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electronics today

INTERNATIONAL

DECEMBER 1980 60p

POLYPHONIC SYNTHESISER PROJECT

Build it for a fraction of the cost of commercial units!

FOUR Circuit design features inside

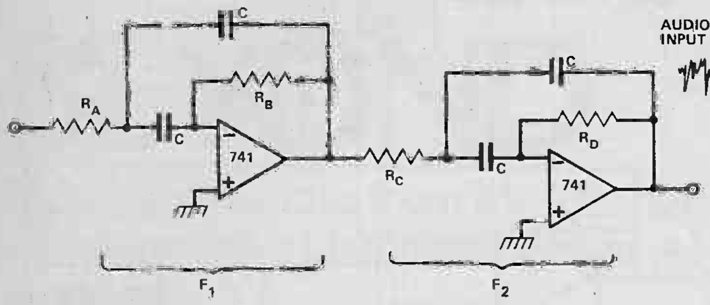
How the Universe began?

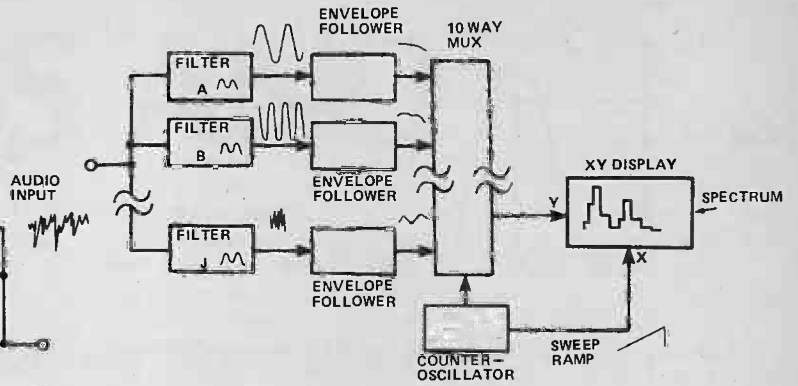
An Alarming Clock design

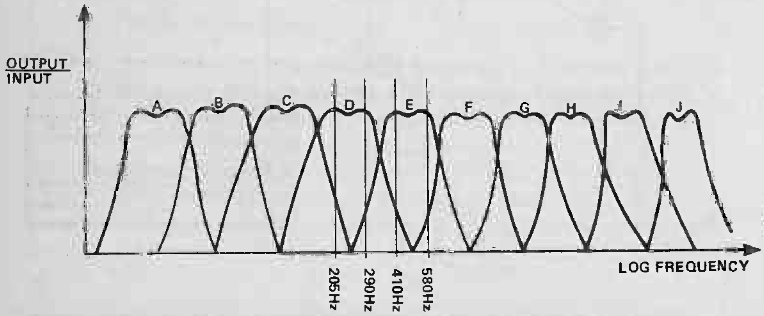
Audio Mixer project

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

R_A	R_B	C	F_1	Q	R_C	R_D	C	F_2	Q	
3k6	360k	15n	290Hz	5	5k1	510k	15n	205Hz	5	FILTER D
1k8	180k	15n	580Hz	5	2k55	255k	15n	410Hz	5	FILTER E







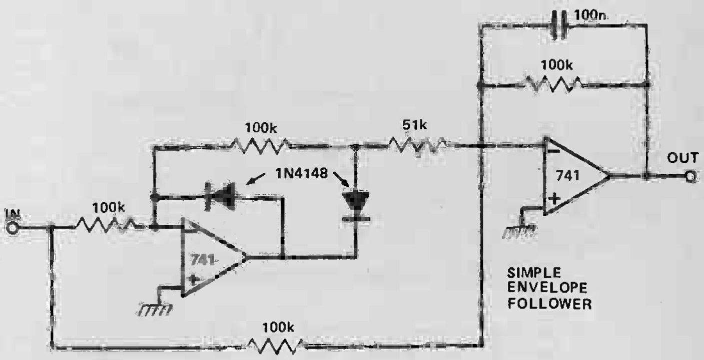


Fig.6. An analysing filter bank suitable for a spectrum analyser. a) Circuit for each of the ten filter stages (A to J), and component values for two of the stages. b) Graph of the ten filter responses. c) Block diagram of the spectrum analyser and a suitable circuit for the envelope follower.

Other Design Applications

Figure 7 shows a design for a parametric audio equalizer. This device has variable cut and lift and a resonance and frequency control. The resonance control is arranged such that as the Q increases, the input signal is attenuated (RV2a), thus maintaining the same overall gain at resonance, independent of the Q setting. The filter is a state variable design which is situated in the feedback/feedforward loop of an op-amp. Thus RV3 controls whether overall response is a bandpass cut or lift.

The resonant frequency is tuned by RV1 and SW1 is used to switch frequency ranges. The Q factor is set by RV2.

The TCA580N (Signetics) is an IC that can be used to simulate an inductance and in doing so may be used to synthesize many conventional LCR filter circuits (Fig.8). The device has a pair of floating input terminals which generate the impression of being an inductor. This inductor is programmed by three passive components, R_{C_1}, R_{C_2}, C_2. By connecting a capacitor (C_1) across the input terminals, a parallel resonant circuit (C_1L) is produced.

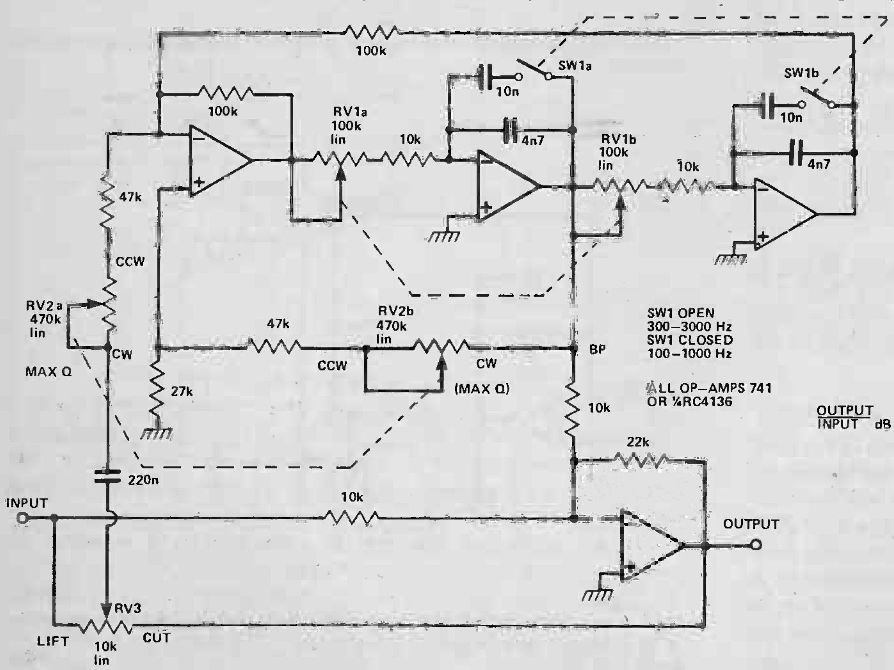
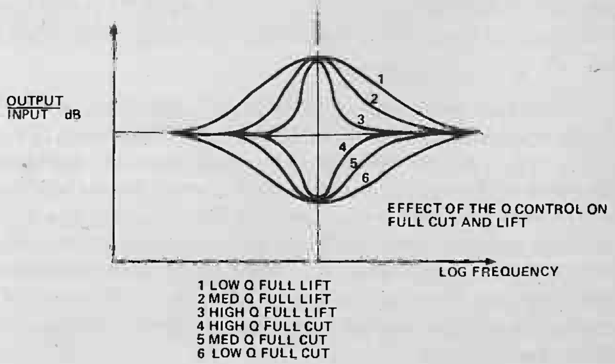


Fig. 7. A parametric audio equaliser with variable cut and lift and a resonance and frequency control.



Moving Story?

The SSM2040 is a four section mobile filter that can be exponentially voltage controlled over a 10,000 to 1 frequency range. The device contains an exponential function generator that controls four variable transconductance amplifiers each having their own output buffers. The IC may be used for electronic music synthesis, musical effects, tracking filters and many other applications where filter mobility is needed.

A four pole lowpass filter for electronic music is shown in Fig.9. Each stage is a single pole mobile lowpass filter, four of these filters are connected up in cascade and fed into an output amplifier. A resonance feedback route is provided so that the Q factor may be manually controlled. The voltage control of frequency is set to -1 V/octave.

By modifying the external components, the device can be transformed into an all pass filter, (Fig.10). This filter has a flat amplitude response and a phase shift that changes by 180° as a function of frequency. As the SSM2040 has four stages, the whole filter has a variable 720° phase shift. When the filter output is mixed with the original signal, two notches are produced in the frequency response occurring when the phase between the original and phase shifted signal is 180° and 540°.

As the phase shift is slowly modulated up and down in frequency, the notches also move producing the characteristic phasing sound.

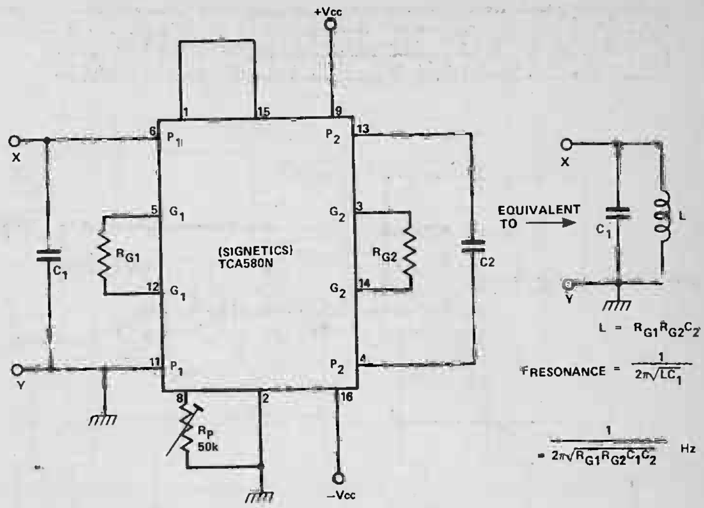


Fig.8. A monolithic gyrator using the TCA580N. It can simulate inductances up to one millihenry.

+Vcc TO –Vcc	14V
SUPPLY CURRENT	0.8mA
Q FACTOR(200Hz)	500 TO 5000
MAXIMUM SIMULATED INDUCTANCE	1mH
FREQUENCY RANGE	DC TO 10kHz

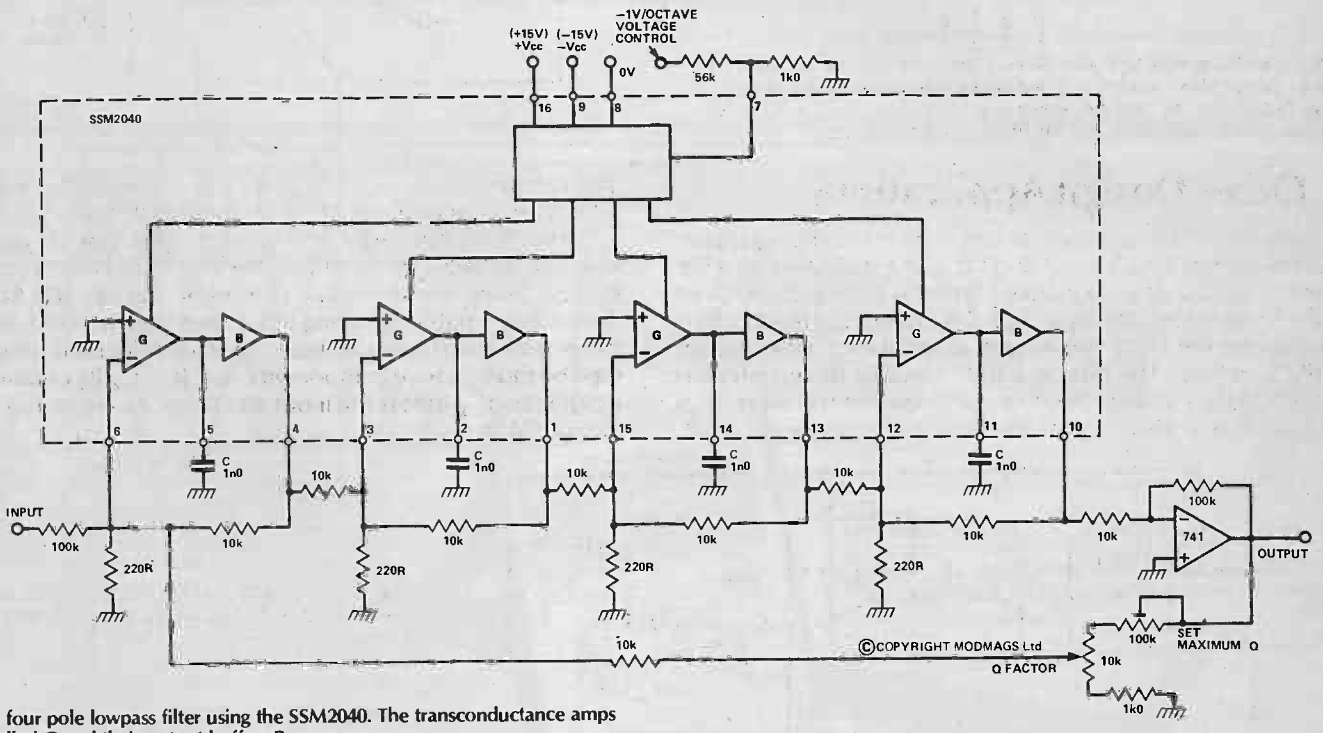


Fig.9. A four pole lowpass filter using the SSM2040. The transconductance amps are labelled G and their output buffers B.

Monomania

Monolithic filters are becoming more and more common. One such device that lends itself to integration is the transversal filter, Fig.11a. This device can produce a steep roll-off slope, a high out-of-band attenuation and most significantly a linear phase response. The transversal filter is a tapped analogue delay line. The input signal is sampled and this sample moves down along a bucket brigade delay line. Each bucket has a separate output so that the signal may be monitored at each stage via a weighting resistor. It is possible to weight the resistors such that they draw out the impulse response of the required filter performance.

When fed with an input signal that is being shifted down a delay line, this impulse response results in it being converted into a frequency response. The filter frequency is directly linked to the clock frequency, and thus it is impossible to make the transversal filter mobile.

It is necessary, as with all sampled data systems, to precede the device with an antialiasing filter and to recover the signal. There are now several transversal filters available, but they are still relatively expensive and are best used only where linear phase response is of prime importance.

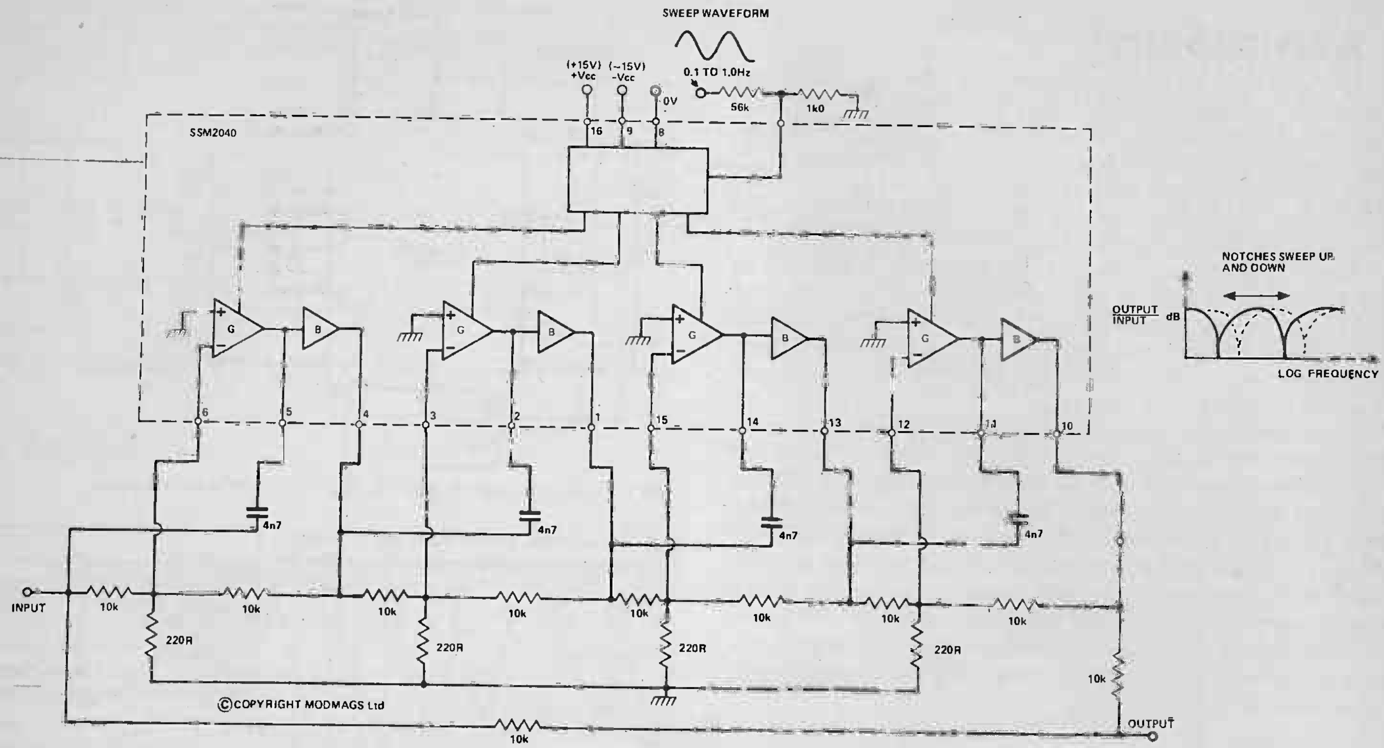


Fig.10. The SSM2040 connected as a two-notch phasing unit.

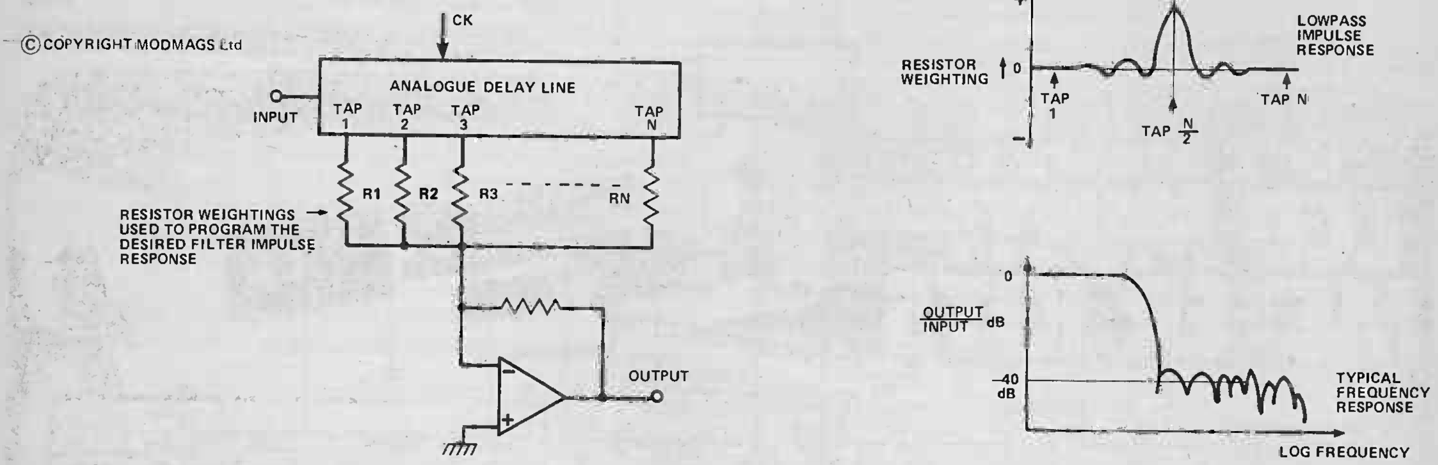


Fig. 11 The transversal filter, with graphs of impulse and frequency responses for a lowpass design.

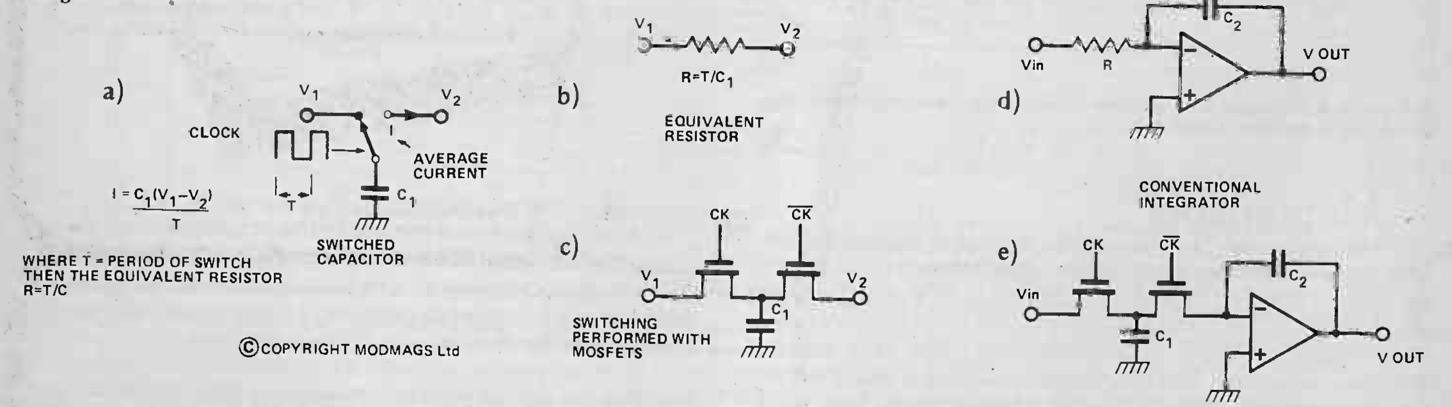


Fig.12. Switched capacitor filters. a) Basic circuit. b) The equivalent resistor. c) Practical design using MOSFETs. d) Conventional integrator. e) Replacing R with a switched capacitor enables a filter to be easily produced in IC form.

Recent Monos

A more recent monolithic device is the switched capacitor filter, which can be used to implement many standard lowpass and bandpass filter structures, Fig.12. The problem with producing monolithic recursive filters is that stable high tolerance components such as resistors and capacitors are very difficult to make, and the filter performance depends heavily upon these tolerances. However, it is possible to simulate resistors with switched capacitor techniques. With the switch as shown in Fig.12a C_1 is charged up to V_1. When the switch is thrown to its other position the capacitor is discharged into V_2. By continually switching the switch, a current I can be made to pass from V_1 to V_2. This simulates a resistance R (where R equals the period of the switch divided by C_1). The switching is performed by two MOSFETs (Fig.12c) driven by antiphase clock signals. The simulated resistor can be used to construct an integrator (Figs.12d,e) which can then be used to build up conventional filter structures. For example a state variable filter would have a resonant frequency Fc, where

$$F_c = \frac{1}{2\pi R C_2}$$

$$\text{but } R = \frac{T}{C_1}$$

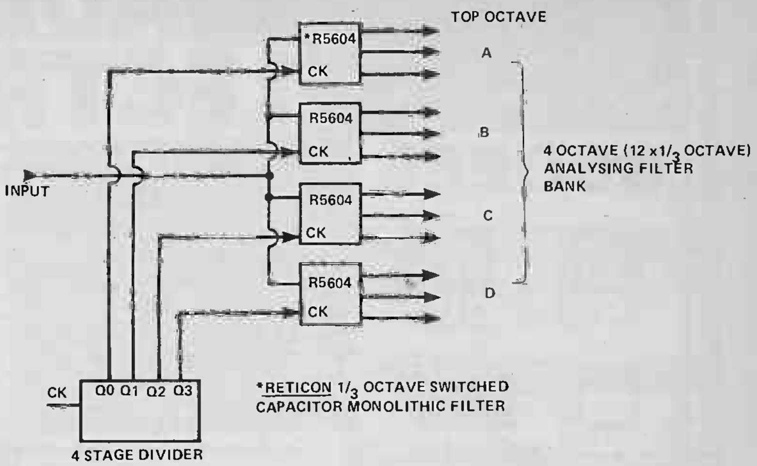


Fig.13. Switched capacitor filter bank using the R5604 from Reticon.

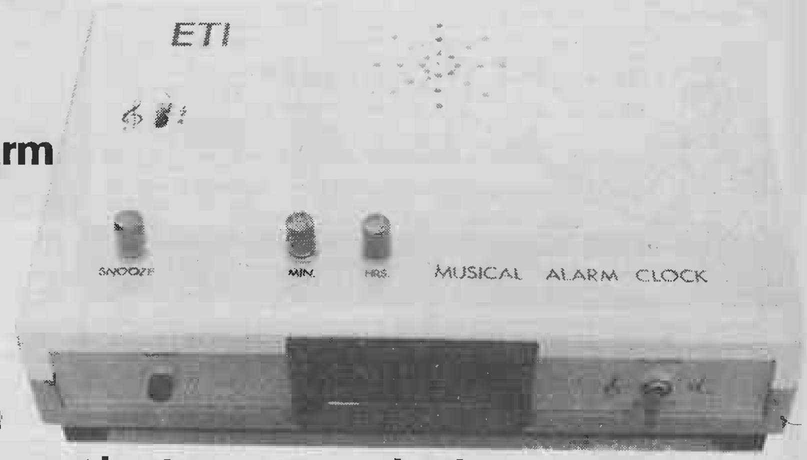
$$\text{therefore } F_c = \frac{C_1}{2\pi C_2 T}$$

Note that Fc is linearly proportional to 1/T, which is the clock frequency.

Reticon make a switched capacitor bandpass filter which contains three filters at one-third octave spacing, thus making filterbank design relatively simple (Fig.13). Maybe in a few years time it will be possible to purchase a wide range of low cost monolithic filters. If that day comes, you won't need to learn how to design active filters!

ETI

MUSICAL ALARM CLOCK



This unique battery-powered alarm clock has all the usual facilities, but in addition it makes a tick-tock sound and plays a random selection of eight tunes under the alarm condition

Design by Ray Marston. Development by Steve Ramsahadeo.

This ETI alarm clock project has a number of unique features. First, the clock is based on a ready-built CM172 module that has a large liquid crystal readout and will give a year or two of continuous running from a single 1V5 cell. The module can be easily programmed for 'time set' and 'alarm set', remembers the 'alarm set' time indefinitely, has ten minute snooze facility, two alarm output terminals and a built-in backlight. The CM172 module forms the basis of the whole project.

The second feature of our clock is that we've provided it with a unique tick-tock sound generator circuit, so that the clock looks electronic but *sounds* like a mechanical device and can easily be located in a darkened room. The tick rate and volume are presettable, to suit individual tastes. The tick-tock circuit typically consumes a mere 9 uA of current and gives a couple of years of continuous running from a standard battery.

Finally, we've fitted our clock with a special and modestly priced microcomputer chip that, in conjunction with a handful of other components, generates a random selection of eight well known tunes under the alarm condition. The tempo, tone and volume of the tunes are variable via preset pots. The microcomputer chip is actually capable of generating a total of 24 different tunes in three selectable groups of eight tunes each (see Table 1).

In our circuit any one of the three groups can be manually selected via a switch. Electronic circuitry then randomly selects and plays complete tunes from that group, automatically inserting a random pause of up to several seconds between the end of one tune and the start of the next when the circuitry is active.

The tune-generator circuitry consumes virtually zero standby current, turns on automatically under the alarm condition, can be temporarily cancelled (for ten minutes) by a snooze button or can be disabled by either of two off switches. It can also be independently played at any time using the tune test switch. We've also provided the circuit with a switch that gives you the option of selecting either the musical alarm or a more conventional pulsed tone alarm signal.

The complete alarm clock project uses a total of three batteries, there being a 1V5 cell for the actual clock module and two 9 V (PP3) batteries for the rest of the electronics. The 1V5 cell can be expected to give one or two years of continuous running. The two 9 V batteries should last about a year under normal usage conditions. The clock is fitted with a special switch-enable slide switch, which ensures that important functions and time settings cannot be inadvertently operated or changed by accidental switch operation at night or when the clock is packed away in a suitcase, etc.

Construction

Construction of the electronics of this project should present few problems. The basic CM172 clock module circuitry is quite independent of the alarm/tick-tock circuitry, so these two major units can be built and tested separately.

Start the construction by tackling the major mechanical problems. If you intend to use the same case as our prototype note that the front panel will need to be split into two and the upper and lower centre-front lips of the case will have to be cut away to accurately accommodate the clock bezel (a separate item, not supplied with the basic CM172 module). The bezel is then used to secure the module to the front panel.

Next, fit all the switches associated with the clock module (PB1, 2, 3, 4 and SW1, 2, 3) to the case and make the interconnections to the clock module solder pads as indicated in the circuit diagram, using a fine-tipped soldering iron. Now connect B1 (1V5) to the module (taking great care to fit it in the correct polarity) and give the clock a functional test as follows.

Turn SW1 on, press SW2 to TIME SET and use PB1 to set MINS and PB2 to set HOURS (when setting HOURS, check that the correct AM or PM indication is given at the left of the display). Similarly, use SW2-PB1-PB2 to select the ALARM SET time, taking care to get the correct AM or PM indication. The alarm facility can be activated by turning SW3 on, in which case a treble clef musical sign will appear at the right of the display. PB4 operates a built-in backlight, for night viewing, and is intended for intermittent operation only.

Alarm/Tick-Tock Circuit

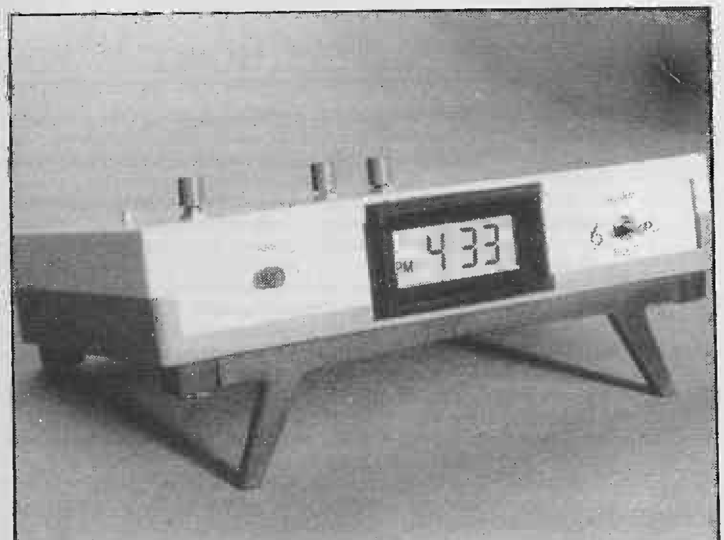
Proceed now to the construction of the alarm/tick-tock circuitry on the large PCB. Note that all ICs should be fitted in suitable sockets. The usual care should be taken to ensure that all semiconductor devices and electrolytics are fitted with the correct polarity. When construction is complete, temporarily fit speaker LS1 in place, together with SW4 and SW6. Connect B2 (9 V) in place, with its negative terminal to −ve and its positive to point A. Check that the tick-tock sound is generated and that its rate and volume are variable via presets PR4 and PR5 respectively.

Now connect B3 (9 V) in place, with its negative terminal to point A and its positive to +ve. Close tune test switch SW6 and check that the circuit randomly generates a selection of tunes, with the tempo, tone and volume variable via presets PR1, PR2 and PR3 respectively. The structure of the circuit is such that the musical alarm randomly runs through a selection of eight tunes, with an undefined pause between each tune. A total of three selections (24 tunes) are available and can be chosen by SW4 (see Table 1).

When you are satisfied that the circuitry is functioning correctly, remove all batteries (including B1), fit the speaker and switches SW4, 5 and 6 permanently in place in the case and complete the interwiring, taking special care over the connections between SW5 and the CM172 clock module. Finally, refit all batteries and give the musical alarm clock a functional check as follows.

