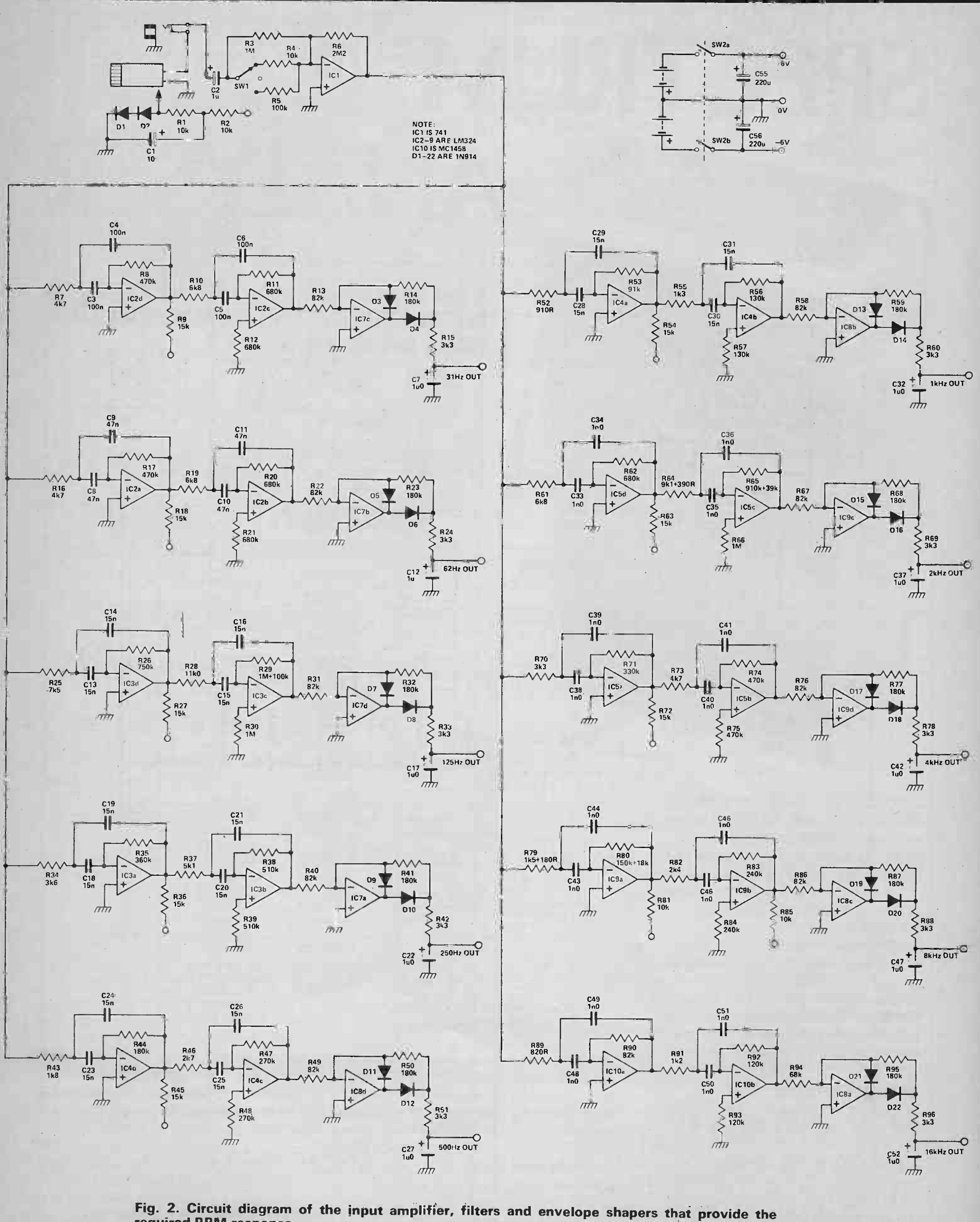


Fig. 2. Circuit diagram of the input amplifier, filters and envelope shapers that provide the required PPM response.

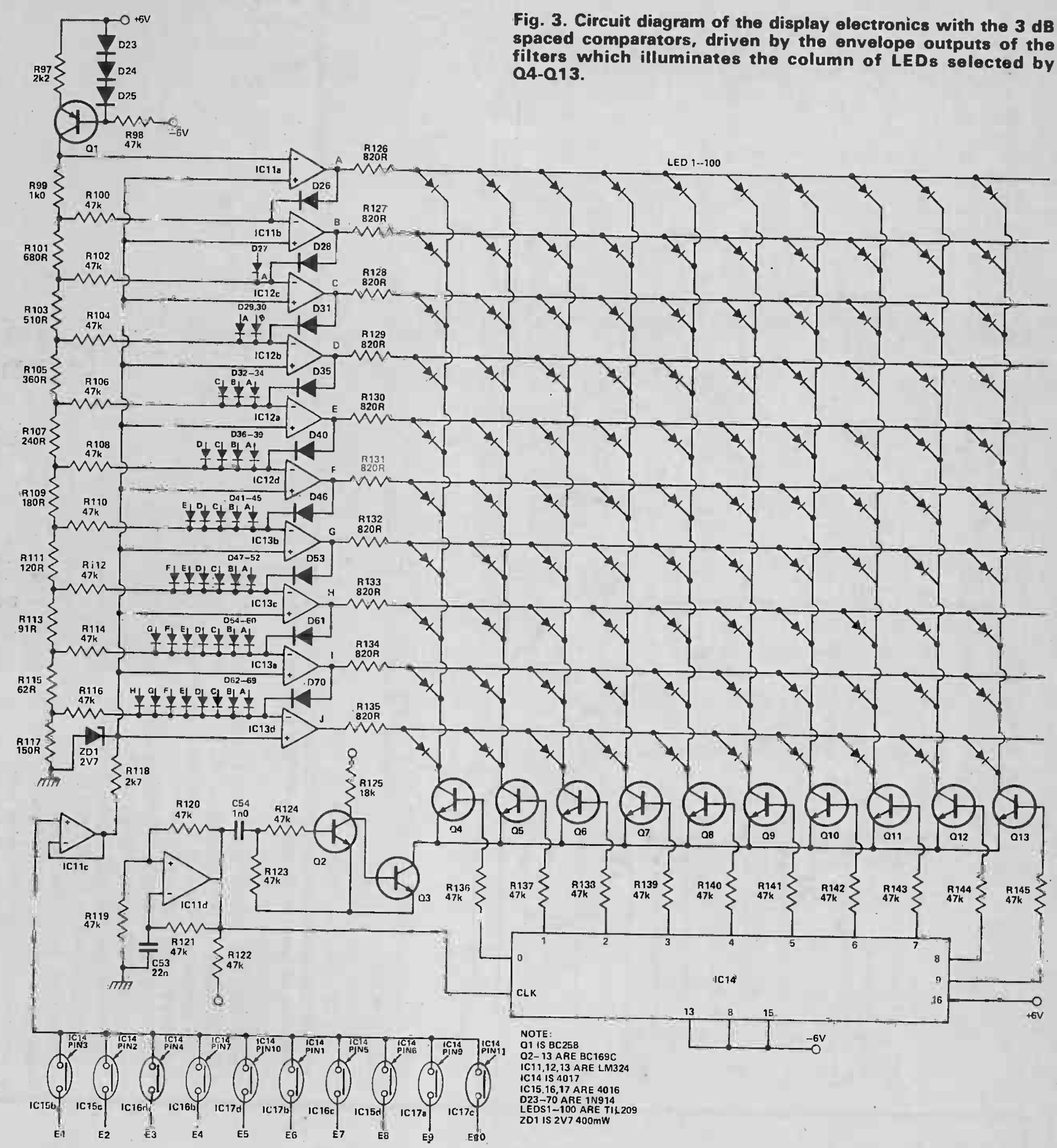


Fig. 3. Circuit diagram of the display electronics with the 3 dB spaced comparators, driven by the envelope outputs of the filters which illuminates the column of LEDs selected by Q4-Q13.

input. While this gives a well-formated and accurate display it is not 'real time' in that if an event occurs at one frequency while the filter is sweeping elsewhere it will not be recorded. For this reason this method is used normally where the spectral content is constant and the sweep is only over a small percentage of total frequency (such as the output of a radio transmitter).

For real time analysis the incoming signal is broken into several frequency bands, just like in a graphic equaliser, and the energy level in each band is displayed on a 'scope or, as in this project, with a vertical column of LEDs. Analysers with anything from ten one octave steps to thirty one third octave steps are available. The display is usually a large matrix of LEDs, frequency along the horizontal axis and amplitude in dB steps vertically. This type of analyser will give a display of the average energy levels that exist and is not capable of discriminating between individual harmonics, this being due to the frequency spectrum having been indiscriminately broken up into octave chunks. Thus the analysis is grainey but it does enable you to instantaneously determine the average spectrum of a sound.

The spectrum analyser described here has ten frequency bands and ten level steps of 3dB each. The first prototype constructed used ordinary dual 741 op-amps and gobbled up

▶

PARTS LIST

RESISTORS (all ¼ watt 5%)	
R1, 2, 4, 81, 85	10k
R3, 30, 66	1M
R5	100k
R6	2M2
R7, 16, 73	4k7
R8, 17, 74, 75	470k
R9, 18, 27, 36, 45, 63, 72	15k
R10, 19, 61	6k8
R11, 12, 20, 21, 62	680k
R13, 22, 31, 40, 49, 58, 67, 76, 86, 90	82k
R14, 23, 32, 41, 44, 50, 59, 68, 77, 87, 95	180k
R15, 24, 33, 42, 51, 60, 69, 70, 78, 88, 96	3k3
R25	7k5
R26	750k
R28	11k
R29	1M + 100k
R34	3k6
R35	360k
R37	5k1
R38, 39	510k
R43	1k8
R46, 118	2k7
R47, 48	270k
R52	910R
R53, 54	91k
R55	1k3
R56, 57	130k
R64	9k1 + 390R
R65	910k + 39k
R71	330k
R79	1k5 + 180R
R80	150k + 18k
R82	2k4
R83, 84	240k
R89, 126-135	820R
R91	1k2
R92, 93	120k
R94	68k
R97	2k2
R98, 100, 102, 104, 106, 108, 110, 112, 114, 116, 119-124, 136-145	47k
R99	1k0
R101	680R
R103	510R
R105	360R
R107	240R
R109	180R
R111	120R
R113	91R
R115	62R
R117	150R
R125	18k

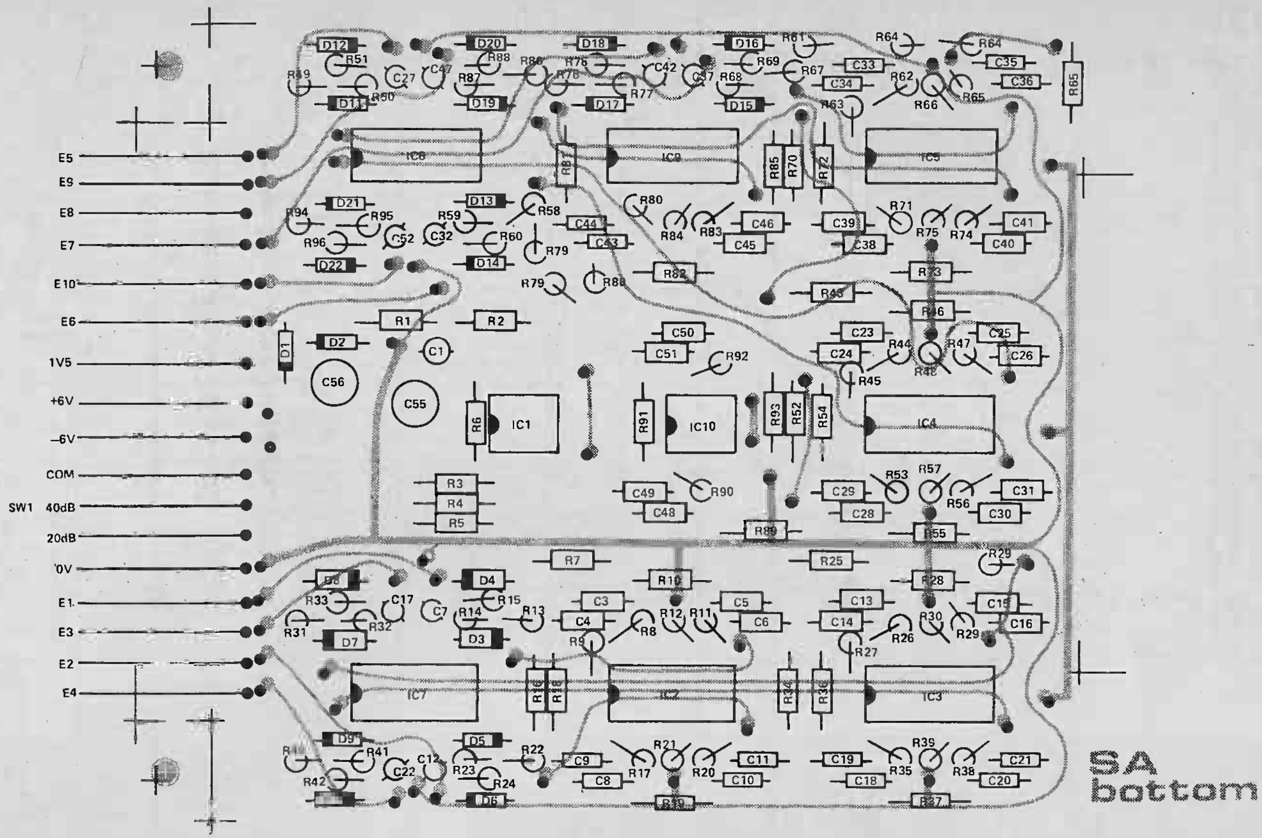


Fig. 4. Component overlays for the Spectrum Analyser boards. For reasons of clarity only the top side foil pattern is shown here. The foil patterns are not given here but are available from ETI. SAE please.

150 mA. This meant that it had to be mains powered and was thus not truly portable. The size of the box used was approximately 7½" by 4¼" by 2¼" and into this space was crowded: 100 LEDs, 70 diodes, 140 resistors, 13 transistors, four 14-pin ICs, 43 op-amps, 60 capacitors, two switches, a socket, a microphone and two printed circuit boards. This was not too much of a problem, the biggest problem was making the unit battery powered. By using the LM324 the problem was solved. This component overlays carefully before starting any work on the project. It is wise to insert, and check, all through hole links first, followed by passive and active devices.

Each board, we complete, should be tested before final assembly.

Our photographs show how our unit went together, but the final appearance is very much a matter of personal taste.

Testing and setting up

The filter bank may be tested with a pink noise generator but preferably with a sine wave oscillator or for those lucky people with a swept oscillator (see ETI sweep oscillator, Aug. '77). This is ideal. The envelope follower output should draw out a contour of the filter, you will have to sweep the oscillator slowly to get a realistic impression of the response curve. If there are any substantial sensitivity changes from channel to channel, then by changing resistors R13, 22, 31 etc you can restore the overall flatness of the analyser. The sensitivity tolerance shouldn't be

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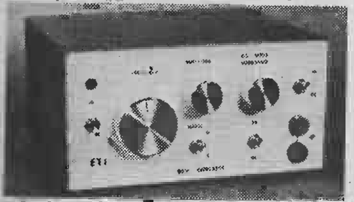
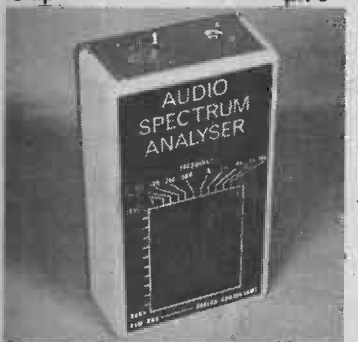


A sine of the times p.90

Chip on the waves p.79



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CAPACITORS

C1	10μ 16V tantalum
C2, 7, 12, 17, 22, 27, 32, 37, 42, 47, 52	1μ0 16V tantalum
C3, 4, 5, 6	100n polycarbonate
C8, 9, 10, 11	47n polycarbonate
C13, 14, 15, 16, 18, 19, 20, 21, 23, 24, 25, 26, 28, 29, 30, 31	15n polycarbonate
C33, 34, 35, 36, 38, 39, 40, 41, 43, 44, 45, 46, 48, 49, 50, 51, 54	1n0 polycarbonate
C53	22n polycarbonate
C55, 56	220n 16V electrolytic

SEMI CONDUCTORS

IC1	741
IC2-9, 11-13	LM 324
IC10	MC 1458
IC14	CD 4017
IC 15, 16, 17	CD 4016
Q1	BC 258
Q2-13.4	BC 169C
D1-70	IN 914
ZD1	2V7 400mW
LED 1-100	TIL 209

SWITCHES

SW1	SPDT centre off
SW2	DPDT

MISCELLANEOUS
PCBs, Electret tie microphone (Eagle), case to suit, display filter, batteries plus holders, 3.5mm jack socket, screws, bolts etc.

BUYLINES

All components used in this project should be physically small — hence ¼w resistors and polycarbonate capacitors.

Try to secure discount prices for the hundred LEDS and possibly for the LM-324S. Watch out for adverts in this issue.

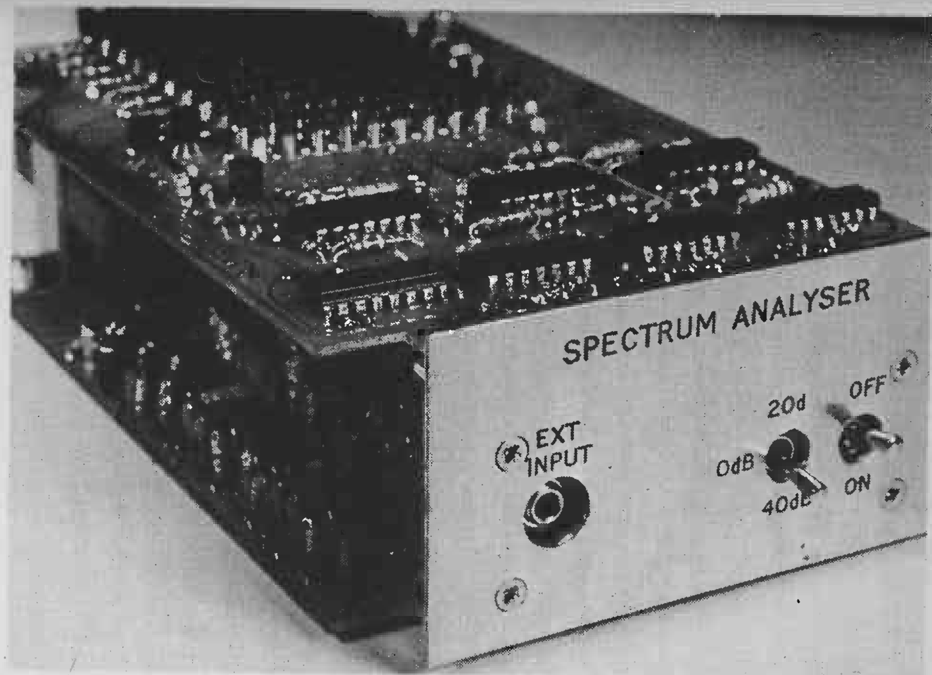
The case we used was from the popular Vero range which is stocked by many local shops nowadays.

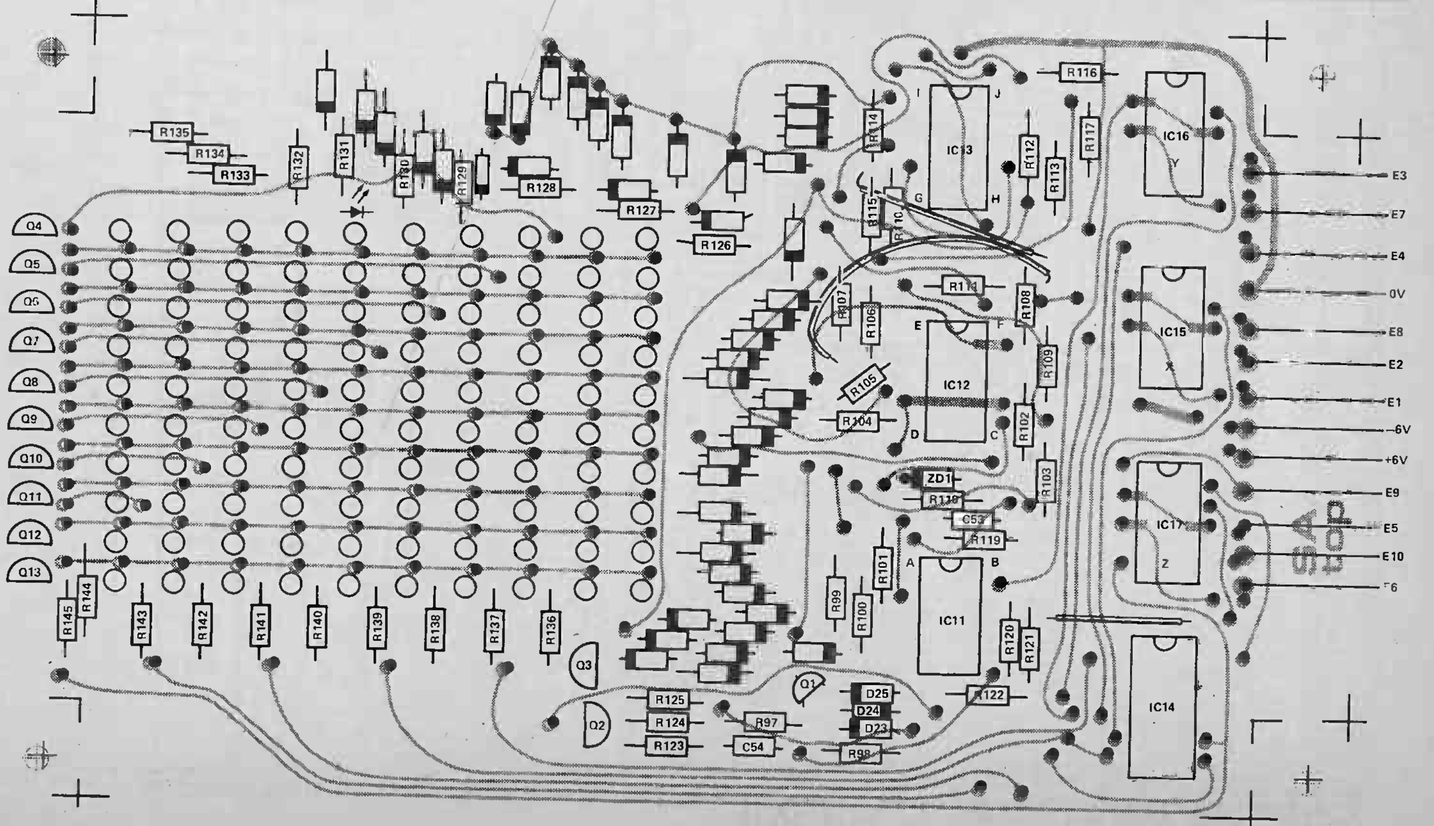
is a low power (0.8mA per package) quad op-amp. Thus, this device consumes one-tenth of the current per op-amp compared to the 741. The electronics consume only 20mA and the LED matrix display, when operating another 10mA. Therefore, if the unit is used for two hours a day, 30 hours of usage will be obtained, as long as HP7s are used. If the usage period is only half an hour per day then 60 hours may be obtained.

Construction

This project is not one to be undertaken lightly in order to keep the size of the unit down to hand held proportions, the two PCBs used have a very dense component layout and much care will be needed during construction.

Study the circuit diagram and





HOW IT WORKS

GENERAL SYSTEM
The general system is shown in Fig. 1. A signal from an electret microphone (this one has quite a good frequency response) is amplified and fed into a filter bank. (An external signal could be plugged in instead of the microphone signal). The filter bank is a set of ten band pass filters, each covering a bandwidth of an octave. Thus a frequency range in excess of 31 Hz to 16 kHz is obtained. The frequencies given are in fact the centre frequencies of the bandpass sections. The filtered signals are then sent to ten peak envelope followers. These units determine the peak signal levels and display a PPM type of response. That is they react quickly to transients and decay relatively slowly. The display that they generate is easier to visually follow than, say, a VU response. The ten envelope signals represent the average signal energy throughout the spectrum. This information must now be displayed on the LED metrix.

To enable a low parts count solution, a multiplexed design was used. That is, the envelope signals are investigated serially in time. A ten-way analogue switch is used to look at each envelope signal in turn. This signal is fed into a set of ten comparators which are logarithmically spaced 3dBs' apart. The comparators drive the LED matrix. The size of the envelope signal determines which LED in the column is lit up. The larger the signal the higher the LED. (However, this machine can be easily modified to give a bar display.)

The comparator that is on tries to light up all the LEDs in its row, but only one LED will light up. This is due to the ten transistor switches that drive the matrix columns. Thus, when the comparators are investigating envelope four, only the switch to column four is on. In this way the information is 'drawn' in the correct frequency column. This multiplexing procedure may seem a little complicated, but had we just used a comparator per LED then 100 comparators would have been needed, this method uses only ten! However you don't get something for nothing and there are a few problems to be encountered. The comparators used are LM324s. In fact they are op-amps, and tend to be rather slow. Their advantages are low power consumption and the ability to drive the LEDs directly. The speed problem means that when the multiplexer changes to the next channel, there is a short period of time when the comparators try to display 'garbage' because they are changing state. To overcome this, a blanking device turns off the LED matrix for a short period whilst the new information in the next channel is analysed. The multiplexing frequency is 500 Hz. This gives an analysis time of 2msec per channel and the whole display is repeated fifty times a second. If you shake the display it will strobe. Try the same thing on your pocket calculator!