Final Testing

Set the correct (real) time and the alarm time with SW2 and PB1-PB2, with the alarm time a minute or two away from real time. Turn the alarm function on via SW3. When the alarm



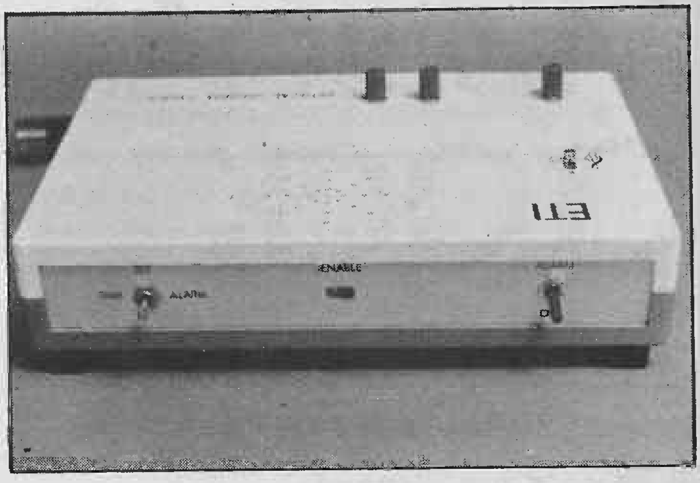


Fig.1. The back panel of the Musical Alarm Clock, showing the special switch-enable slide switch. This ensures that functions and time settings cannot be inadvertently changed.

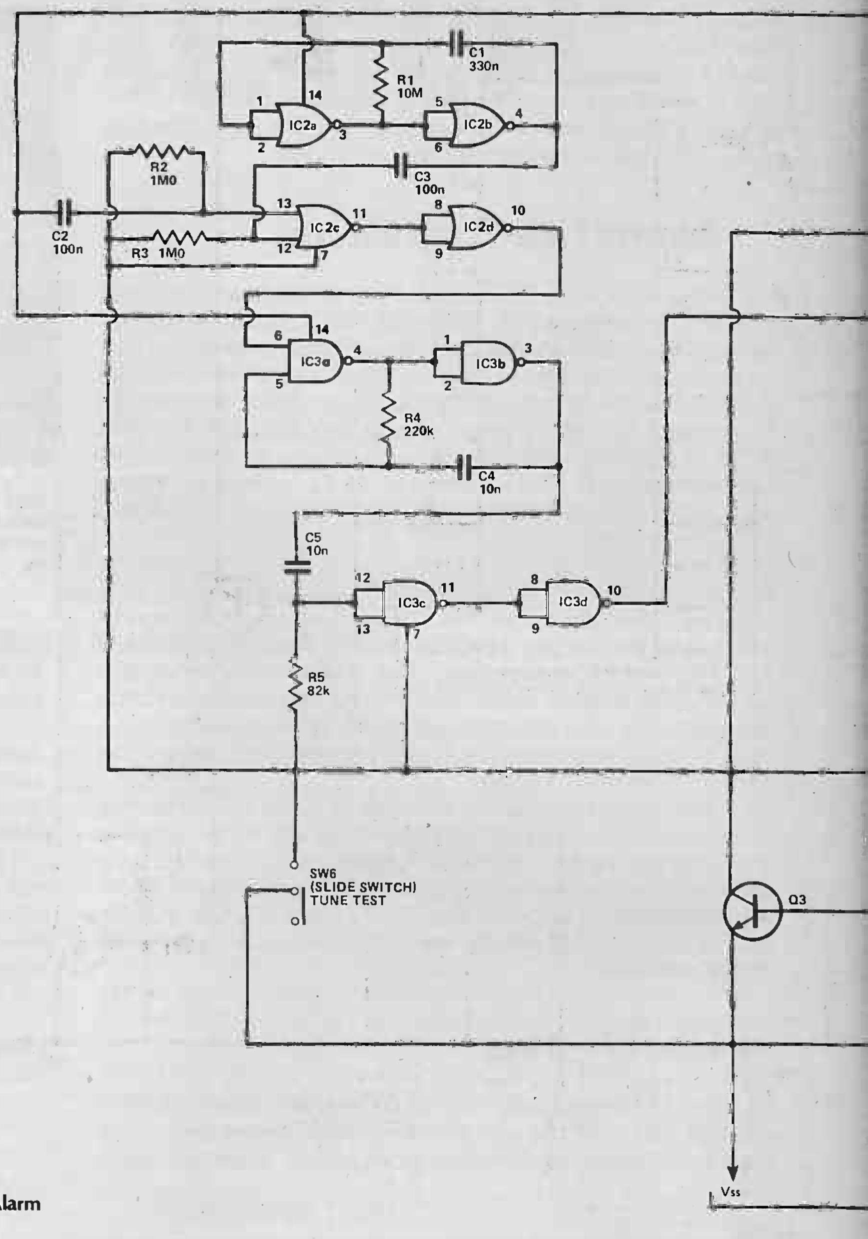
time is reached, the alarm should be generated. The alarm will be musical with SW5 in the TUNE position or will be a pulsed-tone signal with SW5 in the TONE position.

Once the alarm has activated, it will continue to sound for about 12 minutes and then automatically turn off. If the snooze button is pressed when the alarm is activated, the alarm will immediately turn off and then automatically activate again after ten minutes. If SW3 is turned permanently off once the alarm has activated, the alarm will immediately turn off and remain off. If SW3 is momentarily turned off and then on again once the alarm has activated, the alarm will immediately turn off but will re-activate when the set alarm time is reached the following day.

IC OUTPUTS		TUNES
	R_0	Greensleeves
	R_1	God Save the Queen
	R_2	Rule Britannia
	R_3	Land of Hope and Glory
K1	R_4	Sailors' Hornpipe
	R_5	Westminster Chimes
	R_6	Oranges and Lemons
	R_7	Oh Come All Ye Faithful
	R_0	Cook House Door
	R_1	The Stars and Stripes
	R_2	Beethoven's Ode to Joy (9th)
	R_3	William Tell Overture
K2	R_4	Red Flag / Maryland / Tannenbaum
	R_5	Great Gate of Kiev
	R_6	Twinkle Twinkle Little Star
	R_7	Soldiers' Chorus (Faust)
	R_0	Fate Knocking (Beethoven)
	R_1	The Marseillaise
	R_2	Deutschland Uber Alles
	R_3	Toccata in D Minor (Bach)
K4	R_4	The Lorelei
	R_5	Wedding March (Mendelssohn)
	R_6	Colonel Bogie
	R_7	Mozart

Table 1. Tune repertoire of the musical alarm clock.

Fig.2 (Right): Circuit diagram for the ETI Musical Alarm Clock, showing alarm and tick-tock circuitry.



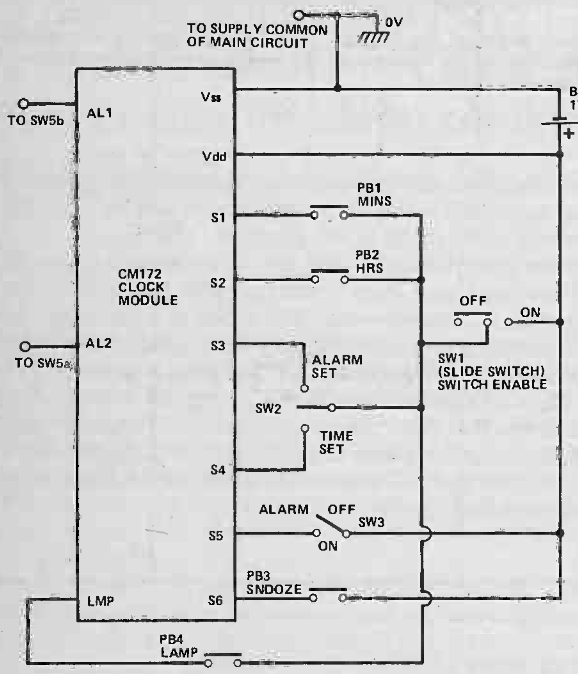


Fig.3 (Above): Connections of the basic clock module for the Musical Alarm Clock.

HOW IT WORKS

The basic CM172 clock module is a ready-built, LCD-readout unit that forms the heart of our project. The module is powered by a 1V5 cell. Note that we've wired an enable slide-switch in series with the alarm-set and time-set switches/buttons and with the back-light lamp button, to guard against inadvertent operation of these mechanisms.

The CM172 module has two alarm-out terminals. In our particular application, a pulsed 1024 Hz tone (1 S on, 1 S off) is available at the AL1 output and a simple high potential is available at the AL2 output under the alarm condition. These outputs are available for a maximum of 12 minutes under the alarm condition; once activated, they can be temporarily cancelled (for ten minutes) by briefly closing snooze button PB3 or can be fully cancelled by momentarily or permanently turning SW3 off.

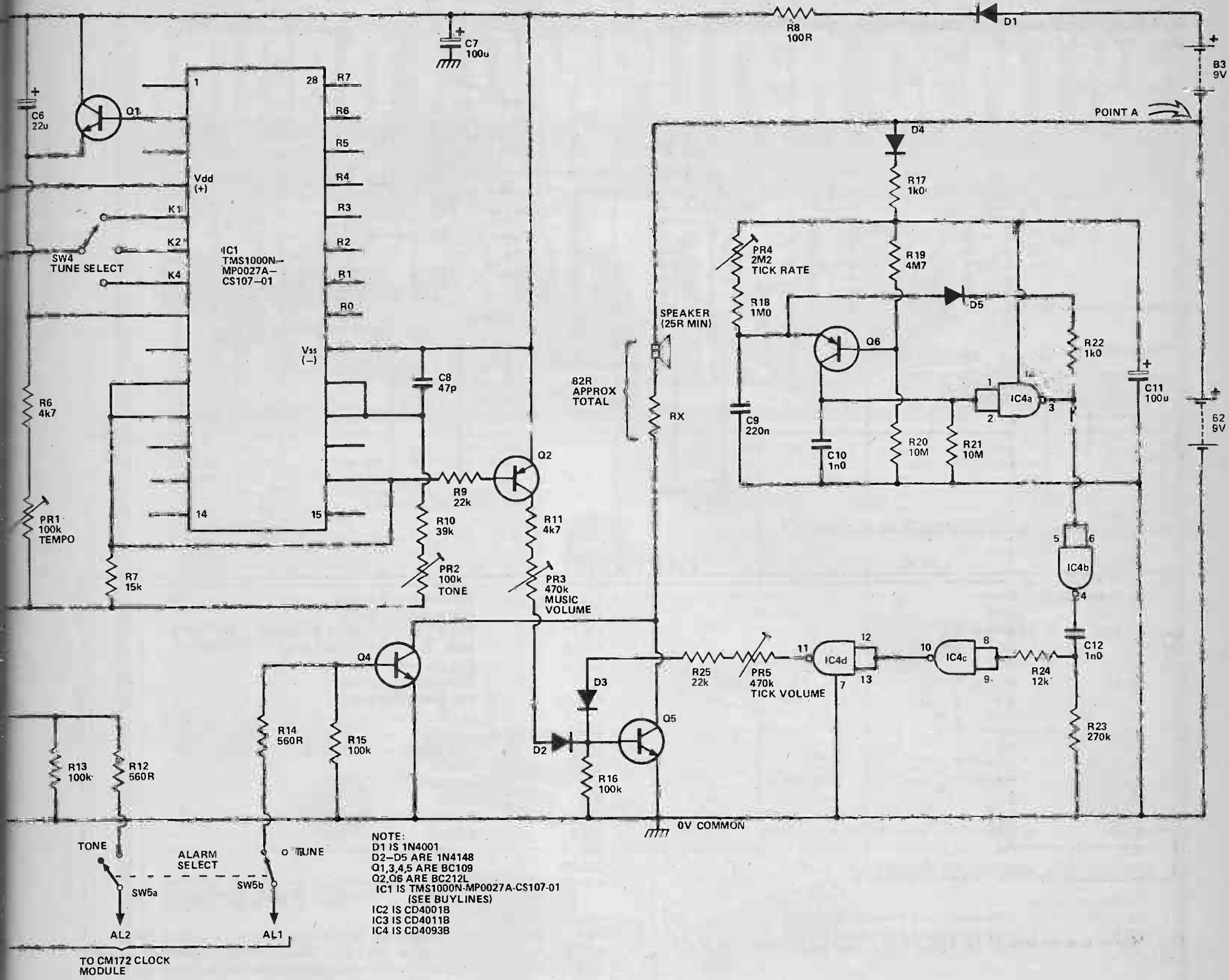
The CM172 circuit can be used as a stand-alone project, its operation being quite independent of the tick-tock and musical-alarm circuits used in the remainder of our alarm clock project.

The tick-tock circuitry is powered from a 9 V supply (B2) and is designed around Q6-IC4 and Q5. A prime consideration in the design of this circuit is that of very low mean current consumption. The complete circuit typically consumes a mere 9 uA or so.

Q6 and IC4a are wired as a special astable circuit, powered from B2 via D4-R17 and C11. At the start of each astable cycle, C9 is discharged, Q6 is cut off and IC4a output is high. C9 then charges exponentially via PR4 and R18 until eventually Q6 becomes forward biased and switches the output of IC4a (a Schmitt gate) low, at which point C9 is rapidly discharged via D5 and R22 and the operating cycle is complete. The cycle then repeats ad-infinitum.

The output of IC4a is then fed to differentiating network C12-R23 via the IC4b buffer. Each differentiated spike is then converted to a clean pulse with a width of about 200 uA by IC4c and IC4d and is fed to pulse amplifier Q5 via the PR5 volume control and gate diode D3. Consequently, a very narrow current pulse is periodically fed to the speaker and causes the speaker (because of its inductive nature) to ring at its natural resonant frequency and produce a characteristic tick sound (note that this design should more correctly be called a tick-tick circuit). The tick rate can be varied by PR4 to suit individual tastes.

The heart of the musical alarm is IC1, a microcomputer chip that contains a simple microprocessor together with a ROM pre-programmed with 24 tunes in digital form. Any one of these tunes can



be selected by connecting the appropriate one of the R_0-R_7 terminals to the appropriate one of the K1, K2 or K4 terminals. Once a tune has been selected, it is made available in digitised analogue form across R7. The tune tempo is varied by PR1, the tone by PR2. The tune generator can be turned on by driving Q3 on or by closing SW6. In our application, the digitised analogue output of the tune generator is fed to the speaker via Q2-PR3 and via Q5 and gate diode D2.

The most unusual feature of our circuit is the method of tune selection and to understand this you need to know how IC1 actually works. IC1 normally selects tunes by sequentially scanning the R_0-R_7 pins to see if a connection exists between any of these terminals and any of the K pins, it being the time-coincidence of the feedback pulse to a K pin that causes actual selection. The scan frame is 10 mS, the scan rate is 450 uS and the scan pulse width is 33 uS.

In principle, therefore, it is possible to select a tune by simply feeding a narrow positive pulse to the appropriate K terminal, ignoring the R pins completely. If this pulse is frame-locked and time delayed, any required tune can be selected. If the pulse is not frame-locked, tunes will be selected at random from the available repertoire. Our circuit uses this latter principle to implement random tune selection.

In our circuit IC2a and IC2b are wired as a slow (several seconds) astable. The output of this astable is converted to a periodic pulse (width about 100 mS) via C3-R3 and IC2c-IC2d and is used to gate a second astable comprising IC3a-IC3b. This astable has a period of about 3 mS and its output is converted to a 0.5 mS pulse via C5-R5 and IC3c-IC3d and fed to one of the K terminals of IC1 via SW4. The K signal thus comprises an occasional (once every few seconds) burst of 0.5 mS pulses with a 3 mS rate and causes successive tunes to be effectively selected at random.

Note that C2 and R2 are used in our circuit to ensure that the very first tune that is selected under the alarm condition is almost invariably the same, with the subsequent tunes being selected at random.

With SW5 in the tone position, the AL2 output of the clock module has no effect on our music generator circuit and the AL1 pulsed-tone signal of the clock module is fed to the speaker via Q4. When SW5 is in the tune position the AL1 output has no effect and the AL2 output is used to drive on Q3 via R12 and so activate the musical alarm circuit.

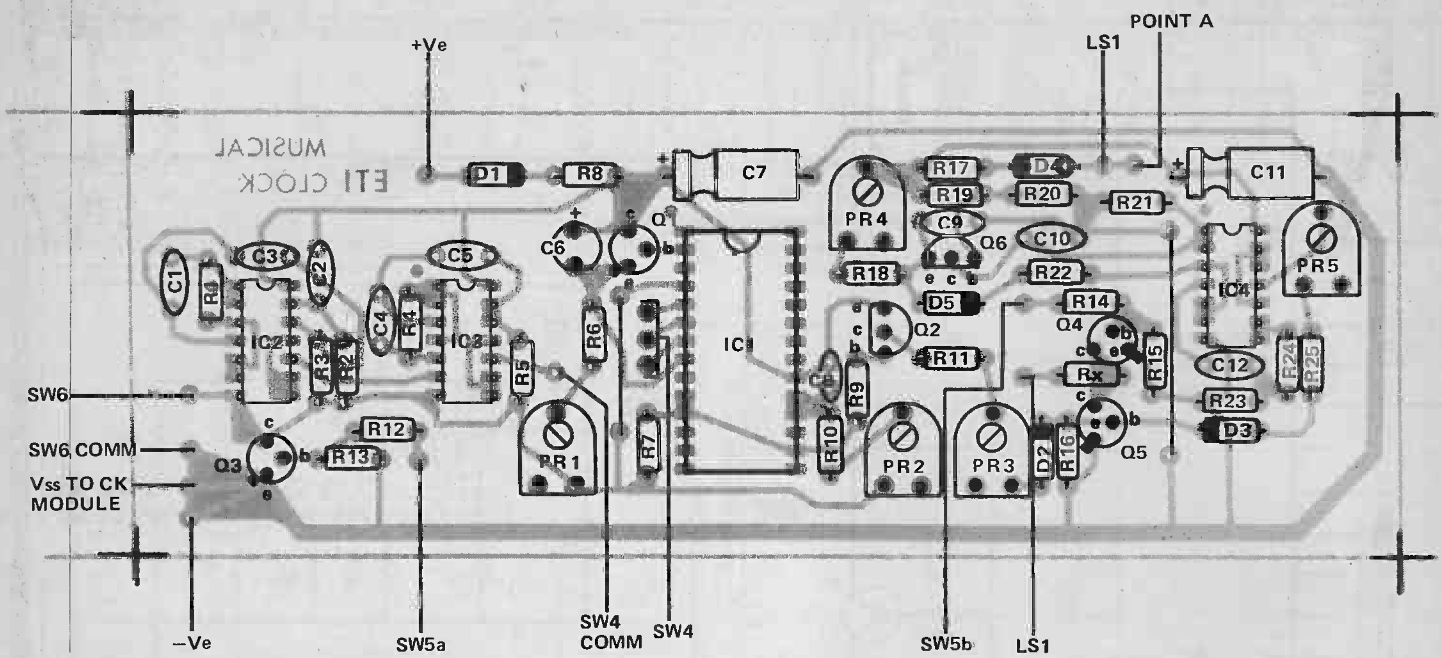


Fig.4 Component overlay for ETI Musical Alarm Clock.

PARTS LIST

Resistors All ¼ W 5%

R1,20,21	10M
R2,3,18	1M0
R4	220k
R5	82k
R6,11	4k7
R7	15k
R8	100R
R9,25	22k
R10	39k
R12,14	560R
R13,15,16	100k
R17,22	1k0
R19	4M7
R23	270k
R24	12k
Rx	(see circuit diagram)

Potentiometers

PR1,2	100k miniature horizontal preset
PR3,5	470k miniature horizontal preset
PR4	2M2 miniature horizontal preset

Capacitors

C1	330n polycarbonate
C2,3	100n polycarbonate
C4,5	10n polycarbonate
C6	22u 25 V electrolytic PCB type
C7,11	100u 25 V electrolytic axial
C8	47p ceramic
C9	220n polycarbonate
C10,12	1n0 polycarbonate

Semiconductors

IC1	TMS1000N — MP0027A
IC2	4001B
IC3	4011B
IC4	4093B
Q1,3,4,5	BC109
Q2,6	BC212L
D1	1N4001
D2-D5	1N4148
CM172	clock module

Miscellaneous

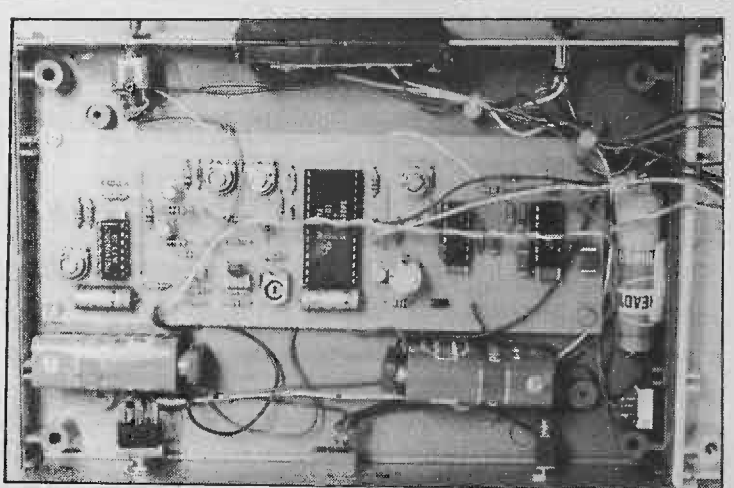
PB1,2,3,4 miniature push buttons; SW1,6 miniature slide switch; SW2 DPDT Centre off biased toggle; SW3 SPST miniature toggle; SW4 three position slide switch; SW5 DPDT miniature toggle; Vero case (order code 202-21034) plus tilt stand (202-21300G), PCB, Loudspeaker (see circuit diagram), Bezel.

BUYLINES

The clock module (CM172) and bezel is available from Ambit International. Most other components (except IC1) should be readily available from major stockists. Ambit International, 200 North Service Road, Brentwood, Essex.

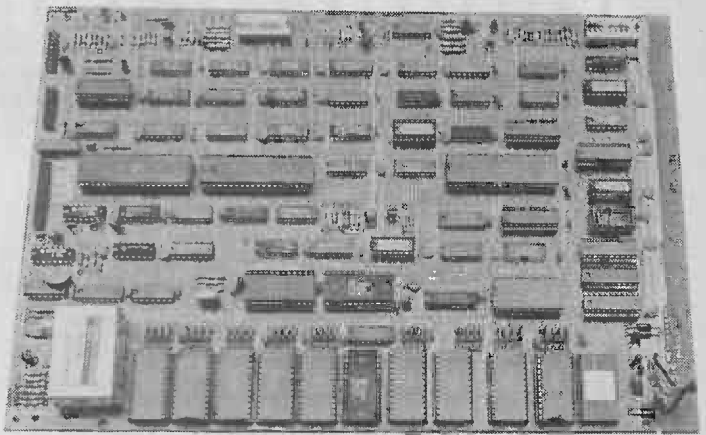
IC1 is a special pre-programmed micro-computer chip and is exclusively available from Chromatronics, River Way, Harlow, Essex, under the designation TMS1000N — MP0027A. The chip costs about £5.

The photograph (right) shows the layout of our clock. Note the inclusion of three batteries.



MICROBASICS

This month Microbasics looks at the Ins and Outs of interfacing with computers and the special chips used. Henry Budgett explains the initials and the innards.



The NASCOM 2 is well equipped with I/O. Both the PIO (a PIA relative) and the UART can be seen centre left.

We have now reached the stage where we have a perfectly useable microcomputer system, except for one small point. We can neither communicate with it nor it with us! The electronic circuits that control these communications with the outside world are called interfaces and may, in general terms, be regarded as being one of three types; parallel, serial or system.

In this part of the series I shall deal with each in turn, leading up to the final part of the series on hardware aspects of microcomputers. Further episodes will deal with the complete system rather than the fundamentals.

Byte Sized Pieces

Parallel interfaces are generally used for local communication. They are extremely useful for transferring large amounts of data over reasonably short distances at high speed. Each transfer consists of a complete byte of data (hence the term parallel) and the interface must therefore consist of at least nine connections (the ninth being an earth or ground plane). In most cases there are more connections. These are controls which ensure that the data is sent only when needed and not sent when the receiving device isn't ready. This controlled transfer of data is known as 'handshaking' and varies in complexity from a simple strobe line, as is commonly found with ASCII keyboards, to a full-blown, defined standard such as the BS4421 interface, seldom seen nowadays outside specific institutions.

Versatile Interface

In implementing a parallel interface on a micro one generally uses a device such as a PIA (Programmable Interface Adaptor) or its close relative the VIA (Versatile Interface Adaptor). These come in a wide variety of guises but are essentially the same. The device contains a number of registers which can be directly accessed as memory locations by the programmer. Into these one may load information which will control the way the device behaves. A block diagram of a typical device is shown in Fig. 1. Basically it consists of two 'ports' (a fairly obvious comparison) which may be set to operate as either inputs or as outputs. This behaviour is determined by the information set into the DDR (Data Direction Register). A logic '1' will select output and a logic '0' will select input.

To control the handshake there are usually two control lines provided per port. Each may be programmed to operate in a variety of modes, but a typical set-up is to have one line operating as a strobe. This signal is generated internally by the device and signals the data acceptor (that's the thing at the other end of the wire) that the data present on the lines at that instant is 'valid'. When the acceptor has taken the data off the lines it then signals back on the other control line that it has done so. This is the 'handshake'.

The programmer can monitor all these goings on by looking at the various registers within the device (they appear to be memory locations) and inspecting the various data patterns. The job is thus made very easy for the programmer and the computer designer, unlike the days of minis.

Prior to the development of these devices, I/O, the ubiquitous acronym for Input and Output, was a complicated job involving logic to decode the various address and control busses and using the mildly complicated system of interrupts to find out which device needed to be serviced. Interrupts come outside the sphere of a hardware-based series because they are mainly dealt with in software.

Bits Of Information

The second interfacing method is serial connection. This, in theory if not in general practice, requires only two wires — one signal and one earth. A typical example of the species is the RS 232C interface, defined by the international telecommunications body CCITT. This (and most other examples) is driven from another fiendishly clever device acronymically known as a UART or Universal Asynchronous Receiver Transmitter.

In many of the modern devices this appears, like its parallel relative, as so much memory to the programmer and all the functions may be pre-defined in software

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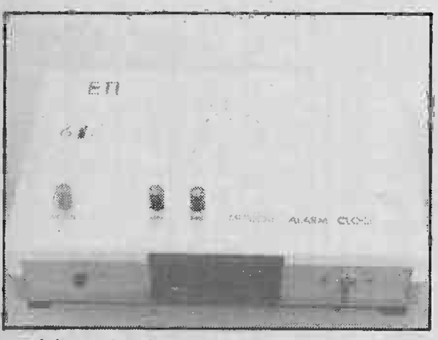
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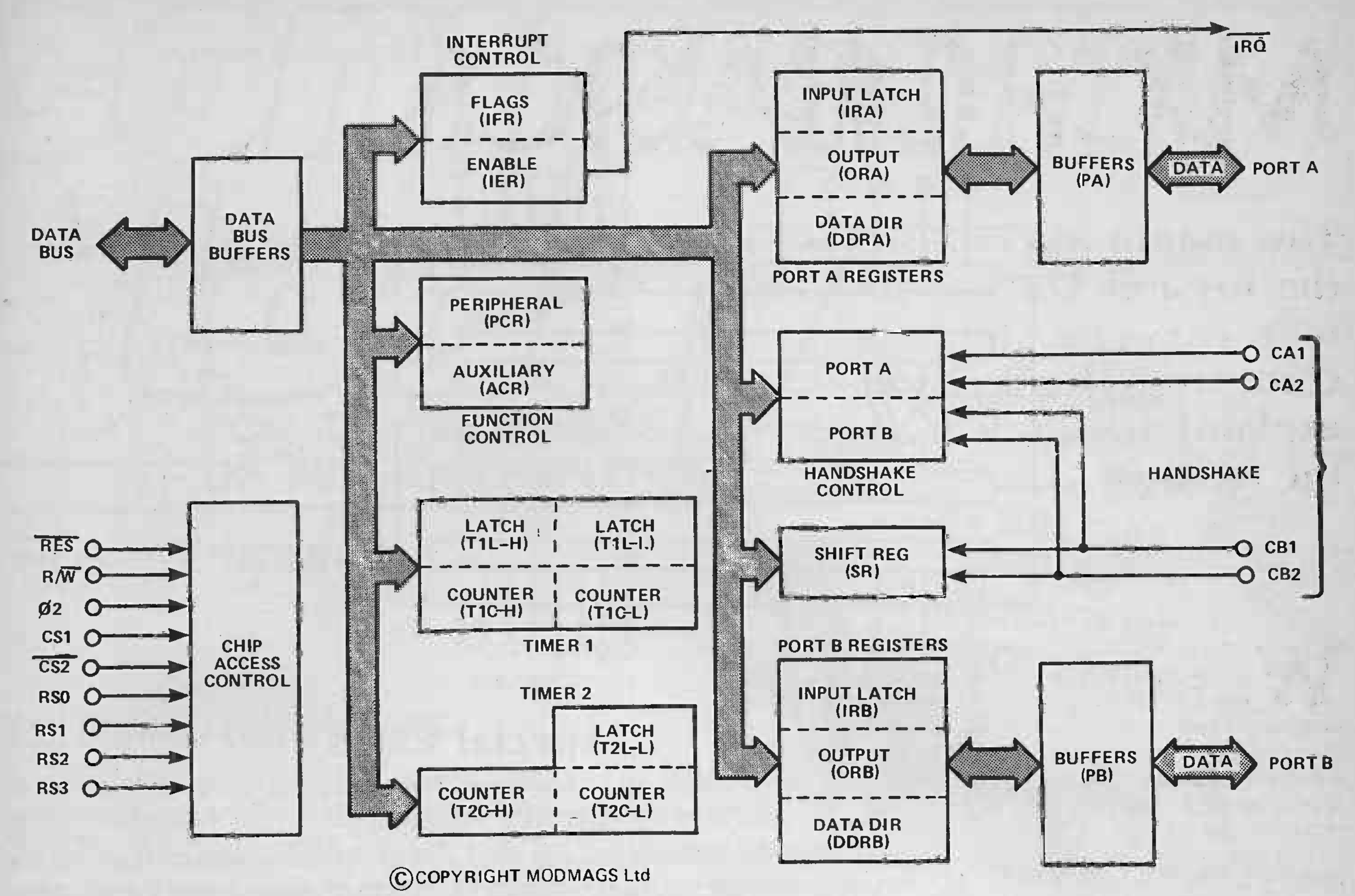


Fig.1 A typical VIA device, the 6522, showing the various internal control registers and handshaking lines.

CONNECTOR PIN	FUNCTION	DIRECTION OF SIGNAL
1	PROTECTIVE GROUND	DATA
2	TRANSMITTED DATA	SOURCE→ACCEPTOR
3	RECEIVED DATA	ACCEPTOR→SOURCE
4	REQUEST TO SEND	SOURCE→ACCEPTOR
5	READY FOR SENDING	ACCEPTOR→SOURCE
6	DATA SET READY	ACCEPTOR→SOURCE
7	SIGNAL COMMON	
8	CARRIER DETECT	ACCEPTOR→SOURCE
9		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		
20	DATA TERMINAL READY	SOURCE→ACCEPTOR
21		
22		
23		
24		
25		

Table 1 The V24 signal interconnection scheme. The source can be a VDU (for example) and the acceptor a modem or even the host computer.

rather than hardware. A typical device is shown in Fig. 2.

Serial transmission has to follow a definite pattern or protocol, its version of handshaking. A typical set-up would be to transmit logic '1' until ready to go, then a single 'start' bit of logic '0' followed by the data and its associated parity bit and then to terminate with two bits of logic '1'. This pattern is shown in Fig. 3. It allows the receiving device to synchronise itself to the data.

The heart of any serial device is its clock. This generally runs at 16 times the actual speed of the transmission. The data is actually sampled at the eighth cycle which allows a certain percentage difference between the transmitter and receiver frequencies.

Quite obviously a protocol as simple as this will not prove very reliable at high speeds and several more control lines are added to perform a handshaking function. The signal connections of the V24, a close and, to all intents and purposes, identical relative, is given in Table 1. The British equivalent is based on BS 4505 Part 5 and is again regarded as compatible. For full details the relevant standards documentation should be obtained.

Serial transmissions are often made via the PSTN or Public Switched Telephone Network (to unscramble yet another acronym). This is done with a piece of equipment known as a MODEM. Yes, that's right, it's an acronym for Modulator/Demodulator. The device converts the serial data stream into two precise audio frequencies, which are sent over the telephone line and detected by a similar unit at the other end, which re-converts them into meaningful data.

Systems On The Move

The system interface is generally called the bus. It is a parallel collection of all the necessary data, address and control signals required to operate not only the main CPU board but all the memory and peripheral devices as well.

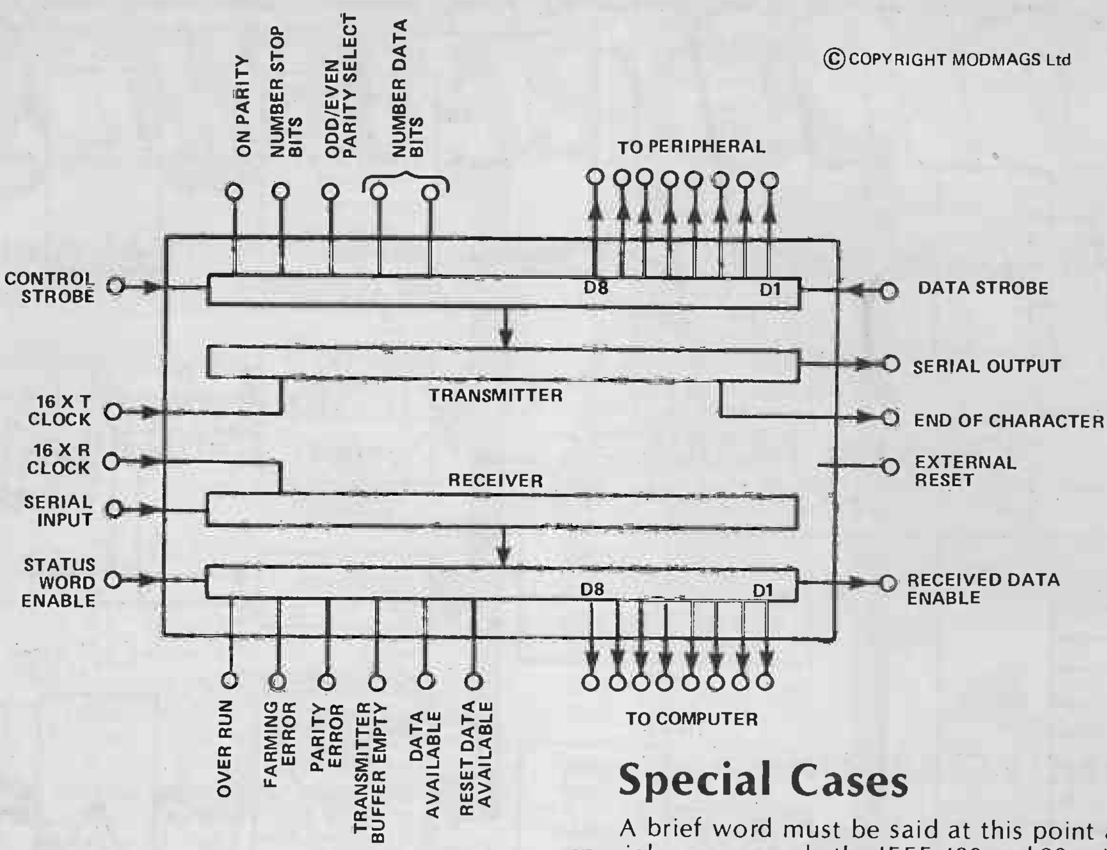


Fig.2 Block diagram of a typical UART.

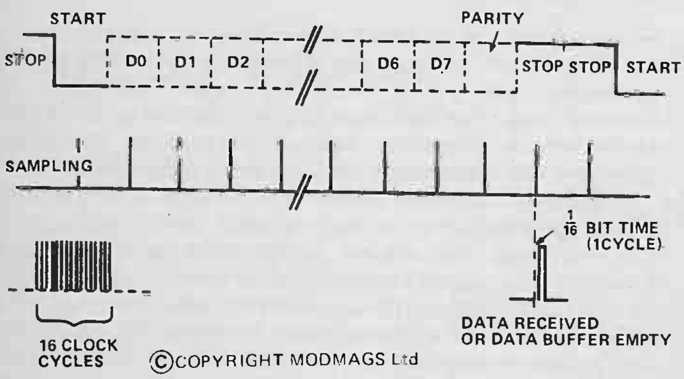


Fig.3 Timing diagram of a serial transmission.

Names such as S100, Eurobus, NASBUS, E78 and many more spring to mind, revealing that the world probably has rather more of them than is really necessary! By rights the system bus should be rigidly defined in both its operation and useage. One of the main reasons that the S100 had a bad name until the IEEE (Institute of Electronic and Electrical Engineers, of course!) got hold of it was its flexibility. This rigid defining process unfortunately takes time. We have now been waiting about three years for the E78, but perhaps even that will be slightly out of date when it arrives. If you are interested in the E78, they have produced a rather nice document, which should be available from The Secretary, E78 Committee, Avante House, 9 Bridge Street, Pinner, Middlesex HA5 3HR.

Special Cases

A brief word must be said at this point about some special cases, namely the IEEE-488 and 20 mA interfaces. The IEEE-488 or General Purpose Interface Bus came to the micro world in a rather mutilated form as the Commodore PET. Designed originally as a high speed data highway it allows multiple talkers (sources) and listeners (acceptors) to communicate along a single eight bit data bus. The bus is controlled by a number of signals and can be accessed by any of the devices. The original useage was in the laboratory where various recording instruments were connected to a single event recorder or data logger. In the case of the PET version, the pruning has resulted in one or two headaches among hardware designers, but it can be genuinely useful.

The original serial interfaces were designed around the teleprinter and teletype devices that are mainly built from solenoids. The device which controls these solenoids is very similar to the uniselectors found in old fashioned telephone exchanges and this is a current driven device. The 20 mA interface is built around a closed loop current source where the drive of 20 mA is turned on or off by a transistor which is in turn controlled by an external voltage. This allowed the interface to be electrically separated from the device, a considerable benefit as the back EMF generated by the coil of the uniselector could fry modern ICs and transistors to a turn if it escaped into the bowels of the computer.

In next month's episode we will be investigating the way in which all this magical, but rather dumb, hardware is kept together. The ingredient is, of course, that tenuous link of binary digits — software.

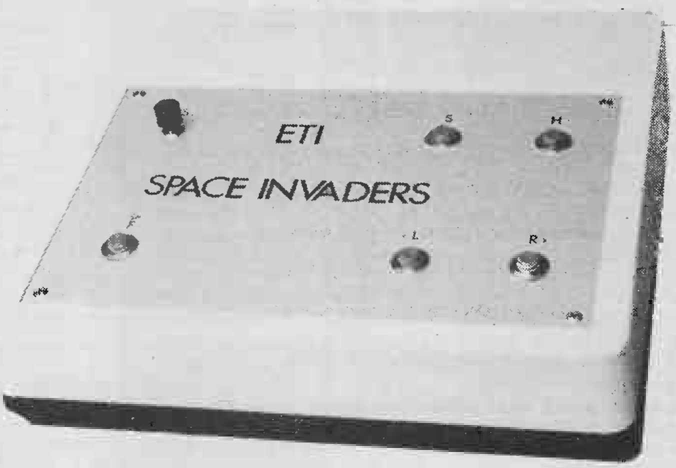
Post Haste

An excellent book has thudded onto my desk this month that is directly relevant to the hardware side of the series. Called 'The use of Microprocessors' by M Aumiaux and published by John Wiley & Sons it costs £12.00 in hardback and is highly recommended reading. The ISBN is 0 471 27689 8.

ETI

SPACE INVASION SOUND EFFECTS

Your first space invasion is but a moment away. We conclude our Invasion Game project with constructional details of the control and sound effects board. Design and development by Steve Ramsahadeo.



Last month we dealt with the main board and power supply, but before you can start repelling invaders you need a handful of controls. Our control board incorporates a sound effects generator — a must for this game.

There are four sounds in all — heartbeat, fire, saucer and explosion. The explosion is not produced by the sound board. It's taken care of by the software. When you press the start button (S on the front panel) the aliens start to advance across and down the screen, accompanied by the heartbeat — an ominous pulsing sound, which speeds up as the aliens gather momentum towards you.

The universe can only be saved for humanity and Nicholas Parsons if you fight back and destroy the invaders. When you fire, the sound generator bursts into life again to let everyone know that you mean business.

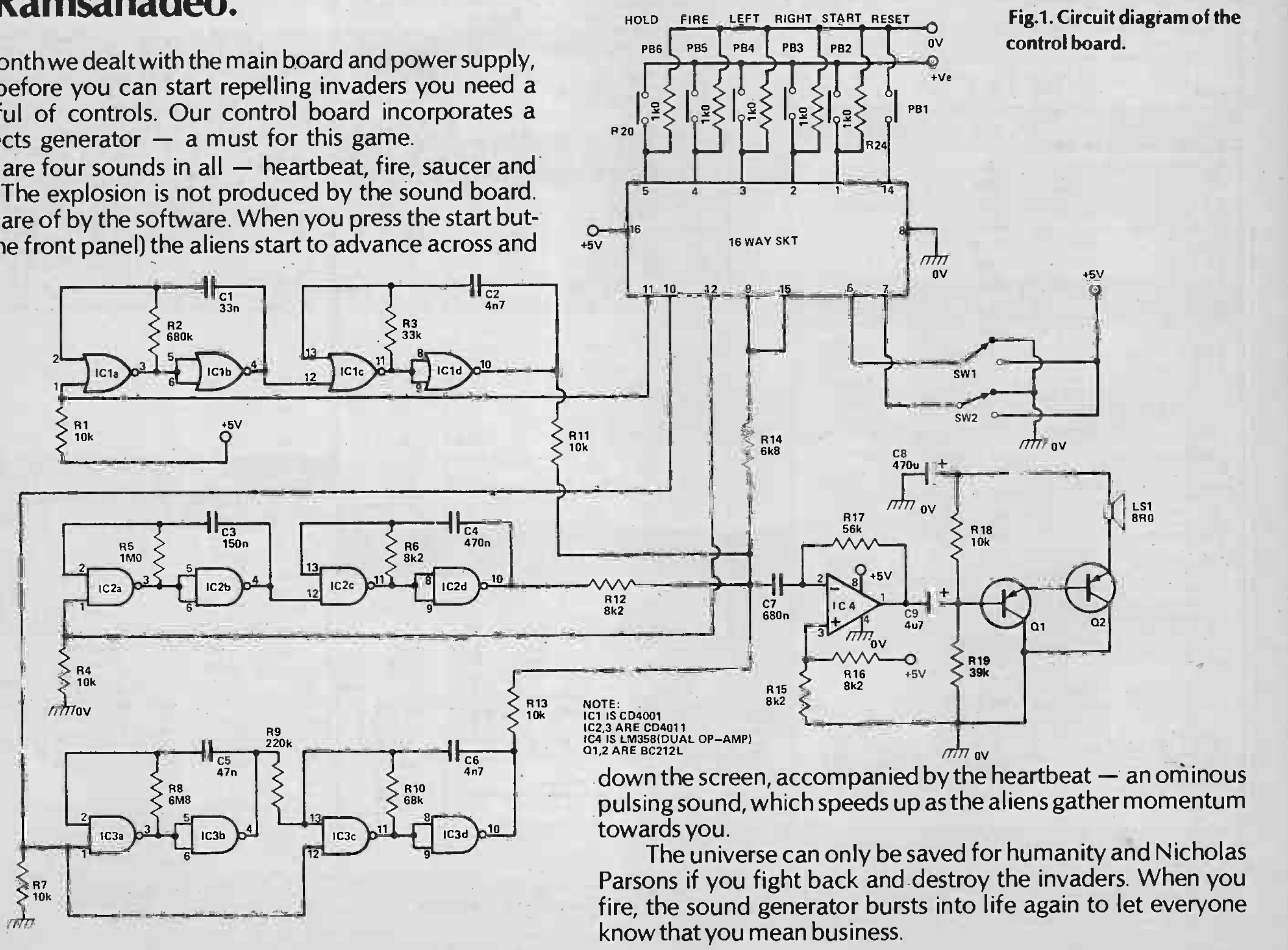


Fig.1. Circuit diagram of the control board.

BUYLINES

Tangerine Computers Ltd can supply the ETI Space Invasion project built for £99.85 all inclusive (£80.85 in kit form). The sound generator and keypad section is available built for £20.25 all inclusive (or £15.38 in kit form).

If you want to shop around for your own components, you can get the main PCB for £21.15 and the sound generator board for £5.60. The ROM is available for £17.75. Prices are all inclusive. Tangerine Computers Ltd, Forehill, Ely, Cambridgshire.

We built the project in a new plastic desk-top, sloping front case from Vero measuring 228x216x76mm (industrial order code 65-5033K). Vero Electronics Ltd, Industrial Estate, Chandler's Ford, Hampshire SO5 3ZR.

HOW IT WORKS

Logic instructions are transferred to and from the main board via a 16 way socket. Instructions go to IC27 on the main board via pins1-7 and 14 and return from IC26, main board via pins 9, 10, 11, 12. For example, when the fire button (PB5) is pressed, the main board detects this and gates pin 1 of IC1a low (logic 0) via pin 11 of the socket. IC1 is wired as a dual gated NOR astable and generates a pulsed tone signal to give the 'fire' sound.

IC2 and IC3 generate the heartbeat and saucer sounds in a similar way except that these are gated on by a high (logic 1) at the respective socket pins (12 and 10), and so NAND astables are required.

The hit sound is programmed into the software and is made available at pin 9 of the 16 way socket.

IC4 acts as a summing amplifier, combining at pin 2 the outputs of the three oscillators and the hit sound (via C7). R11-14 and R17 determine the relative gain of each of the sounds. The output of IC4 is then amplified by transistors Q1 and Q2.

PARTS LIST

Resistors all ¼ W 5%	
R1,4,7,11,13,18	10k
R2	680k
R3	33k
R6,12,15,16	8k2
R8	6M8
R9	220k
R10	68k
R14	6k8
R17	56k
R19	39k
R20 - 24	1k0
Capacitors	
C1	33n polycarbonate
C2, 6	4n7 polycarbonate
C3	150n polycarbonate
C4	470n polycarbonate
C5	47n polycarbonate
C7	680n polycarbonate
C8	470u 25 V electrolytic
C9	4u7 16 V tantalum
Semiconductors	
IC1	CD4001
IC2,3	CD4011
IC4	LM358
Q1,2	BC212L
Miscellaneous	
SW1,2	SPDT miniature toggle
PB1-6	momentary push buttons
LS1	2" diameter speaker
2 off 16 way DIL IDC (Insulation Displacement Connector)	
PCB	
Vero case	

You can gamble on boosting your score to an all-time record by ignoring the aliens for the moment and firing at a flying saucer, which zooms across the top of the screen every now and then. The appearance of the saucer is heralded by its very own unmistakable sonic trademark (in other words — the racket *it* makes is different from the others). Keep banging away at the aliens or divert your attack to the flying saucer? It's your decision.

Construction

The unit is assembled on one PCB with a separate power supply. Construction is straightforward provided that care is taken and attention paid to the orientation of all polarised components.

Begin by inserting all wire links and Veropins followed by the IC sockets, resistors, diodes, capacitors and transistors. The ICs should be inserted last, after all the wiring to the hardware components has been completed.

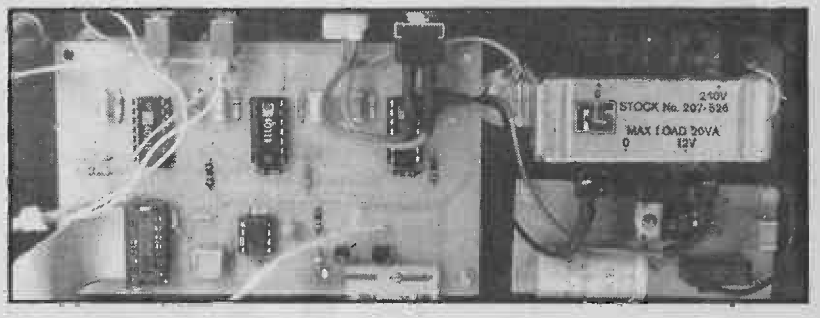
The PCB fits neatly in the top left-hand corner of the specified Verocase as shown in the accompanying photographs.

A 200 mm length of 16 way Speedbloc ribbon cable using DIL (dual-in-line) connectors at each end will have to be made up. This cable will link the main board and sound board, completing the logic and power supply connections.

The pushbuttons (PB1-6) and difficulty switches (SW1,2, mounted at the rear of the case along with the mains switch) can now be wired up.

Finally, the loudspeaker is mounted at the top of the case, a few drops of Super-glue being enough to fix it firmly into place.

After checking all is well, prepare for battle!



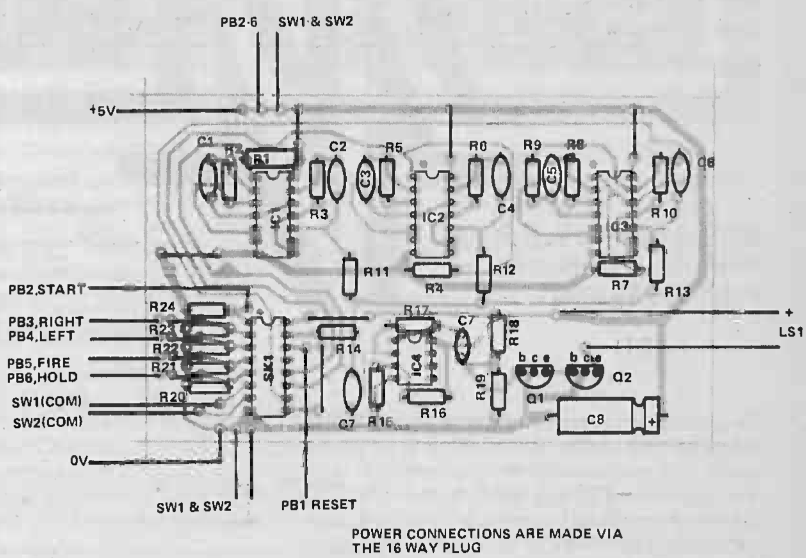


Fig.2. Component overlay.

ETI

AUDIOPHILE

Ron Harris lends an ear to a trio of record decks, from budget to top-of-the-range, and dons headphones to try the latest hip-mounted hi-fi.

This month has seen the launch of the first (of the many) Stowaway imitators — the Binatone "HipFi". (At least it's a better name.) I haven't had a chance to directly compare the two machines yet, but if the demonstration they put on at the press launch is anything to go by, Sony should be flattered but not worried.

The Hipfi sells at £59.95, as against £95 for the Stowaway and, at the launch anyway, the price differential was audibly apparent.

To say it was unimpressive is being polite. To say it sounded rotten is accurate. Impolite, but accurate.

There was a great deal of noise present in the sound, as though the treble was being boosted somewhere along the line, without the programme content at these frequencies to justify it. The headphones in use were comfortable enough, and folded flat for storage.

Overall, though, the sound was very undistinguished — rather like playing a decent cassette with the Dolby switched "out" on an encoded tape. HipFi, like Stowaway, has no Dolby — maybe Binatone were using encoded demo tapes without realising it? Most odd.

Tales Of Three Decks

Ever mindful of the call to consider budget audio more seriously within these pages, Audiophile this month takes a look at both ends of the Dual range of record decks.

The new CS505 has set a rumour rebounding around the halls that it is worthy of a high rank. Consideration was thus made with due interest and deference.

The 731Q is Dual's present top-of-the-range direct drive deck, with which the user is supplied a version of the Ortofon LM30 pickup — last seen in Audiophile hidden within an integral arm.

Just for good measure, Sony supplied me with a PSX 75 manifestation of their art, fitted with the new XL-44 moving coil pickup. Since this and the 731Q are direct competitors I thought the comparison might prove entertaining. The Dual has a slight edge in price, though, as the XL-44 is vended separately and will cost around £45 in most emporiums. It has been designed with the aim of bringing the virtues of the XL-55 Pro into more peoples' lives, by lowering the cost significantly.

CS505 — Table For All Budgets

This design is apparently meant to provide competition for such as Sansui's 222 and JVC's QLA5R decks. In order to reach a comparable price (approx £70) Dual have opted not to supply a fitted cartridge — the first time for them and a welcome development.

Features include a pitch control — unusual for a belt drive such as this — four point suspension for the whole turntable/arm assembly and a claimed ability to track high quality cartridges. Dual make great play of the arm fitted to the CS505 and on appearance it is a claim hard to justify. As you can see from the photo, it looks rather ordinary — old fashioned even.

Overall the deck is well constructed and is in stark contrast to many 'flash' oriental offerings. After seeing it around for a few weeks though, I *still* think it looks old fashioned!

Cartridge fitting is not easy by any means, but then without removable headshells or arms it never is. Dual's system of using a sub-carrier, to which the pickup is attached and which is clicked into place by the finger lift is ingenious and works well.

I fitted the Empire 400TC cartridge initially, as I felt this representative of the units that a 505 would be expected to earn its living carrying around an LP groove. Arm friction was commendably low.

Resulting Results

For a deck at this price the CS505 came through the bench test well and barring accidents, or grievous bodily harm, it is unlikely to intrude audibly to any significant extent.

After a week or so of supplication I managed to insinuate the 505 into a system next to the Sansui 222 — the great competitor. The owner of said deck was not keen at first — methinks he feared the voice of comparison.

Above: the Dual CS505 hi-fi record deck. A good clean sound quality was obtained in play.

The PSX 75 sells as a mid-price (eh?) general purpose record deck, which has full electronic control of the arm. Photoelectric cells are used with small prisms (mounted inside the turntable mat!) to sense record size and there is push button movement control over the arm from outside the dust cover. Quartz-locked speed — naturally — and a 'dial-on' tracking weight control are also present. This means you can set the value while the deck is actually playing a test record. Neat, that.

What I do not like, however, is the lack of a separate bias control. Twiddling the tracking force automatically selects a value for bias and this cannot be altered. I found this meant I was having to set up cartridges to higher forces than were usually needed to obtain secure tracking. Not a vital issue, maybe, but an unnecessary hindrance.

After an evening's listening to the decks — using the Empire 400 — it transpired that both of us preferred the sound of the Dual. It came across more convincingly, with a much better mid range. Without trying a number of different cartridges, there is no ultimate way this can be taken to prove the Dual a better deck than the Sansui. It does seem, though, that the CS505 can be taken as a serious contender at least. Sound quality is undeniably good, being less muddled than most decks at the price.

Finishing Touches

Comparisons such as this allow some standard to be drawn, against which a number of people will be able to reference the facts. Physically the Dual is probably not as nice to use as the Sansui, but I would not expect this to deter many people. The final arbitor should be the sound quality and on that score the CS505 is a good product.

Baby Bio (Tracer) Growth

Biotracers are back and there are likely to be more of them this time. Last year Sony had a deck on the market with an electronic arm, in which all movement was sensed and controlled using linear motors. It cost around £600 and apparently sold "reasonably well".

The idea was obviously too good to waste on the idle rich and since then a horde of little Japanese genii has been beavering away to bring the marvels of bionic record playing to a wider audience. The result is the £270 PSX 75.

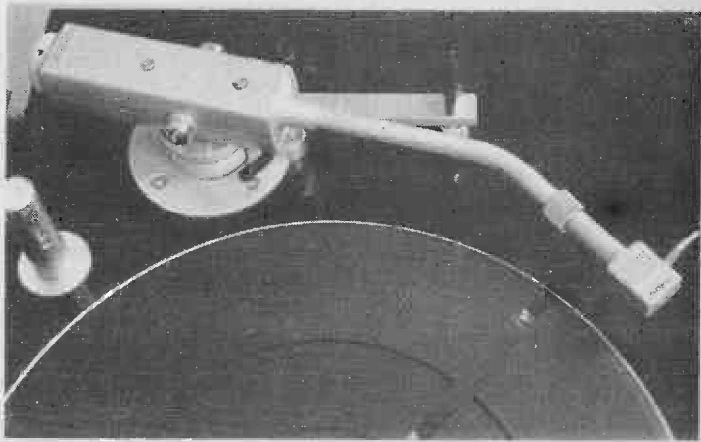
Sony PSX-75 bionic record deck. Note the pillar in the background and the prisms set into the turntable mat. These control record size indication.

Traces of Trackers

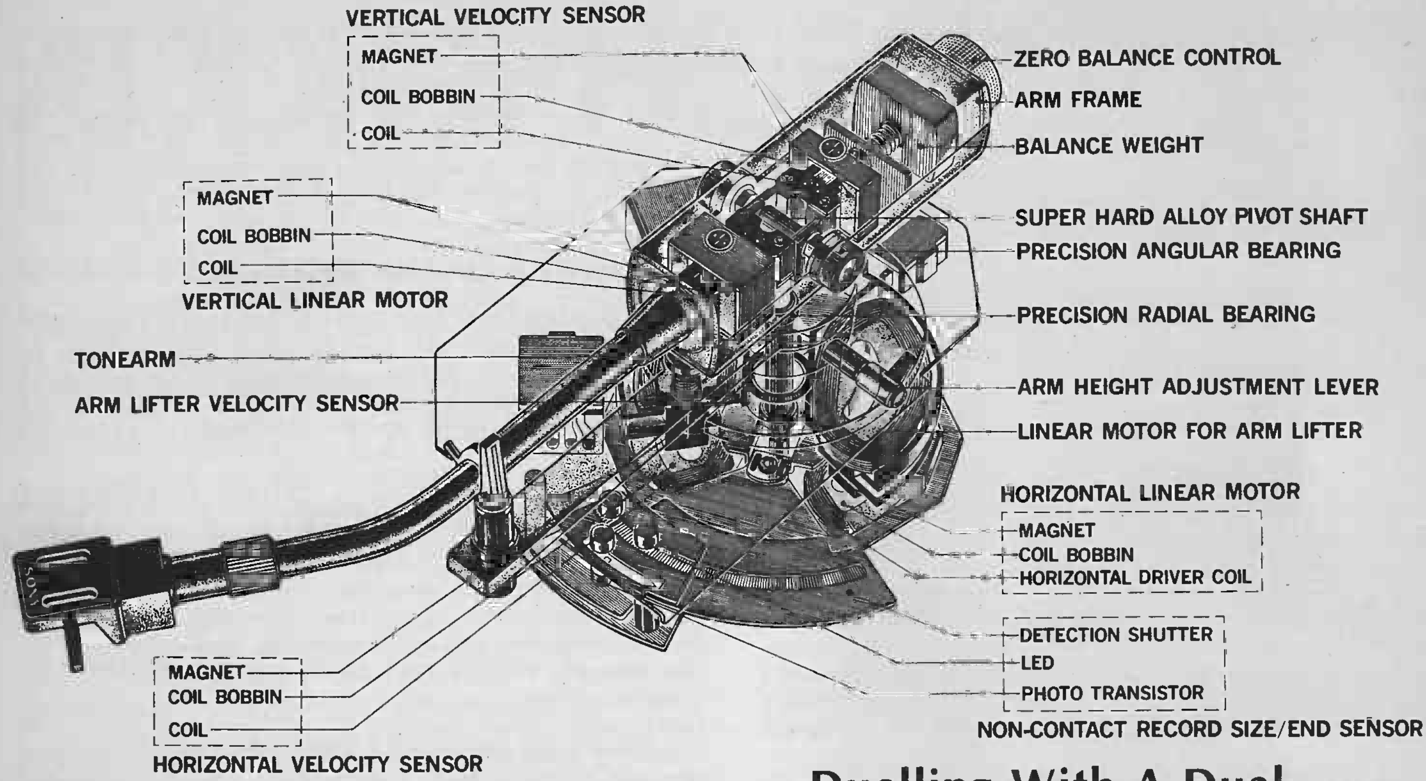
Rather than using up half the words in the Oxford Dictionary trying to describe the Biotracer principle, deflect your eyes onto the drawing for details. Incidentally, you too can have your very own copy of this artwork. All you have to do is buy a PSX 75. The diagram comes stuck on the dust cover as transparent film!

There is no setting up to do with this deck, save that required by the pickup. Arm height is adjustable too, so that vertical tracking angle can be optimised — even the arm rest moves up and down in time with the pillar!

Above: the XL-44 moving coil cartridge. It comes as part of its own headshell.



Above: the mighty linear-motor driven Biotracer pickup arm. On the next page you will find an exploded diagram to explain the clever bits!



Above: the ultimate in complexity? Note the use of two linear motors along each movement axis.

Sony supplied the deck with that XL-44 moving coil cartridge, their new baby. I tried it out with a number of head-amps, including the HA55 which Sony themselves make and recommend, obtaining similar results each time.

Disappointing I'm afraid. I liked the old XL-55 Pro, but I couldn't get on with the XL-44 at all. I found it dull, lacking in detail and with nothing to really justify the exchange of fifty pound notes. Surprising that, considering the pedigree, but it was always a relief to return to my reference pickup, a Coral MC81 (which sells around the same price incidentally) for the extra detail and life.

Loading Arms

The PSX 75 itself is an excellent performer, however. I tried a variety of top flight cartridges, including the Coral, and was returned good results with them all. That arm, bionic or not, has a fairly high mass and very high compliance units, ie Goldring G900 IGC, are best left alone. Moving coils work very well, so the loss is not an horrendous one. If you still harbour any lingering wisps of mistrust about auto-play decks, take a walk into a dealers and have a play with this one.

Despite the little prisms, I'd still recommend anyone who buys a PSX 75 to change the mat and play singles by use of the arm transport controls. Both the Metrocare mat and a GA Soundisc improved the mid-range performance of the deck, with the Soundisc cleaning up the bass considerably, too.

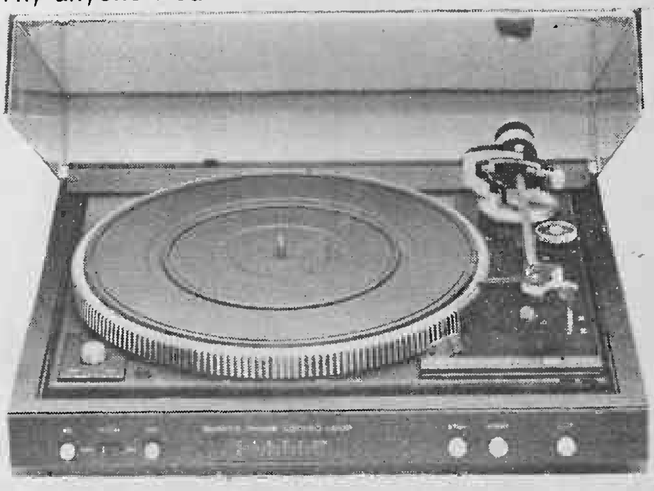
Overall, then, the deck is a good product, well suited to its intended market and sensibly priced. The finish is immaculate as one would expect, with the controls working well at all times.

Duelling With A Dual

As a direct competitor with the Sony, the Dual 731Q could not be more of a contrast. The arm is the exact opposite of the Biotracer. Dual have gone for refining the simplest method possible and concentrated upon resonance suppression. As you can see from the diagram, the counter weight is a complex affair and there is provision to 'tune' the arm/cartridge resonance for individual pickups, to give best bass performance.

Like all Dual decks (except the CS505) the 731Q arrives with resident cartridge — in this case an Ortofon. I reviewed the LM30 in Audiophile only recently (June 80 issue) and would refer Seekers Of The Tiniest Detail thence. Suffice it to say here, that this is a low mass, high compliance design, capable of excellent sound quality.

The 731Q has quartz speed lock and pitch control, just to be different. Here, however, the pitch lock is *not* disengaged if you alter the speed from nominal. Now you can have accurate 32 RPM, if the mood takes you. Why anyone would want a deck that ran at the wrong



Above: the 731Q top-line Dual. Quartz of course.

speed accurately is quite beyond me. Maybe the orchestra were a little slow one day? Speed 'em up lads! It's all quartz-locked!