INPUT AMPLIFIER
An electret microphone has been used as this is relatively inexpensive and yet provides a reasonably flat frequency response. It does however require a 1.5V power supply and this is generated inside the analyser.

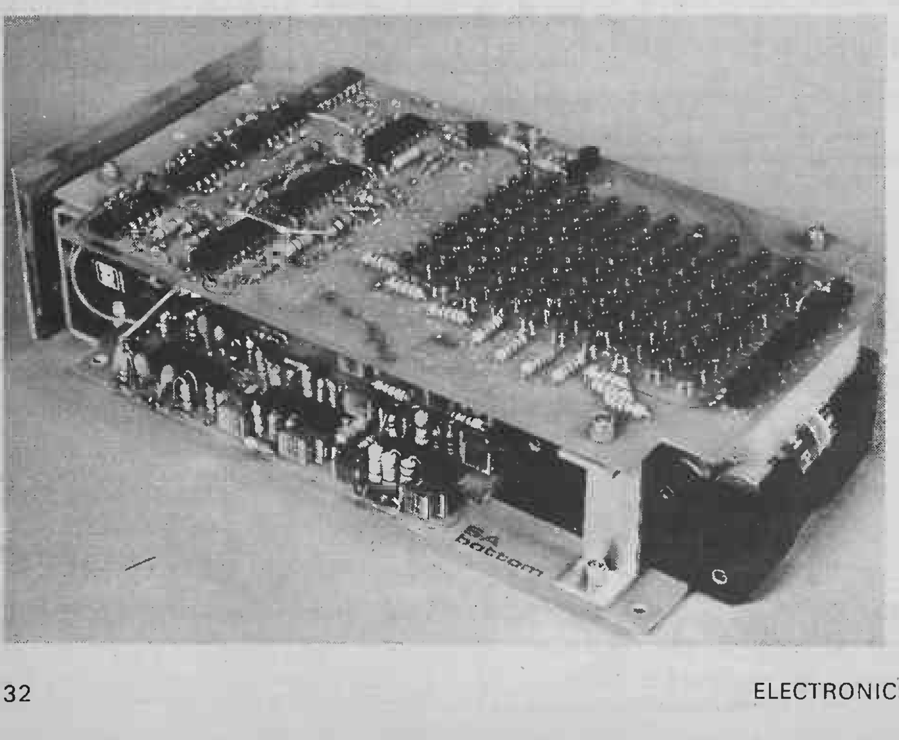
A gain selecting switch SW1 gives a 40dB range of input sensitivities.

FILTER BANK
Each channel of the filter bank is made up of two band pass filters. The filters are double tuned, that is they have slightly different centre frequencies. There is a slight dip in the pass band but at either side roll off slopes are very steep indeed. The filters used are single op-amp multiple feed-back bandpass filters with 'Q's of five each. Each filter pair has a very large signal gain in their bandpass of about 50dB. The tolerance for the resistors and capacitors are 5%. Any components out of tolerance may significantly change the filter response curve. This will cause the gain of each channel to alter. If the gain change is significantly large then it may be necessary to alter the gain of the following envelope follower stage so that the display is not distorted. The op-amps used for the first 9 channels are LM324. These op-amps require a pulldown resistor on their output if gross crossover distortion is to be avoided. There may be some visible crossover distortion at the filter channels output, but this will probably not adversely affect the analyser operation.

PEAK LEVEL DETECTORS
These devices are simple positive peak envelope detectors. The 1μF capacitor is charged up through the 3k3 resistor and discharged through the 180k resistor. Thus the output waveform quickly follows any signal level changes, but then slowly decays, exhibiting a PPM response.

MATRIX DISPLAY
IC11d is a single op-amp oscillator. It generates a square wave output at 500 Hz which clocks IC14, a CMOS decade counter/decoder. This has ten outputs, only one of which is high at any point in time. These ten outputs are used to control ten analogue transmission gates (switches), through which one of the ten envelope signals can pass. The switches are contained in ICs 15, 16 and 17. The output of the switches, this is in fact a multiplexer, is buffered by IC11c, and fed to the ten comparators. The comparators have a fixed reference voltage on one of their inputs (spaced at 3dB increments per device), and the envelope signal on the other. Thus, the comparators with reference voltages lower than the envelope signal try to go high. However, diode logic is used to make sure that only the highest comparator ON is the only one that is on. The logic turns OFF all those comparators below it, so that only one LED is ON at any point in time. The correct column has to be turned ON at the right time and to do this 10 transistors, Q4-13 are used. They are also connected to the counter decoder and are thus synchronous with the multiplexer. To provide the blanking a mono stable period is generated from the counter clock with Q2, 3. On the positive going edge, Q2 is turned on by the 1μF capacitor. However the capacitor quickly discharges and so a short monostable period is generated by Q2. It is inverted by Q3 and used to turn off Q4-13 (and hence the matrix display), for a short period during which the comparators are changing.

By omitting diodes D26-D70 a bar display rather than dot will be obtained.



more than ± 3dB. If it is in excess of this then there is probably a wrong component somewhere.

By feeding pink noise to the device each column should be approximately the same height. Due to the nature of the noise the top of the columns may jump up and down a bit but this should be averaged out by the eye.

The LED matrix may cause some problems. If certain LEDs won't light up then they are either broken or in the wrong way round. If one LED is unusually dim then change it. If a column is unusually dim, then change the transistor that drives that column. If a row is unusually dim, then change the comparator that drives that row. Check that IC11d has a square wave of about 500 Hz (± 30%) at its output. ETI

QUARKS

Truth is Stranger than Beauty — or was it Up is more Charming than Sideways? A guide to the strange world of Atomic Physics by Robin Moorshead.

IT WAS THE GREEK PHILOSOPHERS who coined the term 'atom' as the basic indestructable building brick of all matter. They even realized that matter and energy were closely related when they said all matter was made of the elements fire, water, earth and air. However it was 2000 years before man really began to investigate the subject scientifically.

Modern science began with the parallell developments of physics and chemistry in the seventeenth and eighteenth century. These scientists discovered the true nature of chemical reaction which led to the idea that all matter consisted of a limited number of elements, which they believed were made of indestructable atoms. They thought that all that was necessary was to find out how many elements there were and that part of science was complete. At the same time Newton formulated the "universal law of gravitation" which was the first "force field" to be understood.

By the end of the nineteenth century the picture had become more complex. With the discovery of more elements it was found that they could be laid out in patterns and groups, a fact which strongly suggested that they had internal structure.

When the electron was discovered this single picture of the atom was finally shattered. But a new simple picture was developed consisting of electroncs floating in positive charge clouds like currants in a bun. The physicists had added a second force field, the electromagnetic, to the list which helped them to explain the relationships between the positive and negative charges in the atom.

May The Force Be With You

The discovery of radioactivity gave the physicists a new technique. As the unstable nuclei exploded the high velocity debris could be used to shatter other atoms. This soon led to the discovery of the proton, the neutron and two new forces.

There had to be an incredibly powerful force binding the protons together in the nucleus since the electromagnetic force should blow the tightly packed positive charges apart instantly. Secondly it was discovered that the neutron itself was radioactive, it would only live for about 11 minutes outside the nucleus. This suggested that there was another rather weak force which held together the parts of a neutron. They were named the strong nuclear and weak nuclear forces respectively. The disintegration of the neutro caused tremendous problems, since they calculated that apparently energy was lost when the event happened. The only way to explain this without abandoning the law of conservation of energy was to propose a fourth particle the neutrino. This would carry away the missing energy. So when the neutron disintegrated this happened:

$$\text{neutron} \rightarrow \text{proton} + \text{electron} + \text{neutrino}$$

It was over twenty years before the neutrino was finally discovered.

The nineteenth century concept of a force field had by now given way to the idea that a force was transmitted by an "agent". So when two particles interacted they did so by exchanging the agent of the force. The agent of gravity was called the graviton (which has not yet definitely been detected). The agent of the electromagnetic force was the photon (which is easily detected). This meant that the strong nuclear force needed an agent which was called the 'mesotron'. But there was an important difference between the latter and the first two, since the latter and the first two, since the gravitational and electromagnetic force act over infinite distance but the strong nuclear force only acts over 10^{-13} cm. The agents of the infinite forces had no mass, but the mesotron had considerable mass. A new particle was discovered with the right mass but it didn't behave as it should, a problem not solved until after World War II.

May The Laws Be With You

Also at this time a particle in the 'positron' was dicovered, it was opposite in every respect to the electron, in fact antimatter. Furthermore when it collided with an electron they annihillated one another producing a very energetic photon. This was confirmation of Einstein's equation $E=mc^2$ which mathematically relates matter and energy.

So to sum up the situation before World War II we have:

known particles:
- Electron
- Proton
- Neutron
- Mesotron (with some wrong qualities)
- Photon

Suggested:
- Neutrino
- Graviton

Forces:
- Gravitation
- Electromagnetic
- Weak nuclear
- Strong nuclear

Combined with this they had established several laws which governed particles behaviour when they collided:

(a) Mass-energy is conserved ie: If two particles collide and create two new particles their combined masses may be greater or less than before but this is compensated for by them having more or less kinetic energy.

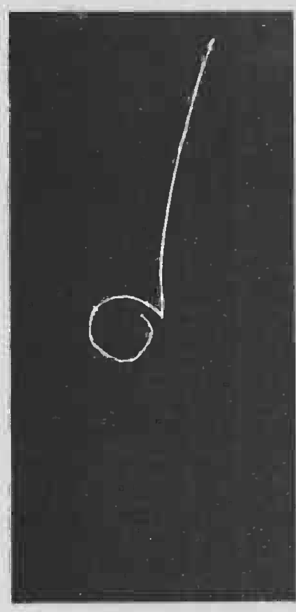
(a) Electric charge is conserved, ie: an electron cannot collide with a neutron and produce a positive particle, it must produce a negative one (and any number of neutral ones).

(3) Most of these particles spin, which again cannot be lost when they collide — like electric charge it must reappear in the new particles.

(4) At the time they believed also that any particle created through the strong nuclear force must disintegrate by it (likewise the weak nuclear). This was later found to be wrong.

The situation was quite satisfactory at the time, with a manageable number of particles and laws which governed all they had observed. But this simple picture was not to last long. Betwen the wars techniques had been developed to accelerate particles to immense velocity, so that they no longer needed to rely on radioactive disintegrations but could produce large numbers of very energetic missiles at will. Also they could see there events happening in "cloud chambers" where the particles would leave vapour trails exactly as high flying aircraft do.

It was with these techniques that the conundrum of the misbehaving mesotron was solved soon after World War II. In fact the mesotron they sought was found and it behaved exactly as expected, but it rapidly decayed yielding the particle they found before the war. So it had to be named and became the μ meson (the first discovered) and the π meson (the new particle). These are now known as muons and pions.



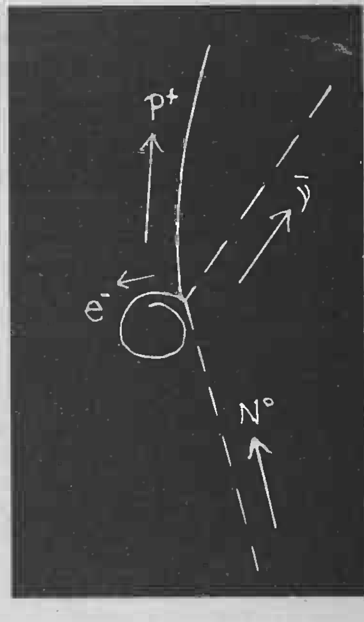
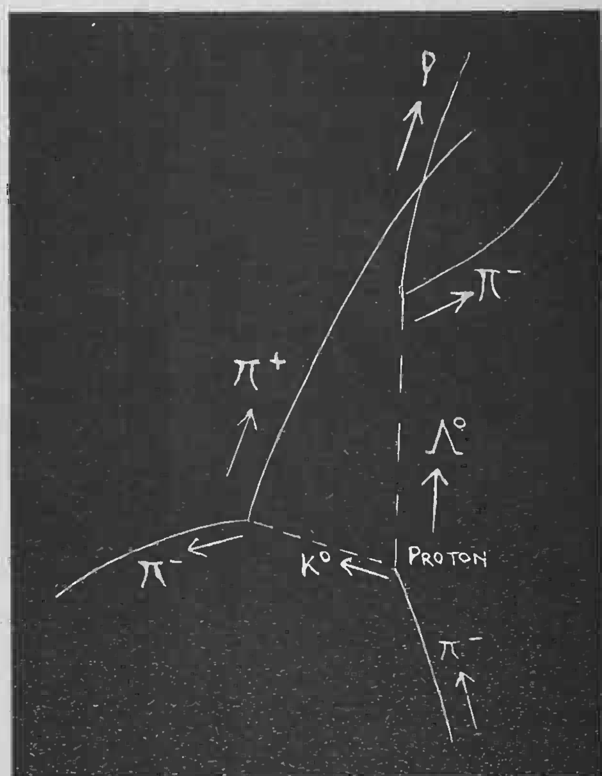


Fig 1 (top left) is what the disintegrations of a neutron would look like, meaningless at first but very meaningful on interpretation. The neutron enters from the bottom leaving no trail as it is incharged, when it disintegrates it yields a light negatively charged electron which has a curved track due to an applied magnetic field, the proton curves in the opposite direction being positively charged but much less than the electron since it has 2 000 times the mass of the electron. So in fact to the physicist it becomes Fig 2 (top right). The hatched lines represent uncharged particles which do not leave trails.

Fig 3. (below) shows the formation of a kaon and lamda hyperon from a pion striking a proton.

Strangely Strange

Now the trouble really started, most peculiar disintegrations were observed producing new particles which had no place in the scheme of things, and worse than that they broke the law that said if they were created by a strong interaction they must decay by it. They were created strongly and decayed weakly. Another feature of their creation was always being produced in pairs. An example of such a happening is shown in Fig 3, here a pion strikes a proton producing two new particles, a 'kaon' and a 'lamda hyperon', these then subsequently disintegrate to yield various known particles. This strange behaviour could be explained by analogy to the law of conservation of electric change. If matter possessed a new quality like charge which had to be conserved when such a particle was produced so an opposite must also be produced. This quality was termed 'strangeness'. The kaon has a strangeness of +1 and the lambda hyperon that of —1, the net result be zero change in total strangeness. Strangeness is now more commonly known as 'Hypercharge'.



The difficulties did not stop there, particles popped up all over the place and everything was in disarray, clearly it was necessary to classify the particles as the elements had been just 100 years ago.

The hadrons ("hard ones") are so named because they respond to the strong nuclear force. They themselves are divided into two subgroups on the basis of the way they spin, into baryons and mesons. Another ▶

difference is that mesons can be created in any number during a reaction, but the number of baryons is constant like total strangeness, ie if a baryon is created so must an antibaryon, and only a baryon can annihilate an antibaryon. The leptons are the 'small charge', little particles only involved in weak interactions. The photon is in a class of its own at present but would be with the graviton if it was discovered.

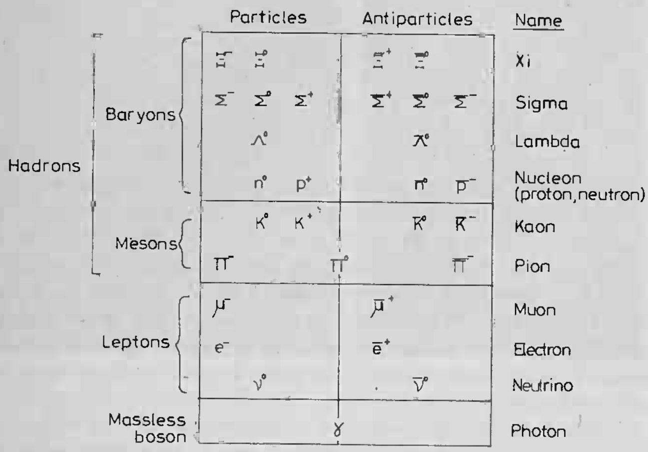


Fig. 4. Classification of particles.

As more and more particles appeared (over 100) they were all classified, and subgroups began to appear within the larger groups. The parallell to the classification of the elements 100 years ago is quite remarkable. Of course this immediately once again suggests internal structure.

Up, Down And Sideways

In 1963 two independent workers came up with a system which would explain all the hadrons in terms of just three particles, the up, down and sideways (or strange) quarks, and their antiparticles. The leptons did not lend themselves to this explanation and are still regarded as truly elementary.

The baryons are said to be composed of three quarks and two mesons, one quark and one antiquark. (No satisfactory explanation of why there are no groups of one four, five or six quarks has yet been offered). The properties of the quarks are such that their sum would be that of the particle they make up.

		SPIN	ELECTRIC CHARGE	BARYON NO	STRANGENESS
QUARKS	U (UP)	1/2	+2/3	1/3	0
	D (DOWN)	1/2	-1/3	1/3	0
	S (STRANGE)	1/2	-1/3	1/3	-1

		SPIN	ELECTRIC CHARGE	BARYON NO.	STRANGENESS
ANTI-QUARKS	Ū (ANTI-UP)	1/2	-2/3	-1/3	0
	D̄ (ANTI-DOWN)	1/2	+1/3	-1/3	0
	S̄ (ANTI-STRANGE)	1/2	+1/3	-1/3	+1

Fig. 5

A proton for example consists of one down (d) and two up (u) quarks. So if the properties of the three quarks are summed up we have;

	U	U	D	OBSERVED QUALITIES OF PROTON
SPIN	1/2	1/2	1/2	= 1 1/2 (FRACTIONAL)
CHANGE	+2/3	+2/3	-1/3	= +1
BARYON NO.	+1/3	+1/3	+1/3	= +1
STRANGNESS	0	0	0	= 0

Fig. 6

The important feature of spin is whether it is fractional or integral, the spin of the baryons is always fractional and that of the mesons integral.

An example of a meson could be the positive pion (π+) which consists of one quark and one antiquark, the up (u) and the antidown (d):

	u	d̄	OBSERVED QUALITIES OF PION
SPIN	1/2	1/2	= 1 (INTEGRAL)
ELECTRIC CHARGE	+2/3	+2/3	= +1
BARYON NO.	+1/3	-1/3	= 0
STRANGENESS	0	0	= 0

Fig. 7

The real justification for the quark theory can be seen by examining one of the "subgroups" found in the baryons. If one compares strangeness, electric charge and number of types per grouping ("istopic spin") we get a group thus:

STRANGENESS	NUMBER OF TYPES
3?	1?
2	2
1	3
0	4

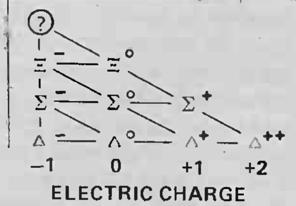


Fig. 8

This suggests another particle which would sit at the apex of the triangle. It would be a baryon, it would have no partners, and it would have three doses of strangeness. In fact it would be a baryon with three strange quarks (S,S,S). And sure enough it was found soon afterwards and is known as the omega minus (Ω—).

So once again the world was simple consisting of the following:

QUARKS	LEPTONS
UP	ELECTRON
DOWN	MUON
STRANGE	NEUTRINO (ELECTRON TYPE)*
	NEUTRINO (MUON TYPE)

*** It was realized in 1962 that there were two types of neutrino, one associated with the electron and one with the muon.**

Fig. 9

Charming Colours

So all particles with mass could be explained by these seven particles (and of course their antiparticles). But it was thought a pity that there was not a fourth quark, so there would be four quarks and four leptons. This was more than just the appeal of four to four symmetry, the leptons were 'paired' into electron plus its neutrino and muon plus its neutrino, so why not the same for quarks. The up and down quarks seemed 'paired' so why not a partner for the strange quark?

However as we progress into the late 1960's despite the fantastic accelerators and other resources, the quarks themselves remained undetected. Some evidence emerged that hadrons behave as a 'bag of bits', but proved impossible to split the bag open. At the same time the fourth quark ceased to hoped for just in terms of symmetry. It was now an essential member of the group of eight, and if it were not found the whole system was in danger of collapse. The reasons for this are very complex, they relate back to an hypothesis put forward as far back as the 1930's. This was that the weak nuclear force was the electromagnetic force 'in disguise'. This would require a new quality like strangeness to exist. Since it was the 'charm' that would ward off the collapse of the quark theory it became known as the charmed quark. It was eventually found in 1976.

Since it is obvious some immense force must bind the quarks together in hadrons, physicists were naturally interested in this. This became known as the 'colour force'. The agents of transmission of this force are called 'Gluons'. The reason it is called the colour force is that physicists have always found it offensive to have two or more identical particles confined together. So it was suggested the quarks assumed colour. (The colour is just a label, there is no implication that they are actually coloured). In the $\Omega-$ for example which has three identical quarks, one is red, one green and one blue, with no net colour since they equal white. The mesons are also 'colourless' since they consist of a quark and an antiquark, and antiquarks have the complimentary colours (cyan, magenta and yellow). So they would always consist of a coloured quark with its complimentary coloured antiquark, again a net colour of white. But pause a moment, we now have four quarks in three colours (and the same number of antiquarks). Life is not as simple as it was in 1964.

Now You See It?

As far as the isolation of individual quarks is concerned there are two schools, one who believes they have done it. The other which believes it is impossible.

Those who believe they have done it claim to have done so by suspending minute balls of niobium metal in an oscillating magnetic field. By this technique they can measure the electric charge on the ball to an accuracy of 1% of that of one electron! So the fractional charge of ⅓ or ⅔ of one electron charge associated with individual quarks should be very apparent, being 33 times greater than the error. This they claim to have done.

There are at least three suggested explanations as to why quarks cannot be isolated. One for example suggests the quarks can be thought of as being on the ends of a piece of string. As energy is fed into the system the string stretches, absorbing energy. So when it breaks it absorbs energy to create a new pair of quarks, so they are never seen in isolation. It is obvious that the validity of the quark theory lies in the isolation of an actual quark, or in a watertight explanation of why they cannot be separated.

However the story does not stop here. In late 1977 it was reported from America that two new quarks appear to be necessary to explain a newly found particle the 'upsilon'. These are named the 'top' and 'bottom' quarks which will have the qualities of 'truth' and 'beauty' (like strangeness and charm). Presumably if we wish to retain the symmetry between leptons and quarks we will expect the appearance of two new leptons!

But On The Other Hand

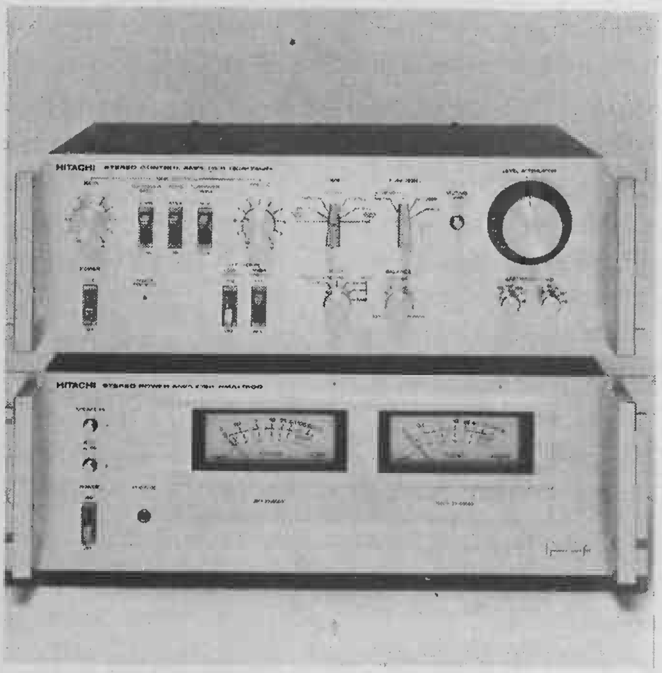
The fact that the search for the 'fundamental particle' seems to go through layer after layer of structure has caused considerable disquiet amongst scientists and philosophers. The Chinese view matters such as this in a rather different way to the West and call such particles 'stratons', implying they are just another layer. Also it has been observed that our approach to the subject may be doomed to failure, since it is in many senses based on the false premises of the original Greek philosophers. Currently there is much discussion about the concept that there is no one way of looking at a subject such as this, nor may there ever be. It may well be that we will always have to use one explanation when we explain one aspect, and another explanation when we wish to explain another. This has been the case for the electron since the 1920's since it is a particle with known mass, but also behaves like a wave. For some purposes it is talked of as a particle and for others as a wave.

The Eastern philosophy that all things are in harmony with one another has not gone unnoticed by the philosophers and scientists, and there have been interesting developments in this field, with respect to forces. Newton unified terrestrial and celestial gravitation, Maxwell unified electricity and magnetism, Einstein at the time of his death was trying to unify these two together. Since then the weak nuclear force and the electromagnetic have been unified, and finally early this year total unification has been proposed with the one force 'supergravity'. Perhaps somebody will come along and unify all matter!

ETI

audiophile......

Gordon King takes over Audiophile this month to explain and review Hitachi's new HMA 7500 MOSFET Power Amplifier, launched amid the Tulips early in March.