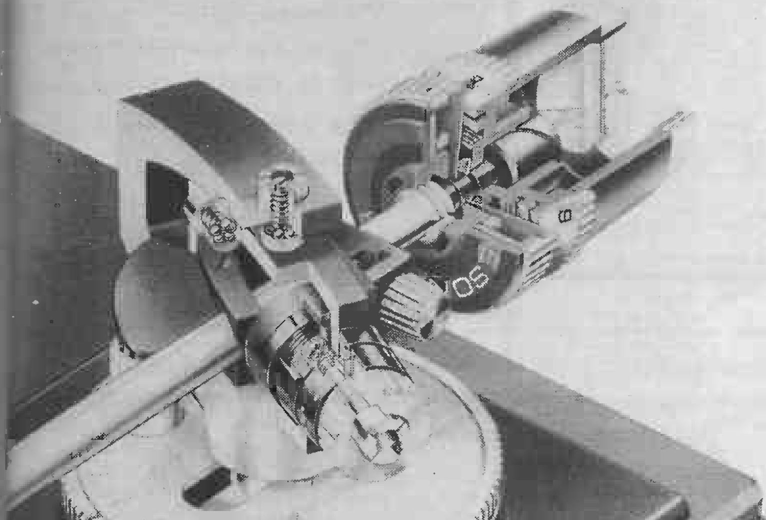
Very silly.

Once again, here is an expensive record deck with a mat fitted to it that would be better employed under a plant pot. In this case it is *really* bad and Dual should sell off their stocks, as floppy frisbees or something, quick, and look to something more substantial. The difference in the sound of the 731Q with a good sensible mat fitted was astounding.

With a Spectra or Metrocare in place the Dual/Ortofon is a fine combination and good value for money — as purchased, however, it simply does not allow the LM30 to do its stuff. Come on Dual, give yourselves a chance!

A cut-away of the Ortofon cartridge fitted to the 737 turntable. Note the spring fitting.

Above: being a bit fond of cut-away photos, Dual supplied this shot of the 731Q circuitry. More complex than a belt of rubber is it not?



Above: even the counter-weights are complex these days. This shows the Dual's ability to mass-tune the resonance of the pick-up arm and cartridge assembly. Dead clever these continentals.

Comparative Studies

I suppose having mentioned the competitive links between the two decks, the question which remains to be answered is which of the two, Sony or Dual actually represents better value for all that money. The nice thing here, for me, is that the two products are so totally different in their approach to a common end that sidestepping a final choice is easy!

The Sony offers a better finish, better ergonomics and leaves the choice (and the cost) of pickup to the user. The Dual offers a very high quality cartridge as part of the package, at a lower price overall, and a very good arm which is tunable for a range of cartridges. Trouble is, having supplied you with a pickup in the first place, chances are you're never gonna use the facility to its fullest extent.

If you like the sound of the Ortofon LM30, then the Dual 731Q will probably be for you. Anyone who doesn't is stuck with it, it seems. Including a cartridge in a deal with a turntable invariably means you're selling the sound of the cartridge rather than the deck. Dual must be happy with the idea at least.

If for no other reason than that, I'd tend to prefer the Sony and go my own way with pickups. Either way you're going to need a new turntable mat and a fairly big shelf to take the deck, as neither of them is exactly compact.

At the price it is worth noting that £210 would buy you a TD160S and SME Series IIIS — a combination which would sound better than either, albeit at the cost of considerably more trouble to set-up, adjust and use.

Trailing Off

Next month we're giving the once-over to the two Pioneer cassette decks, namely the CT-200, which is the new low price metal deck and the CTF-1250, the not-so-new, not-so-low-price flagship design.

	DUAL CS505	SONY PSX 75	DUAL CS731Q	
WOW AND FLUTTER:	±0.05%	±0.035%	±0.025%	(ALL DIN)
SIGNAL-TO-NOISE RATIO:	69 dB	79 dB	75 dB	(ALL DIN B)
START-UP TIME (to 33⅓):	2 S	1½ S	2 S	
SUSPENSION:	4 point — spring	—	4 point — spring	
TRACKING ERROR:	0.16°/cm	+2, −1°	0.16°/cm	
TRACKING WEIGHT RANGE:	0-3 g	0-3 g	0-2 g	
CARTRIDGE WEIGHT RANGE:	2 g-10 g	1 g-15 g	2 g-9 g	
APPROX. PRICE:	£70	£270	£240 (inc. cartridge)	

ETI

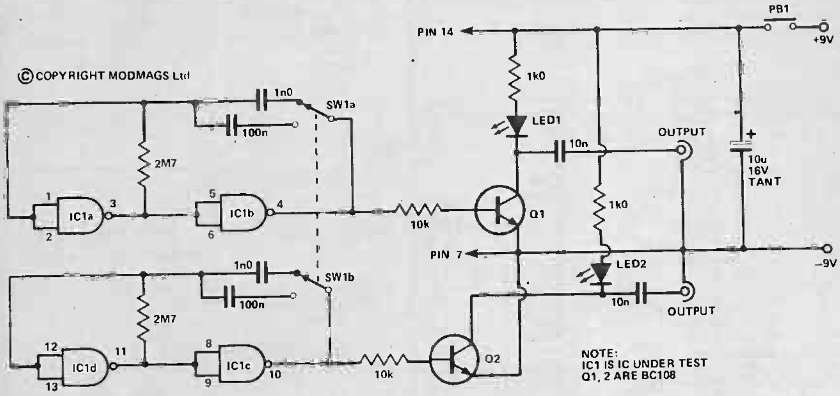
TECH TIPS

CMOS Tester

C. Jordan, Sompting

This circuit was designed to test 4001 and 4011 devices which were suspected of being damaged due to static from careless handling. Two gates on each side of the 14 pin DIL package are tested independently. Each pair is connected as an astable, the timing capacitor being switched to allow the device to oscillate at two different frequencies.

The output of the astable is fed to a transistor which acts as a buffer, driving the LED and providing an audio output to a crystal earpiece. PB1 should be a push-to-make type, so that it is impossible to insert or remove a device with the power on. Although intended to test 4001 and 4011 devices, it will also quite happily test 4030, 4071, 4077, 4081 and 4093.



Digital Clock Switch

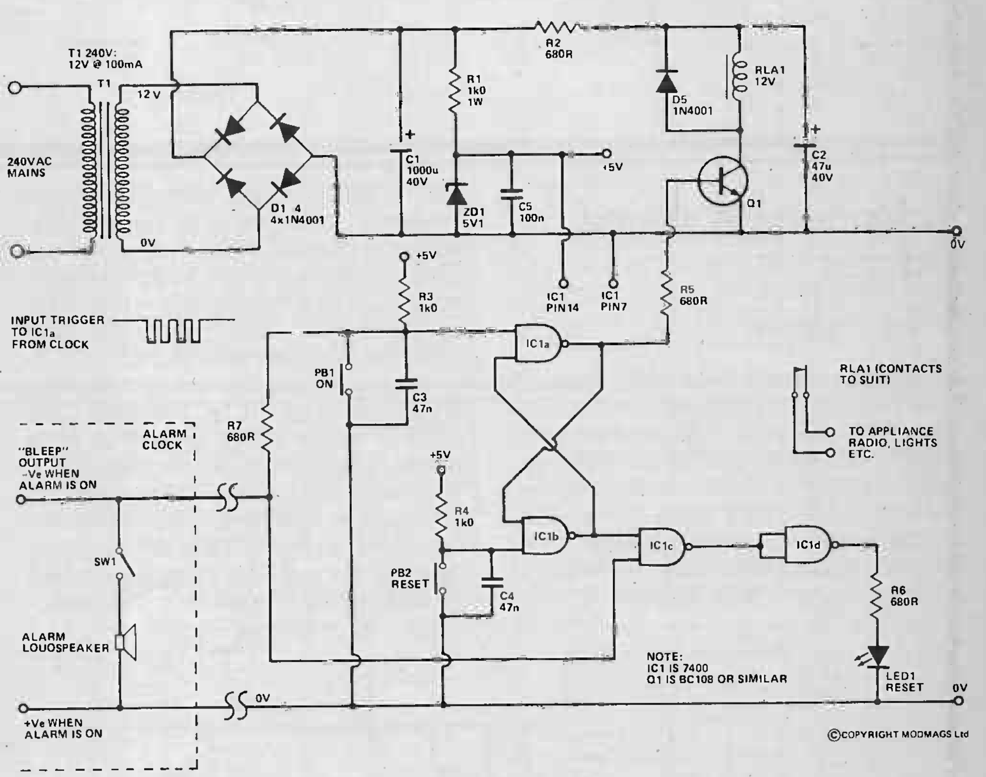
R.D. Pearson, Sheffield

This circuit was designed for a digital alarm clock such as the Hanimex alarm clock sold by ETI's marketplace. It will switch an appliance such as a light or radio on or off manually using push-button switches and will also switch the appliance on when a negative output from an alarm clock triggers it.

IC1a,b form a flip-flop driving the relay via Q1 or an LED via IC1c,d. Input triggers are provided by PB1 and PB2 which are suppressed by R3, C3 and R4, C4 to prevent erratic operation from mains transients and interference.

The ON input trigger to IC1a is driven by a negative pulse from a digital alarm clock. The easiest place to find this is on the loudspeaker of the clock using a multimeter or 'scope across the speaker terminals when the alarm is bleeping. The negative output is connected to the trigger input and the positive output to 0V.

IC1c,d provide a flashing LED indication that the alarm is on. If desired, switch SW1 may be connected in series with the loudspeaker.



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

Noise Limiter

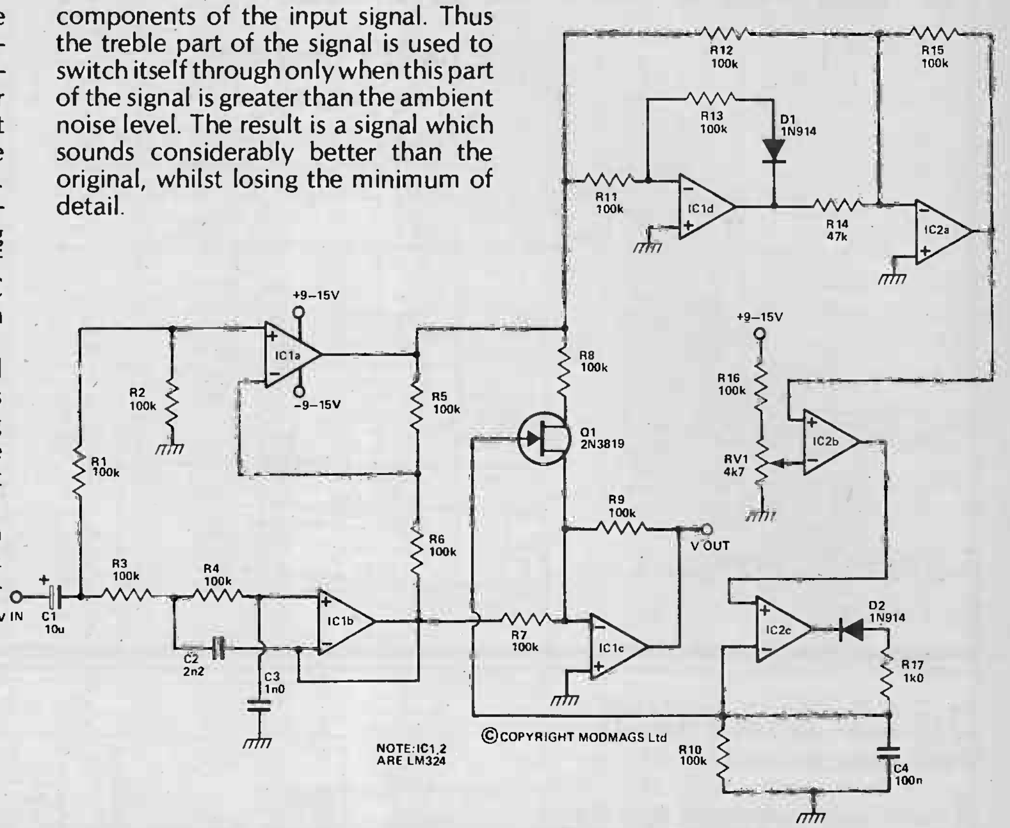
J.P. Macaulay, Crawley

This design depends on the masking effect of high frequency signals on the noise. IC1b, R3, R4, C1 and C2 form a second order filter with a turnover frequency of 1 kHz. Differential amplifier IC1a subtracts the bass-only output at the output of IC1b from the full range input signal to leave treble and noise. This signal is full wave precision rectified by IC1d and IC2a and the resulting DC applied to the non-inverting input of IC2b. The inverting input samples a DC voltage from the slider of RV1, which forms a voltage divider with R16.

IC2b is used without feedback and thus acts as a comparator. When RV1 is properly adjusted, a positive-going pulse appears at the output each time the treble component of the input signal exceeds the noise level.

This pulse is fed into IC2c which, with the associated components, forms a negative peak detecting and hold circuit.

This in turn is fed into the gate of Q1, which is used as a variable resistance in the feedback loop of IC1c. This last op amp is operated as a simple audio mixer which recombines the bass and treble components of the input signal. Thus the treble part of the signal is used to switch itself through only when this part of the signal is greater than the ambient noise level. The result is a signal which sounds considerably better than the original, whilst losing the minimum of detail.



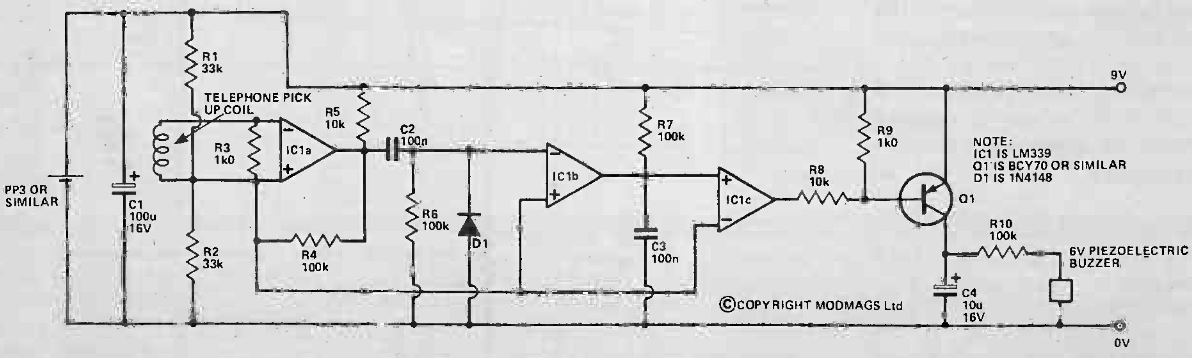
Telephone Warning Buzzer

D. Tate, Melksham

This circuit uses the LM339, a quad comparator with open collector outputs. IC1a has a cheap magnetic telephone pick-up coil (of the type supplied by Maplin which can stick to the side of a telephone using the built-in sucker) attached across its inputs. The positive input is also tied to half supply voltage using a low current resistor chain. R4 provides some feedback hysteresis. When no input signal is present, the output is dependent on the particular device in use, so it is decoupled from IC1b, whose output is then pulled high by R7. Thus comparator IC1c holds the transistor switched off.

When a signal is present, ie the telephone rings, IC1a oscillates, the positive edges being passed to IC1b whose output goes low, discharging C3. The time constant of R7/C3 ensures that the input to IC1c is held low while the telephone is ringing. Thus the transistor (any low current PNP type) is switched on and the buzzer sounds. The piezo-electric device used in the prototypes was a 6 V, 15 mA type supplied by Maplin.

Several of these have been built and a problem from electrical noise was apparent in one example. When the buzzer operated, it injected noise back into the battery supply so that the circuit would not stop buzzing. This was eliminated by using the low pass filter C4/R10. Further, R3 was required in one example to reduce input sensitivity. The circuit consumes about 1 mA in the off state and has the advantage over certain other designs in that it requires no setting up.



DIGEST

Hi-Fi At Your Hip

Yup! It had to happen. Binatone have produced the cheap alternative to Sony's Stowaway portable cassette player. Called the HipFi, it will retail at £59.95. Backed by a nationwide advertising campaign, it has loads of facilities:- separate volume control for each channel, tone control, add-on headphones (optional extra), fast forward and rewind with cue and review, fold flat headphones for easy carrying and a direct talk line for communicating with the outside world. The three pencil alkaline batteries provide approximately eight hours playing time. See this month's Audiophile for further details.

IMPORTANT NOTICE

Readers have recently confused Electronics Today Limited as being associated with the ownership of our magazine, Electronics Today International. Our magazine is owned by Modmags Limited, part of the Argus Press Holdings Limited Group of Companies.

Electronics Today Limited advertises in our Magazine, as "Metac", but so as to prevent any further confusion we wish to make it clear that Electronics Today Limited is not owned or managed by any member of the Argus Press Holdings Limited Group of Companies.

Multi Purpose

Further to our recent Digital Multimeter Survey, we have received details of Thurlby Electronics Ltd's new Model 1503 4¾ digit LCD Multimeter. It has an unusually long scale length of 32,768 counts (± 15 bits) and this enables it to monitor a 1 mV change in a 30 V power rail. Thirty measuring ranges are provided covering the five basic functions of DC and AC voltage, DC and AC current and resistance. Diode test and frequency measurement functions are also included. Maximum voltage input is 1200 V and current can be measured up to 10 A continuous, 25 A short term. The 1503 is an entirely British design and sells for £139. For further information contact Thurlby Electronics Ltd, Suite 1, Coach Mews, The Broadway, St Ives, Huntingdon, Cambs, PE17 4BN.

Breadboard 80

Whatever aspect of electronics you're into, make sure you're in London in November for this year's Breadboard exhibition. From CB to home computing, soldering to synthesisers — demonstrations, special offers — it's all at Breadboard 80.

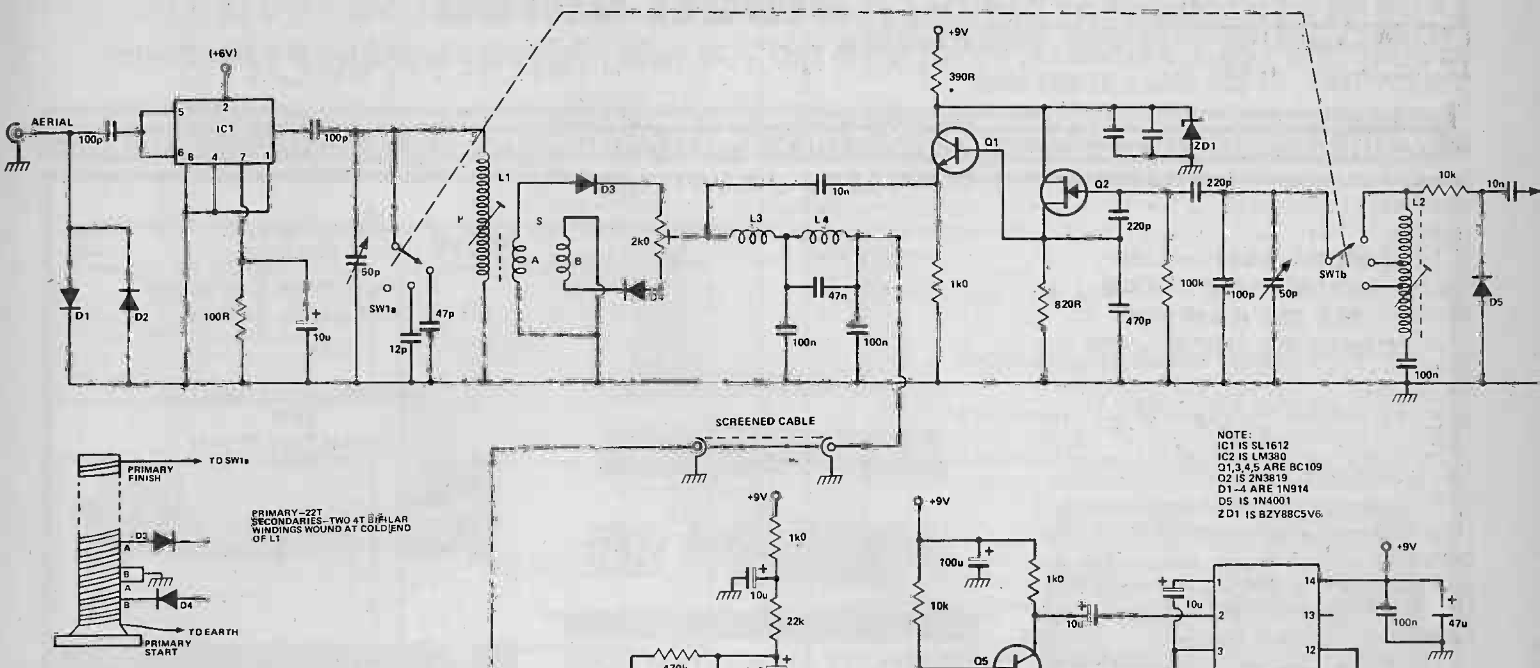
The exhibition runs for five days — the doors open at 10.00 am on November 26th (watch the electronics Press for full details). Don't miss it.

Jaws — OK Style

OK Machine & Tool's new 'AD' wire dispenser comes complete with its own teeth. It's a heavy duty cutting and stripping wire dispenser with precision ground steel cutters and die stamped stripping blade. The strip length can easily be adjusted from 9½-51 mm and the stripping blades can be changed to handle 24 or 30 AWG (0.5 or 0.25 mm) wire. 24 and 30 AWG dispensers are available with 15.2 and 30.5 m of wire respectively. Both offer a choice of insulation colour.

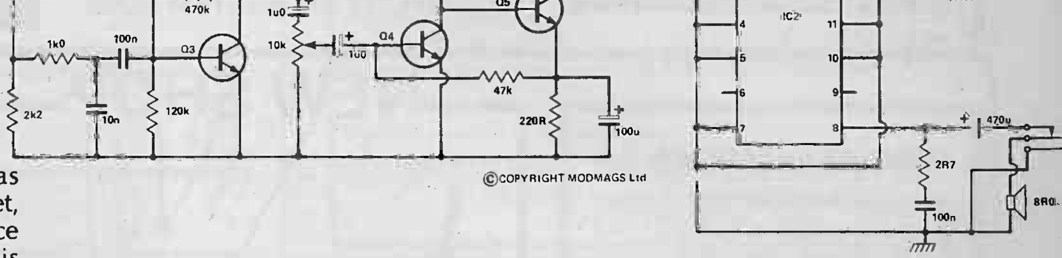
The transparent case nicely avoids that embarrassing situation when you sit down to do a job, pull out a length of wire and find that you're holding the last four inches of wire in your hand. You can see through OK's dispenser how much wire is left on the roll. Refill rolls are available and easily fitted.

For more information on the 'AD' cutting and stripping wire dispenser contact OK Machine & Tool (UK) Ltd, Dutton Lane, Eastleigh, Hampshire SO5 4AA.



Direct-Conversion Receiver

E. Vaughan, High Wycombe



A direct-conversion receiver has many advantages over the superhet, when looked at in terms of performance versus complexity and cost (this receiver can be built for around fifteen pounds or less). The SL1612 is an RF amplifier providing premixer gain. D1,2 protect the IC. The mixer is a balanced type, driving an M-derived single section lowpass filter with a cut-off frequency of 3 kHz. The oscillator is a Colpitts type followed by a buffer stage to prevent pulling of the oscillator when tuning. Although mainly for SSB and CW, AM can be resolved by tuning to zero beat. The circuit employs varicap tuning.

Using a wire aerial 66 feet long, separated at 33 feet by an insulator and fed with 75R coax, it is possible to pick up stations from all over Europe, North and South America and Africa. When correctly set up, this receiver should switch-tune to the 14 MHz, 21 MHz and 28 MHz amateur bands.

L1,2 are wound on ⅜ inch diameter slug tuned formers using 36 SWG wire. Winding details of L1 are shown in the diagram. L2 is 25 turns tapped at 9 and 14 turns. L3 is 1 mH and L4 10 mH.

Automatic Nicad Charger

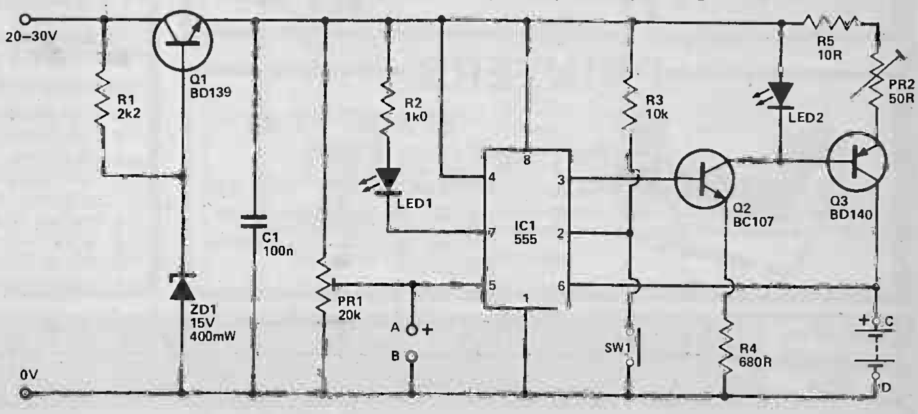
M.G. Baker, Port Elizabeth

This circuit will charge up to eight 1V25 Nicad cells at a constant current of up to 100 mA. When the battery voltage reaches a preset level, the charger trips and charging ceases.

The voltage of a Nicad cell increases as it is charged and reaches a value of approximately 1V45 when fully charged. A battery of eight cells would, therefore, be fully charged when its voltage is 11V6. The recommended charging current of a Nicad cell is usually 10% of its mAh rating ie 50 mA for a 500 mAh battery.

To set up the circuit, connect a voltmeter across terminals A and B. PR1 is adjusted to the desired trip voltage eg 11V6 for eight cells. A milliammeter is then connected between terminals C and D. Reset button SW1 must be momentarily depressed and the required charging current is adjusted by PR2.

The battery may now be connected between terminals C and D. When SW1 is depressed, LED 2 comes on indicating the battery is charging. When the battery voltage exceeds the preset trip voltage, the charging current drops to zero and LED 1 comes on indicating the battery is fully charged.



MUSICAL DOORBELL

An inexpensive programmable doorbell project for your home. This instrument will play any nine-step melody of your choice.

Design by Ray Marston. Development by Steve Ramsahadeo.

Modern doorbells come in two basic types, the simple electrical 'ding-dong' (chime) or the sophisticated microprocessor-controlled multi-tune (Rule Brittania, etc) types. In either case you pay your money and have to accept the sounds that the manufacturers have pre-programmed into your particular device. If you ever get tired of your bell's limited range of sounds, you have little option but to buy a new unit.

We have decided to overcome this by designing a musical doorbell project that the owner can self-program to play any desired (but brief) melody. The essence of our project is that it is simple; it is devoid of hard-to-get micros, PROMs, double-sided PCBs, etc, yet gives an entertaining performance.

Our doorbell is designed to play a nine step melody made up of a selection of five basic notes or tones. The melody lasts for 2-3 S. If the bell-button is briefly operated, the complete melody plays once only; if the bell-button is held closed, the melody repeats continuously. The unit is designed so that the owner can select or 'program' his own melody by hard wiring the interconnections between various pins on the unit's PCB. The nine step, five note choice enables any one of a selection of almost two million (5^9) different melodies to be programmed into the unit!

A feature of our doorbell is that it incorporates a bistable electronic switch that connects power to the unit in such a way that it consumes virtually zero power when in the 'standby' mode. Whenever the bell-button is pressed, the bistable connects power to the unit for the duration of the tune play and then automatically disconnects the power when the melody is complete; this facility ensures long battery life.

Construction

Construction of this unit should present very few problems, if the overlay is followed with care. Note that IC1 and IC3 are CMOS devices and are best mounted in suitable sockets. Also note that an insulated link is connected between pin 3 of IC2 and pin 14 of IC3 on the underside of the board and that Veropins are used to facilitate top-side connections on the PCB.

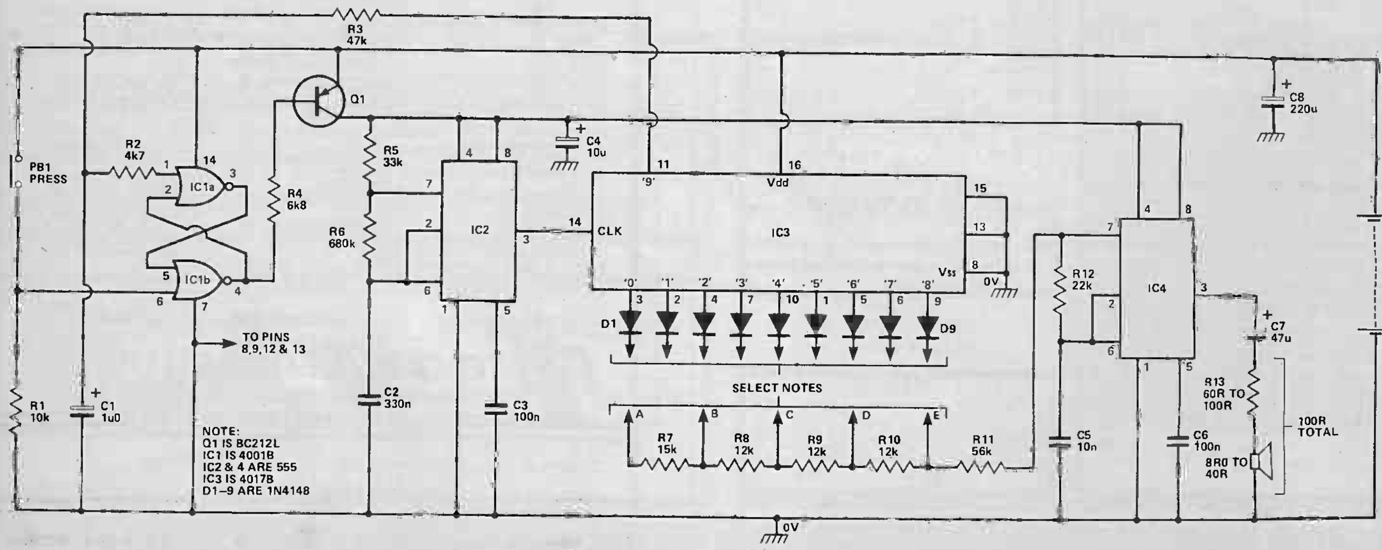
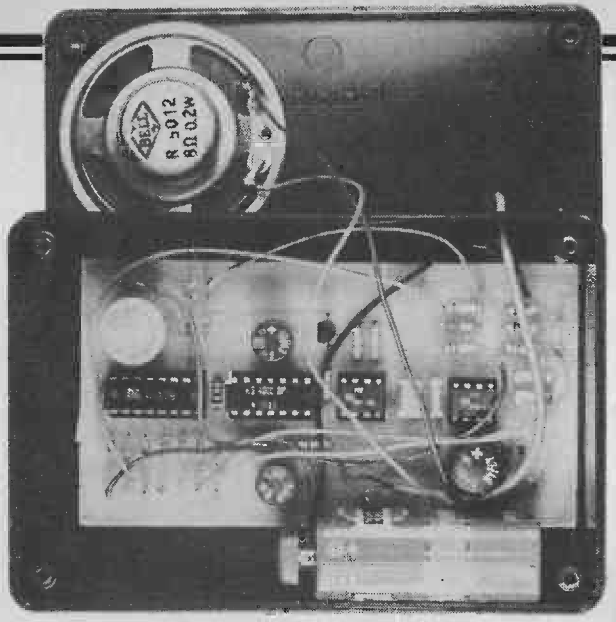


Fig. 1. Circuit diagram of the Musical Doorbell. The connections you make between diodes D1-D9 and the points A-E determine the tune that is played.



Everything fits in the case except the pushbutton; the wire to this can be seen leaving the Verobox.

When construction of the PCB is complete, connect up a suitable speaker, battery and push-button switch and prepare to give the unit a functional check. When selecting a speaker, note that output volume is proportional to speaker impedance and that a high impedance unit will give the loudest results.

When you are ready to try out the unit, connect a flying lead from D1 to one of the A — E note-select points and press PB1 to test the first note in the sequence. You can then wire in the D2 to D9 note-selection connections, one at a time, to establish the rest of the sequence, testing the unit at each step in the wiring sequence.

Once you've finished 'programming' your unit you can fit the PCB, battery and speaker into a suitable box, hang the unit on your front door and finally connect it up to a suitable push-button switch.