The HMA-7500 shown with its companion control amplifier. We understand that Hitachi have decided to make production models in black rather than the finish shown above. Still, what's a coat of paint between MOSFETs?

A NEW RANGE OF MOSFET hi-fi power amplifiers made its debut on the Japanese market at a Tokyo press launching during April 1977 and was introduced to the European market in Amsterdam in March. The range includes Model HMA-7500 with 80 + 80 W steady-state power rating into 8-ohm loads.

Circuit Details

The MOSFETs serve as power amplifiers and are driven by bipolars in differential-pair configuration. A brief, overall picture of the amplifier can be gleaned from the block diagram in Fig. 1. As will be seen, the amplifier includes an input high-pass filter (f_1 circa 3 Hz), drive and overload protection and a meter circuit which monitors the power in the left and right channels on a quasi-logarithmic basis. The meters (one for each channel) are scaled from −40 to +4 dB (0 dB ref. 100 W 8 ohms) and the control circuits endow the movements with a kind of peak programme meter response characteristic. Two stabilised power supplies are used, along with two directly-rectified supplies (one for each channel) from a multi-secondary mains transformer.

FETs are not new for audio power amplification. Junction power FETs are used, for example, by Yamaha and Sony; but the Hitachi amplifiers are the first to use power MOSFETs. Ordinary small-signal FET designs are unsuitable for high current working owing to the rise in conduction channel resistance with increasing current and the relatively low breakdown voltages. These problems were resolved by a new type of junction FET which was developed by Professor Jun'inchi Nishazawa and colleagues of the Electronics Communications Research Laboratory of Tohoku University.

These devices were named V-FETs owing to a large capacity vertical conduction channel and they are the type used in the Yamaha and Sony FET power amplifiers.

MOSST Interesting

The MOSFETs were developed by Hitachi's Central Research Laboratory at Kokubunji, Tokyo. They are made in complementary n-channel and p-channel pairs and are designated HS8401B and HS8402B respectively. Both versions are fabricated for a maximum drain current of 7A and a maximum power dissipation of 100 W achieved, as with the junction FETS, by a wide conduction channel of short length. As is well known, carrier mobility of n-channel devices is greater than that of correspondingly dimensioned p-channel counterparts. This dissimilarity is equalised by the Hitachi p-channel device having a greater channel width and smaller length than the complementary n-channel version being approximately 25% larger than that of the n-channel version. The on resistance of both types is 1 ohm.

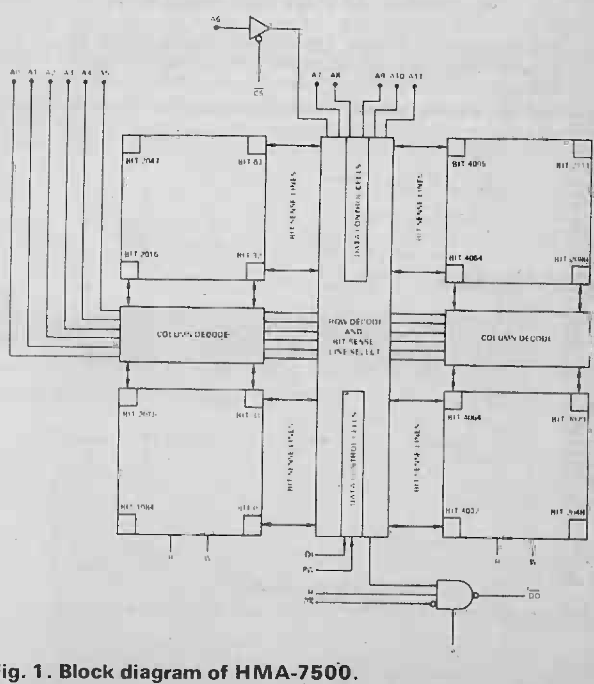


Fig. 1. Block diagram of HMA-7500.

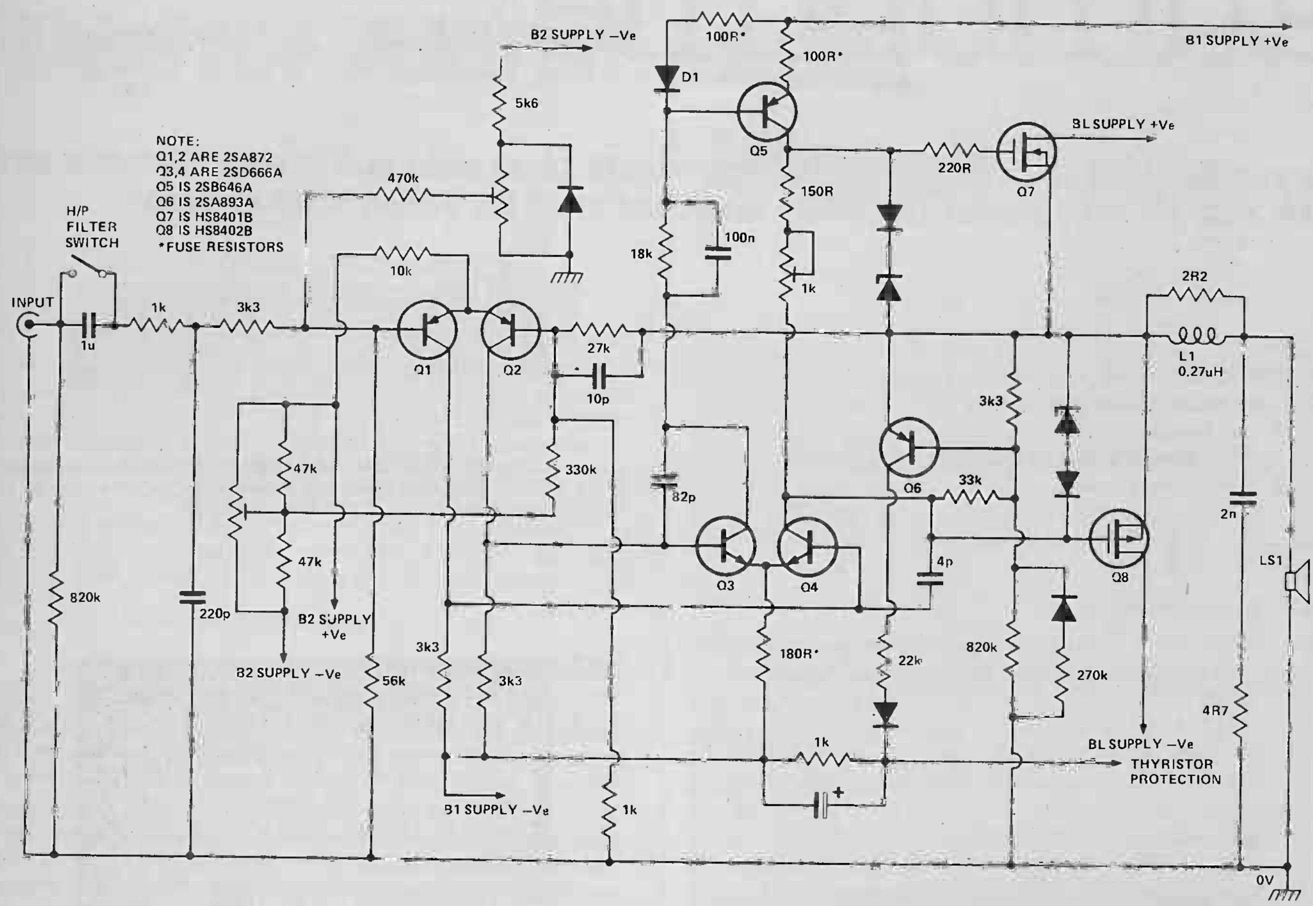


Fig. 2. Circuit diagram of HMA-7500.

The amplifier incorporates very efficient protection circuits rendering it virtually indesctructible regardless of the 'tough' nature of the electrical tests applied! A part of the protection consists of a fast-operating relay for switching the output. The control circuit for the relay also senses the presence of any abnormal rise in DC output off-set and automatically disconnects the speakers in the event of such an aberration. The speakers, of course, are directly coupled to the power amplifiers to minimise low-frequency phase shift.

Advantage FET

Power FETs have a number of advantages over bipolar counterparts, and the table below compares some of the primary audio parameters of the bipolar, J-FET, MOSFET and, for interest, the thermionic valve.

Parameter	Bipolar	J-FET	MOSFET	Valve
Switching speed	2 µS	0.05 µS	0.05 µS	0.1 µS
Upper freq.	20 MHz	100 MHz	150 MHz	~25MHz
Linearity	poor	good	good	fair
Input for 100 W output	1 W	0.5mW	0.5 mW	–
Power ripple effect	small	large	small	large
Thermal stability	fair	fair	good	fair
Circuitry	average	very complicated	simple	simple

FETs also have a very high input impedance, akin to that of the valve, and with MOSFETs because the oxide layer behaves like a high resistance the gate current is almost proportional to the drain-source voltage. The fast switching of FETs results because they employ single type mobile charge carriers.

Minority carriers of bipolars and their storage effects tend to inhibit the switching speed of amplifiers using these devices in the power stages. The very fast mobile carrier control of MOSFETs and the small input capacitance give these devices the edge on upper-frequency response, leading to small slewing times and the minimisation of slewing-induced non-linearity and attendant transient intermodulation distortion (TID).

These are phenomena which have been well disseminated in recent times and which, in certain areas, appear to be over-stated, particularly relative to ordinary programme signal whose rise-time is rarely faster than 50 µS and maximum effective slewing-rate not much greater than 4V/µS.

Runaway

Unlike bipolars whose collector current rises with increasing temperature, quickly leading to thermal runaway unless steps are taken to avoid this, FETs have a negative temperature coefficient which means that the drain current falls with increasing dissipation and hence temperature rise. Protection against breakdown is thus eased. Like thermionic valves, MOSFETs require very much less drive power than bipolars. This is because of the high power gain resulting from the high input impedance. The table shows that while a bipolar power amplifier may require an input drive of 1 W for an output power of 100 W, the same output power from FETs can be achieved from an input drive of a mere 0.5 mW, which greatly simplifies the drive circuits and leads to less drive signal distortion and hence the reduced need for very large amounts of negative feedback.

Another attribute of FETs over bipolars is that their transfer functions approximate a square-law. This means that odd-order distortion is less than that from bipolars while even-order distortion is cancelled by push-pull operation. The transfer functions of bipolars contain more odd-order powers, which means that a relatively high degree of negative feedback is required if the odd-order distortion is to be kept very low.

Resource Drain?

The electric field round the gate electrode is reduced by an ion implanted offset striped gate construction, which gives a source-to-gate breakdown as high as ±14 V and a drain-to-source breakdown as high as +160 V n-channel and −160 V p-channel. Cut-off frequency is limited by the input capacitance and intrinsic gate resistance, being in the order of 600 p and 65 ohms n channel and 900 p and 65 ohms p-channel. The cut-off frequency of the n-channel device is thus a round 3 MHz.

Down to HMA 7500

Almost the complete circuit of the HMA-7500 is given in Fig. 2. The complementary power MOSFETs are mounted on substantial heat sinks and the devices are biased for quasi-class-B operation. The MOSFETs are arranged as source-followers, one for each signal half-cycle, and the optimum bias current for the design is conveniently equal to the drain current. Since this is independent of temperature, the temperature compensating circuits found in some bipolar power amplifiers are unnecessary, which cuts the overall circuit complexity by about 30%. A more powerful model, yielding 100+100 W into 8-ohm loads, uses parallel-connected pairs of MOSFETs, but the front end of the circuit is similar to that of the HMA-7500.

Class EH?

The MOSFETs are driven by the class A differential pair Q3/Q4, which use an active current source consisting of Q5 and D1. It will be seen that both gates are driven together from Q4 collector, so that one MOSFET turns on during the negative half-cycles. The fast switching and optimised bias greatly tame notch and crossover distortion.

The input signal is applied to Q1 of the first differential pair Q1/Q2, while the negative feedback is applied to the base of Q2, which also defines the centre-voltage point. The collectors of Q1 and Q2 drive the bases of Q4 and Q3. Differential circuits are rendered viable because of the high input impedance of the MOSFETs, as distinct from the Darlington circuit requirements of bipolar

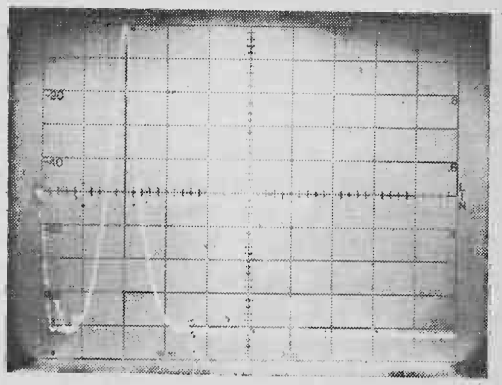


Fig. 3. Harmonic distortion at 80+80 watts into 8-ohm resistive loads at 200 Hz, also showing ripple components. Scale 100 Hz/div. horizontally and 10dB/div. vertically.

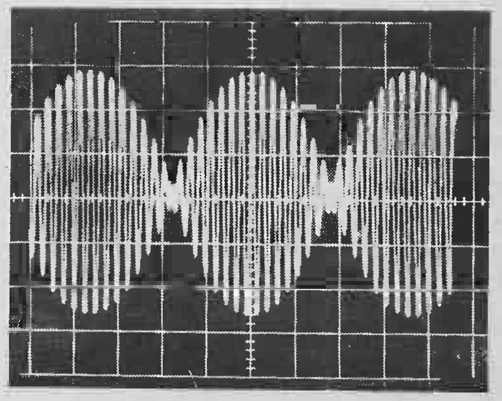


Fig. 4. (a) Composite 15/16 kHz signal at output of amplifier across compex load Z_L of 5 ohms modulus of impedance and 60 degrees phase angle, representing a 'difficult' speaker load. Scale 10 V/div. (b) Intermodulation distortion resulting from signal at (a) across Z_L at an output of 28 V peak composite signal. Scale 2 kHz/div. horizontally and 10dB/div. vertically.

power amplifiers. This means that fairly large amounts of negative feedback can be applied at high-frequency with minimal frequency compensation and without fear of the amplifier going unstable; in other words, a good feedback margin is achieved. The amplifier uses about 40 dB of negative feedback right up to 300 kHz.

Q6 is part of the protection circuit which works in conjunction with a thyristor (not shown). The speaker is connected between the centre-point line and the zero supply line, coupling being made through the 'Zobel' network consisting of L1 and associated components, which improves the total harmonic distortion performance.

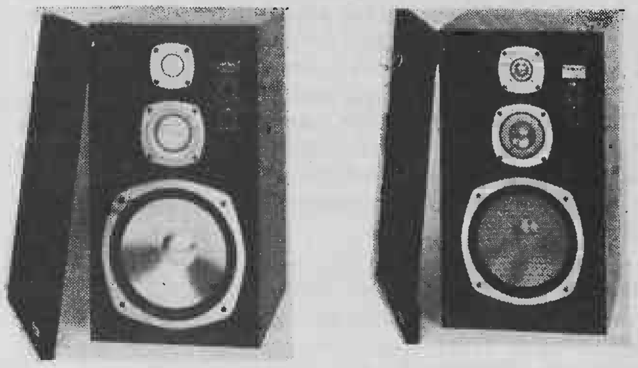
Lab Results

A large number of parameters were scrutinised in the lab by advanced testing techniques, and the results of some of these will now be looked at. The spectrogram in Fig. 3 shows the harmonic distortion and residual ripple components from a 200 Hz pure sinewave driving both channels to 80 W into 8-ohm loads. The spectrogram reveals that even under this high drive situation the amplifier is producing a mere −90 dB of 2nd harmonic (0.003%) and −82 dB 3rd harmonic (0.0079%), with all ripple components (50 Hz plus harmonics) being less than −82 dB.

As the power is reduced the distortion falls, quickly falling *below* our noise floor. In quiescent mode the total hum and noise across 8 ohms corresponded to a mere 1.7×10^{-8}W.

As hi-fi amplifiers rarely drive into pure resistance, we conducted an intermodulation distortion test using equal amplitude signals of 15 and 16 kHz driving into a speaker-simulating load of 5 ohms modulus of impedance and 60 degrees phase-angle at 16 kHz. The two-tone signal at the output of the amplifier is shown at (a) and the result of the test at (b) in Fig. 4. The test was conducted with the composite two-tone signal running at a *peak* value of 28V across the complex load, and the spectrogram at (b), scaled at 2 kHz per main division horizontally and 10 dB per main division vertically, shows that the 2nd order product at 1 kHz is −78 dB (0.012%) and the sidebands of the 3rd order products each about −64 dB (0.06%). This is a remarkably good performance at this high output into a loudspeaker-type load.

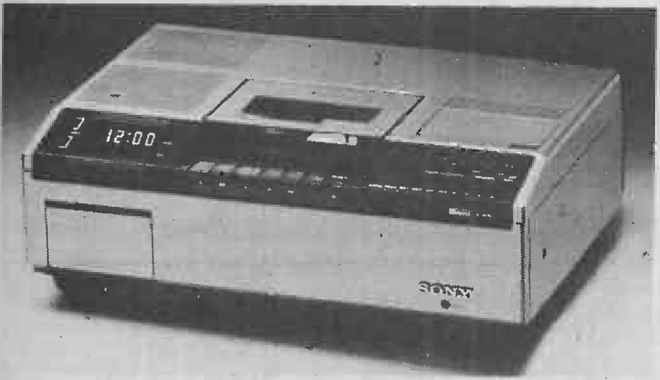
Again, with reduced output the products quickly dissolve below the noise floor. Most speakers require no more than about 16 V peak for 96 to 100 dBA sound pressure level in the listening room.



Pictured here are some more goodies on the way from Hitachi. These include (top) the unitorque direct drive turntable, two new loudspeakers: the HS 330 (above left) and HS 530 (above right), both using metal coned drivers (a GEC idea back in the '50s) and a "gathered suspension". Finally, below is the D900 cassette deck with three heads, solenoid operated mechanics and servo drive dual capstan.

news digest.......

son of u-matic



Not content with swamping civilisation with the U-matic, Sony have now launched their home video recorder — the Betamax. Since being introduced in Japan and America (in 1975) the Betamax system has sold over 500 000 units, but none of these used the PAL system found in the UK (and most of Europe). After three years development the European model (SL 8000 UB) will be in selected Sony dealers from June onwards with an expected price tag of £750 inclusive.

Unlike the U-matic, which uses ¾-inch tape, the Betamax system records on ½-inch tape — at the incredibly slow speed of 15/16ths of an inch per second! Cassettes are available to record from 30 minutes to a maximum of 3 hours 15 minutes, tape cost can work out at as little as 7p per minute.

Some of the many features are: One button record operation, built-in tuner, automatic recording up to three days ahead, pause/still frame facility, etc etc. It's not surprising that so many have been sold of the NSTC version, in fact in America Sony may have been too successful. Universal, Disney Studios and five other major studios are attempting to sue Sony for enabling people to infringe copyright!

For full details contact Betamax Division, Sony (UK) Ltd, Pyrene House, Sunbury-on-Thames, Middlesex.

better late than never

We must apologise to all our readers for the late appearance of the May issue — this was about one week late in most areas. We're almost fanatical about being on time, and it was our first late issue for three years. The reason was entirely due to severe printing problems, over which we have no control. We hope it will be at least another three yeas before it happens again.

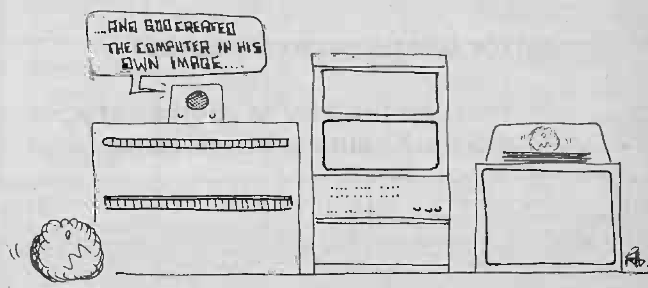
lcd measures ok

Titchy init! The LM300 is the latest thing from Non Linear Systems, you name it they make it smaller, and is distributed in the UK by Lawtronics. With 21 ranges it will measure AC and DC voltage up to 1 kV, AC and DC current up to 1 A and resistance in 5 ranges up to 10 M. The display is a high contrast 3 digit LCD, and the whole thing is only 1.9 x 2.7 x 4.0 inches in smallness. AA batteries provide the power and rechargeable cells are available. It has a 'big' brother called LM350 with 3½ digits in the same size case.

All ranges are protected up to kV DC or AC, prices are £74 and £87 (plus VAT) respectively.

Lawtronics Limited, 139 High Street, Edenbridge, Kent TN8 5AX.

intelligent golf balls



IBM is investigating the shrinking of large mainframes into 2-in cubes, including the several megabytes of main memory that go with them.

Shrinking computers is not a new idea. As far back as 20 years ago, physicists conjectured that the only way to increase computer speed, given fast enough logic and memory elements, was to cut the time delay between the elements by shrinking the interconnection distance. Coupled with its logical extreme, this approach was known as the "hairy smoking golf ball." The "hair" came from the myriad of proposed connectors to the computer; the "smoke," from the thousands of watts the "golf ball" computer would presumably try and fail to dissipate. At that time, various cryogenic approaches were tried but none worked because the superconducting logic proved either too slow or too hard to fabricate or both.

IBM's attempt to revive the "golf ball" is also cryogenic but based on using memory and logic built from Josephson junctions operating at liquid-helium temperature and orders of magnitude faster than previous "golf ball" projects.

Even if the input/output problems can be solved — how do you repair a malfunctioning computer? Because Josephson junctions require several deposition layers with different thermal-expansion coefficients, cycling between room and cryogenic temperatures tends to destroy them.

An IBM spokesman, who is optimistic about finding a solution to the repair problem, says, "We wouldn't have begun the research for the computer if we didn't think we could find a way to repair it without damage."

Power Width

The rated power of the amplifier is maintained from at least 5 Hz to 100 kHz, and the small-signal frequency response is a ruler-straight line well below 5 Hz (high-pass filter out) up to 250 kHz. The −3dB point is 358 kHz. In spite of this extended small-signal frequency response, however, TID could not be incited owing to the high slewing-rate of the MOSFETs which was at least 25V/µsec

With the high-pass filter switched in the lower frequency −3dB point was 3.4 Hz.

Subjective Impressions

The HMA-7500 requires an output of round 1V RMS for full drive so any control amplifier delivering this sort of output can be used with it. Hitachi, of course, make and market a suitable control amplifier; but because this was not at hand at the time of our tests we employed both the Radford ZD22 and the recent Pioneer C-21.

Both of these are very good control amplifiers, the Radford having a number of useful facilities and producing a signal that is so pure that the distortion is virtually unmeasurable. The Pioneer is also of very low distortion and includes a pair of pickup input switches allowing the loading to be adjusted in terms of both resistance *and* capacitance.

Left and right signals are accepted by the HMA-7500 via RCA 'phono' type sockets. Construction is very substantial and the heat sinks are large enough to allow the amplifier to be driven to sustained high steady-state power without distress. A brushed aluminium fascia accommodates the two meters, press-switches for speaker pairs A and B and a mains on/off switch.

The amplifier produces an output signal of virtually the same form as the signal applied to it.

Reproduction is thus essentially uncoloured so any distortion on the source signal will also be reproduced without much modification. The amplifier is not adversely affected by the electrical loading of the speakers, and using studio quality programme signals from master tapes in a controlled auditioning test all the listeners agreed that the reproduction was of a very high standard.

These initial observations have since been confirmed by using the amplifier for a number of months under typical domestic conditions on disc and FM radio sources. Of course, there are times when the distortion on the programme signals themselves detracts from the listening experience. Such distortion can be very much greater than that produced by the amplifier.

Conclusion

Amplifiers which yield a fair amount of distortion, notably even-order distortion, can disguise the source signal distortion and render the reproduction more palatable; but in our judgement it is the job of the hi-fi amplifier accurately to reproduce whatever is fed to it. If colouration is required to disguise the source signal distortion, then this should be introduced by a separate 'black box' connected between the control amplifier and power amplifier!

We vote the Hitachi MOSFET power amplifiers an outstanding achievement in state-of-art hi-fi electronics.

ETI

FM~AM RADIO

Bill Poel of Ambit has been beavering about designing the ETI Chip Monk, a single chip

SINCE THE EARLY 1970s, various attempts have been made at producing a linear IC that performed the functions of AM / FM radio inside one package. Most have not been particularly successful, and those that have found their way into a degree of prominence, quite obviously have not offered any economy or improvement in performance over age-old five and six transistor designs. The Mullard TAD100/TAD110 is an example of this breed of device — and understandably, it never really caught on with volume manufacturers.

Bsic problems were simply those of economy, and when coupled with the notorious instability of combined function radio ICs, with complex switching of live signal paths, the whole idea has been sent back to the drawing board for a re-evaluation. The early euphoria of "ICs for ICs' sake" didn't last long in the consumer market.

Sprague Board To Success

However, after a long lapse in this particular field, Telefunken and Sprague were commissioned by a large manufacturer to supply an IC performing the functions of FM IF, AM RF/OSC/IF, and audio power output stage, in one 16 pin IC, the TDA1083. (ULN2204). It is this IC that has presented the long awaited breakthrough in cost effectiveness, since you will see from the circuit diagram (Fig 1) that the component count is quite dramatically cut from a discrete design, in fact, only essentials such as tuned circuits and decoupling capacitors seem to be left, leaving the European manufacturer a chance — at last — to think about competing with the imports from the Far East, since the biggest cost, in the form of labour, is now reduced to a bare minimum. Testing is greatly simplified, since the chances of incorrect assembly are reduced in proportion to the parts count — and thus the TDA1083 is destined for a sparkling future.