HOW IT WORKS

The circuit comprises a bistable (IC1) and a transistor power switch (Q1), two 555 astable multivibrators (IC2 and IC4) and a 4017 decade counter/divider (IC3). The bistable (IC1) is designed around two gates of a CMOS 4001B and controls the base bias of Q1, which in turn controls the positive power supply connections to IC2 and IC4, the two 555 chips.

Normally, the output (pin 4) of the bistable is high, so Q1 receives no base drive and is cut off. Under this condition, IC2 and IC4 consume no power: IC1 and IC3, being CMOS devices, also draw negligible power under this condition. The entire circuit, in fact, consumes a typical 'standby' current of only a microamp or so.

The circuit is activated by briefly pressing PB1, thereby causing the IC1 bistable to change state and connect power to the IC2 and IC4 astables via Q1. IC2 is wired as a low frequency astable (a few Hertz) and delivers clock pulses to IC3. IC3 is a 4017B decade counter/divider; it has ten decoded outputs, which sequentially go high on the arrival of successive new clock pulses, only one output being high at any given time. In our application, the first nine decoded outputs are used to sequentially select (via D1 to D9) timing resistors in a second astable, the IC4 tone generator, which drives a speaker via C7.

The first nine clock pulses from IC2 thus cause nine tones to be sequentially selected via the R7-11 resistor network. On the arrival of the tenth clock pulse, pin 11 of IC3 goes high and resets the IC1 bistable via R3 and C1, thereby cutting off Q1 and removing power from IC2 and IC4, thus completing the operating cycle.

The action of the IC1 bistable is such that, if PB1 is briefly pressed, the instrument plays a single sequence of nine notes (total duration is 2-3 S) and then automatically switches off. If PB1 is held closed, however, the sequence continuously repeats. Note that the owner can set up any tone sequence that he wishes by suitably interconnecting the diode outputs of IC3 to the 'A' to 'E' selection pins on the R7-11 note-selection chain.

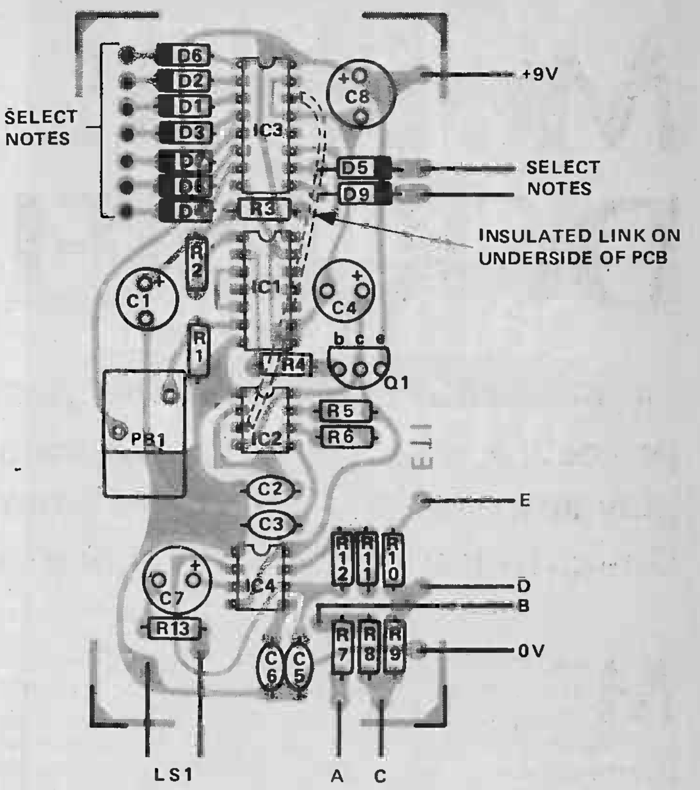


Fig. 2. Component overlay.

PARTS LIST

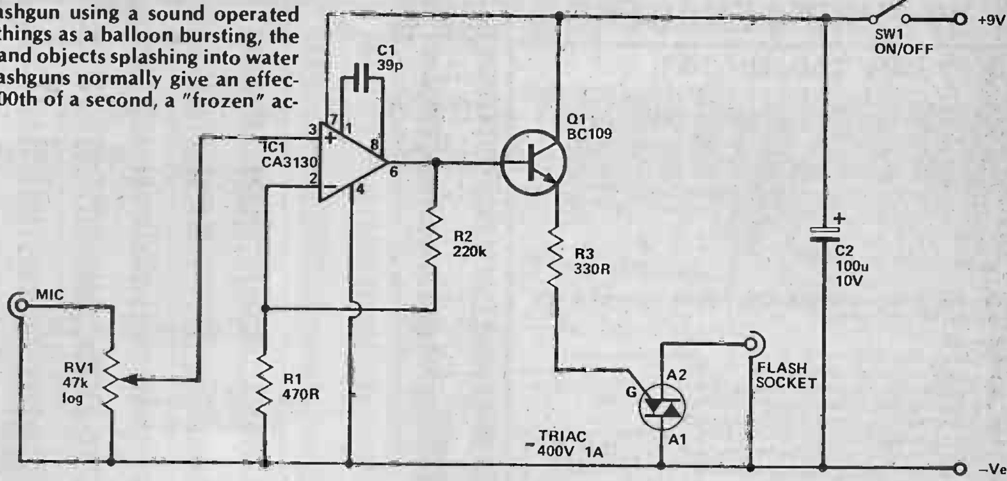
Resistors All ¼W, 5%	
R1	10k
R2	4k7
R3	47k
R4	6k8
R5	33k
R6	680k
R7	15k
R8,9,10	12k
R11	56k
R12	22k
R13	100R
Capacitors	
C1	1u0 63 V electrolytic PCB type
C2	330n polycarbonate
C3,6	100n polycarbonate
C4	10u 63 V electrolytic PCB type
C5	10n polycarbonate
C7	47u 25 V electrolytic PCB type
C8	220u 25 V electrolytic PCB type
Semiconductors	
IC1	4001B
IC2,4	555
IC3	4017
Q1	BC212L
D1-9	1N4148
Miscellaneous	
LS1	Any 8R0 to 40R speaker: see text
PB1	momentary action
B1	PP3
Case	

BUYLINES

All components used in this project are common types, readily available from many of our advertisers.

Sound Triggered Flash Unit

By triggering an electronic flashgun using a sound operated switch, photographs of such things as a balloon bursting, the cork leaving a champagne bottle and objects splashing into water can be taken. Since electronic flashguns normally give an effective shutter speed of around a 1000th of a second, a "frozen" action photograph is obtained.



The photograph must be taken under fairly dark conditions so that the ambient light does not give an exposure if the camera's shutter is set to "B" and opened.

The circuit is based on operational amplifier IC1 which is used in the non-inverting amplifier mode. R1,2 are a negative feedback network which set the gain of the unit at about 500. RV1 (sensitivity) biases the non-inverting input to the negative supply rail. Ideally the input should be fed from a crystal or high impedance dynamic microphone, but the unit will work quite well using a low impedance dynamic microphone or even a high impedance speaker as the signal source.

Q1 is used as a discrete emitter follower output stage which provides the relatively high trigger current required by the triac. R3 is a current limiting resistor. Under quiescent conditions the output of IC1 will be at virtually negative supply potential, and the triac, therefore, receives no gate current. When a signal is received by the microphone, positive going signals are amplified by IC1 to give an output that is a few volts positive. The triac then receives a strong gate bias, causing it to trigger and give a low resistance across its A1 and A2 terminals. These terminals connect to the flashlead via a suitable socket (or flash extension lead with the unwanted plug removed) and the flashgun is, therefore, fired. The circuit operates almost instantly, giving very little delay between the commencement of the sound and the flashgun being triggered. Sometimes more interesting photographs can be obtained by introducing a small delay.

This can be achieved by moving the microphone a metre or two away from the object(s) being photographed.

The current consumption of the unit is approximately 4 mA. It is advisable not to advance RV1 much more than is absolutely necessary in order to give reliable triggering, as frequent spurious operations of the unit could otherwise result.

ASTROLOGUE

Satellites in Surrey and spying on storms. Ian Graham has news of a British spacecraft and the latest addition to the World Weather Watch network.

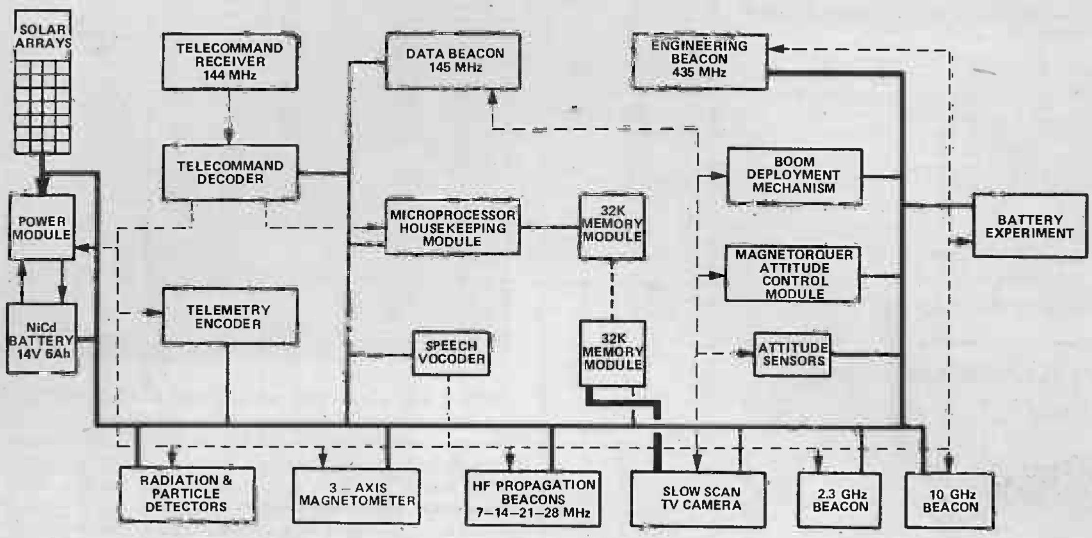


Fig.1 System diagram of the UK amateur scientific spacecraft — UOSAT.

Just over a year ago scientists and engineers began building Britain's first amateur satellite — UOSAT. It isn't a British copy of its American predecessors (the OSCAR series), whose function was to improve long distance VHF and UHF communications. UOSAT has much broader and far-reaching mission objectives — to allow radio amateurs and amateur scientists to study the electromagnetic propagation properties of the atmosphere from HF to microwave frequencies, stimulate active participation in space science (particularly in schools, colleges and universities), establish sources of hardware for the AMSAT programme and pave the way for future amateur spacecraft, possibly operating at new frequencies. That's an ambitious list of objectives.

The project is supported by British industry and research organisations, AMSAT, RSGB (Radio Society of Great Britain) and the University of Surrey, where the spacecraft is being built. A great deal of the design work is complete and systems testing is under way.

On Board

The Propagation Studies Experiment will carry beacons transmitting from 7 MHz up to over 10 GHz, particle radiation counters and a magnetometer. The Education Experiments comprise an Earth-pointing slow-scan TV camera and a synthesised voice telemetry system. The payload will also include two Future Systems Experiments — a two axis, Earth-pointing, gravity gradient spacecraft stabilisation system and an on-board microcomputer to investigate image processing, telemetry/command management and data store/dissemination.

Even when the Space Shuttle is operational, a satellite launch will be an expensive operation in terms of a project such as UOSAT. Fortunately, NASA have agreed to launch UOSAT free of charge as a piggy-back 'passenger' with the Solar Mesosphere Explorer (SME). SME is due to be placed in a circular polar orbit in September 1981 by a Delta 2301 launcher. UOSAT will be placed in a 530 km Sun-synchronous orbit.

Although Astrologue is naturally largely devoted to the multi-million dollar/pound/rouble headline-grabbing space shots it makes a pleasant change to hear of projects like UOSAT. If your university or business is managing or contributing to a space programme in any way, I'd like to hear about it.

One Small Launch . . .

GOES-D, the latest American weather satellite, is now safely in a slightly elliptical orbit about 40,000 km above the equator following a successful launch from Cape Canaveral on September 9th. Over the next few weeks GOES-D's engines will be used to circularise the orbit at 35,600 km. During its planned lifetime of seven years, Geostationary Operational Environmental

Satellite D will, like its predecessors, gather and transmit information on weather conditions over Canada, the United States and South America and a large area of the Atlantic.

Although GOES-D is concerned with American weather, the GOES satellites are part of a worldwide system — the World Weather Watch (WWW) project. Europe has contributed its Meteosat series to WWW, which, as the name suggests, maintains a global weather monitoring service.

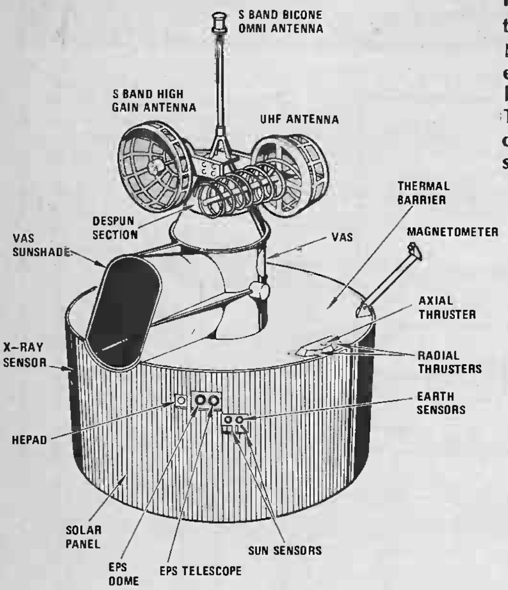


Fig.2 A quick guide to the GOES satellite. Notice the main experiment (VAS) behind its sunshade. The antennae are carried on the despun section (top).

GOES-D has now been redesignated GOES-4. It's the first satellite to be put in orbit as a joint operational and research mission of the National Oceanic and Atmosphere Administration (NOAA) and NASA. This is the first picture to be received from GOES-4, transmitted on September 24th. South America is clearly visible on the right. The remnants of Hurricane Hermine are still swirling over Southern Mexico and Honduras. Storms which begin far out into the Pacific Ocean (two can be seen here) can be tracked, giving ample time to warn major population centres.

. . . But A VAS Difference

The Hughes Aircraft Company are building three of the spacecraft (GOES-D, E and F), the main instrument of which is a visible and infra-red spin-scan radiometer (VISS) atmospheric sounder. The whole instrument is called a VAS for short (thank goodness). The VAS will look down on the atmosphere and gather information on temperature and moisture at different altitudes.

It doesn't sound very exciting, but in fact the onset of very severe weather conditions is characterised by the development of higher and higher cloud tops. Up to now the instruments used could not look down through the highest clouds and watch cloud building at lower altitudes.

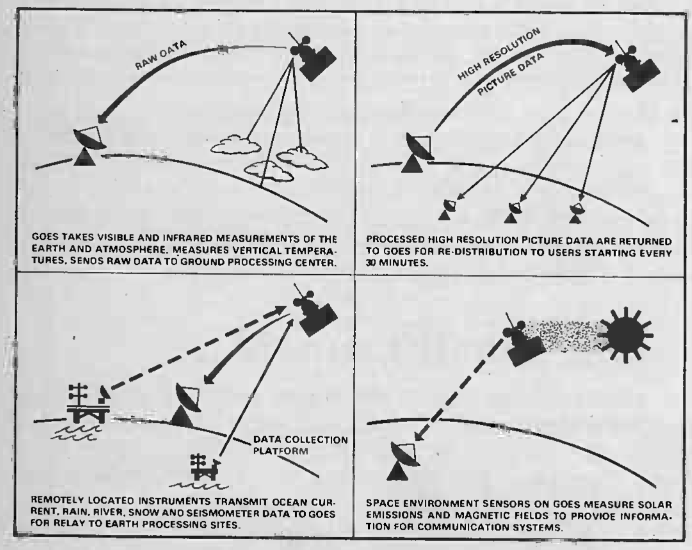


Fig.3 On the GOES data trail, complete with American spelling.

On The Data Trail

The information collected by VAS is processed and transmitted to the Wallops Island (Virginia) receiving station, where it is further processed and sent back up to the satellite for relay to US users.

The satellite also carries instruments to detect solar protons and electrons, alpha particles, X-rays and magnetic fields — information useful in telecommunications and power distribution.

The Hardware

GOES-D weighs in at a shade under 400 kg (orbital weight). It's basically a cylinder covered with solar cells and surmounted by antennae (overall height 12 ft, by 7 ft in diameter). The main body spins at 100 RPM, but the antennae are held stationary with respect to Earth.

The spinning motion is used to photograph the Earth. A frame is exposed each time the craft spins round. After each spin, a scanning mirror in the camera turns slightly, so that the camera gradually scans from pole to pole. It takes around 1800 steps to produce a full image of the Earth in 18 minutes.

During the recent eruption at Mt St Helen's a GOES satellite tracked the cloud of volcanic ash across the US.

ETI

THE BIG BANG THEORY

Over billions of years gas clouds condensed into stars and planets. The primeval soup bubbled. Self-replicating organisms writhed. Fish struggled onto the land and took their first breath of air (dramatic stuff). The result of it all — A.S. Lipson. Here, he discusses how it all began.

One of the many questions that has intrigued man since he first learned to speak is that of the origin of the universe. How did it all begin? Where did our world come from and what caused its existence? Scientists, being only human, (or so we are told) have not been immune to this type of curiosity — even Isaac Newton hypothesized about the origins of the stars. However, it is only fairly recently (during the second half of this century) that any research on this topic has been viewed as 'respectable', or fit material for a serious investigation. During this time, two main opposing theories as to the origin of the universe have developed; the Steady-state theory and the Big Bang theory. It is the latter which tends to be generally accepted these days, as we shall see later. But first we'll need to look at some of the background information . . . . .

The Red Shift Mystery . . .

It was found during the 19th century that when light from the Sun was passed through a narrow slit and then split into a spectrum by a prism, the spectrum showed hundreds of tiny dark lines across it. The reason for this was not known until the advent of quantum mechanics this century, but it was noted that the lines always occurred in the same positions in the spectrum, corresponding to set frequencies or wavelengths of the light. In 1868, it was found by Sir William Huggins that not only were all the same lines found in the spectra of stars, but in some stars, the lines were shifted very slightly from their positions in the solar spectrum. Sometimes the shift was towards shorter wavelengths; the blue end of the spectrum, and sometimes to longer wavelengths; the red end of the spectrum. With a disappointing lack of originality these two changes became known as the blue shift and the red shift, respectively. In order to explain the shifts, Huggins used an analogy with sound. When you are standing still, and are suddenly passed by a fast moving car (of course Huggins, working in 1868, did not explain it in terms of cars, but anyway . . .) which is emitting some sound, you may have noticed that as the car passes you the pitch of the sound drops. (Producing the eeeee-owwwww sound beloved of motor sport enthusiasts.)

This change in pitch, or frequency of the sound waves is caused by the relative velocity between the car and yourself. It follows that light, which is also a wave, is affected in the same way by relative motion between the object emitting it and the object receiving it. In fact, the light from a star moving away from us at great speed is shifted slightly to the red end of the spectrum, and a star moving towards us has its light shifted very slightly to the blue end of the spectrum. This explains the red and blue shifts. Now, it so happens that the wavelength of the dark lines in a spectrum is one of those quantities which physicists find relatively easy to measure with extreme accuracy. By doing this, and comparing the wavelengths of dark lines in the spectra of stars to the wavelengths of the same lines in the spectrum of the Sun, it is possible to calculate fairly precisely just how fast a star is moving towards or away from the Earth.

In The Beginning . . .

Things really began to get interesting, though, when astronomers looked at the shifts in the spectra of other galaxies. They discovered that the distant galaxies appear to be moving away from our own galaxy — the Milky Way. There are one or two exceptions; for instance, the Andromeda Nebula, the closest large galaxy to our own, appears to be moving towards us at about 300 kilometres per second. In general, however, the other galaxies seem to be moving away. In fact it appears that almost every galaxy we can see is rushing away from every other galaxy. This can be simply expressed by saying that 'the universe is expanding'. As a general rule, distant galaxies show a distinct red shift in their spectra and the further away the galaxy, the greater the red shift tends to be, indicating that the further away a galaxy is, the faster it is likely to be travelling away from us.

It began to look as though a long time in the past, all the galaxies were squashed up together and then a massive explosion sent them flying apart. This is the bare bones of what became known as the Big Bang theory and various calculations have shown that if this is indeed what happened, then the 'beginning' — the creation of the universe — took place about 10-20 billion years ago.

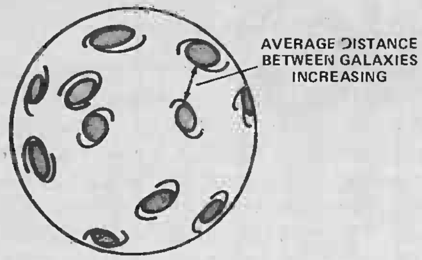


Fig.1 If you visualise our three-dimensional universe as being on the two-dimensional surface of an expanding balloon, you can see that, although the galaxies are getting further away from each other, the centre of expansion is not on the surface.

Before Genesis

Some cosmologists, however, were somewhat unhappy with this explanation of the expansion of the universe. It involves a 'beginning' and therefore raises the awkward question of what was 'before'. In fact, it was reasoned, it would be much more satisfying philosophically if a theory could be found which did not involve a 'beginning' for the universe, (this idea, that a theory ought to be philosophically satisfying, is not quite as silly as might be thought. Time and again in physics, the theory which *feels* best has been the correct one). In the late forties Hoyle, Bondi and Gold proposed the Steady-state theory. This takes care of the expansion of the universe in a most ingenious manner; although the various galaxies are receding from each other all the time, new matter is continuously being created to 'fill up the gaps'. As more matter is created, it collapses by gravitational attraction to form new galaxies. Thus there is no need in this theory for there ever to have been a beginning — the universe is as it is simply because it has always been the same. According to the Steady-state theory, there never was a beginning to the universe, and presumably there will never be an end — it will just keep expanding, old galaxies dying, new ones forming. This theory does have a certain 'neatness' about it that is rather satisfying.

As a first impression it might seem that it would be impossible to tell which of the two main theories — Big Bang or Steady-state — is correct. The only real difference to the universe now would be that, if the Steady-state theory is correct, the rate of expansion would be constant, whereas if the Big Bang theory is correct, the expansion would be slowing down somewhat, as gravitational attraction attempts to pull the galaxies back together again. This slowing-down, however, is far too slight for us to be able to measure. So how can we decide which theory is correct?

Well, for a start, there are one or two things which can only be explained in terms of the Big Bang theory. One of these is the abundance of the element helium in the universe — there is far too much of the stuff around for it to be explained in terms of the Steady-state theory (exactly why doesn't really concern us here). Another is the 'three degree Kelvin microwave background' — which we will consider later. Finally, there is this; according to the Steady-state theory, the universe has always been much the same as it is now, whereas according to the Big Bang theory, it has only evolved to its present state slowly, and it was different in the past. If only we had some way of looking at the past of the universe, we could compare it with the present. If the two were largely similar, we could conclude that the Steady-state theory is roughly correct. If, however, there was a noticeable difference in, say, the structures of galaxies then and now, we might conclude that the Big Bang theory is correct. But we can't look at the past. Or can we? When we look at the stars, we do not see them as they are, but as they were when they emitted the light we see. Light takes only about eight minutes to reach us from the Sun, but nearly four and a half years from even the closest star. When we look at the more distant galaxies, we see them as they were many millions of years ago. Evidence is not 100% conclusive (it rarely is in cosmology) but weighing the facts one against the other, it seems it is the concept of the 'Big Bang' that is correct.

The Microwave Background

Now it is time, then, to elaborate a little on the Big Bang theory. A common misconception is that this theory states that about 15 billion years ago, a massive explosion occurred at one point in space, throwing out matter which eventually condensed into stars, galaxies, planets and (finally) us. In fact, this is not correct. The explosion is not imagined to have occurred at one particular point in space. It took place at every point in space, occupying the entire universe. It makes no sense, then, to ask "Where was the explosion?" The best way of understanding this is to imagine our universe as being on the two-dimensional surface of a balloon, which is being inflated. It makes no sense to ask where on the surface of the balloon is the centre of expansion; every point is just as much the centre as any other.

We will now see what it is thought the precise beginning of the universe was like. Nobody actually knows what the universe was like during the first few fractions of a second; our knowledge only starts after this. After the first tenth of a second or so, the vast and intense quantities of energy that had just sprung into existence with the universe were making the temperature of the universe an incredible 30 billion degrees on the Kelvin scale (at temperatures as high as this, the Kelvin and Centigrade scales are virtually identical). Apart from the pure energy in the form of photons, a lot of electrons and positrons were in existence, together with equally large numbers of particles called neutrinos. In addition, there was a slight contamination of heavier particles, like protons and neutrons. After a second or so, the temperature had dropped to only ten billion degrees or so and this was still far too hot for protons and neutrons to form atomic nuclei. This process didn't begin until three or four minutes after the beginning,

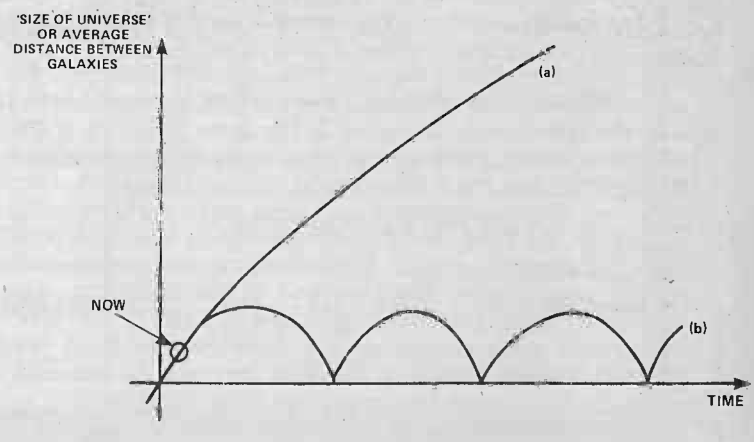


Fig.2 Will the universe keep expanding (a) or will the gravitational attraction be strong enough to make it collapse again, only to start another expansion (b)?

when the temperature had dropped to a mere (. . . . a mere. . . !!! . . .) 900 million degrees. Even though nuclei had been able to form, there was still far too much energy for electrons to be able to join up with the nuclei to form stable atoms. It took nearly three quarters of a million years for that to occur and by that time, most of the original electrons and positrons had vanished. (When an electron meets a positron, the two disappear, giving off energy. This is what is thought to have happened, leaving just a few particles behind.) Gradually, gravity clumped the atoms together, and then clumped the clumps, to form stars and galaxies. Eventually, life developed: but that happened much later.

Cold Radiators

So how can we test this theory? Well, if it is correct, there should still be some radiation hanging around from this beginning. The appropriate calculations have been performed, and it turns out that the radiation ought to be roughly equivalent to that emitted by a perfect radiator at a temperature of about three degrees Kelvin (minus 270° Centigrade). This doesn't seem to be a lot, (things *that* cold don't radiate much heat) but despite this it is measurable. In the mid-sixties, Penzias and Wilson measured this 'three degrees Kelvin radiation background', more or less by accident. At first they blamed poor readings on their equipment and on a pair of pigeons which had nested inside the horn-shaped antenna they were using!

That's it then. It looks very much as though the Big Bang theory is in fact the correct explanation of the origin of the universe. There are still unanswered questions, however.

What's on Next?

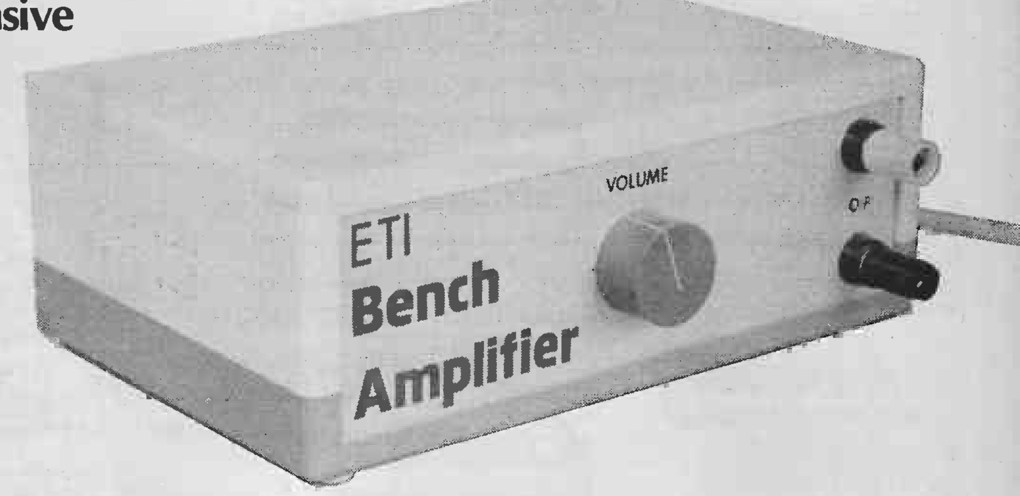
We've seen an explanation of how the universe began, but how will it end? Will it just keep expanding, getting larger and larger and cooler and cooler, or will gravitational attraction pull the galaxies back together again, the expansion of the universe slowing and eventually stopping, then 'going into reverse'? This depends on exactly how much matter there is in the universe. If there is enough, then the gravitational pull will be strong enough to make the universe collapse back in again. If not, then the expansion will continue. In the former case, the universe is said to be 'closed', and in the latter case, 'open'. Either way, the human race will certainly be long extinct before it happens. So we may never know which is the case. Some evidence seems to indicate that the universe is closed; some that it is open. Until fairly recently, it seemed that the universe was probably closed. However, it is now thought possible that the sub-atomic particles known as neutrinos might have mass, contrary to what has been thought for many years. There are so many neutrinos in the universe that, if this is the case, it might be enough to make the difference between an open and a closed universe.

We will finish with one more fascinating possibility. It has been suggested that, if the universe does collapse back on itself, it would first return to its original state of intense heat, and then possibly explode outwards again, beginning the whole thing all over again. We can imagine the universe forever exploding outwards, contracting again, exploding, contracting . . . . Perhaps the universe we live in is formed from the remnants of the cycle before . . . . Sadly we shall never know . . . .

ETI

BENCH AMPLIFIER

Yet another useful piece of test gear from ETI. This time it's an inexpensive but good quality bench amplifier for use with an external speaker. Design by Ray Marston. Development by Steve Ramsahadeo.



Bench amplifiers are fairly simple pieces of workshop test gear which are useful for testing the functional performances of tuners, signal sources and speaker assemblies, etc. They also act as useful standby power amplifiers when not being used in the 'test gear' mode.

Our bench amplifier has been designed with low cost and good functional performance in mind. The unit has a high-impedance (10 M) input stage which is accessed via a pair of panel-mounted terminal sockets. The unit is mains powered and can deliver approximately 4 W into an 8R0 speaker. The bench amplifier is designed to be used with external speakers, which are connected to the instrument via a second set of panel-mounted terminal sockets.

The amplifier works best when used with external 8R0 speakers, but can in fact be used with any speaker impedances in the 4R0 to 16R range. Fidelity is reasonably good, with distortion typically being less than 0.1% at 1 W at 1 kHz. The frequency response is virtually flat to 200 kHz.