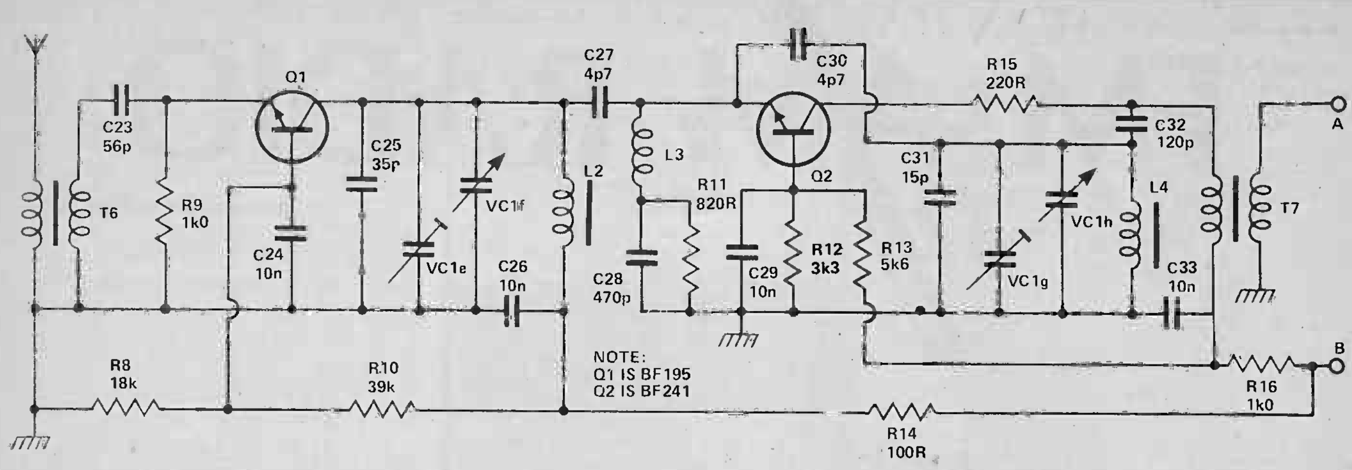
Supply And Current Demand

But more than that, the TDA1083 uses the advantages of IC complexity to reduce AM / FM switching to DC functions, and to provide an overall circuit that operates with a supply voltage as low as 2 V (in the AM mode, the FM oscillator stops at about 4V), and has a current drain of some 8-10 mA, including the audio output stage! (quiescent current drain conditions). The specifications, whilst not admittedly 'HiFi' are nonetheless quite excellent for this class of radio — which in any case usually finds its scope somewhat limited by an indifferent loudspeaker in a non-ideal enclosure.

What is good for the manufacturer, must also be good for the home constructor / enthusiast, and this article sets out the basic application of the TDA1083 in an easily made and aligned wireless, based on a DIY tunerhead for the FM range. The coils used are selected from the universally renowned TOKO range, and are sufficiently predictable to enable the constructor to switch on the set in a non-aligned state, and expect to be able to hear a sufficiently wide selection of stations and general 'noises' to permit more accurate alignment without the aid of specialized RF test gear.

Construction

If the specified components are used and assembled on the PCB according to our overlay, construction should not pose any problems. The wire links on the underside of the board should be wired last. ▶



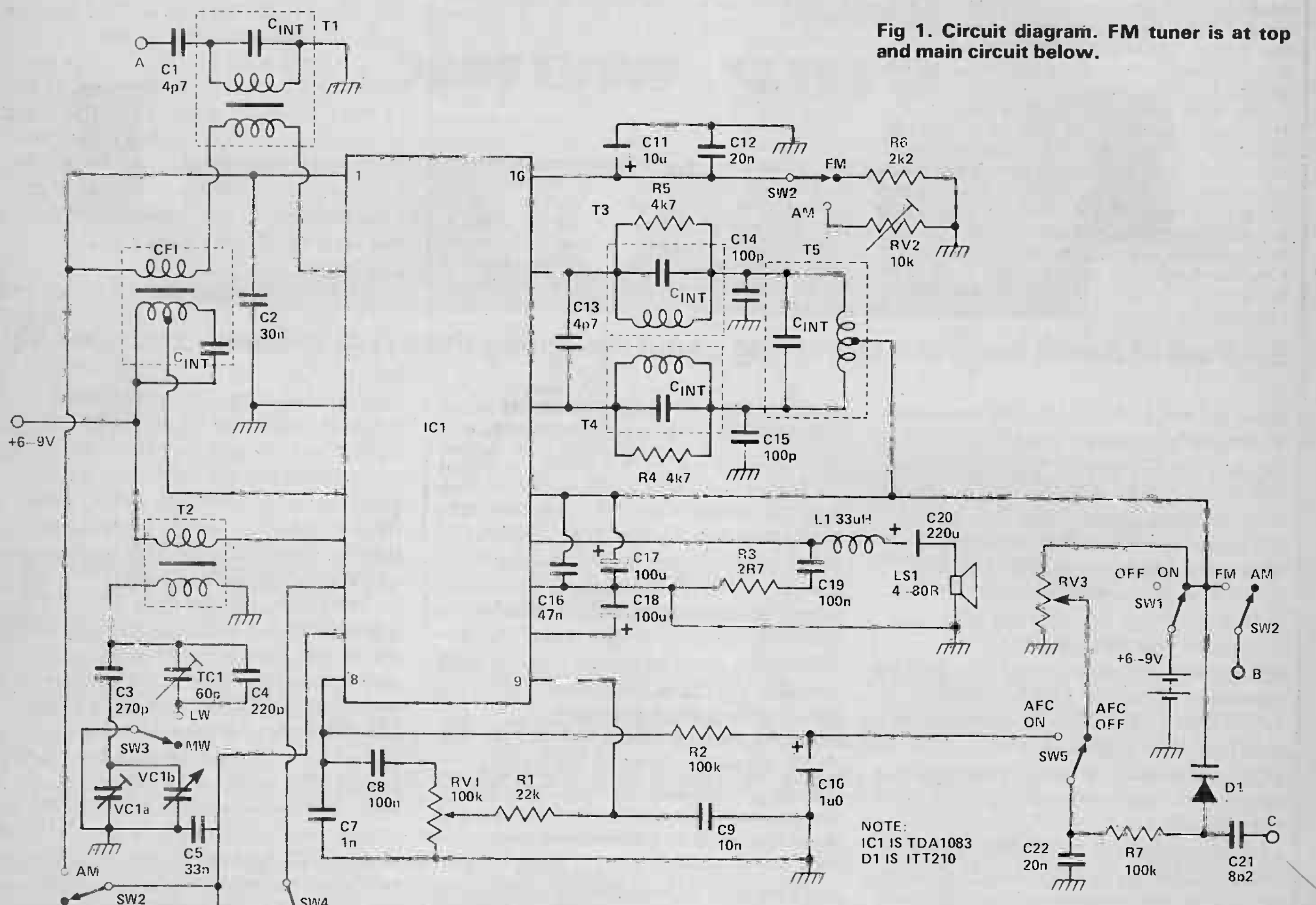
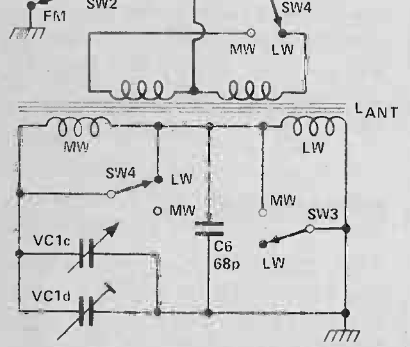


Fig 1. Circuit diagram. FM tuner is at top and main circuit below.



Testing and alignment

Testing and alignment is delightfully simple, when compared tothe usual AM/FM set. On switch on some noises must be heard on AM if the unit has been correctly assembled — and the best start point to check the circuit out is on MW. The local oscillator is set to 2075kHz (470kHz) at the HF end, and with antenna coil flush on the road, the MW is just about optimized already! The LF end should be set for 995kHz, and the antenna coild peaked on the rod, with the capacitive trimmer adjusted at the HF end of the band. A meter may be used connected to pin 16 to read the reverse AGC voltages, but the ear should be just as suitable an indicator.

The Long Wave (150 to 275kHz) is tuned in a similar fashion, except that there is no antenna trimmer provided since the basic inductance adjustment is quite sufficient to track the rod over this relatively narrow band. The LW coil is about

HOW IT WORKS

In portable applications, selectivity in the FM tunerhead is not put to a very stringent test unless you happen to live within a couple of miles of the transmitter. The circuit adopted here is almost universal (with minor modifications only) amongst manufacturers of portables, table radios — and most 'non-HiFi' wireless. The input stage is a common base NPN stage, with untuned input via a broadband ferrite transformer, and the output is a single tuned circuit, feeding the mixer/oscillator stage.

The mixer/oscillator employs the usual emitter/collector feedback to provide the oscillation 10.7MHz high of the signal input frequency, enabling the IF of 10.7MHz to be taken off via the IF transformer. You will see the oscillator coil is coupled into the collector (after the 220Ω spurii 'stopper' resistor) via 120pF, which is a low impedence at the oscillation frequency. At 10.7MHz, this capacitor is used to resonate the primary of the output coupling IF transformer — where the VHF oscillator coil is a low impedance to ground for the IF signal.

The most important spect of this stage is the tracking, which is the term used to describe the way in which the RF tuned circuit and oscillator tuned circuit are made to remain a constant 10.7MHz apart when tuning the range 88-108 MHz. Much padding capacitance is required in the tuned circuits to enable the tracking to remain reasonably consistant over the range – and if omitted, the tuning range would exceed some 30MHz due to the reduction in residual capacity with the tuning capacitor at minimum. This technique is used to improve the stability of the circuit, since any errors in the manufacture of the tuning capacitor etc. are less emphasized than if the tuning capacitor were simply made to vary the few pF necessary to cover the range 88-108MHz in the case of low residual capacity — say 5-8pF since

$$\text{Frequency} = \frac{1}{2\pi\sqrt{LC}}$$

where at 108MHz, C is residual – say 7pF (tuning capacitor at minimum)

$$\text{so } L = \frac{1}{(2f\pi)^2\, 7 \times 10^{-12}} = 0.310\text{uH}$$

to tune to 88MHz with a 0.310uH requires a capacitor of

$$\frac{1}{(2 \times 88 \times 10^6 \times \pi)^2 \times .310 \times 10^{-6}} = 11\text{pF}$$

Thus 88-108MHz can be tuned by just 4pF swing, but that is impracticable due to stability of components, and accurate matching of the stray capacities affecting the RF and oscillator circuits.

The various values used in this design are derived from a combination of calculation and experimentation, and represent a suitable choice for the tuning capacitor specified.

Once out at 10.7MHz, the signal passes through the IF bandpass filter, which uses two High Q IFTs, coupled in such a manner as to provide adequate selectivity for the IF stage in the TDA 1083. The signal emerges at the detector stage at pin 14 and 15, to be demodulated in a form of quadrature, and thence passes to the audio amplifier via the volume control. The DC level at the detector output of pin 8 is also proportional to the IF frequency, and thus provides a suitable reference for the AFC to operate. The AFC voltage must be fully decoupled from audio frequencies, otherwise the AFC will simplv track the FM of the carrier, and nullify the modulation. If insufficiently decoupled, the tone will appear excessively treble, as the bass frequencies will be removed by the AFC.

On AM, the procedure is carried out entirely in the TDA1083, with the ferrite rod antenna signal being mixed with the local oscillator 470kHz above the signal frequency, to produce the 470kHz IF signal at the input to the AM IF filter.

Tracking considerations also apply to AM, in much the same way as to the FM section. Those of you with scientific calculators can apply the same formulae if you like, but suffice it to say that the values used in the AM tuned circuits are right for the job. The oscillator coil has an inductance of 156uH, which if tuned with all the available capacity from the tuning condensor, would cover plenty more than the 525-1605kHz that comprises the MW. In fact, the values are chosen because of the simplification of LW, where coverage is made possible with the simple addition of a capacitor across the oscillator tuned circuit although the antenna coils are switched. This is the European convention. and has grown up over the years as being the optimum compremises in sets where both MW and LW are required. Those countries not requiring LW tend to use AM tuning capacitors with 80pF swing in the oscillator section, and 180pF in the antenna, providing what is known as parallel tracking – since the oscillator frequency is always 470kHz above the RF frequency, it requires proportionately less capacity to tune, if the values of the RF and oscillator inductance are chosen to be the same.

The oscillator padding (or 'tracking') capacitor that is placed between the 26pF swing of the tuning condensor, and the 156uH oscillator coil in this design, is designed to reduce the effective capacity swing 'seen' by the oscillator inductance, and thereby create a situation where the oscillator and RF stages track together. Those of you with programmable calculators can see that the tracking cannot be entirely accurate in this way, and in fact, the tracking is only spot-one at three points along the tuning scale. However, from FIGURE THREE you will see the actual curve, and by careful design, the tracking error can be kept insignificantly small in the context of this type of receiver.

A word about pin 16— this provides access to the IF gain of the TAD1083, and a 10k preset here provides adjustment of the maximum gain on AM, which is generally rather too great, causing excessive noise pickup and general hash. The lower the resistance, the lower the gain. A little experimentation after final alignment will select the optimum point for any particular unit.

Finally, a word about the audio stage, since IC AF stages are notorious for RF instability. Early units such as the SN76023 suffered severely from ultrasonic instability, brought about by general positive feedback, poor earth layout or reactive loading. The TDA 1083 has come a long way since then, but still is slightly prone to RF instability when driving into high current loads. The use of the 33uH choke and famous Zobell network at the speaker output pin are obligatory to prevent RF getting into the IF stages and causing the whole thing to break up and crackle on audio peaks. The instability takes the form of an FM signal at about 18-20MHz and so the choke is selected from the TOKO TBA series to be self resonant at those frequencies. Other types of 33uH choke may not necessarily be as effective due to different self capacities.

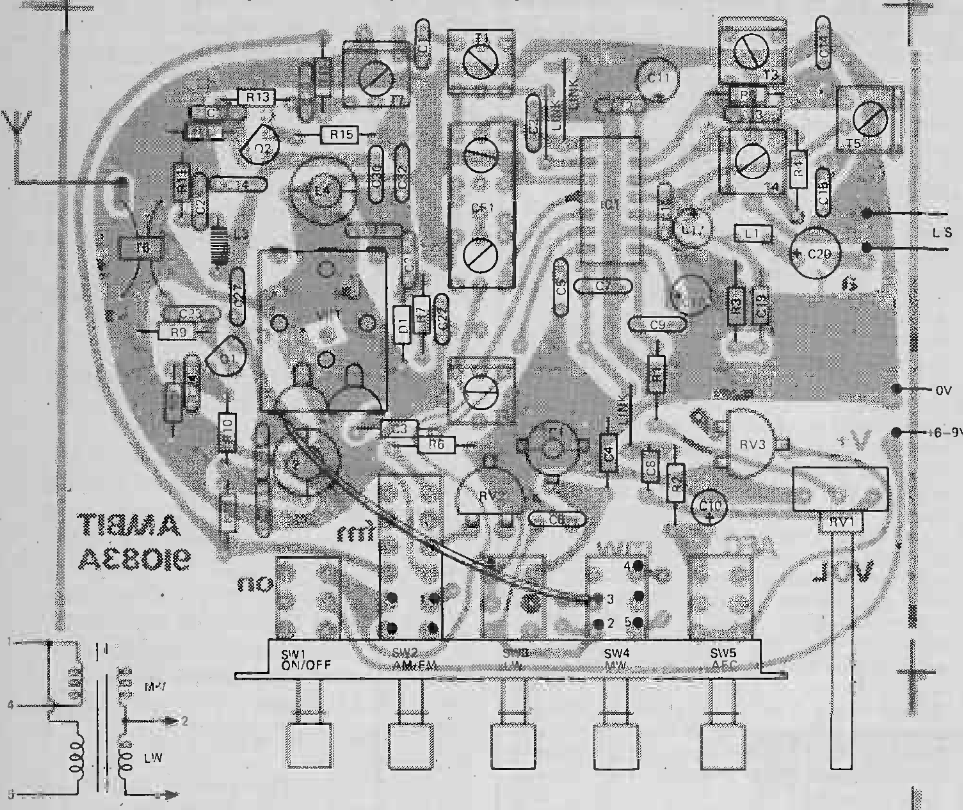
7.5-10mm from the rod end in most cases. The IF filter requires virtually no adjustment — only occasionally will peaking the Blue core have an effect.

The detector coil should be set for best AF, and with most devices, the coil does not require a damping resistor, though certain manufacturers data advocates the use of something in the region 10-22k. If too heavily damped, the audio on strong signals becomes distorted.

FM is slightly more troublesome. A 10.7MHz signal source is a useful aid to set the IF, but once again, it is possible (with patience) to adjust by ear. The oscillator coil will be approx. 3-4mm above the top of the 2½ turn S18, and the RF coil flush with the top of the S18. Such is the reliability of the S18 style, presetting the coils in this fashion has always provided sufficient initial assistance to enable further alignment to continue. It is very difficult to get completely lost in the wastes of MHz using this approach.

The FM detector coil T2 should be set for best AF, on a relatively weak signal — and then the other FM IFs can be adjusted for best quieting. With the IF aligned, the tracking procedure for the RF and oscillator coils is now a great deal easier, and can be carried out with the knowledge of the local transmitter frequencies as your basic datum points. Those of you with signal generators, spectrum analyzers etc. to hand, should not require further instruction on their application to this particular task. Impressive performance should result, with 5uV or better FM sensitivity, and AM sensitivity to match any other portable radio you can lay your hand on. (In the under £50 region). Familiarity with radio design only comes with long experience. More so than other area of electronics, since there is no real "go/no go" state, as 'go' is very much a matter of degree. 'No go' can be obvious enough, but there will always be conditions of instability where the unit will operate delightfully well at one point, and not at all further along the band. The TDA1083 brings radio a little closer to the 'Go/no go' wireless, but there are still many areas of degree of 'go', so once you have achieved a satisfactory state of 'go' you continue to try to squeeze a little extra out of the circuit at your peril. The last dB is always the hardest to achieve. ETI

Fig. 2. Component overlay for the Chip-Monk radio.



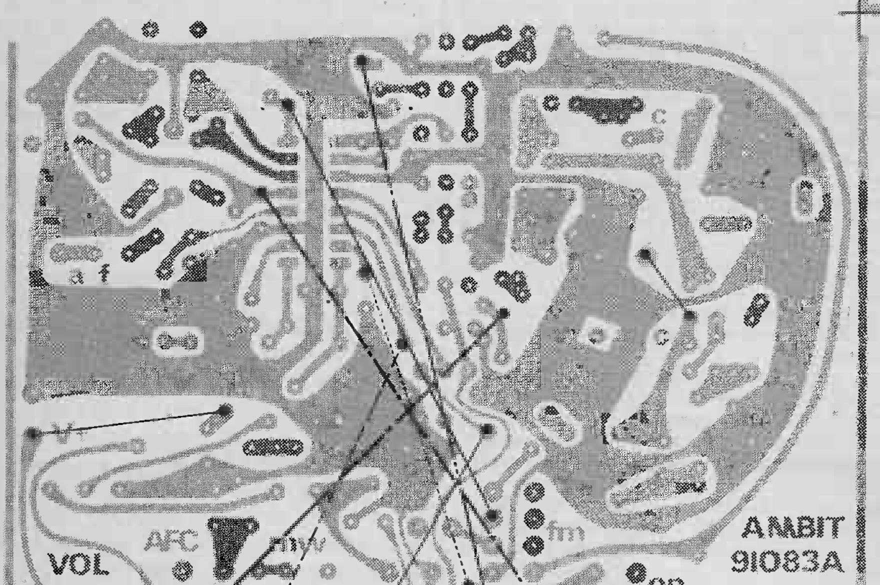
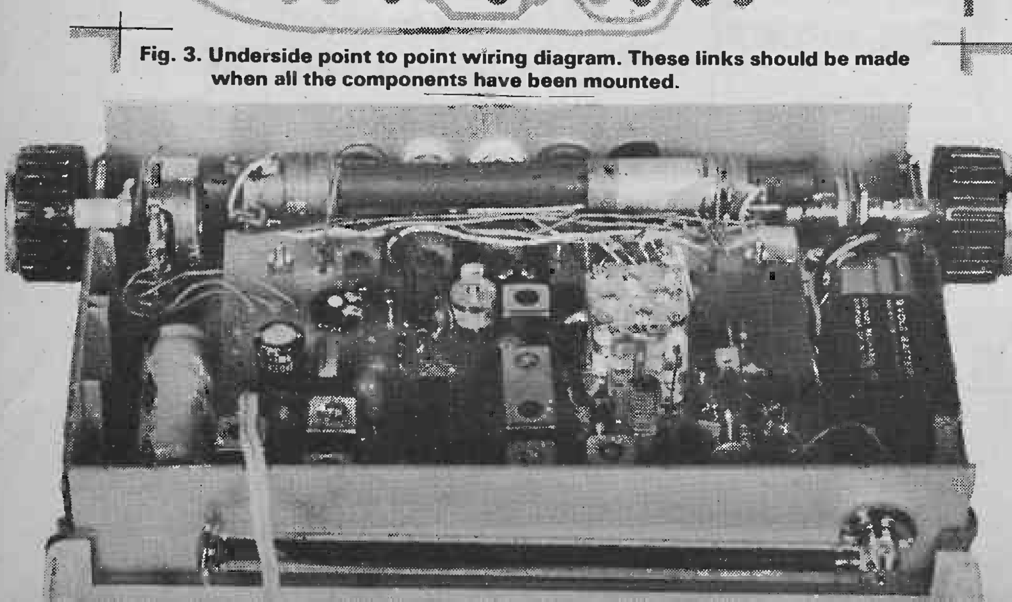


Fig. 3. Underside point to point wiring diagram. These links should be made when all the components have been mounted.



PARTS LIST

RESISTORS (all ¼W 5% unless stated)		
R1	22k	
R2, 7	100k	
R3	2R7	½W
R4, 5	4k7	
R6	2k2	
R8	18k	
R9, 16	1k0	
R10	39k	
R11	820R	
R12	3k3	
R13	5k6	
R14	100R	
R15	220R	
CAPACITORS		
C1, 13, 27, 30	4p7	ceramic
C2	30n	polyester
C3	270p	ceramic
C4	220p	ceramic
C5	33n	polyester
C6	68p	ceramic
C7	1n0	polyester
C8, 19	100n	polyester
C9, 24, 26, 29	10n	polyester
C10	1µ0	10V tantalum
C11	10u	10V tantalum
C12, 22	20n	polyester
C14, 15	100p	ceramic
C16	47n	polyester
C17, 18	100u	10V tantalum
C20	220u	10V electrolytic
C21	8p2	ceramic
C23	56p	ceramic
C25	35p	ceramic
C28	470p	ceramic
C31	15p	ceramic
C32	120p	ceramic
VARIABLE CAPACITORS		
VC1	CY2-22177	
TC1	60p	trimmer
SEMICONDUCTORS		
IC1	TDA 1083	
D1	ITT 210	
Q1, 2	BF 195	
INDUCTORS		
T1	94AE530465	
T2	YMRS16726	
T3, 4	KACS 9339PFV	
T5	YJCS 17105	
T6	FXH 1	
T7	94AES 30465	
L1	33uH	
L2, 3	red s18 2½t	
L4	16t/13mm dia 26 SWG air core	
SWITCHES		
S1, 3, 4, 5	2 pole change over	
S2	4 pole change over	
MISCELLANEOUS		
PCB as pattern, speaker, case to suit		

BUYLINES

Ambit International at 2 Gresham Road, Brentwood, Essex, are to offer a kit of parts for this project. The kit will include the PCB, electronic components, Ferrite rod and coil but will not cover the speaker, case, mounting hardware etc. For two months ETI readers can buy the kit at a special offer price of £10.95, the regular price is £13.95.

Getting hold of the hardware for the tuning drive might prove difficult but although not included in the kit Ambit should be able to help in this area as well.

THE TEXAS TI 59 REVIEWED

As technology increases in leaps and bounds, calculators become more powerful and computers more stupid — The gap in the middle ever decreasing. Les Bell has taken a look at the new Texas Pocket Programmable (PPC), the latest in calculator technology .....

THE MAJOR DIFFERENCE between the TI59 and previous PPCs is its use of 'Solid State Software'. If you flip the calculator over and slide out the panel in its base, you will find a 0.85" x 0.7" x 0.35" block of plastic labelled 'Master Library Module'. This is, in fact, a read only memory containing anything up to 5,000 steps of program, which in the case of the Master Module provide 25 programs designed to solve a variety of problems.

The Master Library Module can be changed easily for different modules in the fields of Surveying, Business, Navigation, Aviation and Statistics (no Mathematics or Electrical Engineering modules as yet). A spare module can be carried in the wallet supplied along with 40 magnetic cards.