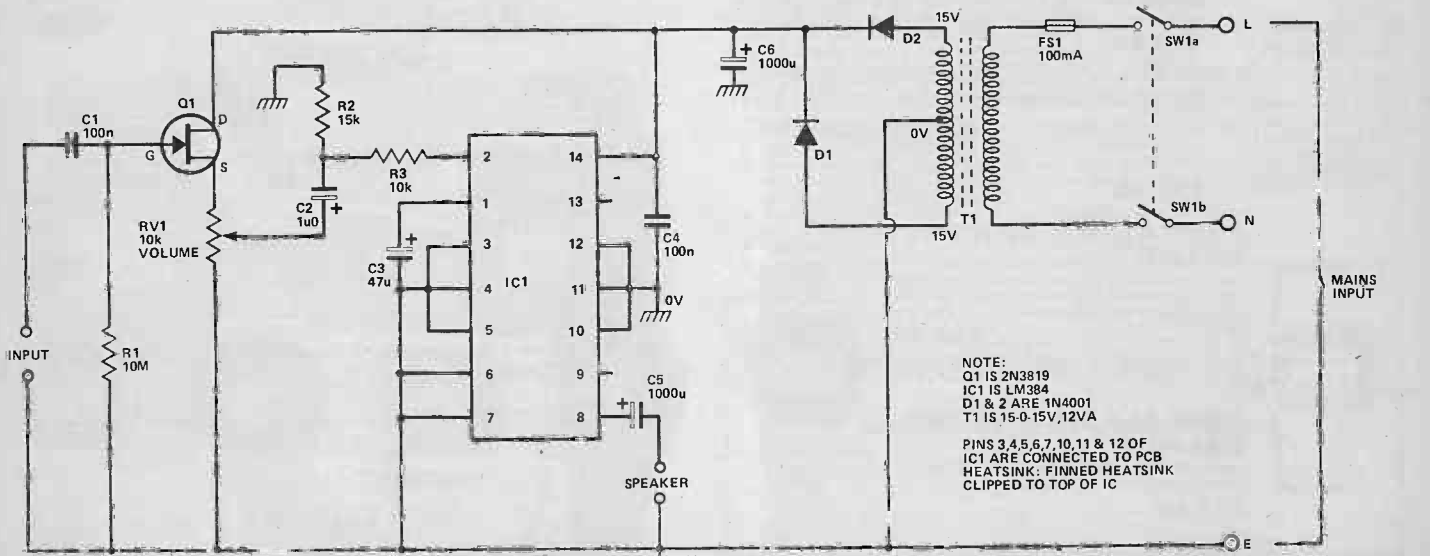


Fig. 1. Complete circuit diagram for the ETI Bench Amplifier.

Piece Work

With the two latest additions to the successful Chess Challenger series of games, the Challenger is moving away from the concept of a chess board with a computer tacked onto the side, towards the truly electronic board. If you're wary of computers and keyboards, you may have been put off buying a chess machine up to now because of the business of entering your move on the keypad. The Sensory 8 gets round that by featuring a touch-sensitive playing surface, by means of which the computer can 'see' the pieces and automatically enter each move. A light on each square shows the 'from' and 'to' positions. Eight levels of play can be selected and skill level can even be changed in mid-move. All the usual Chess Challenger features (position verification, problem-solving, etc.) are featured.

Following on from the Voice Chess Challenger, you can now also buy a Voice Sensory Chess Challenger. In addition to all the Sensory 8's features, it speaks its own moves, repeats yours, calls out captures and rattles off all the board positions on command. If this dedicated dalek's voice gets a bit too much you can switch it off (unlike human opponents).

The voice version also has a built-in chess clock showing the time left for you or the computer and the elapsed time for the game. You can buy a plug-in printer to provide a copy of every move in the game. Sensory 8 sells for £129.95 and Voice Sensory for £279.95.

Aimed at chess players with itchy feet, Ingersoll Electronics have introduced the Chess Traveller, featuring eight levels of play. The pieces push into holes on the board (with spare holes for taken pieces) and can be covered by a perspex top at half time or between games. Through the keyboard you can select skill level, verify board positions, solve problems and set up specific openings. Power is from mains or batteries. Chess Traveller retails at around £50.

Oops

Space Invasion Game

In the Space Invasion Game main circuit diagram (November 1980, p.66/67) pin 12 of IC11b is shown connected to earth. It should be connected to pin 34 of the CPU, IC15. All new boards supplied by Tangerine Computers will have this modification, so check your board before making any changes.

On the same diagram, the ICs are identified in two ways. Our own IC1, IC2, etc. notation identifies the components as they appear on the Parts List. The circled legends (A1, A2, B1, H4, etc.) are the positions of the ICs shown on the PCB supplied by Tangerine Computers.

IC20 pins 20 and 21 should be swapped over. The label adjacent to IC7c was ringed in error — it is diode D2 not position D2.

Cassette Interface

In the component overlay of the Cassette Interface project (October 1980 p.65 Fig.5) the junction of R4-R5-R6-C4 is shown connected to pin 1 of IC6. Instead it should be connected to pin 2 of IC6. The circuit diagram is correct but the foil pattern on p.109 is incorrect. Simply break the track connecting the above components to pin 1 and solder a new link across to pin 2. The Buylines information omitted from this project was included in Digest in the November issue.

Radio Control

In the Radio Control Buylines section (October 1980) the prices quoted for the transmitter and receiver do not include PCBs, receiver case or SLM three-pin gold-plated servo connector block. Prices including these items are £8.57 for the transmitter kit and £14.26 for the receiver kit (including VAT) from Ambit International.

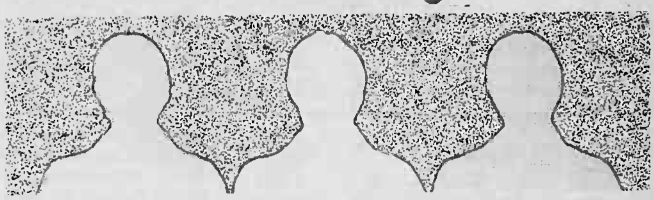
The Inch War

Length may not be everything, but JVC want to know how important it is to video tape recorder owners. They are shortly to test market four hour VHS tapes in the UK. As it is a test exercise, the new E240 tapes will be in very short supply until the experiment is complete and demand is established.

JVC expect the demand to come from owners of their new HR7700 recorder, with which every absentee landlord worth his salt can tape up to eight programmes from different channels in a fortnight.

So, if you like the idea of four hour VHS tapes, seek out the elusive JVC test retail outlets and form an orderly queue with piggy banks at the ready.

HELP! ETI NEEDS YOU!

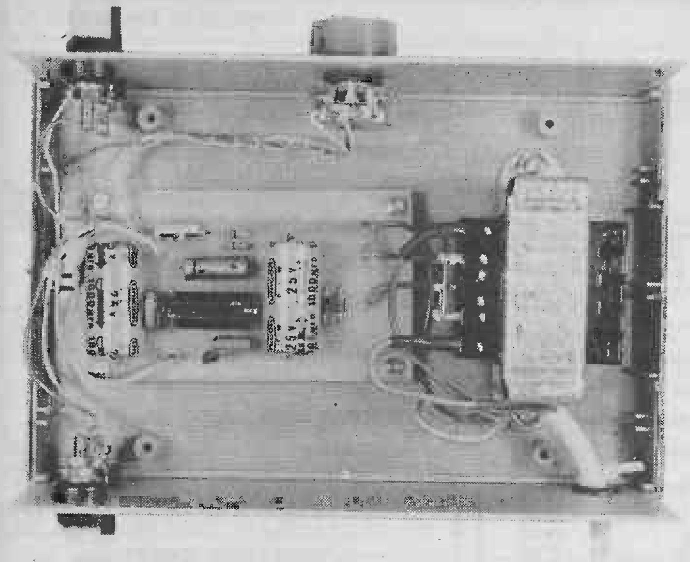


Could you be a Project Engineer for ETI? — the person who fills this position will be able to design and build up projects to the standard of finish ETI readers are used to seeing in their magazine. This calls for someone with a good knowledge of circuit design and with the patience to carry the design through to a finished state. Existing staff are available to assist in all aspects of design work. The easiest part of the job will be writing up the project once it is completed. None of the present ETI technical staff were journalists previous to joining, and no-one has found the writing a difficult task.

We have no preconceived notions of age required.

Applications should reach us as soon as possible with C.V.

Apply in writing to:
The Editor,
Electronics Today International
145 Charing Cross Road,
London WC2H 0EE



This photograph shows everything fitted, wired up and ready to go. Volume and output sockets are on the front panel, input sockets are on the back. Note the heatsink on the IC.

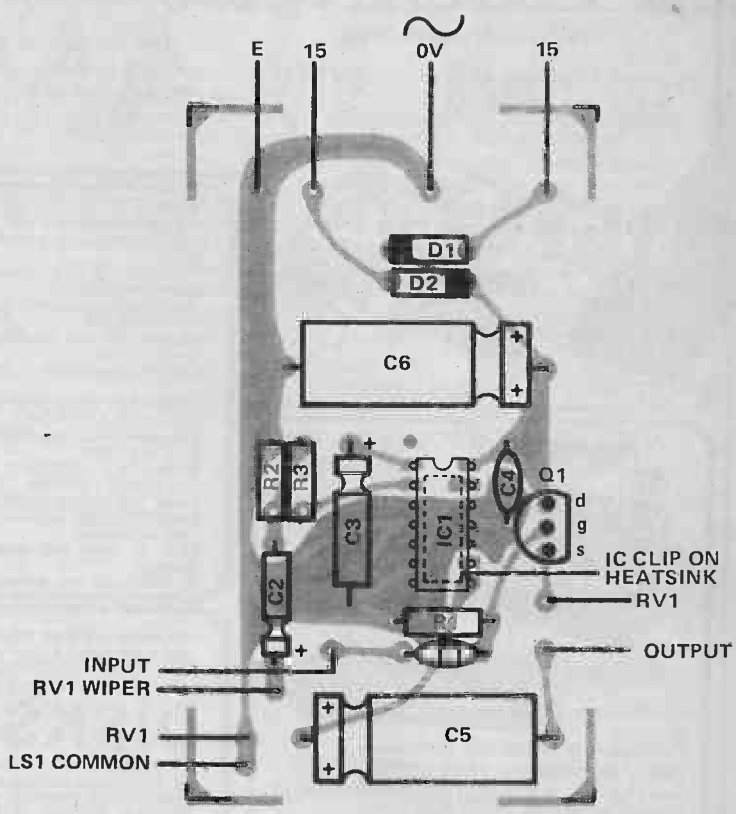


Fig. 2. Component overlay of the Bench Amp.

Construction

The circuit is very simple, so construction should present few problems. IC1 should be soldered directly to the PCB, without the use of a socket. This IC uses a modest area of PCB foil as a partial heat sink and has additional sinking provided by a finned clip-on heat sink (see Buylines). The PCB should be provided with ten Veropin terminals, to facilitate connections to T1, RV1 and the input and output terminals.

When PCB construction is complete, fit it into a suitable cabinet (we used a Verobox on our prototype) and make the inter-connections to T1, RV1 and the input and output terminals. On our prototype we omitted FS1 and SW1 from the actual unit, since these facilities are already provided by the fused mains plug and the switched power socket of our bench installation. We fitted volume control RV1 and the speaker terminals to the unit's front panel and fitted the input terminals to the rear. Note that holes should be drilled in the sides (or top) of the completed cabinet, to aid heat dissipation.

When construction is complete, fit a suitable speaker (ideally 8R0) in place, connect a suitable input signal and switch the unit on. Check that good quality reproduction is obtained (the unit will drive roughly 4 W into an 8R0 speaker) and that the volume can be varied via RV1. The unit is then complete.

HOW IT WORKS

There really is not a great deal we can say here, as the circuit is so simple. Q1 is a high-impedance FET (field effect transistor) and is used as a unity-gain buffer amplifier. Its output is fed to the input of IC1 via RV1, the volume control.

IC1 is a 5 W power amplifier (actually an up-rated version of the LM380 2 W IC), with an integral heat sink that is coupled to the three centre pins on either side of the IC package. The output of the power amplifier is passed to an external speaker via C5 and the whole shebang is mains powered via T1-D1-D2 and C6.

The output of the IC is (according to the manufacturers) short circuit proof and can work into any speakers with impedances in the 4R0 to 16R range. In practice, the circuit works best with 8R0 speakers. To ensure adequate power handling capability, our IC is soldered to a PCB heat sink and is also fitted with a finned clip-on heat sink.

PARTS LIST

Resistors all ¼ W 5%	
R1	10M
R2	15k
R3	10k
Potentiometer	
RV1	10k logarithmic
Capacitors	
C1, 4	100n polyester
C2	1u0 63 V axial electrolytic
C3	47u 25 V axial electrolytic
C5, 6	1000u 25 V axial electrolytic
Semiconductors	
Q1	2N3189
IC1	LM384
D1, 2	1N4001

Miscellaneous
FS1 100 mA and holder, SW1 DPDT miniature toggle, T1 15-0-15 12 VA mains transformer.

BUYLINES

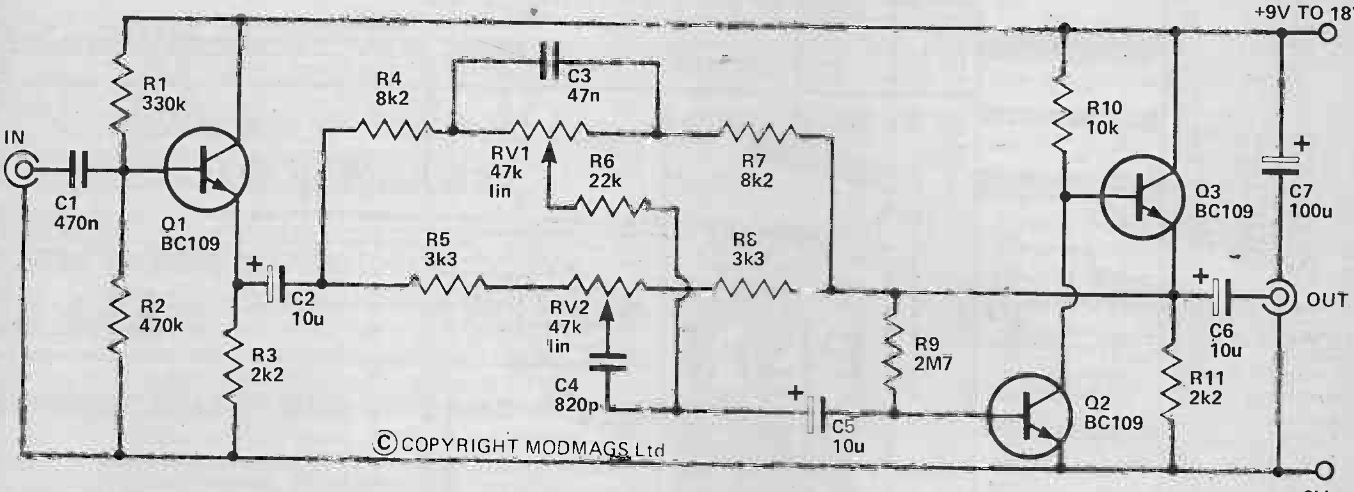
All components used in the ETI Bench Amplifier should be readily available from major stockists. The clip-on heat sink can be obtained from Watford Electronics.

Active Tone Controls

This tone control circuit is easy to incorporate in a stereo amplifier, disco unit, or whatever, as it has a high input impedance (over 100k), a nominal voltage gain of unity and a low output impedance. The usual bass and treble controls are included in the unit, with about 12 dB of boost and cut being available at 100 Hz and 10 kHz. The noise and distortion produced by the circuit are both extremely low due to the large amount of negative feedback used and the unit can handle output signal levels of several volts RMS without clipping.

Q1 is used in a straightforward emitter follower buffer stage that gives the unit a high input impedance. C2 couples the output of Q1 into the tone control circuitry. This is an active circuit which provides frequency-selective negative feedback over an amplifier. The amplifier uses Q2 as a conventional common emitter stage direct-coupled to emitter follower output transistor Q3. The latter gives the unit a low output impedance.

The tone control networks are slightly simpler than the usual "Baxandall" configuration, but give a perfectly acceptable level of performance. RV1 controls the bass while RV2 is the treble control. Feedback is at a maximum with the sliders of the potentiometers to the right and at a minimum with the sliders set fully to the left. Of course, the gain of the circuit is inversely proportional to the level of feedback. Maximum feedback therefore corresponds to maximum cut and not to full boost. The current consumption of the circuit is a little under 1 mA per supply volt.



DESIGNER'S NOTEBOOK

This month Ray Marston takes an in-depth look at the VCO section of that ubiquitous CMOS chip, the 4046B phase-locked loop.

The 4046B CMOS chip is probably one of the most versatile and least-used of all the ICs in the CMOS range. The device glories in (or suffers from) the descriptive title of 'micropower phase-locked loop' and there is a widespread misconception amongst many electronics amateurs and professionals that the device can only be used in PLL-type applications. In fact nothing could be further from the truth.

The Inside Story

The 4046B actually contains a pair of phase-comparators, a zener diode and one VCO or voltage-controlled oscillator. All of these sections are independently accessible via the IC pin-outs. The VCO section of the device is probably the most versatile and cost-effective voltage-controlled oscillator on the market. It produces a well-shaped symmetrical square wave output, has a top-end frequency limit in excess of 1 MHz, can be voltage-scanned through a 1,000,000:1 range (1 Hz to 1 MHz) when used with a single timing resistor or through any range from 1:1 to infinity (0 Hz to 1 MHz) when used with a pair of timing resistors.

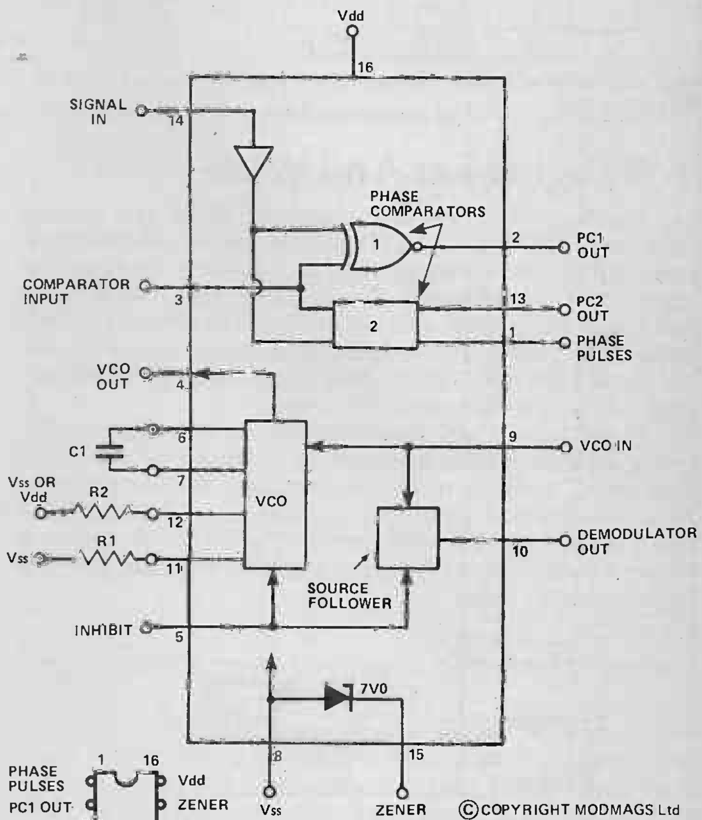
If that were not enough, the voltage-controlled oscillator can also be independently gated on and off via an INHIBIT terminal, can be operated from any supply in the range 3-18 V and can, when used in conjunction with one of the 4046B's phase comparators, produce a two phase output. The linearity of the VCO is typically a healthy 1% or so.

Basic 4046B VCO Circuits

Figure 1 shows the internal block diagram and the pinouts of the 4046B phase-locked loop IC. The device contains two types of phase comparator, a VCO and a zener diode. In practical PLL applications, the VCO and one or other of the comparators are interconnected to form a 'loop', which causes the VCO to lock the the mean frequency of an input signal connected to pin 14.

For our present purposes the most important element of the IC is the VCO, or voltage controlled oscillator. The operating frequency of the oscillator is governed by the value of a capacitor connected between pins 6 and 7 (minimum value 50 pF), by the value of a resistor wired between pin 11 and ground (minimum value 10k) and by the voltage applied to VCO-input pin 9 (any value up to the supply voltage in use).

Figure 2 shows the simplest possible way of using the VCO section. Here, the pin 9 'voltage control' input is tied permanently high and the circuit acts as a basic square wave oscillator, with its frequency variable over a 10:1 range by RV1. Note at this point that the VCO output (pin 4) is tied directly to the pin 3 phase comparator input. If pin 3 is allowed to float, the comparators tend to self-oscillate at about 20 MHz and superimpose an HF signal on the top part of the VCO output waveform.



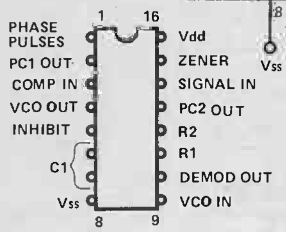


Fig.1 Internal block diagram and pin-out of the CD4046B micropower phase- locked loop CMOS IC.

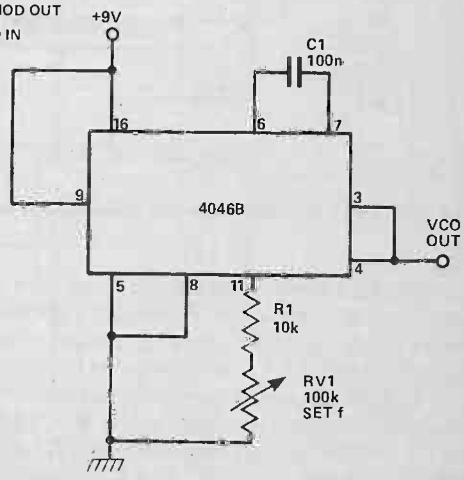


Fig.2 Simple variable-frequency (200 Hz to 2 kHz) square wave generator.

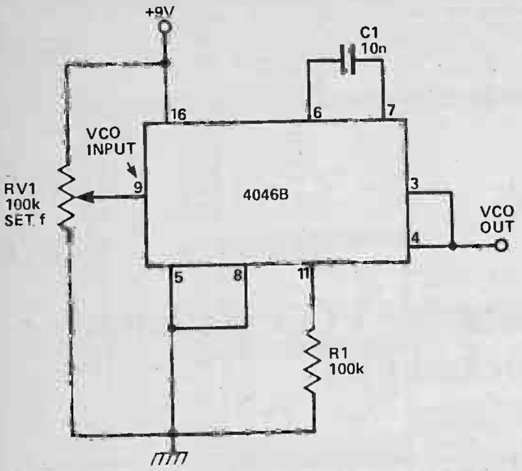


Fig.3 Wide range VCO with frequency variable from near zero to 1.4 kHz via the pin 9 voltage.

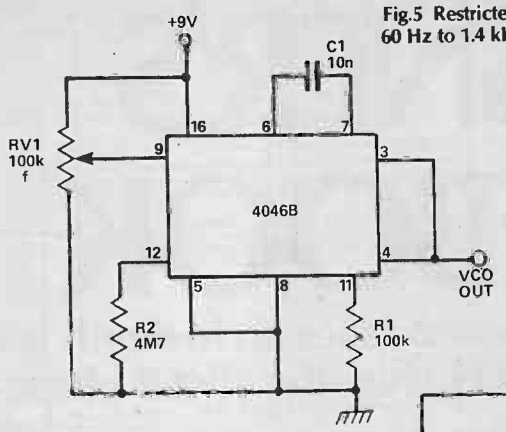


Fig.5 Restricted range VCO with frequency variable from 60 Hz to 1.4 kHz by RV1. R2 acts as an offset resistor.

Ranging Far And Wide

Figure 3 shows how to connect the 4046B as a genuine wide-range VCO. Here, R1-C1 determine the top (maximum) frequency that can be obtained and RV1 controls the actual frequency via the pin 9 voltage. The frequency falls to near-zero (a few cycles per minute) with pin 9 at 0 V. The effective control range of pin 9 varies from roughly 1 V below the supply value to 1 V above zero, ie RV1 has a 'dead' control area of several hundred millivolts at either end of its range.

Figure 4 shows how these 'dead' areas can be eliminated by wiring a silicon diode in series with each end of RV1. The circuit also shows how the minimum operating frequency can be reduced to absolute zero by wiring a high value resistor (R2) between pins 12 and 16. Note here that, when the frequency is reduced to zero, the VCO output randomly settles in either the logic 0 or logic 1 state.

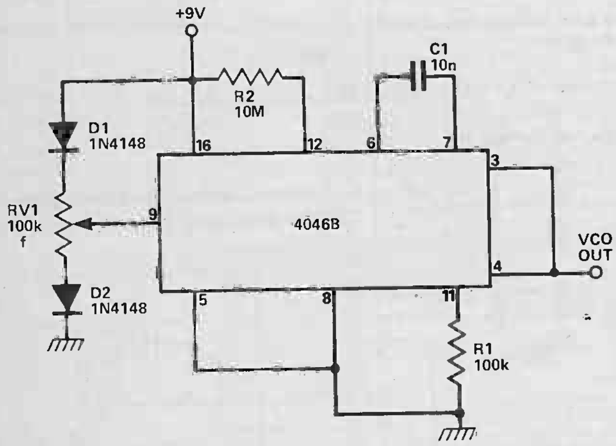


Fig.4 Wide range VCO with frequency variable down to absolute zero.

Figure 5 shows how the pin 12 resistor can alternatively be used to determine the minimum operating frequency of a restricted-range VCO. Here, f_{min} is determined by R2-C1 and f_{max} is determined by C1 and the parallel resistance of R1-R2.

Figure 6 shows an alternative version of the restricted-range VCO, in which f_{max} is controlled by R1-C1 and f_{min} is determined by C1 and the series combination of R1 and R2. Note that, by suitable choice of the R1 and R2 values, the restricted-range VCO can be made to span any range from 1:1 to near-infinity.

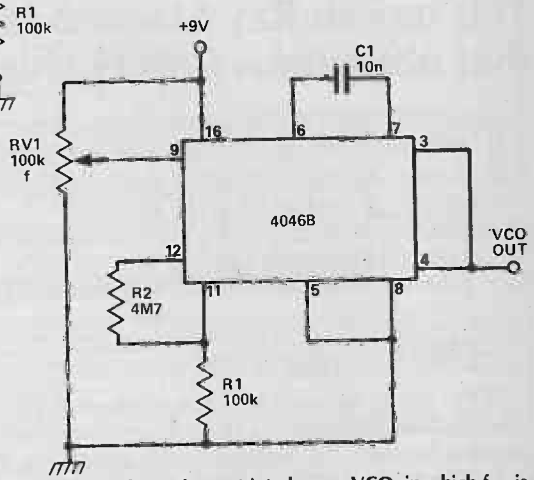


Fig.6 An alternative restricted range VCO, in which f_{max} is controlled by R1-C1 and f_{min} by (R1 + R2)-C1.

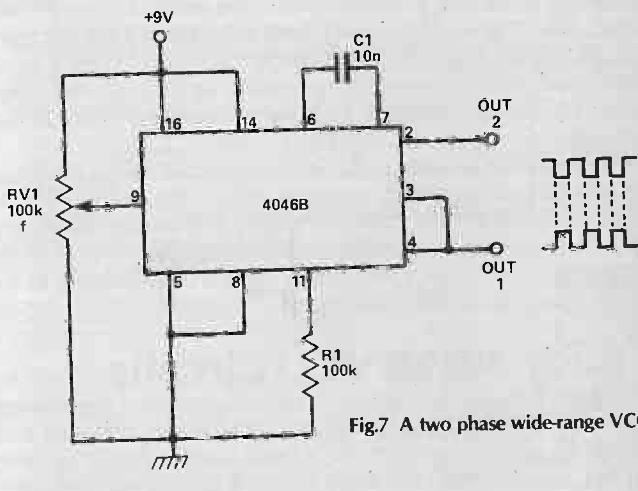


Fig.7 A two phase wide-range VCO.

Square Pear

The VCO can be made to generate a pair of anti-phase square wave outputs by connecting its output to the phase-comparator input, taking the signal input (pin 14) high and taking the anti-phase output from pin 2. Figure 7 shows the connections. Note that this circuit makes use of the IC's built-in EX-OR gate (phase comparator 1).

The VCO section of the 4046B can be disabled by taking pin 5 of the package high (to logic level 1). This feature enables the VCO to be gated on and off by external signals. Figure 8 shows how the VCO can be manually gated via a push button connected directly to pin 5, while Fig.9 shows how the circuit can be gated electronically by an external gate inverter. Alternatively, if the two phase output facility is not required, the internal EX-OR phase detector can be used to provide gate inversion, as shown in Fig.10. Note in this latter case that pin 4 is not connected to pin 3.

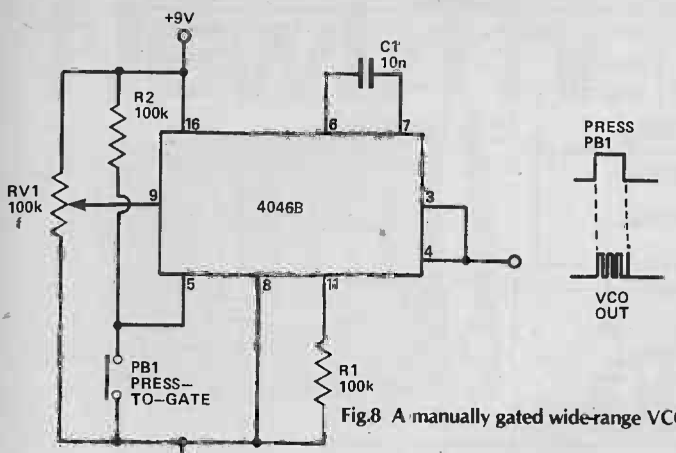


Fig.8 A manually gated wide-range VCO.

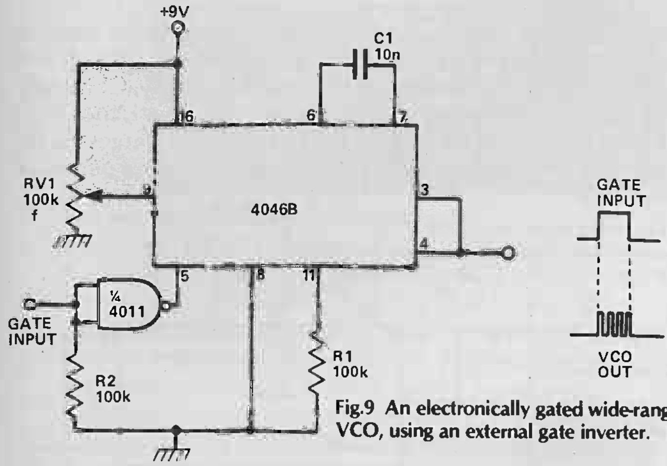


Fig.9 An electronically gated wide-range VCO, using an external gate inverter.

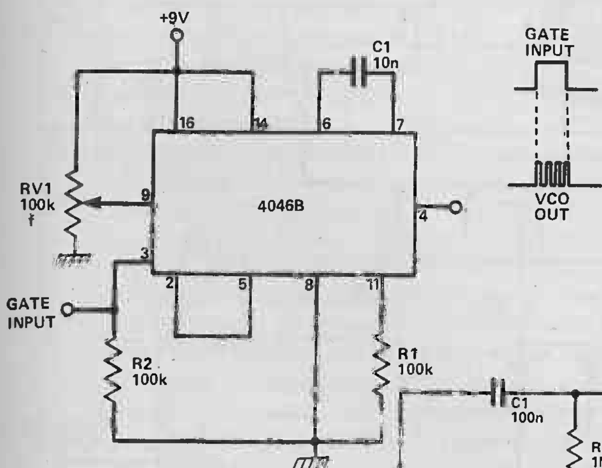


Fig.10 An electronically gated wide-range VCO using the internal EX-OR phase detector for gate inversion.

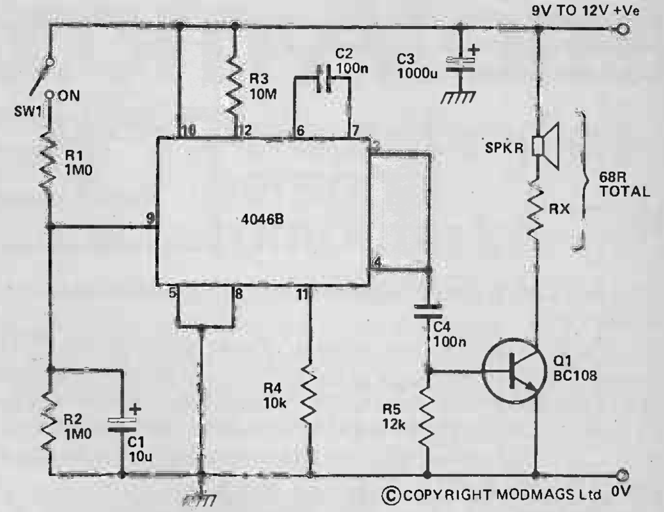


Fig.11 An electronic siren giving slow rise and fall of its operating frequency.