Programs are called up from the Module by the keystroke sequence '2nd Pgm nn,' where nn is the program number, and the user-definable keys can then be used to run the programs. In addition, Module programs can be called as subroutines from user programs by the same sequence of keystrokes since the Module programs do not occupy the same address space as the read/write memory in which the user's program runs. The Solid State Software can also be downloaded into the RAM section for examination or modification, using the keystroke sequence '2nd Pgm nn 2nd Op 09.' The calculator can then be put into the 'Learn' mode and the program modified.

User Memory

This leads us naturally into a discussion of the block of memory available to the user in the TI59. Here again TI's semiconductor memory expertise has come to the fore; the P59 is, in terms of memory, way ahead of its competition, with a possible 960 steps of program memory.

Why 'possible'? Well, the TI59 has inherited an organizational hangover (if that's the word!) from its predecessor, the SR52. In that calculator, program memory and data memory are physically the same, and, as many owners discovered, spare program space can be used for data storage. The TI59 employs a similar scheme, but now TI openly admit to it, and partitioned memory has become what PPC owners call a 'supported feature'. Another SR52 unsupported feature which has turned up respectably in the P59 is the ability to store data on magnetic cards. ▶

When initially turned on, the TI59 has 480 steps of program memory and 60 data registers. However, the user can repartition memory, trading off 80 program steps for every 10 data registers, so that one may have 800 steps/20 data registers or 320 steps/80 registers-or one of several other combinations.

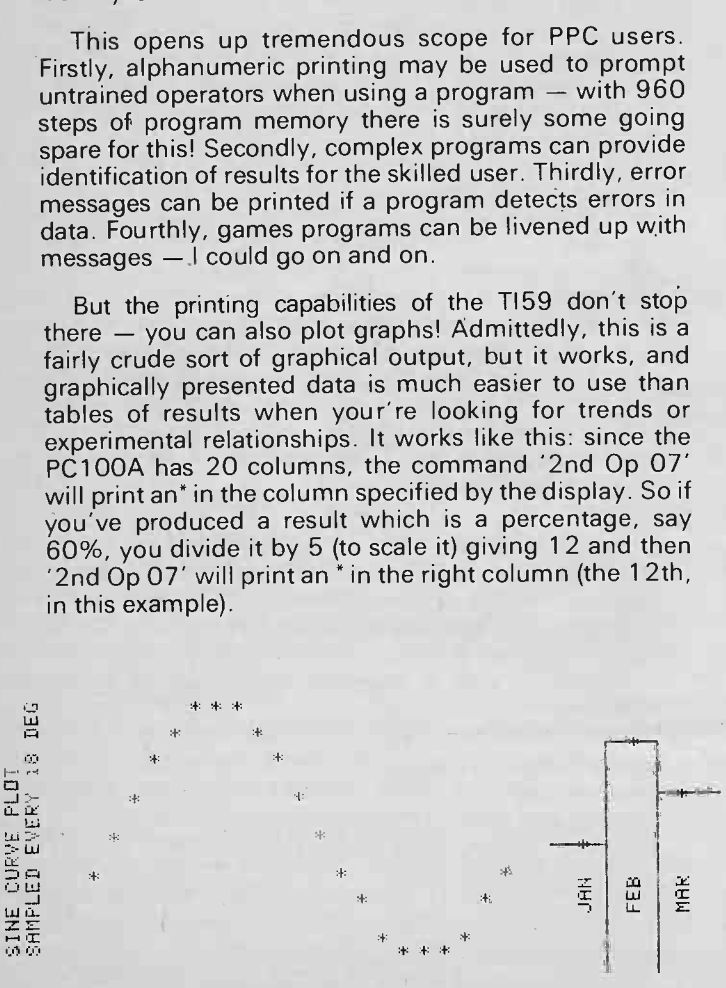
The TI59 has a kid brother, the TI58, which has identical features, including the Solid State Software, but less memory (240 steps/30 registers on switch-on) and no magnetic card capability. Except where memory size or magnetic cards are concerned, all my remarks apply equally to both PPCs.

Printing

The third main area of advance is in the incorporation of printing facilities in the P59. Like the SR52, the TI59 is designed to operate with the PC100A print cradle. The important difference between this and previous PPCs is that the P59/PC100A combination can print alphanumerics. The PC100A can print 20 characters wide, and this can be divided up into 5-character quarters, with each character being represented by a two digit code, e.g. A is 13 and (is 55. Five characters therefore fill a 10-digit display, and four such displays are successively loaded into a print buffer, which is then completely printed. Alternatively, the current answer can be printed along with four characters on the right to identify it.

This opens up tremendous scope for PPC users. Firstly, alphanumeric printing may be used to prompt untrained operators when using a program — with 960 steps of program memory there is surely some going spare for this! Secondly, complex programs can provide identification of results for the skilled user. Thirdly, error messages can be printed if a program detects errors in data. Fourthly, games programs can be livened up with messages — I could go on and on.

But the printing capabilities of the TI59 don't stop there — you can also plot graphs! Admittedly, this is a fairly crude sort of graphical output, but it works, and graphically presented data is much easier to use than tables of results when your're looking for trends or experimental relationships. It works like this: since the PC100A has 20 columns, the command '2nd Op 07' will print an* in the column specified by the display. So if you've produced a result which is a percentage, say 60%, you divide it by 5 (to scale it) giving 12 and then '2nd Op 07' will print an * in the right column (the 12th, in this example).

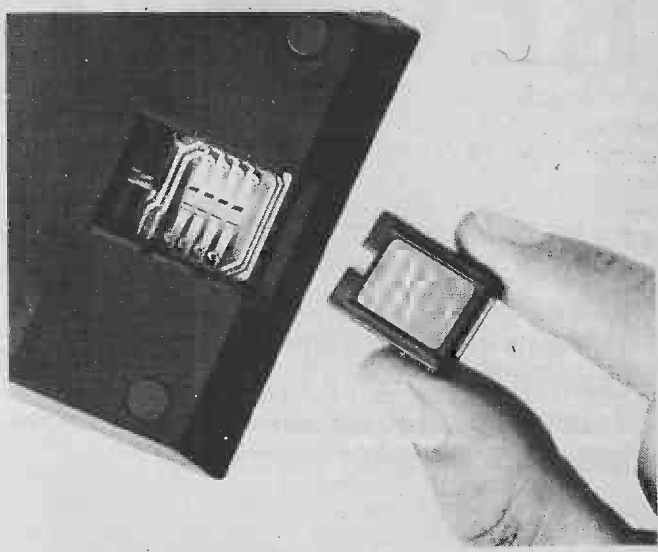


The printer even plots curves:

The printer can also be used to produce a listing of the labels in your program, a listing of the program itself, results (obviously), and, in the trace mode, all intermediate results and the instructions that generate them as a program executes.

Functions

From the technological advances of the P59, we move on now to the design of the machine, the way it operates, and its ease of use — all functions not of the technology, but of the time, effort and ingenuity/insight of the design team.



A 4k interchangeable programme just plugs in:

The appearance and construction of the P59 are pretty well standard, as you can see from the photographs. The keys have a good 'tactile feedback' feel, and are spaced at what is probably the minimum spacing for convenient, fast and accurate operation. This brings me to the only bugbear I found with the calculator — the visibility angle of the display. A PPC, by the nature of the beast, spends a lot of its time on a desk-top, but I discovered that working with a notepad on the desk in front of me and the P59 to the right of that (say, 7" from dead centre), I had to constantly lean over to read the display. Now that's bad — are you listening, TI? Mind you, with the TI59, I could learn to live with it!

As a manual calculator, the TI59 performs very nicely indeed, although the keyboard is perhaps a little crowded for occasional heavy sessions of keybashing; but if you use it a lot, you'll get to know it like the back of your hand and if you use it a little, the busyness won't bother you. I've experienced no difficulties in finding my way around the keyboard, but some colour-coding might have helped.

The TI59 uses TI's 'Algebraic Operating System which makes use of parentheses to over-ride the rules of algebraic hierarchy, and enables you to enter calculations as they are written.

The TI59 sports a tremendous array of functions, including all the usual trig, exponential and scientific functions. In addition, there are also two-variable mean and standard deviation, and although there is no sign of it on the keyboard, the statistical capability is further extended by functions accessed by the key sequence '2nd Op nn', where nn is a two-digit code assigned to each function. Other special operations include the print functions, library program downloading the signum function, memory partitioning, error flagging and a set of operations which can increment or decrement data registers.

While this scheme is slightly awkward to use manually, it does give an additional 40 infrequently used functions without cluttering the keyboard. And of course, most of these functions will be used almost exclusively from programs, so their ease of use is not very important. A list of special operations on the back of the calculator would have been handy, though.

Programming

As a programmable calculator, the TI59 performs extremely well. Program entry is extremely easy, and simple programs can be made up as they are entered. For longer programs, it is, of course, advisable to at least sketch out a program on paper before commencing entry.

Programs consist basically of the same set of keystrokes as you would use to solve the problem manually. However, in order to let a program run without the need for human intervention, PPCs have a number of instructions not found on conventional calculators, such as go to (GTO), label (Lbl), and conditional branches (x=t, x≥t, etc.). These instructions are used to structure the program and transfer control between sections.

The TI59 allows the use of 72 labels to identify program sections: these are the usual 'Lbl A,' 'Lbl 2nd A' type as well as others created using virtually any other key as a label, e.g. 'Lbl CLR,' 'Lbl × .' This permits the creation of extremely large programs in sections, each with a specific function.

There are four different tests which can be made in order to decide program branching (x=t, x#t, x≥t, x<t), which are fairly standard on PPCs. In addition, a Decrement and Skip on Zero (DSZ) instructions can be implemented on registers 0.9 to control program looping, as well as the inverse function, Decrement and Skip on Non-Zero.

The power of most memory referencing instructions can be multiplied by the use of Indirect addressing. For instance, it is possible to branch indirectly, to store and recall data indirectly, to call Library Module programs indirectly, to set flags indirectly, all manner of tricks. A good example is the instruction 'If flg Ind 02 Ind 22', which will recall register 2, and on finding the value 5 there will test flag 5. If that flag is set, it will recall register 22, giving the value 64 and will then jump to step 64. If flag 5 is not set, the program will continue normally. As you can see, instructions of this type pack real programming power, but only 'STO Ind' and 'RCL Ind' are used often.

Programs can be written as subroutines, so that they can be called by other programs, simply by avoiding the use of = (which completes all pending operations) and terminating the program with a subroutine return, 'INV SBR'. If this technique is used, you can have up to six levels of subroutines, which is probably enough to process three-dimensional arrays in quite complex fashions. (I haven't tried it yet though!)

Editing a program is very easy, as you can over-write, insert or delete steps and can single-step, backstep or jump about in your examination of the program. If you use the PC100A printer, then its trace mode will let you see what is happening as each instruction is executed, as well as providing complete program listings (it can't be easy to write down 960 steps!).

The Card Reader

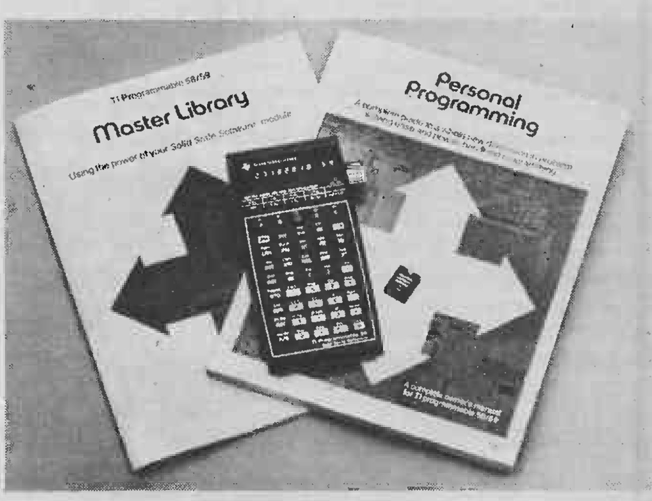
Since there are 960 steps of program memory in total (regardless of whether they contain program or data) it is just not possible to put the whole memory onto one magnetic card. To get round this, the memory is divided into four banks, each of which may be separately written onto one side of a magnetic card. The bank number should be in the display, and the key sequence '2nd Write' will then record that bank onto a card. Each card has two sides, consequently two cards are required to store the whole memory.

It the bank number in the display is negative, when the program is subsequently reloaded, it will be found impossible to list it, or to enter the 'learn' mode to examine or modify it. This provides a means of protecting software from accidental (or deliberate) modification, and ensures security of confidential data.

Cards can be read under program control, enabling large amounts of data to be entered for processing.

Documentation

The most incredible calculator ever devised would be of dubious value without the knowledge of how to use it, which is the result of experience and a long session with the owner's manual. The P59 manual is called 'Personal Programming' and is an A4 format book almost ¾'' thick. This provides plenty of examples to explain both the operation of the various calculator functions, and the rudiments of programming.



The TI59 comes with very good instructions and a manual with details of the programs in the software module.

With a PPC of this complexity, there is just no way you can sit down and start writing programs — even display control takes two pages of explanation. The only way to do it is to sit down with the manual and start at the beginning, working through every example. Programming is a skill you learn by doing, not by reading, and 'Personal Programming' is well organised for this. In short, the manual doesn't let the machine down.

Also supplied is a programming pad, and a guide to the programs in the Master Library Module. This guides the user through the keystroke sequences needed to enter data and run the programs, as well as explaining the program operation and providing necessary information on registers used, parentheses levels etc. Again familiarity breeds ease of use: you have to sit down and play with the machine to learn how to really use it.

Summing Up

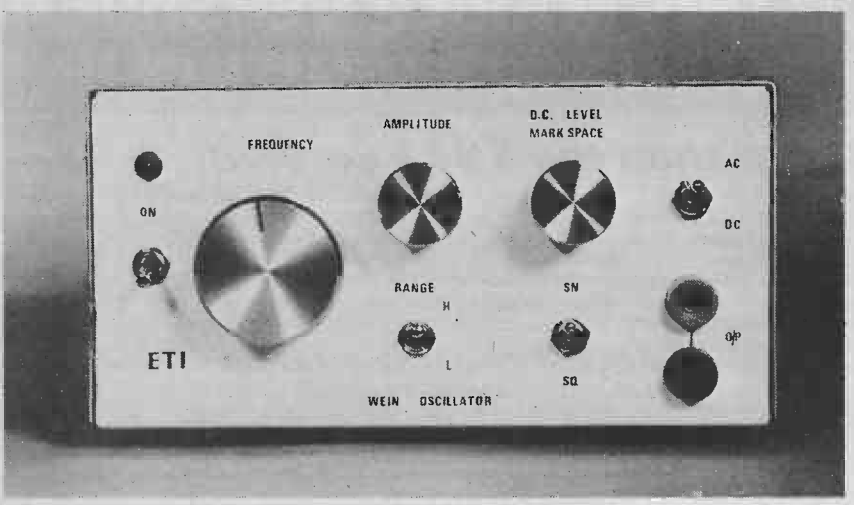
The Programmable 59 incorporates several major advances over previous PPCs, specifically in terms of memory. The basic calculator has a more than adequate range of functions, but the addition of the 'Solid State Software' modules converts it into a general- or special-purpose calculator of extraordinary power.

Probably the greatest compliment I could pay the P59 is to say that as a long time HP and RPN user, I would never have contemplated any other kind of calculator. I'll probably still use my HPs (I don't need another calculator), but if I was a first time PPC buyer, the TI Programmable 59 would be top of my list.

Both calculators are available from Texas Instruments retailers. The Programmable 58 retails for around £80 inc. tax and the TI59 for £210. These are recommended retail prices — discount prices may be considerably lower. The PC100A is yours for only £175 and extra Library Modules are £25 each. ETI

WIDE RANGE

A source of sine and squ... of equipment to have aro... team design



A LOW COST (cheap is what birdies do) source of sine and/or square waves is an invaluable aid to many an electronic hobbyist. So often when testing anything from grannies old valve job to brother's multi-kilo watt stage warmer a source, any source, of sound is useful — well perhaps using Radio 1 would show a certain desperation.

Crest Of A Wave

Our design provides an instrument that, in two ranges, covers the audio spectrum supplying both sine and square waves. The amplitude of the output is continuously variable and can be AC or DC coupled. When generating sine waves in the DEC mode, the DC level of the output can be varied. This latter control alters the mark/space when in a square wave mode.

In order to keep the cost down some compromises have had to be made. The major of these being that the amplitude of the waveform does not remain constant as the generator's frequency is varied over its range. This effect is caused by mismatch in the dual ganged pot that alters the frequency of oscillation, and the simple nature of the amplitude control network.

The oscillator should, nevertheless, prove a valuable addition to many a test bench and, being battery powered, can be used anywhere.

Construction

If the overlay is followed carefully the on board components should present no construction problems. Take care that the ICs are fitted the right way round.

There is a considerable amount of wiring between the PCB and the front panel. Follow the overlay, in conjunction with the circuit diagram, carefully and everything should be OK. ETI

BUYLINES

The CA 3019 diode array is an RCA device that, although not seen all that much, should be available from most of the large mail order semiconductor suppliers.

The other components should all be familiar to you and readily available.

HO

To explain circuit action, we must, as is often the case, assume that the circuit is operating and that we have a fixed amplitude sine wave at the output of the op-amp. The ratio of this output fed back to the non-inverting terminal of the op-amp is given by the ratio:

$$\frac{Z_2}{Z_1+Z_2}$$

$$\text{where } Z_1 = R + \frac{1}{j\omega C}$$

$$Z_2 = \frac{R}{1 + j\omega RC}$$

The ratio may thus be expressed as

$$\frac{R}{3R + j(\omega R^2C - 1/\omega C)}$$

As the op-amp will maintain zero volts between its input terminals there will be no phase difference between the op-amp's output and the divided down feedback signal, ie the complex part of the above expression must be zero at the frequency of oscillation.

This means that

$$\omega R^2C - \frac{1}{\omega C} = ?$$

$$\text{or } \omega = \frac{1}{RC} \quad \text{therefore } f = \frac{1}{2\pi RC}$$

Note also that the attenuation factor of the network — given by the real part — is equal to

$$\frac{R}{3R + j(\omega R^2C - 1/\omega C)}$$

As long as the gain of the amplifier is about three the oscillator will function satisfactorily but if the gain varies from this value performance will be degraded. The gain control network formed from the diode bridge and series diodes keeps the gain at the required value.

The oscillator's output is rectified by the bridge and fed to the diode and zener. As the oscillations increase in amplitude the diodes begin to conduct, lowering their impedance. This tends to increase the amount of negative feedback thus reducing the op-amp's output this stabilising the systems gain.

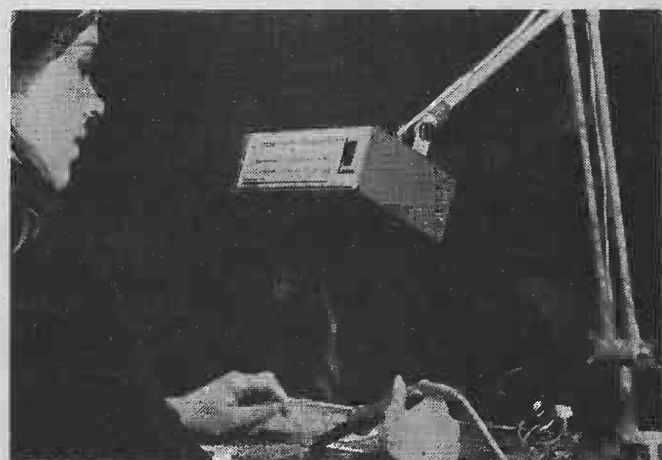
. . news digest . .

terrorists smell?

They certainly reek of explosive. Pye Dynamics hope business will go like a bomb with their PD3 explosives detector. The hand held unit weighs in at 750 gm and works by ionising a continuous stream of air with a high voltage. The continuous air stream is produced by a miniature electric fan, and heavy ions (ie: explosives) are separated from the light air ions. Pye claim that explosive concentrations as low as one part per billion (1 per 10^9) can be detected with the unit.

In use the device has only three controls — trigger on/off switch, volume control and set zero control (for the audible warning tone). It comes complete with all the usual sorts of accessories (briefcase, charger etc.), they even give a sample of explosive! At about £1 500 it's probably going to start appearing in a lot of places in the near future. Pye Dynamics Ltd., 459 Park Avenue, Bushey, Watford, Herts.

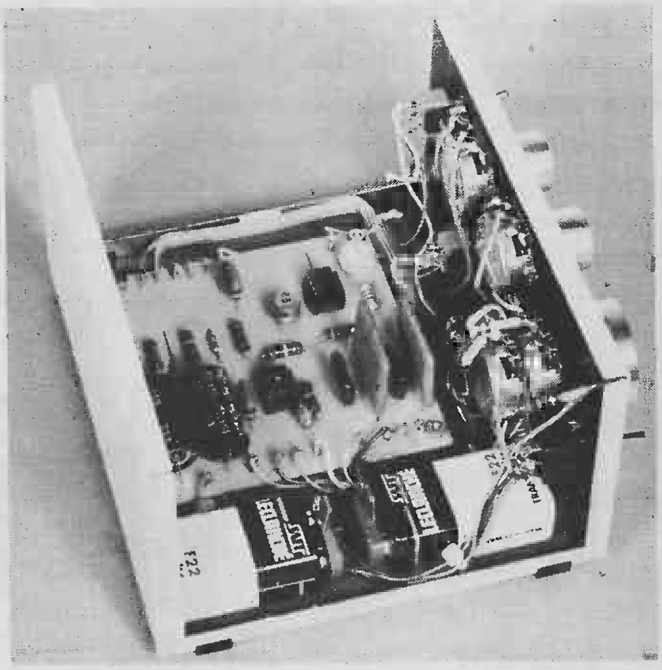
electronic deodour



One of the joys of electronics is the acrid smell from the flux . . . well some people get their kicks in strange ways. Pleasant though it may be in small quantities, the smell can really get up your nose. Vero Systems have introduced an interesting device to help take the nasty bits out of the atmosphere. Called the Komax 77 Soldering Steam Absorber (a bit of a nosefull) the unit contains a replaceable, chemically impregnated, plastic filter — fumes are sucked through it at a rate of 46 litres per second. Vero Systems (Electronic) Limited, 362 Spring Road, Sholing, Southampton SO9 5QJ.

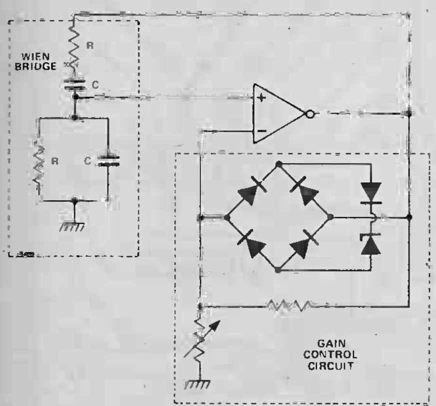
E OSCILLATOR

...ves is always a useful piece ...so sign on with this ETI project



T WORKS

The full circuit diagram of the oscillator is shown in Fig 1. The resistors in the feedback network have been replaced by a ganged potentiometer to allow the frequency of oscillation to be varied. The value of the capacitor C in the bridge can also be altered, by parallel connection of another capacitor, making the range covered by the circuit encompass the audio spectrum.



Block diagram of the oscillator elements.

RV2 sets the gain if the amplifier IC2 whose gain is then dynamically modified by the matched diodes of IC1.

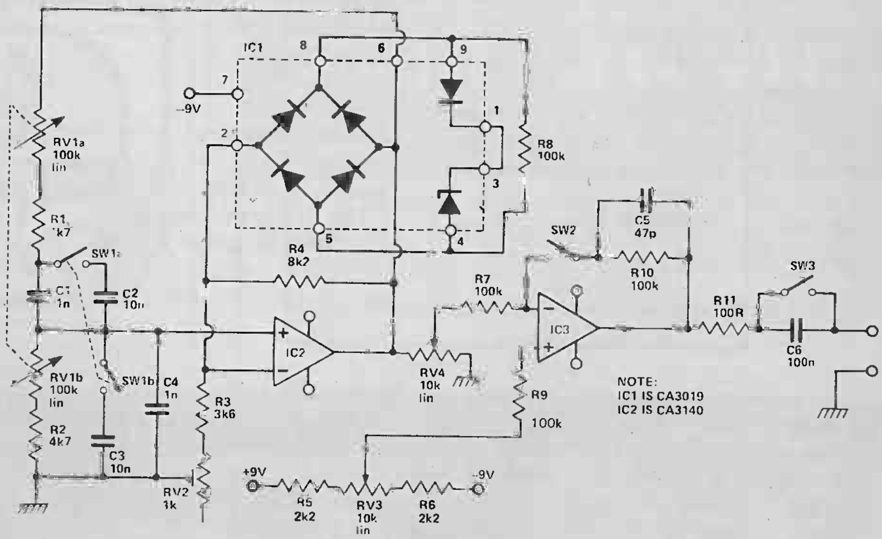
The output of the oscillator is fed, via level control RV4, to a unity gain output buffer. Op-amp action will ensure that the DC level at the output of IC3 will equal that set at the wiper of RV3.

By removing the feedback resistor, R10, by opening SW2, IC3 will act as a comparator converting the input sine wave to a square wave.

In this mode the potentiometer RV3 acts as a mark space control by varying the reference voltage applied to the comparator.

The buffered signal is taken via R11, to limit any short circuit current, directly to the output terminal if SW3 is closed (DC coupled) and via capacitor C6 (AC coupled) with SW3 open.

Power to the unit is derived from batteries. C7-9 are decoupling.