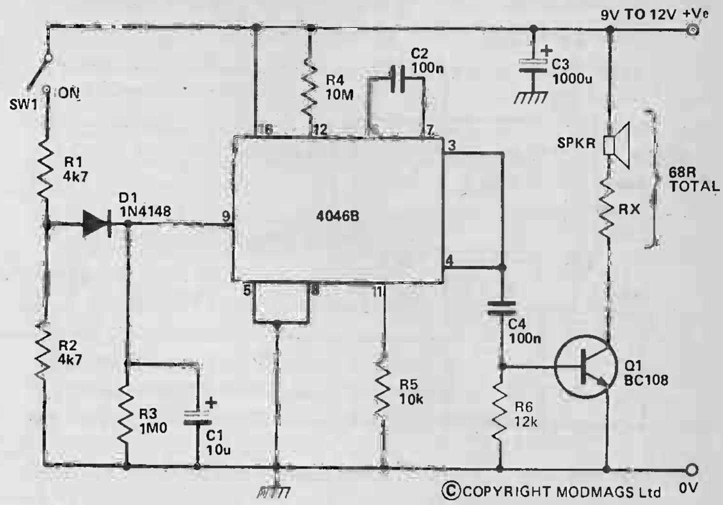


Fig.12 This quick-start siren gives a rapid rise and slow fall of its operating frequency.

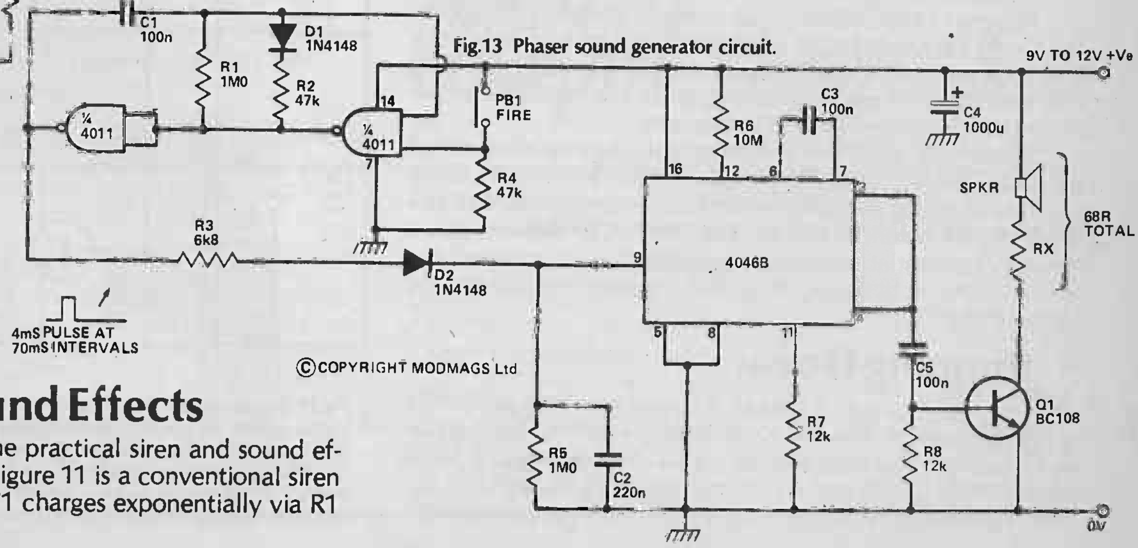


Fig.13 Phaser sound generator circuit.

Sirens And Sound Effects

Figures 11 to 14 show some practical siren and sound effects generator VCO circuits. Figure 11 is a conventional siren circuit. When SW1 is closed, C1 charges exponentially via R1 and the VCO frequency rises slowly from zero to a maximum value. When SW1 is opened, C1 discharges via R2 and the operating frequency slowly decays to zero. The VCO output is AC-coupled to the speaker via C4 and Q1.

The Fig.12 quick-start siren is similar to the above, except that C1 charges rapidly to half-supply volts via R1-R2 and

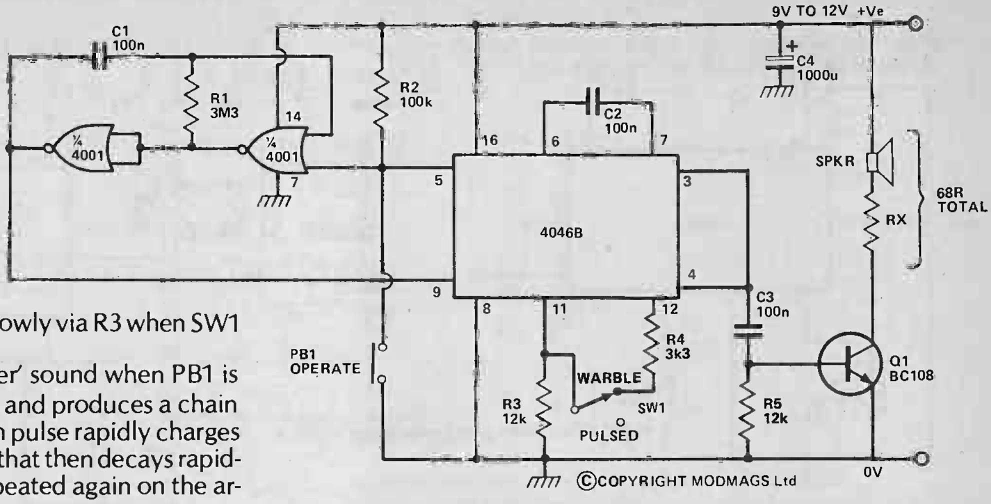


Fig.14 Combined pulsed tone/warble tone alarm generator. The high tone is determined by R3, the low tone by (R3 + R4).

D1 when SW1 is closed and discharges slowly via R3 when SW1 is opened.

The Fig.13 circuit produces a 'phaser' sound when PB1 is closed. The 4011 astable is gated by PB1 and produces a chain of 4 mS pulses at intervals of 70 mS. Each pulse rapidly charges C2 via R3 and D2, to produce a high tone that then decays rapidly as C2 discharges via R5, only to be repeated again on the arrival of the next pulse.

The Figure 14 circuit generates either a pulsed tone or a warble tone signal (depending on the setting of SW1) when PB1 is closed. PB1 is used both to enable pin 5 of the 4046B and to gate on the 4001 astable, which then applies a rectangular (alternatively fully-high and fully-low) waveform to pin 9. In the pulsed mode the VCO generates zero frequency when pin 9 is low. In the warble mode it generates a tone that is 20% down on the high tone when pin 9 is low.

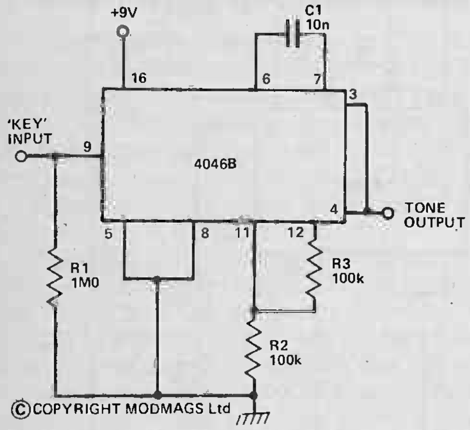


Fig.15 FSK generator — logic 0 = 1.2 kHz, logic 1 = 2.4 kHz.

Miscellaneous VCO Circuits

Figures 15 to 19 show a miscellany of 4046B VCO circuits. The Fig.15 circuit is that of a FSK generator which produces a 2.4 kHz tone when a logic 1 signal is applied to pin 9 and a 1.2 kHz tone when a logic 0 signal is applied. The high tone is controlled by R2 and the low tone by R2 and R3.

Figure 16 is a 220 kHz FM generator. The internal zener of the 4046B (pin 15) is used to provide a stable 7V0 supply to the x20 3140 inverting amplifier, which is quiescently biased at 3V5 by the R2-R3 potential divider. The pin 9 VCO signal is thus a mean 3V5 potential amplitude modulated by an amplified version of the AF input signal, which thus frequency-modulates the output of the VCO.

Running Down

The Fig.17 circuit is that of a run-down clock generator of the type used in dice and roulette games. When PB1 is pressed, C1 charges to a high voltage via D2. Simultaneously, Q1 is biased on via D3-D4 and effectively connects R6 between pin 11 and ground. Under this condition, the VCO operates a high frequency (tens of kHz) and effectively generates a random number of clock pulses. When PB1 is released, Q1 turns off and the VCO timing is governed by R8. Simultaneously, C1 rapidly discharges to half-supply volts via R1-R2-D1, so the VCO operates at only 100 Hz or so. C1 then slowly discharges via R3 and the VCO frequency slowly decays to zero over a period of about 15 S.

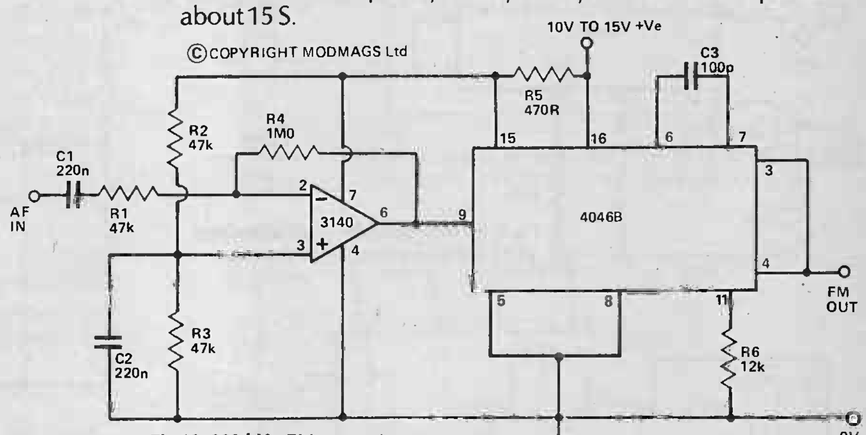


Fig.16 220 kHz FM generator.

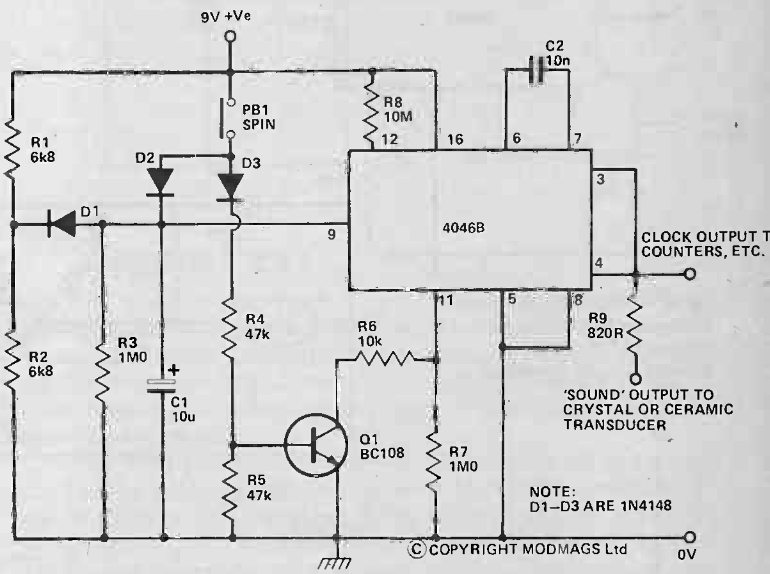


Fig.17 Run-down clock/sound generator for use in dice/roulette games. The circuit is suitable for use with edge-sensitive clock circuits only.
The output can be used to directly clock most types of counter and can be fed, via R9, to crystal or ceramic transducers to directly produce 'run down' sounds. When the run-down is complete this circuit may settle in either logic 0 or 1, so it cannot safely be used to clock level-sensitive circuitry.

ETI

POLYSYNTH

The Transcendent Polysynth is no ordinary synthesiser — it's a family of them. Each of its voices is a complete synthesiser in itself with two VCOs, two ADSRs, a VCA and a VCF in addition to all the usual synthesiser functions. Design and development by Tim Orr.

The Polysynth is a four octave polyphonic music synthesiser. The standard unit has four voices making it possible to play up to four notes simultaneously. Each voice is a complete synthesiser in itself, having two VCOs, two ADSRs, one VCA and one VCF. The voices are totally voltage controlled which enables them to be ganged up in a bus system. So when the master sustain level is adjusted, the sustain level on all voices is set to the same value. However, by providing independent pitch and gate signals, it is possible to control each voice from the keyboard and yet have a common control over the other parameters. The machine can be expanded (with an extra mother board (PS4), four voice boards (PS7) and a panel board (PS6)) to a system with eight independent voices.

The design has minimal wiring (Fig.1), most of which is made with preformed ribbon cable links. Also, the four voice boards, which require nearly 50 signals each, plug into a Molex connector bus system.

All the common controls, such as ADSR parameters, modulation oscillators, volume and noise level are located on the left hand panel board (PS5).

The right hand board (PS6) handles the individual parameters of oscillator tuning and the voice on/off control. Both the panel boards deliver their signals to the mother board (PS4) which then distributes them to the voice slots. Pitch and gate parameters are independently fed to the voices, these signals being generated by PS1, 2 and 3.

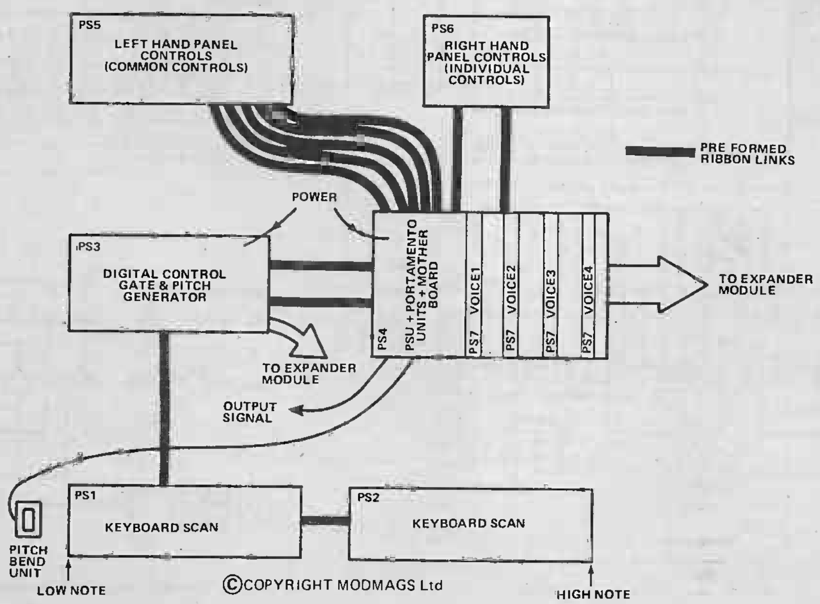
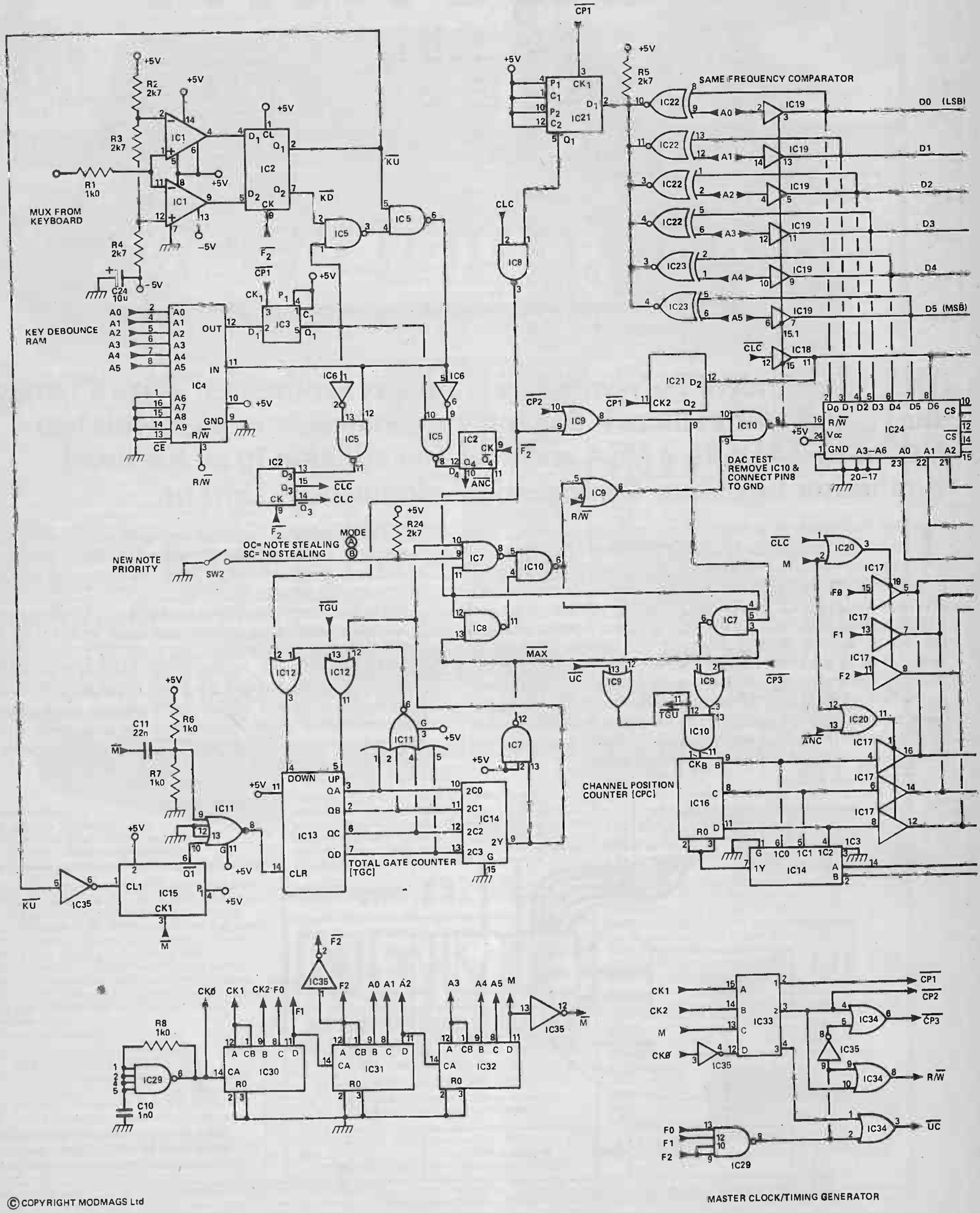
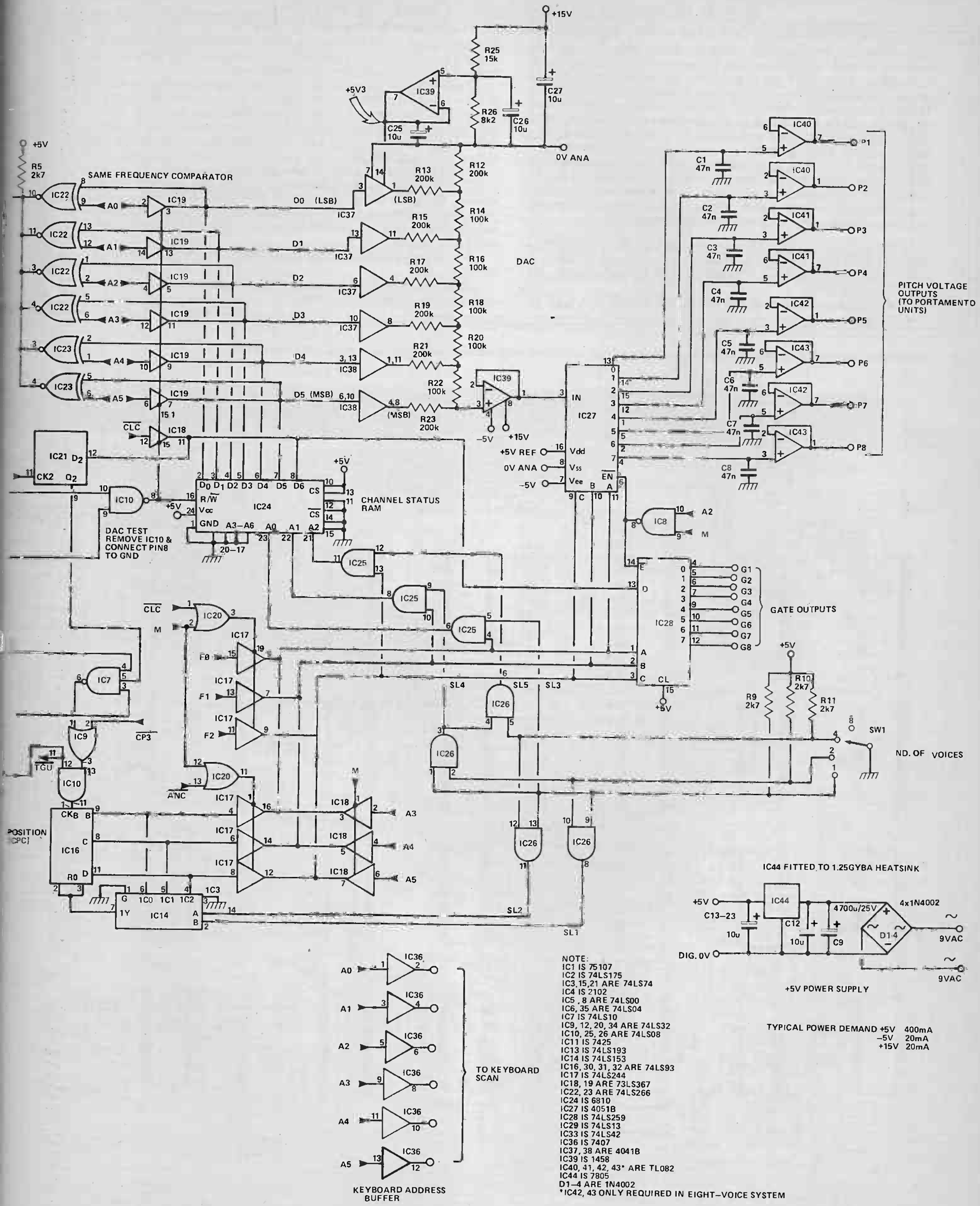


Fig.1 Board-to-board connections. The voice boards (PS7) simply plug into the mother board (PS4) using Molex connectors.

→

Fig.2 Circuit diagram of PS3, the digital board. In this and other diagrams there is a central overlap to make things easier to follow.





NOTE:
IC1 IS 75107
IC2 IS 74LS175
IC3,15,21 ARE 74LS74
IC4 IS 2102
IC5 ,8 ARE 74LS00
IC6, 35 ARE 74LS04
IC7 IS 74LS10
IC9, 12, 20, 34 ARE 74LS32
IC10, 25, 26 ARE 74LS08
IC11 IS 7425
IC13 IS 74LS193
IC14 IS 74LS153
IC16, 30, 31, 32 ARE 74LS93
IC17 IS 74LS244
IC18, 19 ARE 73LS367
IC22, 23 ARE 74LS266
IC24 IS 6810
IC27 IS 4051B
IC28 IS 74LS259
IC29 IS 74LS13
IC33 IS 74LS42
IC36 IS 7407
IC37, 38 ARE 4041B
IC39 IS 1458
IC40, 41, 42, 43* ARE TL082
IC44 IS 7805
D1–4 ARE 1N4002
*IC42, 43 ONLY REQUIRED IN EIGHT–VOICE SYSTEM

HOW IT WORKS : DIGITAL CONTROL PS1, 2, 3

The synthesiser decides which notes are being played by digitally scanning the keyboard and analysing the received data. A high frequency oscillator (Fig.2, IC29 pin 6) generates the master timing for the system, which is then divided down by a 12 stage divider (IC30,31,32). The signals generated by this section are used to construct various timing waveforms. A0 to A5 are sent to the keyboard scan circuitry, PS1,2 (Fig.4). This is a 49-way multiplexer. The six bit code can address 64 locations, but as the keyboard only has 49 notes the other 15 locations remain unused. The bottom part of the code (A0,1,2) addresses each eight bit multiplexer whilst the top three bits (A3,4,5) are decoded by IC8. The outputs of IC8 then enable each of the seven multiplexers in turn so that a complete keyboard is produced. When a note is unpressed it generates − 5 V and when it is between contacts it is an open circuit.

Figure 5 shows a typical keyboard scan output. The unused top 15 addresses have been arranged to generate a key unpressed signal. A power clear circuit has been included to clear the system when the machine is powered up.

The keyboard output is fed to a voltage comparator, IC1, Fig.2. Integrated circuits IC1,2,3,4,5 form a circuit block that decides when to assign a new channel to a voice (ANC) or to clear an existing one (CLC). The problem of channel assignment is very complex.

The synthesiser has only four voices, so the question to be answered is 'what should happen if more than four keys are pressed at once'? Should the system ignore the extra keys or should it reassign voices to the extra notes? If reassignment (referred to as note stealing) is to be used, then which notes should be taken, the first or last note played, or the highest or lowest in pitch?

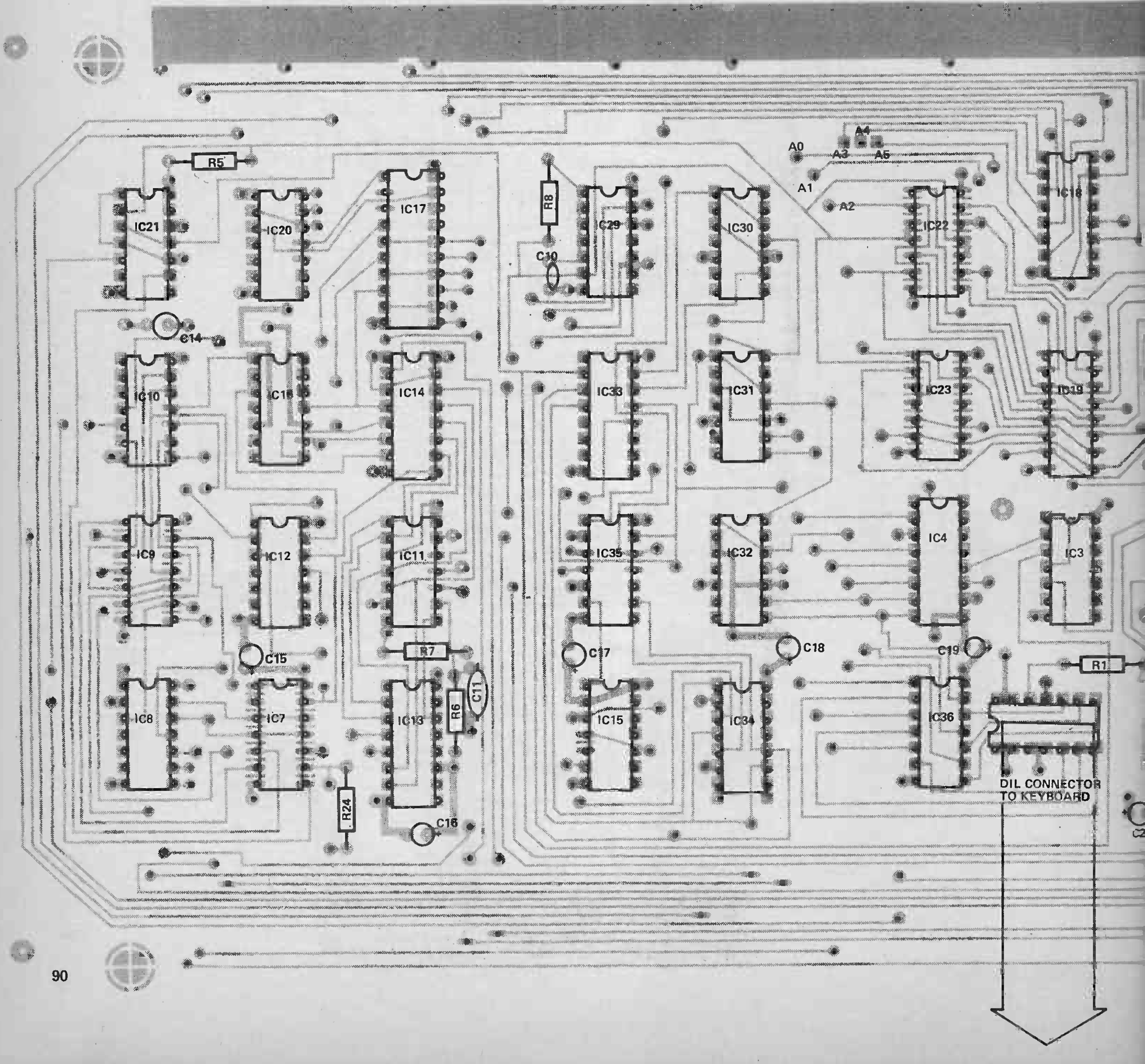
There is no 'correct' solution other than a voice per note.

Two modes of operation have been provided. Mode A permits note stealing, where the oldest channels are reassigned to new notes, when the selected number of notes is exceeded on the keyboard. Mode B does not allow more than the selected number of notes to be assigned. Extra notes are ignored.

The digital electronics is best considered as consisting of a series of modules.

The 'Total Gate Counter' is used to keep count of the number of channels in use at any point in time.

Fig.3 Component overlay of the master clock/timing generator. Dots indicate through-board connections on the double-sided PCB.



Airwaves

Sony technologists have recently developed what they consider to be "the most advanced portable communications receiver in the world." The ICF-2001 receiver is no bigger than a hard-cover book and has the capability of capturing English-language short-wave programmes from Australia to Finland, foreign language services from all four corners of the earth, ship-to-shore, amateur and Citizens Band as well as standard medium-wave, long-wave and FM broadcasting. It has a built-in minicomputer instead of a tuning dial, so instead of searching for the frequency of your favourite (or a new) station, with this model you simply punch in the frequency you require on a pocket calculator style keyboard, or instruct it to scan wave bands and stop automatically each time it reaches an 'on air' transmission. If it's a station you want to listen to frequently, you simply enter it into the computer memory and recall it by pressing one of the six 'preset' buttons. And that's not all! Other features include dual PLL synthesiser circuit for stable reception, auto-scan tuning, manual tuning, preset tuning, liquid crystal multi display, electronic 90 minute sleep timer and powerful 6 W output. The ICF-2001 sells for a mere £150. For further information contact Sony (UK) Ltd, Pyrene House, Sunbury Cross, Sunbury on Thames, Middlesex, TW16 7AT.

Boffin Badge

The following is not to be mistaken for an ETI special offer. Address all your orders clearly to the Physics Dept. at Imperial College.

The Physics Dept.
Imperial College

Dear Sir,

This is an unsolicited Press Release to inform you of my latest, and rather wacky, entrepreneurial venture.

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'I am A. S. Lipson'.

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The Mysterious Physicist

P.S. Discounts are naturally available on large orders.

Royal Award

On September 11th this year HRH the Prince of Wales launched a new award — a challenge to industry and people with new ideas. The Prince of Wales Award for Industrial Innovation will be presented in June 1982 following a competition promoted and organised by the BBC programme 'Tomorrow's World'. The aim is to encourage people with innovative ideas to get them accepted by an industrial backer, eventually to have their ideas taken into production. The first step will be to invite people to send their ideas to 'Tomorrow's World' for scrutiny. The competition is open to institutions and individuals alike. The closing date for this first stage is December 31st 1980.

Certificates will be presented to the winners (who will be expected to have found a commercial backer to qualify) at the end of the current series of programmes. During the following twelve months those ideas will be developed and by June 1982 the judges will decide who has made the most promising, enterprising and exciting commercial development. The winner will be awarded the Prince of Wales Award for Industrial Innovation and Production for that year. For further details contact: Unit Manager, 'Tomorrow's World', BBC TV, London, W12 8QT.

Project 80

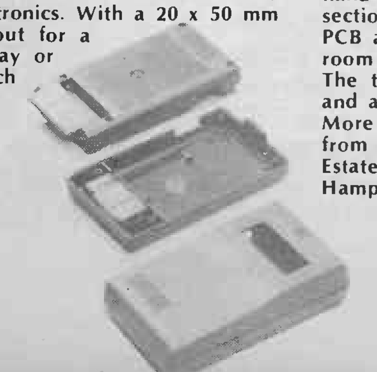
The number of inquiries we receive on the Project 80 modular synthesiser seems to be inversely proportional to the number of shopping days left to Christmas. Is it going to finish in the December edition? Will it be miraculously transformed into Project 81 as from the January 1981 edition? Are we going to continue to support it?