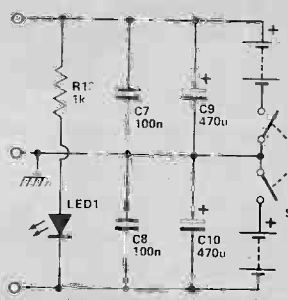
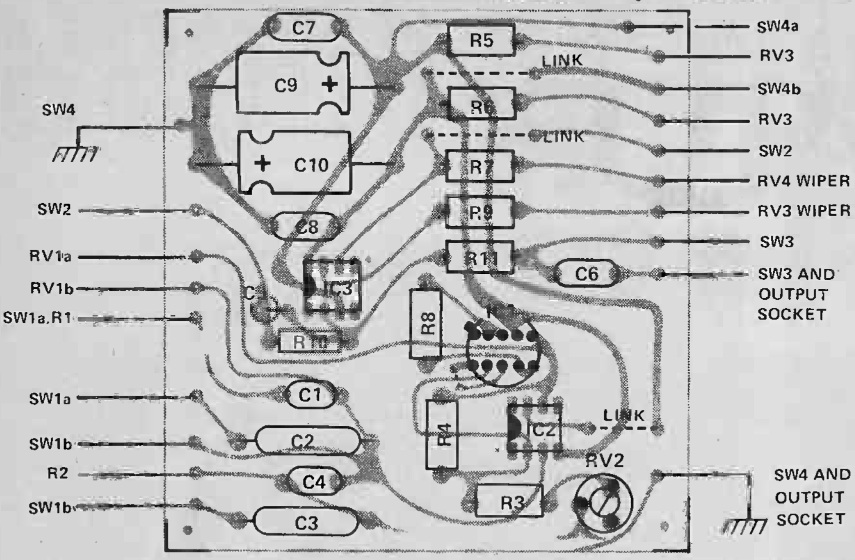


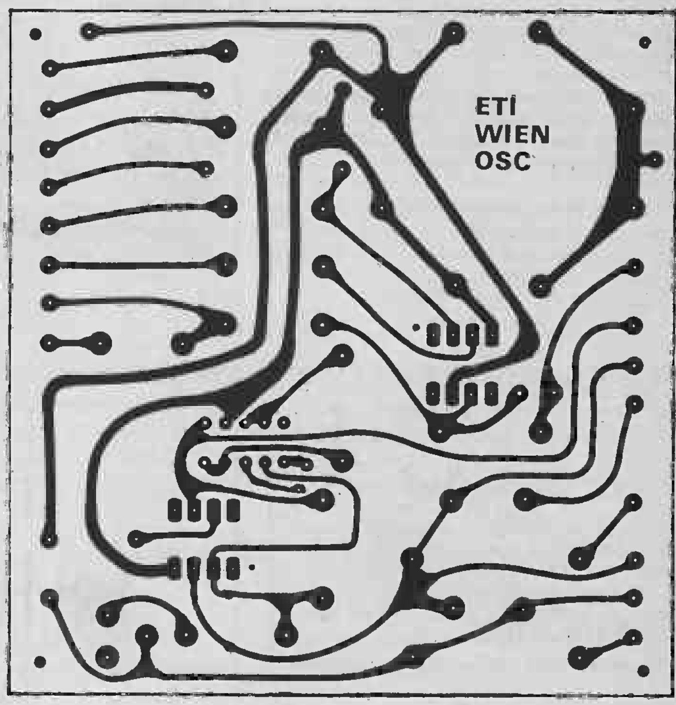
Fig. 1. Full circuit diagram for the Wien bridge configuration, wide-range oscillator. The power supply and output stages are also shown on the diagram. Components within the dotted line are part of ICI.

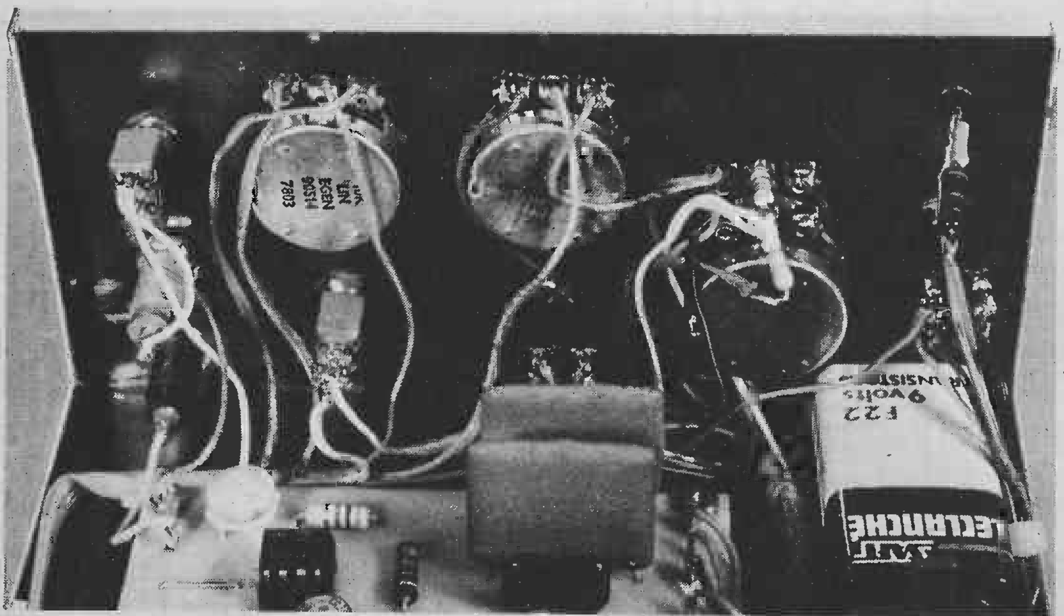


Left: Fig. 2. The component overlay for the Wien bridge oscillator PCB. The power supply components too are included on the board. Check the IC orientation carefully before soldering in.
Below: The foil pattern for the oscillator PCB, shown full size at 86mm x 88mm.

PARTS LIST

RESISTORS (all ¼w 5%)	
R1, 2	4k7
R3	3k6
R4	8k2
R5, 6	2k2
R7, 8, 9, 10	100k
R11	100R
R12	1k
CAPACITORS	
C1, 4	1n Polystyrene
C2, 3	10n Polystyrene
C5	47p Polystyrene
C6, 7, 8	100n Polyester
C9, 10	470u 16V Electrolytic
SEMI-CONDUCTORS	
IC1	CA 3019
IC2, 3	CA 3140
LED1	0.2" type
POTENTIOMETERS	
RV1	100k lin dual guage
RV2	1k preset
RV3, 4	10k lin
SWITCHES	
SW1, 4	DPST
SW2, 3	SPST
MISCELLANEOUS	
PCB as pattern, case to suit, 2x PP3 plus connectors.	





A view of the back of the front panel of the completed oscillator project. This gives a good idea of the wiring of these to the PCB components and positioning of the battery within the box.

SYSTEM 68 UPDATE

Many readers have now got System 68 working well and are very happy with it. Many have pointed out errors and ambiguities. We list here all known errors even those of a very minor nature and those that have appeared previously.

VDU Board A.

The system 68 VDU board A contains the main timing counters for the whole of the VDU. Most of the System 68 problems have been found to be on this board. The circuit diagram and the PCB layout contain several errors which can be very difficult to diagnose and correct without the right equipment. Most of these have been printed in later issues as corrections but we will reprint them here together with some other points which have come to our attention.

a. The PCB layout shown omits the ground connection to IC2 and IC4.

b. The LINK from IC10pin 1 to IC9 pin 6 is omitted on the component overlay.

c. On the circuit diagram the Line Sync output (LS) is shown as being connected to IC5c this should be connected to IC5d. The PCB is correct.

d. The horizontal Display Enable (DISEN) is shown connected to IC7d, this should be connected to IC7c. If this is not done then the first character on each line will be duplicated and the last character lost.

e. The output from the master oscillator (IC1 gates a and b) is sometimes insufficient to drive the two TTL loads connected to it (IC2, IC14). This manifests itself on the screen as a series of lines rather than characters, the lines often look like 'hyphen' or 'underscore' characters. To overcome this it is necessary to buffer the output through the spare gate of IC12. First connect IC12 pin 1 to IC12 pin 14 to ensure that it is always at logic '1'. Break the copper track joining IC1 pin 8 and IC2 pin 14 where it passes RV1, join IC1 pin 8 to IC12 pin 2, join IC2 pin 14 to IC12 pin 3. Thus IC1 now only needs to drive the single TTL load presented by IC12 pin 2.

f. One of the main problems with the VDU does not become apparent until the CPU card has been installed and tested. Ghost characters will appear at various places across the line, these are usually at 8, 16 or 32 character locations from the beginning of the line, the ghost character will overwrite any other character at that point. The problem occurs if the CD4040 character counter is not operating fast enough. In our original prototype we used a National Semiconductor device which we found out later is significantly faster than its competitors. Even so, a lot of NS CD4040 devices are not fast enough. The best answer is to replace this CMOS device with the equivalent function in TTL which will thus operate at much faster speeds. Two four-bit binary counters such as 7493s can be used or there is a single 8 bit binary TTL counter called the 74393 which has been announced in the past few months. Neither option is pin compatible with the CD4040 and so a slight external bodge is required.

g. Of the three extra character options available (invert, grey or flash) only one can be used at a time, not two as stated. This allows full 7 bit ASCII to be stored in the VDU RAM rather than the 6 bit which appears to be sufficient. The 8th data bit can thus be used to enable one of the extra character options to denote a cursor or special message area.

h. Some printing errors make the checkout a bit difficult.
June 77 p35. IC2 pin 1 9.375MHz should be IC2 pin 14.
June 77 p35. The 'Greater Than' and 'Less Than' signs were missed out near the end of column 1. The 'Greater Than' sign will appear before the 'Question Mark' and the 'Less Than' sign will appear before the 'Equals' sign.

i. TTL and CMOS. It was suggested that the 74C devices on the board could be replaced with 74LS devices, in fact the majority of the 74 and 74C devices could be replaced with 74LS devices. (NB there is no 74LS75).

VDU Board B.

a. The PCB layout does not have a connection under IC28 which should connect IC24 pin 16 to IC28 pin 7.

b. The suggestion in the text that the DM8679 could be used in tandem with the DM8678 to give upper and lower case characters is unfortunately false. National Semiconductors have no plans to produce the 8679 shown in their data book but have produced an 8678 CAH which gives lower case characters. The 8678 CAH can be used with the 8678 CAB but an additional latch (eg 7474) is required to latch the extra data bit and drive the two chip enables.

c. The only other problems with VDU board B appear to have been component shortages. The 74C157 can be replaced with a 74LS157 but not 74157. There is no equivalent to the DM81LS95.

The above errors account for most of the problems with the System 68 VDU, Tamtronik Ltd offer PCBs with most of the track errors corrected.

CPU Board

a. Links D0-D7 are shown incorrectly marked on the component overlay. For D0 read D7, for D1 read D6, for D2 read D5, etc. Note that the error is at both ends of the links, many people misread the correction and changed the links at one end only thus getting into even more trouble. It is not necessary to rewire the links at all, simply relabel the ends.

b. The clock phases from the MPU chip are reversed and should have a 22R resistor in series with each line. The simplest way to correct this is to break the track between IC1 pin 3 and IC3 pin 9, also that between IC1 pin 37 and IC3 pin 7. Now using a 22R resistor link IC1 pin 3 to IC3 pin 7, using another 22R resistor link IC1 pin 37 to IC3 pin 9.

c. Link IC2 pin 11 to IC2 pin 5 and 13.

d. There is a small amount of track missing near IC8 pins 9 and 10, this track should link IC8 pin 15 to the point marked D7.

e. The NRDS and NWDS strobes shown at IC5b are incorrect and the data buffers IC10 and 11 will not operate correctly as shown. The following modification is required.
Connect IC4 pin 12 to IC3 pin 13, also IC4 pin 7 to IC3 pin 12.
Disconnect all connections to IC5 pins 3, 4, 5, 6, 7, and the existing NRDS and NWDS lines to ICs 10 and 11 and ICs 6 and 7.
Connect IC3 pin 11 to IC5 pin 3 (seperate pins 2 and 3)
IC5 now decodes as follows —

pin 1 Enable	pin 2 A	pin 3 B	pin 4 O/P 0	pin 5 P/P 1	pin 6 O/P 2	pin 7 O/P.3
VMA.Ø2	R/W	W3d	NWDS	NRDS	INWDS	INRDS
1	x	x	1	1	1	1
0	0	0	0	1	1	1
0	1	0	1	0	1	1
0	0	1	1	1	0	1
0	1	1	1	1	1	0

IC3d output (pin 11) will be low whenever both the RAM enable and the PROM enables are high and thus an enable of an off-board device is required, if an on-board enable is required then this output will be high.

The R/W signal from the MPU is applied to IC5 inpu B and with IC5 enabled by VMA.Ø2 the outputs on pins 4-7 will be as follows —

IC5 pin 4	Low when an external WRITE is required (NMDS).
IC5 pin 5	Low when an external READ is required (NRDS).
IC5 pin 6	Low when and internal WRITE is required to RAM (INWDS).
IC5 pin 7	Low when an internal READ is required.

Thus the output on pin 7 is not needed in this system, pin 6 will drive the R/W inputs of the on-board RAMs (ICs 6, 7 pins 14), 4 and 5 will be output on the 31 way connector and also drive the direction pins of the buffers (ICs 10, 11).

TTY Board

There appears to be a problem which can occur when trying to write data to the UART. The UART spec requires that the data be stable for the complete duration of the DS pulse. This pulse derives from a decode of the address lines and VMA. Ø2 in ICs 1 and 4 (or 7) and should thus occur during NWDS and the data should be static during this time. This problem does not seem to occur on our prototype where we have 74LS42s in place of 74C42s. The problem has only arisen with one or two readers and it may be that either—

a. the CMOS devices are delaying the DS strobe, or

b. the decode on the CPU card described above has not been done.

To date these are the only errors we have found in System 68, apart from these most problems appear to have been caused by insufficient checking of the completed PCBs to ensure that all through hole links are OK and that there are no short circuits or track breaks.

ETIBUG2

An address and data is missing from the listing of ETIBUG2, at address ED8A the data is E5, this is the offset for the jump instruction.

ETI

data sheet

L911 MICROPOWER COMPARATOR/DETECTOR — SILICONIX

The L911 is a monolithic bipolar-PMOS integrated circuit intended to meet the system requirements of a micropower detector alarm or ON/OFF control system. The circuit will operate from a supply as low as 6 V and is tested for 15V conditions. The alarm output can source sufficient current to activate an NPN or SCR buzzer alarm driver and can also interface to TTL or CMOS Logic. The L911 can easily operate with a standby supply current of less than 10µ A from a 9 V battery, and less than 17 µ A and 15 V. The high input impedance MOSFET comparator easily interfaces to high impedance sensor devices.

Low Battery/Threshold Detector (Pin 1). Pulses the output when the voltage at this input falls below the internal reference of =2V2. An internal comparator turns on the Low Battery Timing Oscillator, which in turn produces 'trouble signal' pulses at the output.

This Low Battery/Threshold detector can be used for any application requiring a pulsed output alarm. To defeat action of this input, tie it to the positive supply.

—Input (Pin 4) — This input is connected to the negative (inverting) input of a MOSFET-input comparator. The output (pin 11) is low when this pin is more positive than the +input, (pin 2 or 6). When voltage to this pnput falls below the +input, the output (pin 11) goes high, which can be used to trigger an alarm. External connections can force a system to be either latching or non-latching. This input is protected against static electricity by a zener diode.

+Input (Pins 2 and 6) — This MOSFET input, which is also zener protected, is connected to the positive (non-inverting) input of the input comparator. Pins 2 and 6 are internally common to allow flexibility in RCB layout. The common mode range of the input comparator is from ground to 4V below the positive supply.

Bias (Pin 7) — Current flowing into this input determines the standby current drain of the L911 since the internal current sources are multiples of this current. Normally 8MO is connected between pin 7 and V+ to provide approximately 4µ 5 A of standby current for V+ = 9 V, but any value between 0M5 and 10M will work.

Noise Suppression (Pin 8) — Noise suppression is connected internally to a high impedance point in the comparator. An optional capacitor connected between this pin and Ground (pin 9) effectively gives the system hysteresis by incorporating an input time delay. This capacitor forms a low pass filter, preventing false triggering in RF fields by reducing input noise sensitivity. A 4µ 7 capacitor acts as 2 second delay (0.5 Hz low pass filter). Under normal operating conditions, however, this capacitor is not needed. The voltage at pin 8 is normally 50 mV above ground; the alarm triggers when this voltage reaches a diode drop (0.55 V) above ground. Therefore, the output ON condition (due to a low inverting input) can be inhibited by keeping pin 8 < 200 mV above ground. Any switches or circuits connected to this pin should have leakages of less than 100nA.

Low Battery Timing (Pin 10) — This pin allows timing of the alarm oscillator when the system goes into the 'Low Battery Alarm' condition. In normal standby or output alarm conditions, this pin is open. In the Low Battery Alarm condition, this pin starts sourcing current equal to 0.4 x I_{SET}, to begin the Low Battery Timing Period. When a capacitor from Low Battery Timing (pin 10) to Ground is charged to approximately 2V6, the alarm is pulsed ON and the capacitor is discharged to approximately 0V7. The charging current and the value of C determines the period of the warning (making C_T equal to 4 7 will give a 7.5 ms alarm ON pulse every 30 seconds).

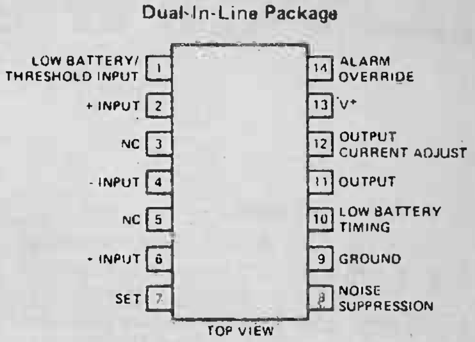
- **Minimizes System Power Requirements**
  - Supply Current Less Than 10µ A
  - Allows Greater Tan 1 Year of 9V Alkaline Battery Life
- **Simple Maintenance of System**
  - Includes Weak Battery Threshold Voltage Detection Circuitry
  - Outputs Trouble Alarm Signal
- **Reduces External Component Requirements**
  - No Buffers Required for High Impedance Transducers (R_{IN} > 1000M)
  - Directly Interfaces to TTL or CMOS Logic
  - Directly Drives Power SCR's Transistors or VMOS
  - Built-In Reverse Battery Protection

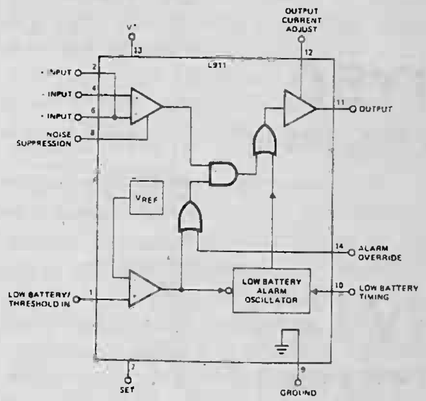
Output (Pin 11) — This output is triggered HIGH by the inverting input (Pin 4) going LOW or by a low battery alarm condition. This alarm output is constantly HIGH for an imput alarm and pulsed HIGH for Low Battery Condition. The output will source at least 0.5 mA of current (with V+ =6 V) to an external driver during the alarm condition. The output normally returns to LOW when the alarm condition clears, but by rearranging external circuitry, the Input Alarm can be made to latch ON even after the input has cleared. The output (pin 11) can be connected to ground to allow logic to be driven at the output current adjust, pin 12. The output must be kept below 5 V to prevent breakdown to the chip substrate. There is an internal shunt resistor to ground of typically 20K to 100K.

Output Current Adjust (Pin 12) — Pulling this pin up to V+ through a resistor increases the output source current capability from its minimum 0.5 mA to a maximum of 30 mA. For example, a 2KO pullup resistor gives an output current of 9 mA for 9 V. The output current adjust can safely be pulled to ground or to V+ if the 30 mA maximum current limit is observed.

This pin is connected to the positive supply from 6 to 15 V. The low standby current allows use of a 9 V alkaline transistor radio battery.

Dual-In-Line Package



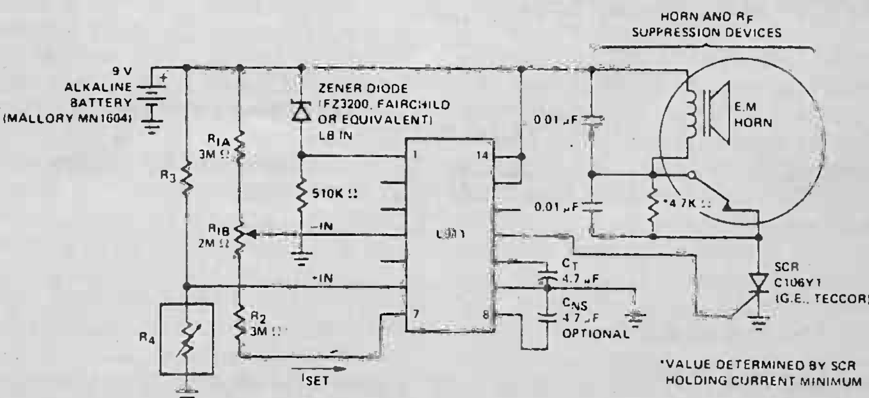


Reverse battery protecton is built in and no damage will result from reverse battery voltage being applied.

Alarm Override (Pin 14) — This pin allows the "Low Battery Alarm" condition to override a constant 'input condition' alarm when Alarm Override is connected to Ground as shown in Figure 1, when the Alarm Override is connected to V+, and there is a Low Battery condition during an input alarm, the output will continue to be ON constantly until the condition clears or the battery dies.

For full data sheets and price/availability information contact **Siliconix Limited, Brook House, Northbrook Street, Newbury, Berks RG13 1AH**

Fig 1: A battery Powered Temperature Alarm (Non-Latching). Sounds a Buzzer Whenever the Temperature Rises Above A Preset Level

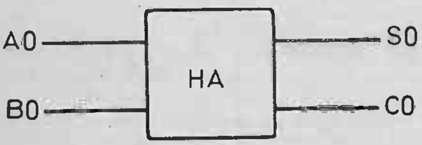


DIGITAL ELECTRONICS

Arithmetic Units

SO FAR, THE WORK which we have carried out on the blob-board has covered gating, flip-flops, counter and display stages and the use of a register. Within the limitations of 8 IC's, we cannot, of course, hope to cover every possible principle of digital electronics, and the IC's which were selected for the board were designed to reflect the applications of digital electronics most often seen in published circuits.

The two important topics of arithmetic and memory have not been specifically mentioned, partly because small projects seldom need arithmetic or memory (and large projects can make use of the more flexible facilities of a microprocesser, particularly if this incorporates a memory) and partly because the building blocks of arithmetic units (gates) and some types of memory (flip-flop) have been covered.



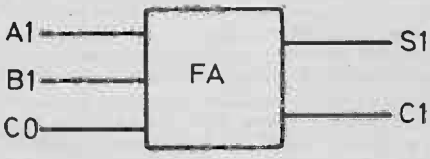
A0	B0	S0	C0
0	0	0	0
0	1	1	0
1	0	1	0
1	1	0	1

Fig 1. Half-adder symbol and truth table.

Nevertheless, in this last part we shall look at some of the circuitry we have not covered previously, and also at some systems which can be tried out in the board. In addition, it is useful to note that the board can now act as a very useful intermediate unit for experimental work on more advanced systems, since it can provide up to six clock oscillators, four flip-flops, four NAND gates, one register, and a complete circuit-and-display for one set of BCD digits.

Adding:

Binary addition can be **serial** or **parallel,** of which parallel addition is more common. The **half adder** has the truth table in Fig. 1 and is used for the least significant digits of two numbers. Its output will be the sum (the digit which will appear in the



A1	B1	C0	S1	C1
0	0	0	0	0
1	0	0	1	0
0	1	0	1	0
1	1	0	0	1
0	0	1	1	0
1	0	1	0	1
0	1	1	0	1
1	1	1	1	1

Fig. 2. Full adder symbol and truth table.

final figure) and the carry which will be added to the next significant figure. The **full adder** circuit is used for all the next stages of the adder unit and has three inputs and two outputs; its truth table is shown in Fig. 2. The inputs to the full adder are the two digits A_1, B_1, and the carry C_0 from the previous half-adder stage. The outputs once again are the sume and another carry C_1 which is taken to the next stage. The total number of adding stages which will be needed must equal at least the total number of binary digits in the sum of the numbers.

Half-adders and full adders can be made up using gates (Fig. 3) but once the principles have been checked it is easier to use IC's made for the job. The 7482 is a two bit full adder, whose internal circuitry, with truth tables, is shown in Fig. 4. From the diagram, we can see that the inputs are Co from the previous half-adder (which would be either an integrated full adder with no carry input, or made up from gates) and the second significant digits A_1 and B_1. The sum of this stage is obtained at the terminal marked S1, and the carry is internally connected into the second stage of the adder, whose inputs are B_2 and A_2 with outputs sum S2 and carry C_2. The next step up is the 7483, which is a four-bit adder and any requirements greater than this is dealt with by arithmetic units of much greater complexity.

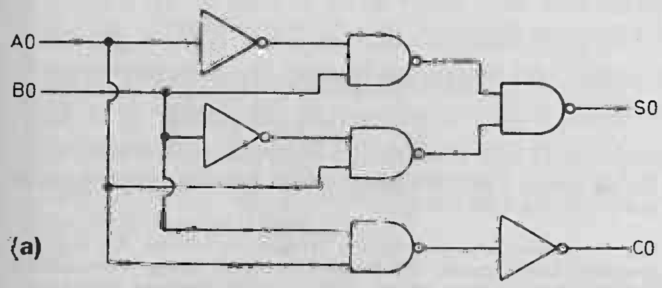
In general, if more than a simple addition is needed, it is more economic to use LSI arithmetic units.

Memories

Memory units which are used in digital work come in several varieties. One class of memory is the volatile memory, based on flip-flops, which is cleared wherever power is switched off; this type could be used in pocket calculators. Non-volatile memories are the types using pre-set registers (such as read-only memories or ROMs) or which use magnetic tapes or cores or other types of storage which are not erased when power is switched off. A simple type of volatile memory is a SISO shift register with its output connected back to its input so that the information is read back in after one complete set of clock pulses; this type of memory can only deliver its contents in the order in which they are stored. If the register has parallel outputs with gates, however, it becomes possible to find which digit (0 to 1) is stored in each flip-flop, so that, in the language of computing, random access is possible. This is a simple random access memory (RAM).

At this point it is worth pointing out that most memories in general use permit random access. The type of memories which we refer to as RAM are random access memories

BY EXPERIMENT PART 9



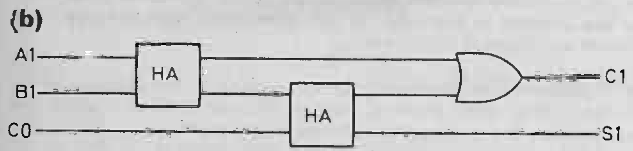
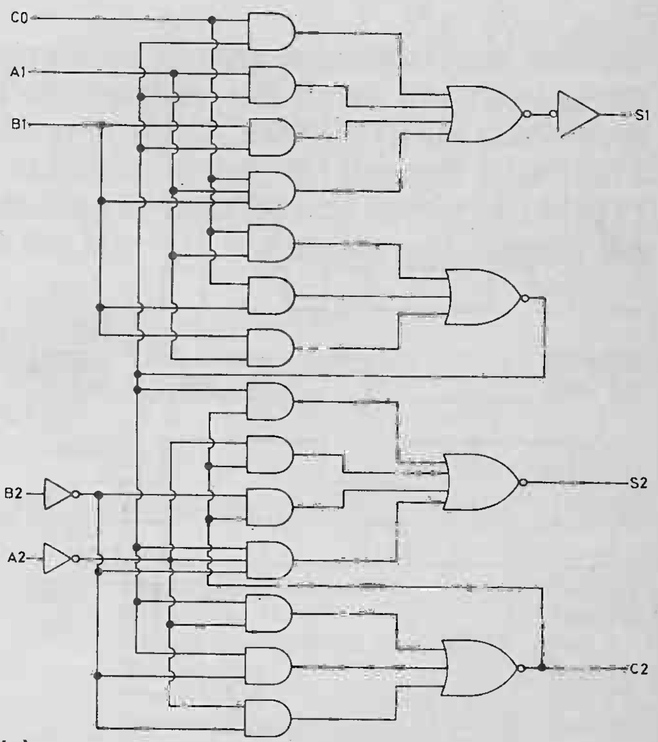


Fig. 3. Above: Adders (a) Half-adder circuit, using NAND-gates and inverters. (b) Full adder, using half adders and OR-gate.



(a)

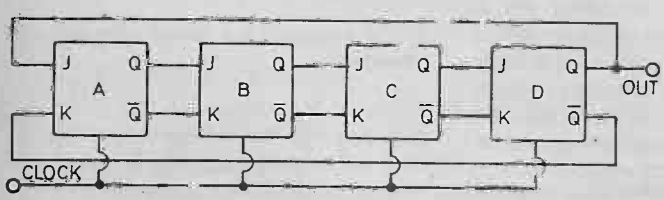
Fig. 4. Right: (a) Schematic of 7482 two-bit full adder. Note again the advantages of medium scale integration. (b) Truth table.

(b)

TRUTH TABLE

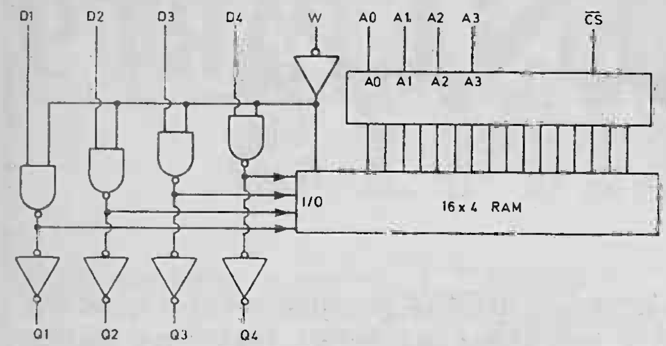
INPUTS				OUTPUTS					
				C0 = 0			C0 = 1		
A1	B1	A2	B2	S1	S2	C2	S1	S2	C2
0	0	0	0	0	0	0	1	0	0
1	0	0	0	1	0	0	0	1	0
0	1	0	0	1	0	0	0	1	0
1	1	0	0	0	1	0	1	1	0
0	0	1	0	0	1	0	1	1	0
1	0	1	0	0	1	0	1	1	0
0	1	1	0	1	1	0	0	0	1
1	1	1	0	0	0	1	1	0	1
0	0	0	1	0	1	0	1	1	0
1	0	0	1	1	1	0	0	0	1
0	1	0	1	1	1	0	0	0	1
1	1	0	1	0	0	1	1	0	1
0	0	1	1	0	0	1	1	0	1
1	0	1	1	1	0	1	0	1	1
0	1	1	1	1	0	1	0	1	1
1	1	1	1	0	1	1	1	1	1

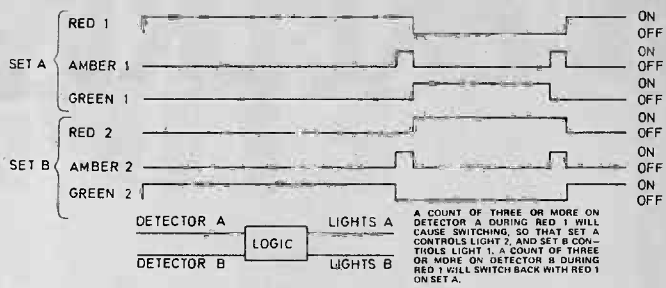
Fig. 5. Below: SISO shift register connected as a memory — the information must be read out in serial form.



which can be written as well as read when suitable inputs are applied.

They should properly be called random access read/write memories. Read only memories are usually also random access, but the information which is stored has been put there either by the manufacturer (in the design stage) or by the user (as with PROM) when the memory is first used. In the older types of PROM, using fusible links, the memory cannot be altered once programmed, except by fusing a few more links. The more modern UV erasable PROM's permit complete erasure and re-programming.





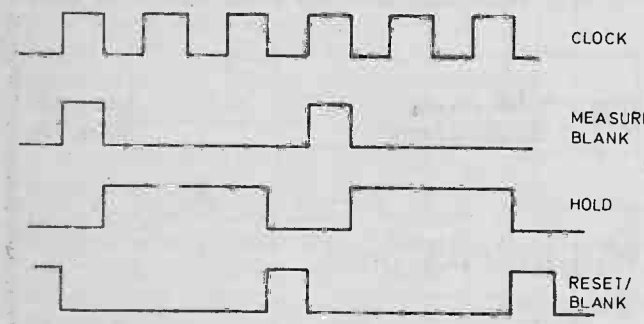


Fig. 6. Above left: 7489 RAM schematic, showing addressing system for 16 4-bit words.

Fig. 7. Below left: Pulses for frequency meter. During the measure/blank cycle, the input frequency being measured is gated to the counter, but the display is blanked out. During the hold cycle, the display is on, showing the count, but the display is blanked out. During the hold cycle, the display is on, showing the count, but the input frequency is gated out, so that the reading is steady. On the reset/blank cycle, the counter is reset and the display is blanked. If the repetition rate is more than 50Hs or so, there is no flicker.

Fig. 9. Above: Priority traffic lights problem. This scheme gives priority (long term period) to the longer line of traffic, as measured by the pulses from the detector pads.

RAM and Address

For either type of memory, the inputs will consist of address lines which locate positions in the memory. We can think of these address lines as grid lines on a map, with each pair of crossing lines locating a point. When a point is addressed by voltages on the lines which 'cross' at the point, then the output will be the digit, 0 or 1, stored at that point.

As an example of addressing, Fig. 6 shows the arrangement of the 7489 RAM which is a 64 bit memory which uses four rows of 16 columns of storage. The rows are addressed by the inputs D_1, D_2, D_3, D_4, so that a four bit word can be read into each of sixteen columns. The columns are addressed by another four-bit word which is decoded (1011=column 11;0110=column 6) by a decoder stage which then drives the column.

To write, a four-bit word is placed on the D inputs, and the write gate is activated, with the appropriate column slected by A_0—A_3. To read, no signal is present on the D lines, and selection of a column places a four-bit word on the output Q1—Q4.

Suggestions for Future Board Work

Figure 7 shows the sequence of pulses which are needed by a frequency meter. The system here is that pulses are counted for one unit, count is held on display, then cleared so that the system can be cleared for another (updating) count. The ICs on the board enable you to try this system out for one digit of counter.

Figure 8 shows the pinout of the 74141 BCD-decimal decoder. This IC, not used on our board, can be connected to the BCD output of the 7490 and will give outputs on ten pins, according to the state of the count. The active state is represented by a **zero** output on a pin, so that a zero output on the '7' pin (pin 10) represents a count of 7, and so on. Using this, could you design a ten-note jingle player?

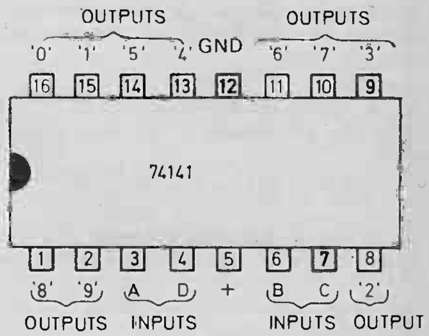


Fig. 8. Pinout of the 74141 BCD-Decimal decoder.

Finally, Fig. 9 shows the operation of priority traffic lights. These lights operate with a longer red phase on one set than on the other, but this can be reversed if more than three vehicles cross a detector strip during the long red period on one set of lights. This scheme needs a clock pulse, counters, register and gates, could you make one? ETI

electronics tomorrow.....

by John Miller-Kirkpatrick

SOFTWARE HURTS

Hardware: The touchable parts of a computer system, ie chips, PCB, VDU, printer, keyboard, case, etc.

Firmware: Variable parts of a computer system, usually in some form of variable programs or switch selectable options. The Firmware is a part of the overall system which assists in the running of the whole system. Examples of Firmware are ROM Monitor programs, Paper or Mag tape monitors, I/O handling routines, etc. Firmware is usually an interface between the Hardware and the Software.

Software: Programs written to use the facilities offered by the Hardware and the Firmware to do the job required by the user.

Take for an example the CBM PET computer or one like it. Its Hardware consists of a case with a keyboard, cassette recorder and VDU. Inside the case is a set of chips, power supply, a scattering of capacitors and resistors, PCB and I/O interfaces. Inside some of the ROM chips is a BASIC interpreter, subroutines for interogating the keyboard, reading or writing the cassette, writing to the VDU, etc. These are the Firmware of the PET, the programs could be changed by replacing the ROMs but then the unit would not be a PET, it would become something else. The Software is the program you write in BASIC to play noughts and crosses or chess or do calculations, etc. This is variable and will change from one user to another. Note that the program to play chess on a PET is defined as Software whereas the program to play chess on a CHESS CHALLENGER is Firmware, if you change the firmware on a Chess Challenger it may become a Checkers (Draughts) Challenger or a Backgammon Challenger, ie you would change the main function of the unit.

It is perfectly possible for Software to become Firmware and in fact this very often happens. With a lot of microcomputers the firmware supplied is in the form of a simple Monitor program which facilitates the writing of programs in machine code and then possibly storing these on paper or magnetic tape. If you write or buy a BASIC to run on that machine then the BASIC is a form of Software. If, however, you always use the machine with BASIC for running games or business programs then the BASIC can be assumed to be a permanent component of the machine and as such is Firmware even though it may be stored in RAM. Another example of Software becoming Firmware is in a development system such as SCRUMPI 3 where the end product is going to be using the same Hardware as the development system. Here the original Firmware is a development Monitor program in PROM, the user uses this to develop his own Software which can then be put into PROM and replace the Monitor. Whilst the Monitor is in the unit then the user's program is Software, when the user's program replaces the Monitor then the SCRUMPI 3 becomes a different product with the user's program as Firmware. The user's program could be a control system, a games system or another form of Monitor.

SOCKET TO ME

Some aspects of Hardware can also be considered as Firmware, take for example a socket provided on the main PCB for expansion by the insertion of an additional PCB or IC. The additional unit plugged into this socket could add to or change the use of the main system, by plugging different units into this socket the use of the main system can be changed from day to day or from one system to another. Is the socket not there for a Firmware feature? Other forms of sockets may enable expansion from the basic system by the simple addition of ICs in the form of extra RAM or PROM. In these cases all of the necessary interfacing and wiring already exists, a simple example is the socket for the ETIBUG2 PROM on the System 68 CPU card — is this feature Hardware, Firmware or Socketware (or Whatware)?

Why then, if Software is an intangible feature, does Software hurt? If you write your own Software you will find that it hurts your brain when it does not do what you expected it to do. You can look forward to many happy hours spent trying to work out why the program insists on overwriting itself or going into an untraceable loop. If you have your Software written for you then it can hurt your pocket, some of the consultants around are more used to writing software for mainframe or mini computers and the cost of having a very simple system written for a micro can be very high. Other consultants specialise in micro programming — probably specifically for a small range of machines. This type of consultant will know his machine and its capabilities and will thus be in a better position to write Software faster and at a lower cost. One of the advantages of using a consultant is that he may well have a set of software similar to your requirements already in existance — all you have to pay for is the modifications and an overhead to cover your share of the cost of the original Software.

Some Software/Firmware recently advertised in an American magazine gives some examples of what is available, in what form and at what price (in USA dollars).

Disk Extended BASIC on disk	$300.00
10 Games in BASIC on CUTS tape	$ 20.00
Multi-Tasking Operating System	$175.00
Extended BASIC (16K) in disk	$ 95.00
in PROM	$800.00

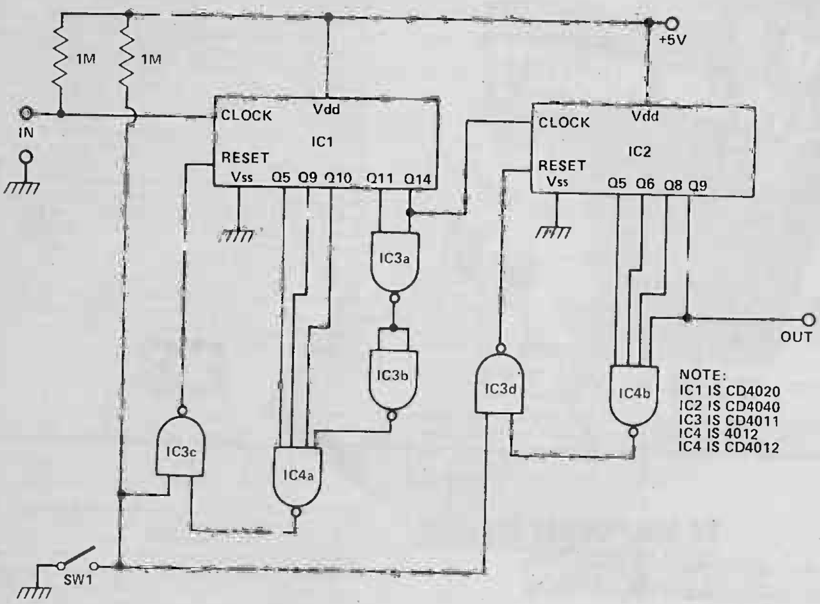
The last example above shows the difference in cost in PROM Software which in theory reflects the cost of the PROM ICs. When comparing costs of this kind remember that the disk version requires 16K of RAM to operate in, a typical US cost for this is $500.00. Software or Firmware in mask programmed ROM is probably going to be cheapest in the long run but this requires high volumes of sales.

Thus Software Hurts your head and/or your pocket, possibly the worst type of Software for pain is the program you use for running your personal budget — it can tell you how long it is going to take you to pay for itself!

ETI

Divide by 4,320,000 Counter

J. Stark



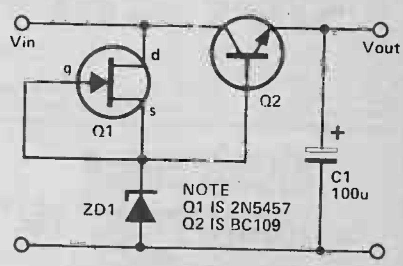
So what is a 4320000 counter good for? Well, 50x60x60x24 = 4320000 so that if you feed in 50 Hz at the input the counter will give 1 pulse per 24 hours, e.g. it can form the basis of an extremely accurate 24 hour alarm. Such an alarm never requires setting once the counter has been reset to zero at the required time of day and will thereafter give the alarm at exactly the same time every day. It can thus be used for instance to wake oneself up every morning without fail.

Such a circuit is very easily built using just 4 cheap CMOS chips. IC1, a 14 stage binary counter is set to divide by 10000 (binary 10011100010000) by reseting to 0 on the count of 10000. Similarly IC2, a 12 stage binary counter divides by 432 (binary 110110000). IC3 and IC4 provide the necessary decoding to reset the counters (which are reset by a logic '1' unlike TTL where a logic '0' is usually required). Additionally the gating allows the counter to be reset to 0 by SW1.

Voltage Stabiliser

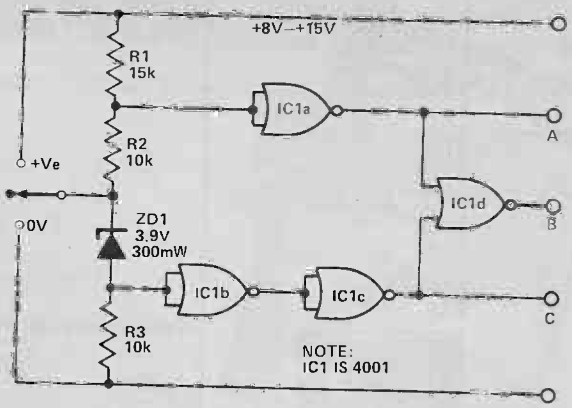
J. Nicholls

Here is a voltage stabiliser with good performance and low component count which will operate well, even when $V_{in} - V_{out}$ drops to 2 V. Only a few milliamps are dissipated through the zener, making it suitable for battery operated equipment.



Most circuits of this type (but with the FET replaced by a resistor) suffer from zener saturation when V_{in} is getting low, or excessive zener current when V_{in} is high.

Actual component values can be varied to suit individual applications.



3-way CMOS switch

G. Warburton.

When the input is switched positive the voltage across the zener is sufficient to bias the junction between R3 and the zener high, producing a high output at C.

With the input unconnected, the junction between R1 and R2 is high while the junction between the zener and R3 is low. This will produce a high output at B.

Connecting the input to 0V causes output A to go high.

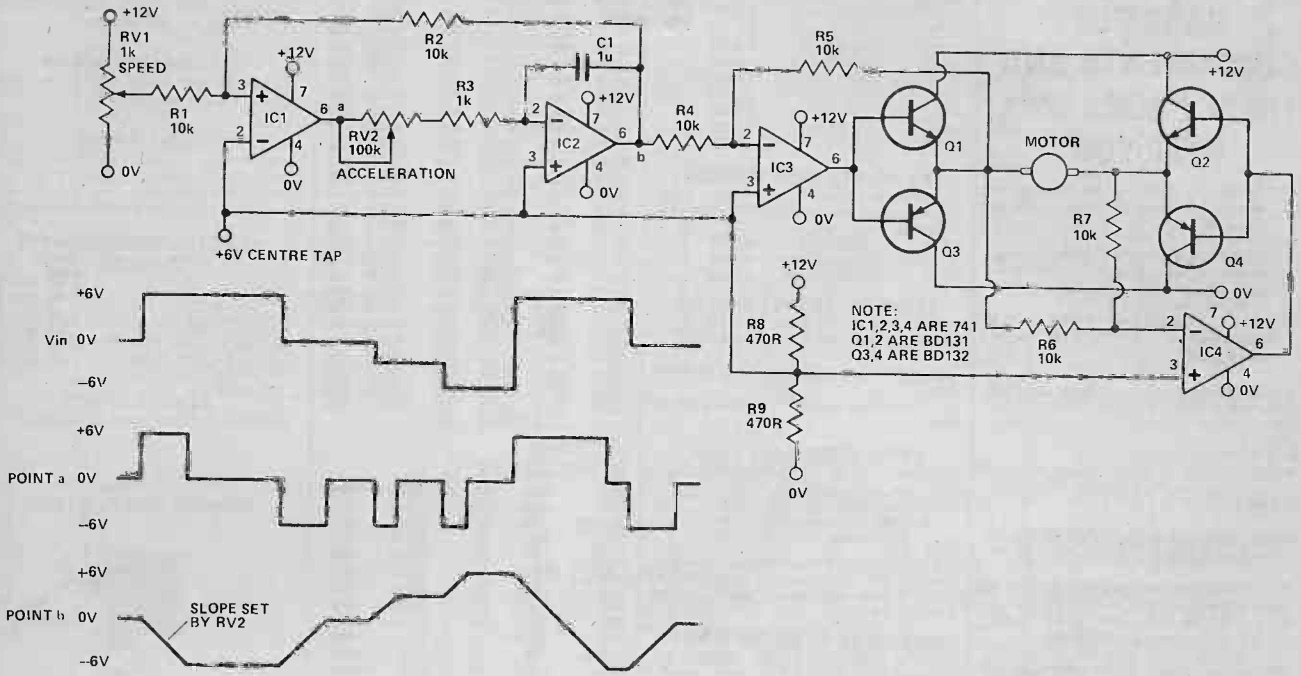
The circuit was primarily designed to be used with quad CMOS switches (i.e. 4016, 4066) for audio switching but can be used for a variety of applications.

Tech-Tips is an ideas forum and is not aimed at the beginner. We regret we cannot answer queries on these items.

ETI is prepared to consider circuits or ideas submitted by readers for this page. All items used will be paid for. Drawings should be as clear as possible and the text should preferably be typed. Circuits must not be subject to copyright. Items for consideration should be sent to ETI TECH-TIPS, Electronics Today International, 25-27 Oxford St., London W1R 1RF.

Controller For Model Trains

E. Parr

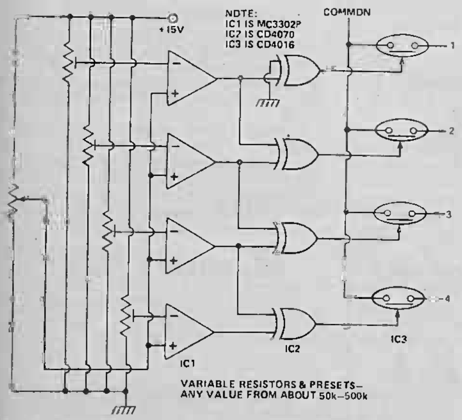


Most model railway controllers have the unfortunate characteristics of giving instant starts and stops to the train which would be very unnerving for the model passengers. The circuit described gives a steady acceleration or deceleration on speed changes, and the speed and acceleration controls do not interact.

The power supply is 12V split by R8 and R9 so it appears to the op amps as a ± 6 V supply. Voltages in this description are referenced to the 6V centre tap. IC1 and IC2 together form a unity gain inverting amplifier, with the gain determined by R1 and R2. The slope of IC2's output, is determined by C1 and R3/RV2. The output of IC1 will thus take up one of three states: +6 V (hard positive), 0 V (balanced), −6 V (hard negative) dependent on the output voltage being more positive that equal to, or more megative than the voltage set by RV1. The output voltage will thus ramp up or down at a constant rate until it is equal in magnitude (but opposite in sign) to the voltage on RV1. This is summarised on the waveform drawing.

Voltage b drives buffer amplifiers IC3 and IC4 to give a push pull 12 V drive to the motor for forwards and reverse. Note that the feedback resistors R5 and R7 are taken from the transistor emitters to compensate for the transistor V_{be} drops. The motor should have some current cut-out or limit connected in series with it to protect the transistors.

In use RV1 sets the speed, and RV2 the acceleration: it gives a very realistic train control, although much more skill is needed to stop a train accurately at a station platform. In this respect it is very close to driving a real train.



Slide Switch

C. Jordan

One of the disadvantages of slide pots is the unavailability of matching slide switches, as with rotary switches and pots, but slide pots can be given switching action by the use of this circuit.

Each analogue switch is only turned on when the comparators driving the respective EX-OR gate are in opposite states, i.e. when the voltage on the slider wiper is between the appropriate two preset voltages.

The example is a 4-way, 1-pole switch with off but any-way, any-pole switches can be made, using 741s as comparators if economic. A little mechanical ingenuity can provide click stops, if required.

. . . . news

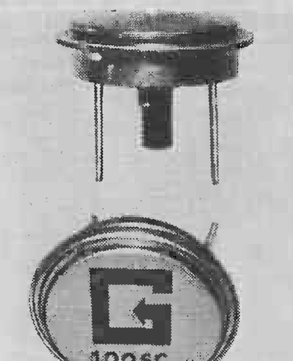
watch out! satellite about

General Electric Company of the USA's researchers have demonstrated that space technology could assist narcotics and immigration agents in stemming the flow of contraband drugs and illegal aliens across remote stretches of America's borders.

In field tests ranging across the US GE(USA) communications experts have demonstrated that a geostationary space satellite, orbiting at an altitude of 23 000 miles over a fixed spot on the earth's surface, could keep field agents in constant mobile radio contact with a base station — even from isolated points thousands of miles apart.

The tests involved two National Aeronautics and Space Administration experimental communication satellites — ATS-3, hovering above the mouth of the Amazon River, and ATS-1, over the equator south of Hawaii — and a station wagon equipped with a special antenna and radio equipment.

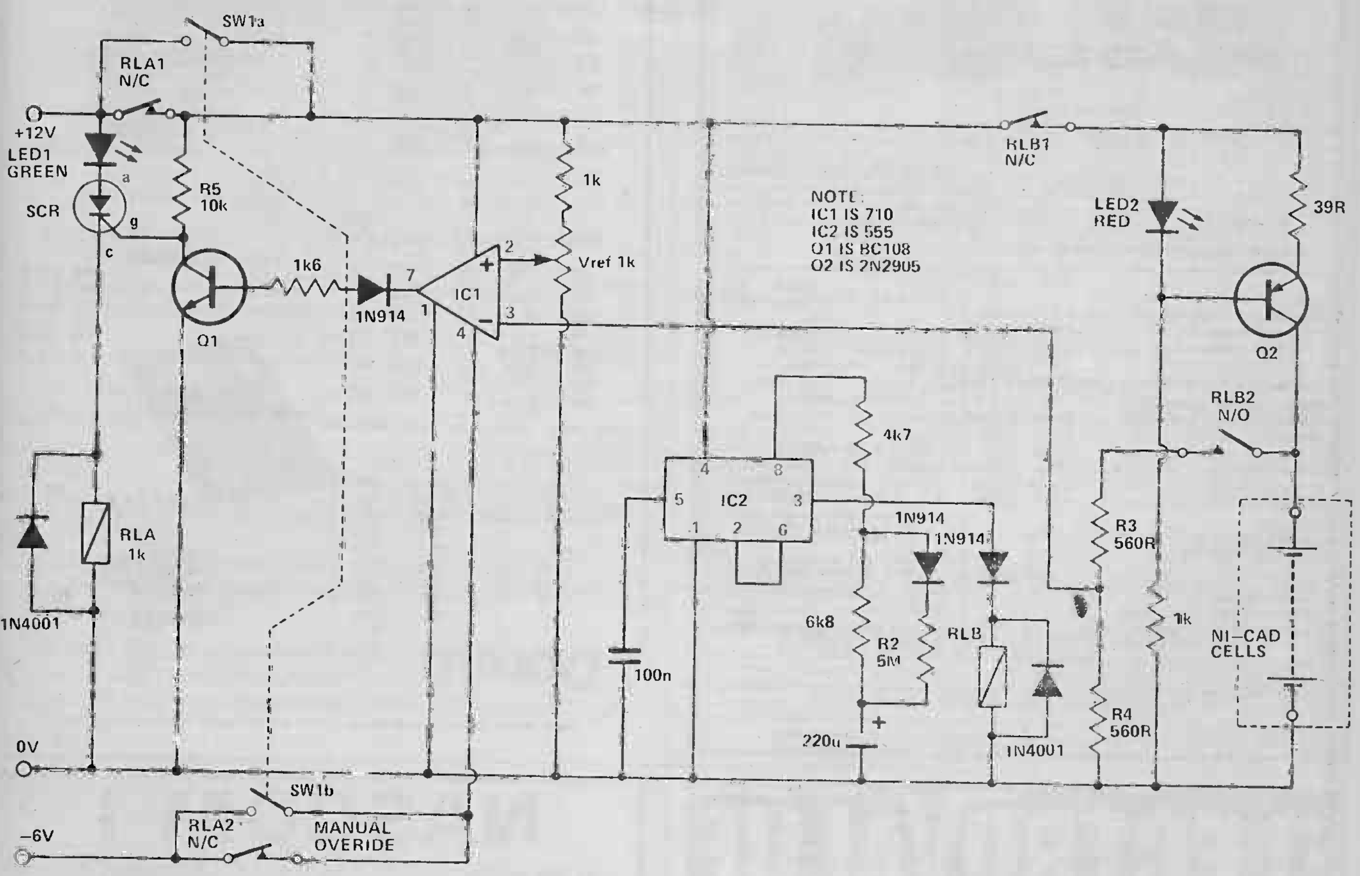
germanium power



Who said Germanium was dead? Available from Wintronics is the World's first 100 amp transistor on a single chip, and it's made from a Germanium junction nearly half an inch in diameter. Called the GPD 100SC series the devices are made by (wait for it) the Germanium Power Devices Corporation. Packaged in a standard TO 68 box, the devices have a typical gain of 120 at —60 A I_c and will be available with various operating voltages. Wintronics, Southon House, Edenbridge, Kent.

socket & see

Galatrek Engineering have just introduced a new 13A socket tester. Instead of sticking your fingers in the hole, you can now stick a Galatrek in . . . and it lights up rather than you! It uses two neon lights to indicate the condition of the wiring, and can detect faults on the live, neutral or earth. Retail price will be £3.95. Galatrek Engineering, Scotland Street, Llanrwst, Gwynedd LL26 0AL.



Automatic Nicad Charger

P. Reynolds

The problem of ensuring that expensive Nicad cells are not damaged in the recharging process is twofold. First, as the cells have a low internal resistance, they are susceptible to damage by excessive charging currents. Second, damage will also occur if the charging process is carried out for too long a period.

The first problem may be overcome by charging at a constant current. The second problem may be overcome by the use of an automatic sensing comparator circuit, which compares the voltage from the cells with a preset voltage, related to the fully charged value. In practical terms, the circuit appears as shown in Fig 1. A red LED supplies the voltage drop to ensure that Q1 passes a constant current of about 25 mA to the cells under charge. This charging current may be adjusted, if desired, by changing the value of R1. The 555 runs in the astable mode. However, the duty cycle is adjusted to be *less* than 50%, by incorporating a diode and resistor in parallel with R2. How this is accomplished may be easily understood if one remembers that charging of the capacitor takes place through these paralleled components, whereas, due to the blocking diode, discharging current only flows through R2. The 'off' time is around 15 mins. and the 'on' time less than 0.5s. The relay coil, RLB, thus receives a positive pulse of short duration every 15 mins. Contact RLB1 opens, disconnecting the charging supply and contact RLB2 closes. A sample of the total voltage across R3 and R4 is applied to the variable input of the 710 comparator. This input voltage is compared to the preset reference voltage and if found to be greater, the output will drop to −0V5 (from +3V2). The inverting action of Q2 causes the gate of the thyrister to undergo a positive transition, via R5. The gate causes the device to conduct, causing the contacts RLA 1 & 2 to open and disconnect the supply from the rest of the circuitry. The green LED is illuminated, indicating the termination of the charging period.

This circuit may be used to charge a total of six 1V5 cells. Of course V_{ref} may need adjustment so as to be commensurate with the voltage across R4. A manual override switch is also provided.

digest...

if at first...

One of the most used machines in the ETI office is the Telex, now the PO has come up with a computer controlled international exchange. In the past you had to keep redialling if for some reason (war, flood or act of God) there was no answer. As you can imagine it could become a pain in the finger with up to 13 figures. Now the computer will make the repeat attempts for you, the auto redial stops after 3 unsuccessful attempts though.

Another facility provided by the new exchange is an informative fault message system. Instead of an indecipherable +X- spluge, the computer will print a message along the lines "The lines to Neasden are down, try again in 3 weeks", well even though it still is impossible to make a connection — it's nice to know why!

whoops

ETI would like to point out an error which crept into the April and May Maplin advertisements. The Maplin Touch — Sensitive Piano is **now available**, the advertisement said that it was not (amazing how gremlins can creep in!). The error has now been corrected, and the typesetter shot.

Tank battle — May 78: ZD1 should be 8V2, C1,4,5,6,16 — 100n and C2,3,11,12 — 220n.

listen to the cracks

Ever felt that you were cracking up? Unlike an aircraft you can tell somebody — now a company called Dunegan-Endevco is actually listening to aircraft to detect when there is too much stress about. They call the technique acoustic emission and it is claimed to have advantages over X-ray or ultrasonic systems. It seems that materials under stress emit high frequency sound which when analysed can be used to pin-point areas of corrosion and stress. The system is catching on fast in the States, the US Air Force has used it to check out planes while actually in flight! Dunegan Endevco, United Kingdom Division, Melbourn, Royston, Herts.

new part part

Tamtronik, the PCB people have just started a components division specifically to supply parts for projects published in electronics magazines. Further details from Tamtronik Ltd., 217 Toll End Road, Tipton, West Midlands DY4 0HW. (021-557 9144.)

heavy heavy hole

A giant black hole has been discovered in the Virgo constallation. At the centre of the Mersier-87 galaxy the hole was identified by astronomers at California Institute of Technology and Kitt Peak Observatory in Arizona.

Billions of stars appear to be slowly circling the hole in ever-diminishing orbits — the weight of the hole is estimated at about 5 000 million of Earth Suns, and growing! Don't worry quite yet, Mersier-87 is 50 million light years away.

breadboard 78

At long last a show for the electronics enthusiast — Breadboard '78 will be held at Seymour Hall in London from the 21st to 25th of November — mark it in your diary now 'cos ETI will be exhibiting! If you are a firm and would like more details contact: Breadboard '78, Abbey Mead House, 23a Plymouth Road, Tavistock, Devon PL19 8AU.

r.s.p.c.hi-fi?

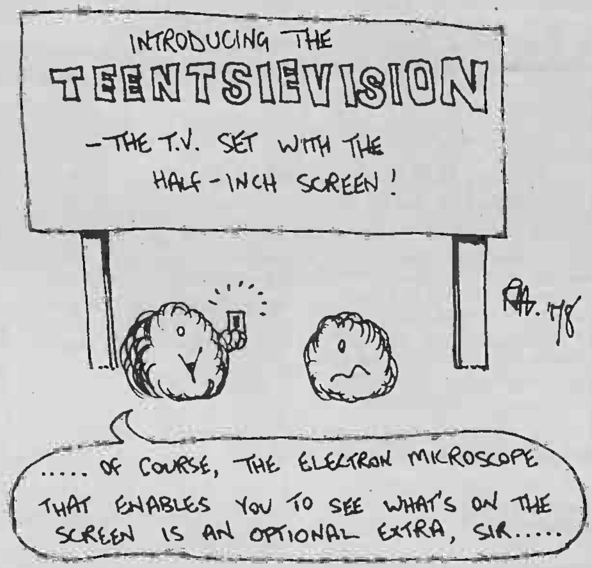
We get a huge quantity of press releases at ETI, everybody wants publicity. Well this item we thought was an April spoof — but it's real — so with a straight pen:

Sanyo have sponsored Harvey Smith and Team Sanyo to compete in UK horse jumping and riding shows. The names of Harvey's horses have been changed to the following (old name in brackets): San Mar (Olympic Star), Sanyo Video (Upton), Sanyo Blender (Countdown), Sanyo Microwave (Spooky), Sanyo Cadnica (Salvador), Sanyo Music Centre (Graffiti) and last but by no means least Sanyo Hi-Fi (Graf).

Anyone for 50p each way on Sanyo Hi-Fi?

news... ...digest

matchbox tv camera



Fairchild have succeeded in producing a CCD camera with 400 000 elements. The camera is rugged enough to be built into the tip of a munitions shell and survive the blast when fired. Defence experts in London were recently shown video tapes made from a shell-borne camera as it descended from 5 000 feet with a parachute slowing it down. Weight of the camera is only 2 ounces.

odds & ends

★ SERT (Society of Electronic and Radio Technicians) are to hold a Hi-Fi seminar on Wednesday, 7th June. The seminar will be held at the Institute of Marine Engineers in London, cost £15 (£10 for SERT members). Further details from SERT, 8-10 Charing Cross Road, London WC2 0HP.

★ Oertling Ltd have supplied the golf ball manufacturers Penfold with electronic weighing machines — next customer IBM?

★ Two new catalogues received this month, first from Technomatic Ltd — 20 pages of part numbers and prices, send large SAE to Technomatic Ltd, 54 Sandhurst Road, London NW9 9LR. Second catalogue costs money (but worth it) — 40 pages from Marshalls for 35 pence sent to A. Marshall (London) Ltd, 42 Cricklewood Broadway, London NW2 3ET.

★ The Post Office is trying to export Viewdata to the USA, and is having talks with AT&T to find out how it can be connected to the US phone system.

★ Burr-Brown have introduced two interesting operational amplifiers. The 3573 has a power rating of 40W (100W peak) and the 3528 is a FET input type. The FET device has a typical bias current of 75 femto amps (that's 10^{-15}). Burr-Brown International, 17 Exchange Road, Watford, Herts WD1 7EB.

★ Visitors to Piccadilly in London will be dazzled by large-scale laser light shows. The idea is to replace the famous Coke signs with a projection screen, and write advertisements with lasers. The scheme is to operate as a trial for a year.

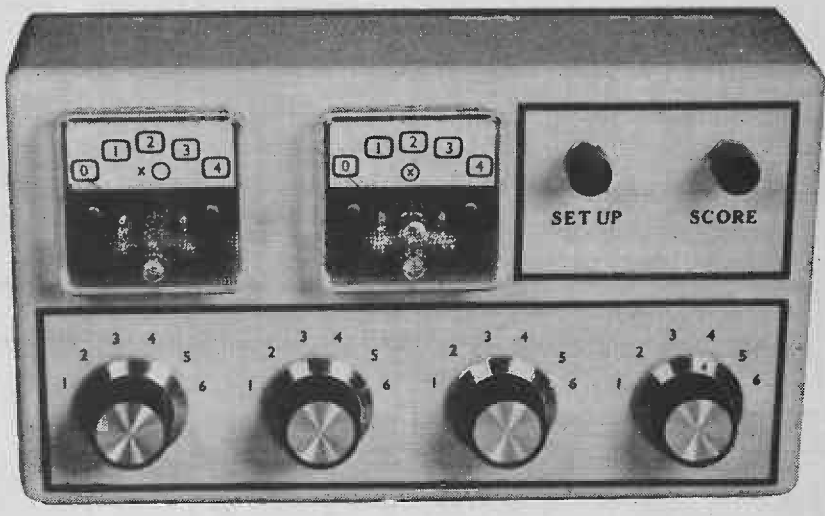
★ For a scoop preview of the Commodore Systems printer see Microfile this month.

barclay electronics & tritron

Will readers please contact ETI before sending any money to the above companies who have advertised in previous editions of the magazine.

STARS & DOTS

A. Willcox has ditched dogged reliance on digital devices to come up with a new version of the popular clear thinking game.



THIS CODEBREAKER game is based on the traditional pencil-and-paper game known variously as 'Stars and Dots', 'Bulls and Cows', or 'Moo', and which has recently become popularised as 'Mastermind', and is usually played as follows. The first player sets down a four-digit code which his opponent must try to duplicate by a series of guesses for which he is awarded points. In one version of the game a star is given for each correct digit in the right position, and a dot for each correct digit in the wrong position.

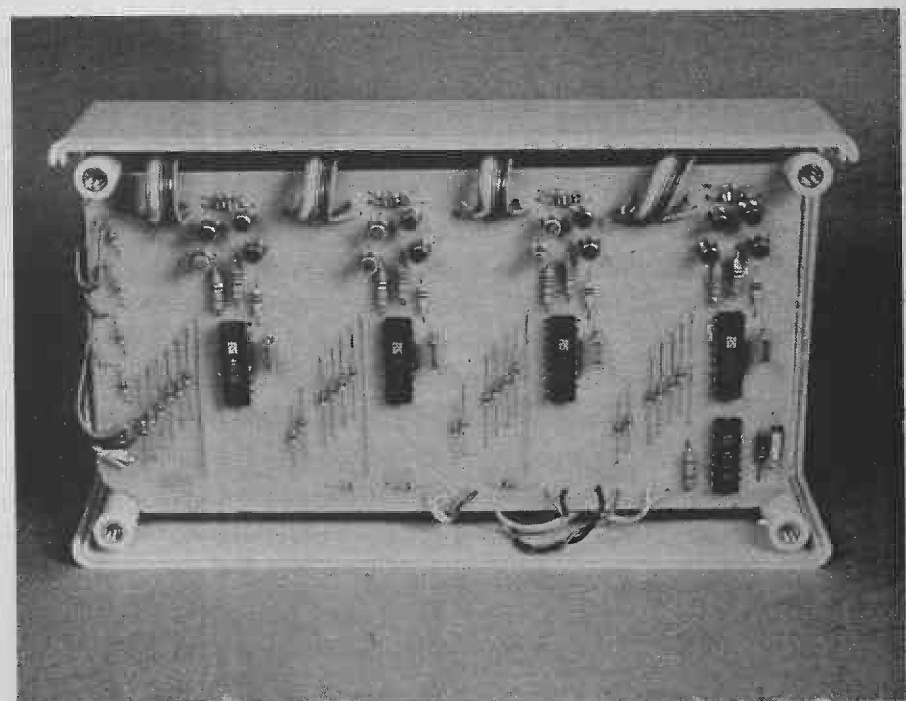
Analogue Stars

In the following illustrative game the hidden code is 1633:

	STARS	DOTS
1234	★★	
5634	★★	
3434	★	●
5233	★★	
5244	★★	
1633	★★	

The object of the game is to crack the code in the least number of guesses, and in order to achieve this it is necessary to analyse carefully the results of previous tries.

This electronic version of the game sets a random code and awards the appropriate score for each attempt, thus allowing the game to be played solo fashion. A pen and paper record is kept as before, or, if a 'Mastermind' board is used, the switches may be marked with colours rather than numbers. Each attempt is duplicated on the switches and the score is shown on the two meter movements. There is no indication in the score as to which of the digits is correct.



A look at the inside of our stars and dots game.

Construction

The version of stars and dots that appears here uses analogue circuits techniques rather than the digital methods of ten used to implement the game. This has resulted in a number of improvements over other electronic version of the game — namely simple circuitry and low power consumption, so low in the idle state that there is no need to fit an on/off switch to the project the overlay shows the arrangement of the onboard components but it can be seen there is a lot of interwiring to-from and between switches. Study the circuit diagram to familiarise yourself with what is going on and take care with this stage of construction as this is where mistakes are likely to occur.

ETI ▶

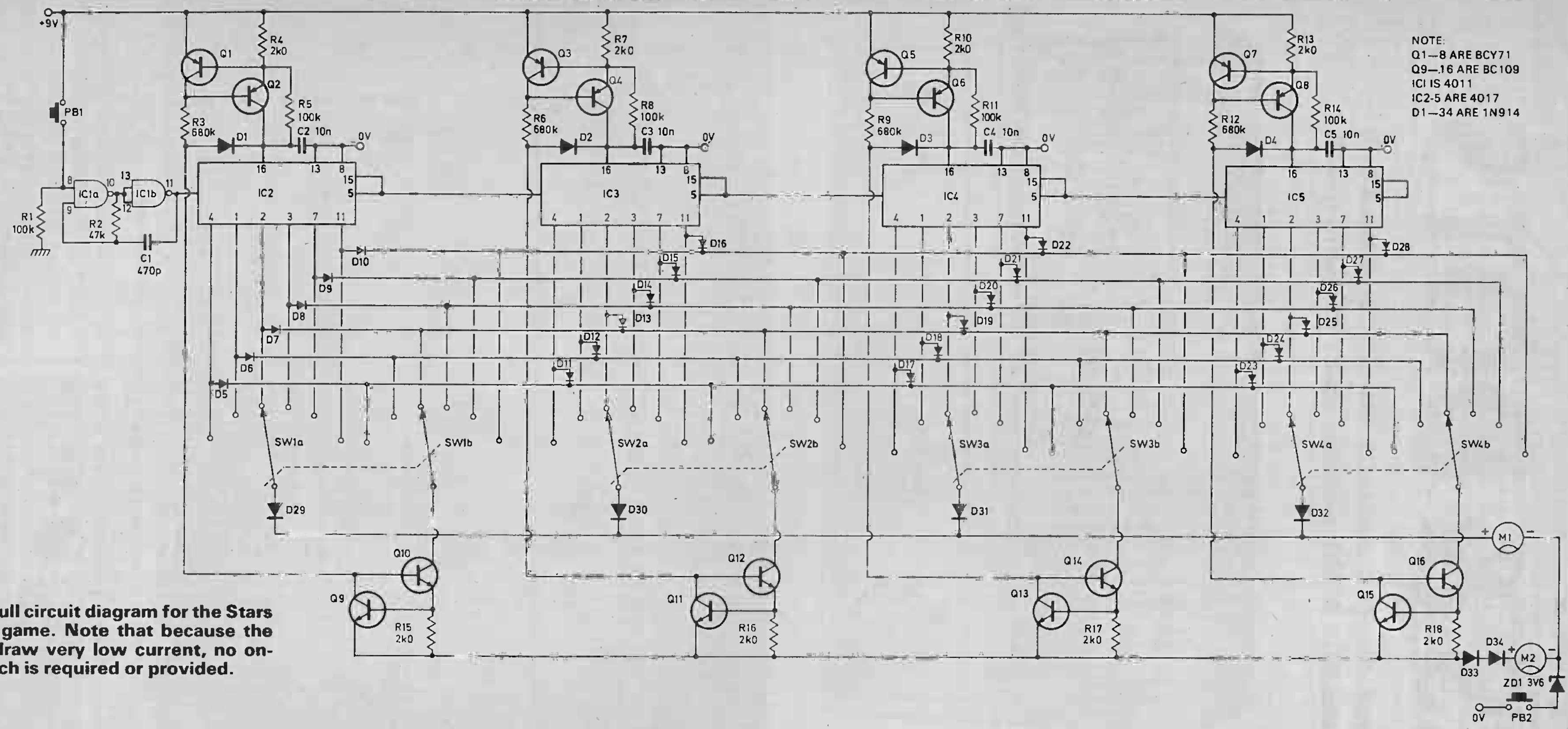


Fig. 1. Full circuit diagram for the Stars & Dots game. Note that because the games draw very low current, no on-off switch is required or provided.

HOW IT WORKS

The random number generator consists of a four stage counter using 4017 decade counter ICs, locked by IC1 whenever the set-up switch PB1 is depressed. The state Q6=1, is unstable, resetting the IC to Q0=1, and also 'carrying' to the next stage. Supply current to each stage when the score button is depressed is limited to 0.25mA by constant current circuits, which means that this 'unit' of current is all that is available at whichever Q output is enabled. Continuity to the ICs at other times to enable the code to be stored is provided by the 100k resistors R5, 8, 11 and 14. The current drawn in this situation is so small as to make an on/off switch unnecessary, and the number remains stored until the set-up switch is again operated.

If any of the switches is in the correct number/correct position situation a unit of current will be available at each relevant switch's 'a' section to be 'added' by M1. This use of analogue circuitry avoids the need to randomize the score—there is no indication of which of the stages is correct. Note also that when a switch is in this correct number/correct position situation none of this current is available to pass on to the common lines connected to the 'b' sections, because the diode connected to the active Q output becomes reverse biased. This is so because the voltage at this output is taken down to a little over ZD1 voltage, incidentally lowering V_{DD} also, to which it is tied internally.

Turning now the 'b' section of the switches, these deal with the situations where the correct number/incorrect position has been selected. Observe that Q0 - Q5 is connected in each case by diodes to common lines, making its particular unit of current available to the other stages. And so if a switch selects the correct number of another stage, current will flow through its 'b' section, this time to M2 via another constant current stage. These extra constant current stages are necessary for if, say, a switch selects a number which corresponds to the code in two other stages (e.g. code=1233; switches=3456) then two units of current would otherwise flow through M2 instead of the one unit due. If on the other hand two or more switches select a number which only appears once in the code, only one unit will flow because this is all that is available from the IC. The use of analogue circuits in this way, by allowing the IC outputs to be commoned, results in a very simple circuit.

When a switch has selected a correct number of its 'a' section, its 'b' section must be inhibited, otherwise it will also score on M2 if the same number exists in another stage. The voltage drop on VDD mentioned earlier is utilised to achieve this by removing, via a diode, the bias to the lower constant current stage concerned. The base voltages of the lower constant current stages are raised by the presence of the two diodes in series with M2 to facilitate this.

The 3V6 Zener diode maintains a reasonable working voltage for each IC under all conditions, and without this the V_{DD} connection would not follow closely any fall in a Q output level as described earlier. Total consumption is so low that the battery requirement is met by a PP3, and if this doesn't last at least a year there's something wrong.

Note that if both buttons are depressed together the input level rating of the 4017 IC5 is exceeded due to the changing level of V_{DD} so although perhaps this is to be avoided no damage has resulted in practice when this has been done.