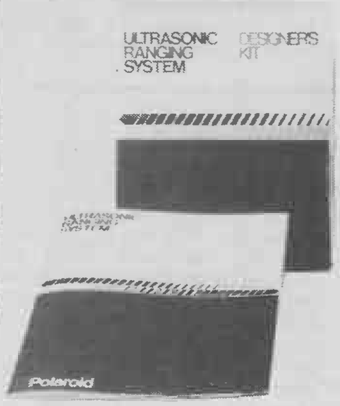
Well, it won't be finished in this issue and it won't be transformed into Project 81, but we certainly are going to continue supporting this very, very popular series. We are going to bring together in one magazine all the Project 80 articles published to date (with new additions or amendments, where necessary) together with three new modules, the keyboard design, the keyboard controller, setting-up procedures and a brand new section on module and whole-system applications written by the system designer, Charles Blakey.

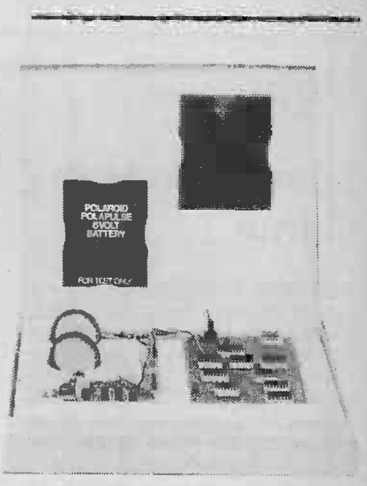
Watch out for the Project 80 handbook on the bookstalls in January.

Pocket Box

A new pocket-sized, hand-held case is now available from Vero Electronics. With a 20 x 50 mm cut-out for a display or switch panel and an integral battery compartment (PP3), it's ideal for remote control handsets and hand-held instruments. The top section will take a 71 x 107 mm PCB and the bottom section has room for a 56 x 105 mm board. The two sections snap together and are secured by four screws. More information is available from Vero Electronics, Industrial Estate, Chandler's Ford, Eastleigh, Hampshire SO5 3ZR.







Sound Designing

The ultrasonic rangefinding system used in Polaroid's SX-70 sonar focusing cameras since 1978 is now available separately as a designer's kit. Polaroid can supply an attractive kit containing two sonar transducers, the necessary circuitry to convert time into distance, two 6V Polapulse batteries, a LED readout and a technical manual. Both of the transducers emit and receive four-frequency ultrasonic signals.

Applications? The system has already been used in the unlikeliest of places. When Gossamer Albatross crossed the Channel by pedal power, it had to be flown at a precise height above the brine. However, none of the commercially available altimeters give accurate readings below about 20 m. Gossamer wave-hopping was achieved using a custom-built altimeter incorporating the Polaroid system.

Ultrasonic rangefinding could be useful to a pilot of any aircraft manoeuvring in limited air space close to the ground or to a driver who doesn't have all-round vision from his cab. In fact, any situation where a measurement of distance has to be converted into an alarm signal or a control signal could put the Polaroid kit to good use. Several medical applications spring to mind immediately — automatic focusing glasses (especially for cataract patients), sonar cane for the blind, obstacle warnings for wheelchair-users, etc.

For more information on the Polaroid Ultrasonic Ranging System Designer's Kit, contact Polaroid (UK) Ltd, Ashley Road, St Albans, Hertfordshire.

HOW IT WORKS : DIGITAL CONTROL PS1, 2, 3

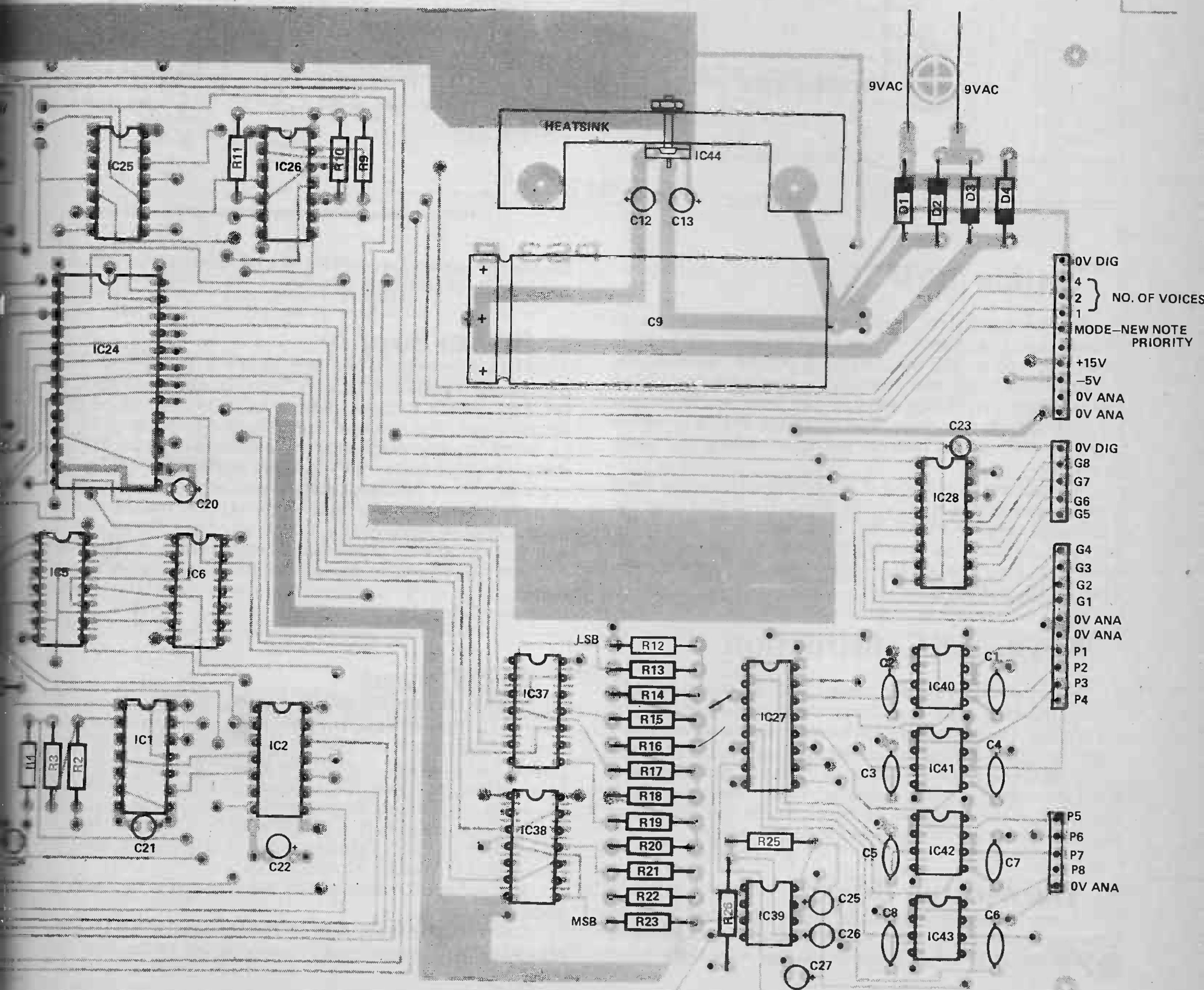
The 'Channel Position Counter' decides which will be the next channel to be assigned. In mode B, if a channel is already in use, then it will skip it and continue on until it finds a free one. In mode A, the channel position counter indicates the oldest channel which will then be reassigned if no free channel is available.

The 'Channel Status RAM' stores the pitch and gate parameters, the number of parameters stored being determined by the selector switch SW1. Data bits D0 to D5 generate pitch and bit D6 is the gate signal. The 'Same Frequency Comparator' comes into operation when a note is released. When this occurs, the circuit removes the gate signal from the respective memory location but rewrites the pitch data which would otherwise be cleared!

The electronics in Fig.2 generate gate and pitch signals which are fed to the synthesiser voices. Some typical waveforms are shown in Fig.9 for a four note selection. Each note pressed generates a gate and pitch signal and assigns a voice channel. When a note is released the gate signal is lost, but the pitch value remains until the channel is reassigned. The gate signals are distributed via an eight way addressable latch, IC28. The pitch signals are distributed via an eight way multiplexer driving eight sample and hold units (IC27,40,41,42,43). In a four voice system IC42,43 are omitted.

The pitch data is generated by a precision DAC (IC37,38,39). This converts the six bit code into an analogue voltage using an R/2R network.

A typical DAC error is shown in Fig.11. When the MSB of the code changes from 0 to 1 the step size is too small, which results in an error for all codes where the MSB is 1. Generally the worst errors are generated at the changeover point of the high bits of the code, Fig.12. The synthesiser is exceptionally sensitive to errors of this nature and so the DAC must be accurate to ten bits even though it is only converting six bits. A ten bit accuracy will give a worst error of 1 part in 32 (3% of a semitone) per step, which is only likely to occur when bits D5 and/or D4 change state. To obtain this performance 0.1% tolerance, 25 ppm/°C metal film 100k and 200k resistors are used. For superior performance the resistors can be matched for a 2 to 1 ratio using the best matched pairs at the MSB end of the DAC. Using this technique worst step errors of about 1.5% can be obtained. The DAC is powered from a + 5V3 reference voltage, which results in a + 1V/octave output.



PARTS LIST : PS1, 2

Keyboard Multiplexer

Resistors all ¼ W 5%

R1,2	100k
R3	1k0
R4	5k6
R5	4k7
R6-18	2k7

Capacitors

C1-5	10u 16 V tantalum

Semiconductors

IC1-7	4051
IC8	74LS42
Q1	BC212L
D1	1N4148

Miscellaneous

PCBs PS1 and PS2, 8 off 16 pin DIL sockets, 14 pin DIL socket, 49-note keyboard, 49 keyboard contacts (single pole), full length bus bars, self-adhesive cable clamp.

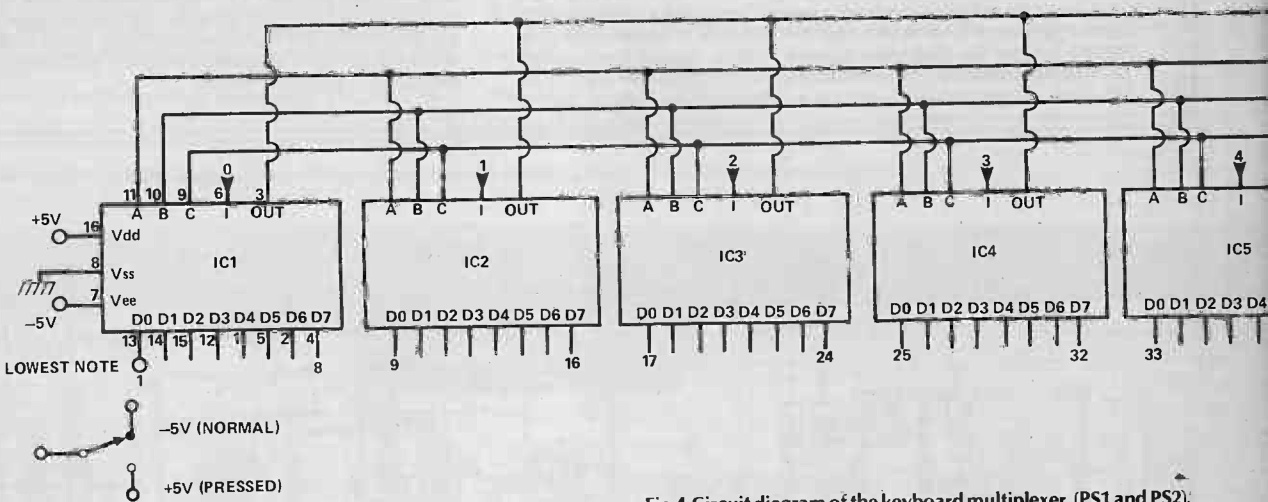


Fig.4 Circuit diagram of the keyboard multiplexer. (PS1 and PS2).

Testing the DAC

Set the number of voices to one and measure pitch voltage one. Use note C to generate octaves. The voltage should be 1 V ± 2% per octave. If possible measure the voltage for each note of the keyboard, using a 4½ digit DMM. The semitone step change should be 83.3 mV ± 3%. Repeat for pitch outputs two, three, four. If you are unable to do this you can rely upon a musical ear when driving the voice modules!

Another test is to remove IC10 and link pin locations 7 and 8 with a piece of wire. The DAC (IC39, pin 1) will then draw out a full range ramp (64 steps) which must have NO VISIBLE step errors. Be careful when inserting the precision resistors. Don't bend them too close to their body and don't overheat them when soldering them in.

Keyboard Construction

Bend the end wire on the contact blocks as shown in Fig.13. Thread the 49 contact blocks onto the two bus bars, making certain that the contact wire is between the two bars. Use a clean cloth for handling them to avoid grease contamination.

Glue the blocks into position on PS1 and 2 and then solder the bent wire ends into position. When the assembly is mounted on the keyboard adjust each wire contact so that there is a 1 mm clearance between it and the plunger.

The Mother Board

The mother board distributes the common and independent synthesiser parameters from the panel boards to the individual voices, which plug into the 50-way voice slots. It also houses the power supply, LED drivers, portamento circuits and output volume control.

Power Supply

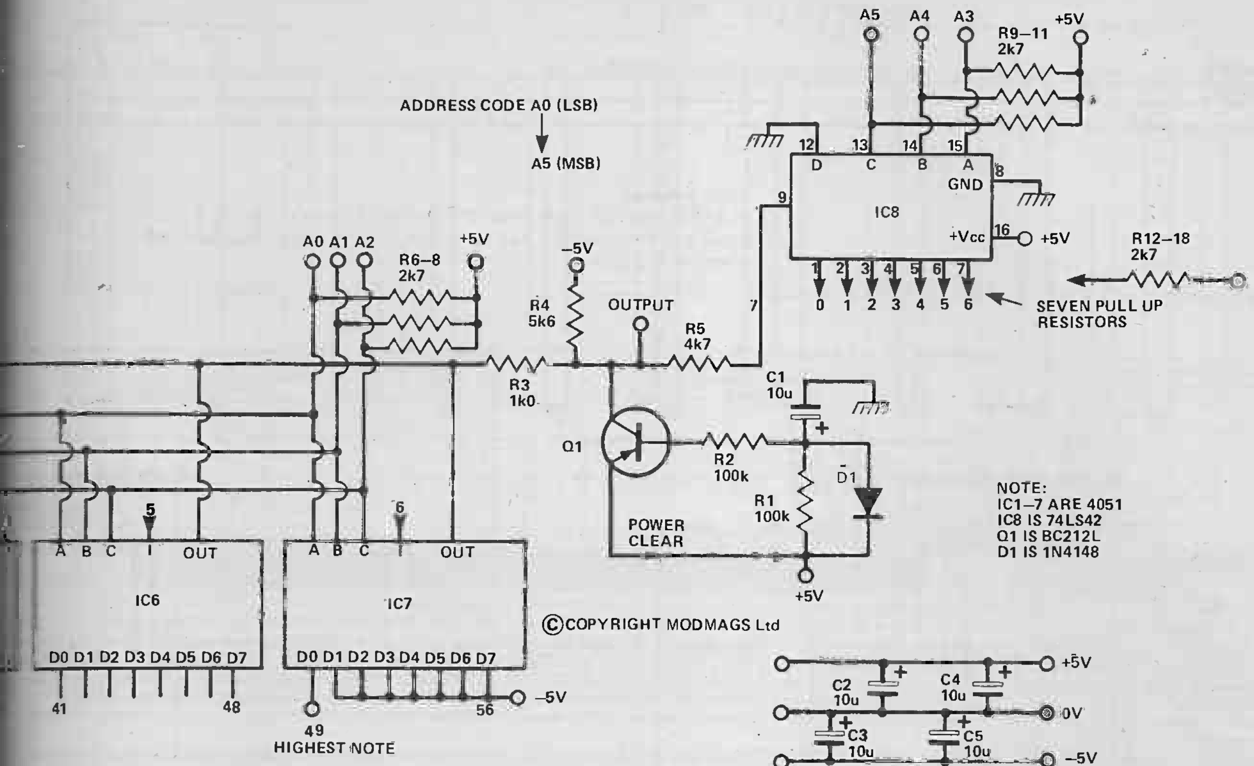
The power supply produces ± 15 V and −5 V. The positive rail is generated by a precision voltage regulator (IC1, Q4, Fig.7) which is mirrored by IC2,Q2,Q3 to produce the negative rail. The positive rail should be set to +15 V (± 10 mV) by adjusting PR1, with no voices plugged in. Check that the negative rail is −15 V ±75 mV and that the −5 V rail is −5 V ± 200 mV. Now plug in the voices and recheck the supply rails. Allow the unit to 'burn in' for 24 hours and then readjust the positive rail to +15 V if necessary. Lock PR1 into position with a small blob of nail varnish. All the oscillator frequencies and pitch spreads depend on the power supply being stable and so if PR1 is altered then the tuning of the whole machine will be lost!

LED Drivers

The LED drivers (IC4, IC5 Fig.8) buffer the TTL gate signals from PS3 to the voice LEDs on PS6. These LEDs can be used to test the digital control section. Set the 'Nos. of voices switch' to four. A new LED will come on, in sequence, as notes on the keyboard are held down. The LEDs will go off when the respective notes are released. Repeatedly tap a single note. This will cause an illuminated LED to cycle around the four voices. If you tap two or three notes, then two or three LEDs respectively will cycle around. Now switch to two voices.

Voices 1 and 3 turn on and off together and so do voices 2 and 4.

Switch to one voice. Now all the LEDs will act in unison. Next switch to eight voices and repeatedly tap a note. The four



DIL	CONNECTIONS
1	+5V
2	+5V
3	–5V
4	–5V
5	A2
6	A1
7	A0
8	A3
9	A4
10	A5
11	+5V
12	OUT
13	GND
14	GND

PARTS LIST : PS3

Resistors all ¼ W 5% unless otherwise stated

R1,6,7,8	1k0
R2,3,4,5,9,10,11,24	2k7
R12,13,15,17,19,21,23	200k 0.1% metal film
R14,16,18,20,22	100k 0.1% metal film
R25	15k 1% metal film
R26	8k2 1% metal film

Capacitors

C1-8	47n polycarbonate
C9	4700u 25 V electrolytic
C10	1n0 ceramic
C11	22n polycarbonate
C12-27	10u 16 V tantalum

Semiconductors

IC1	75107
IC2	74LS175
IC3,15,21	74LS74
IC4	2102
IC5,8	74LS00
IC6,35	74LS04
IC7	74LS10
IC9,12,20,34	74LS32
IC10,25,26	74LS08
IC11	7425
IC13	74LS193
IC14	74LS153
IC16,30,31,32	74LS93
IC17	74LS244
IC18,19	74LS367
IC22,23	74LS266
IC24	6810
IC27	4051B
IC28	74LS259
IC29	74LS13
IC33	74LS42
IC36	7407
IC37,38	4041B
IC39	1458
IC40,41,42,43*	TL082
IC44	7805
D1-4	1N4002

*IC42,43 only required in eight-voice system

Miscellaneous

PCB PS3, 5 off 8 pin DIL sockets, 28 off 14 pin DIL sockets, 9 off 16 pin DIL sockets, 20 pin DIL socket, 24 pin DIL socket, DIL connectors and ribbon cable, Redpoint 1.25GYBA heatsink, cable clamp.

LEDs will come on in sequence and then there will be a four note gap whilst the unit addresses the other four voices that live in the expander module.

Now the mode selection. Select four voices and mode A. Play and hold down a four note chord with the left hand. Now press four notes at once with the right hand, and then release the right hand. The LEDs will go out, indicating that the right hand 'stole' the four voices and then released them, in doing so turning them off. Repeat this procedure in Mode B. The LEDs will remain on.

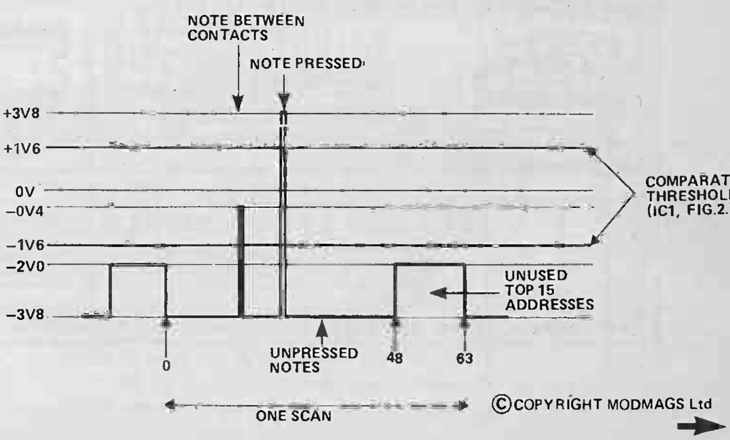


Fig.5 Output of keyboard multiplexer.

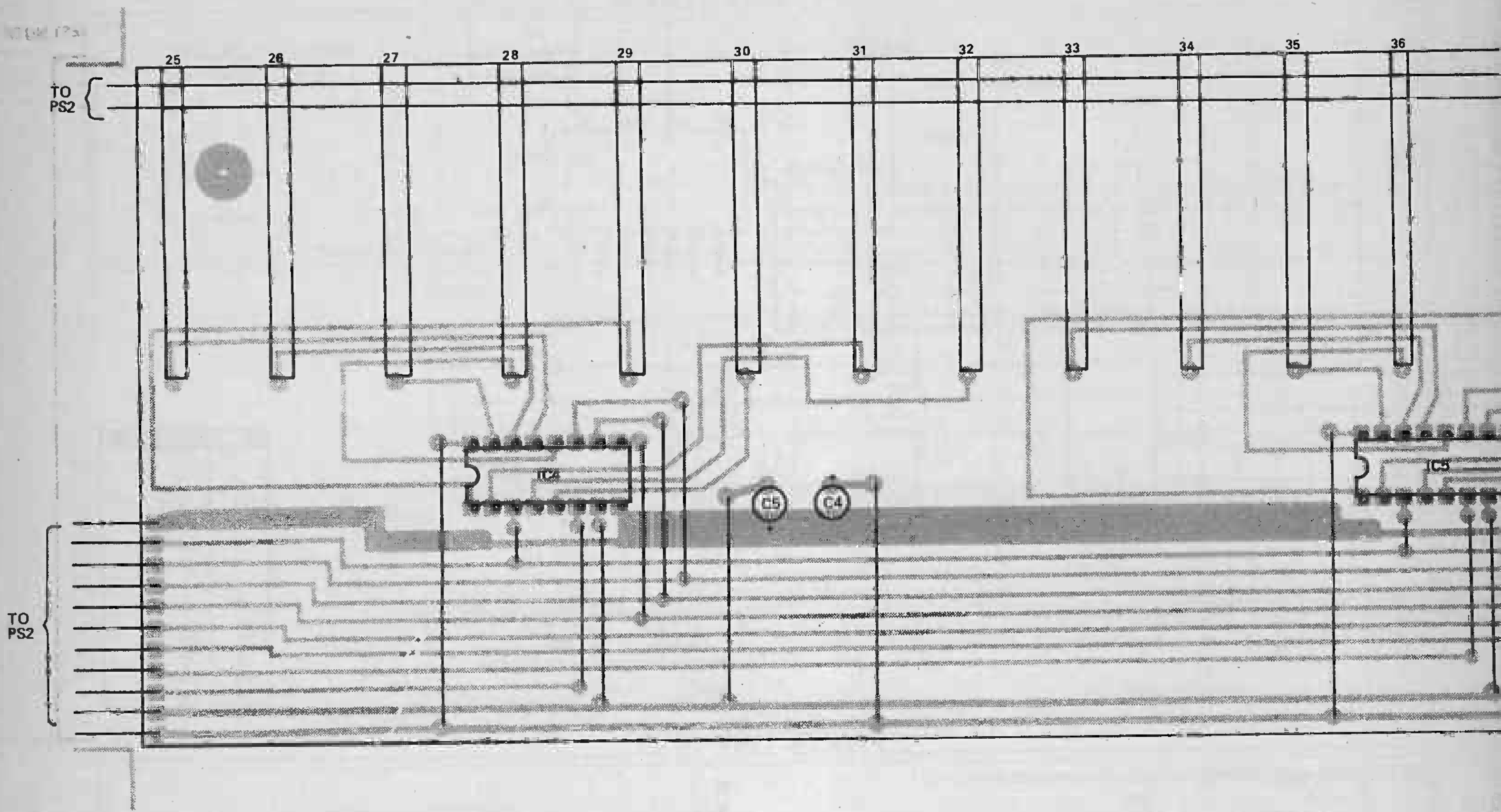
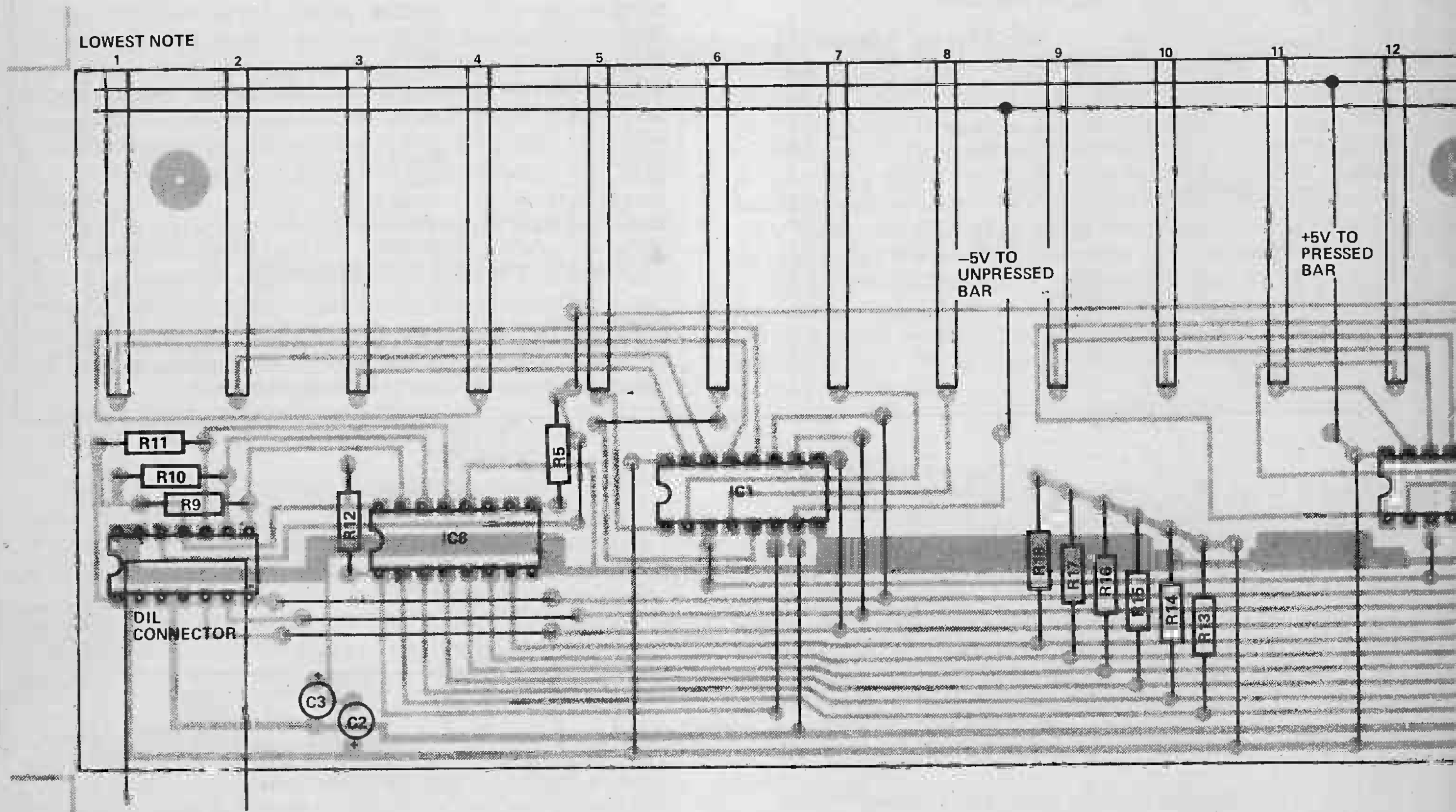
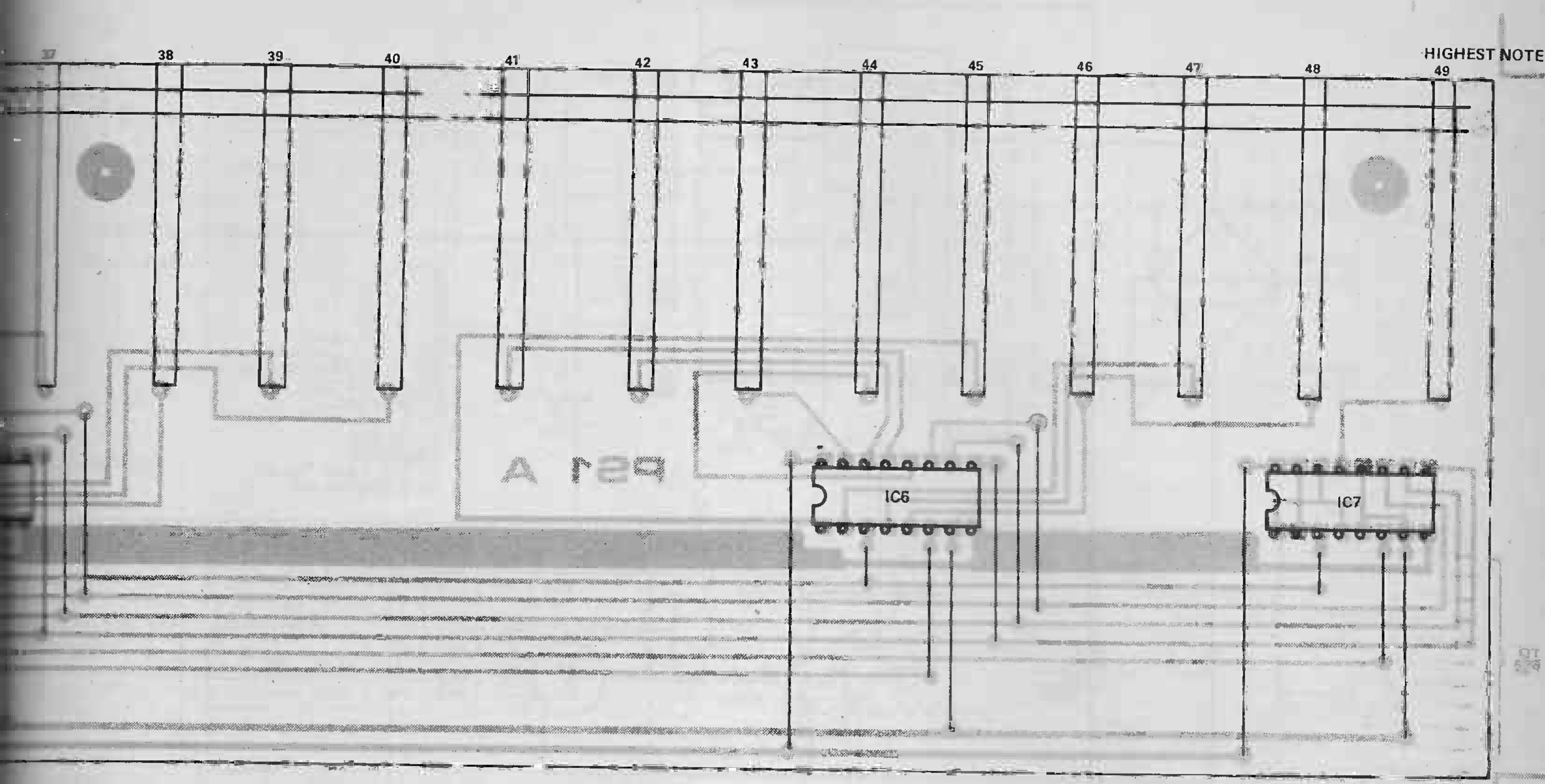


Fig.6 PS1 (above) and PS2 (below) together form the multiplexed keyboard.




37
38
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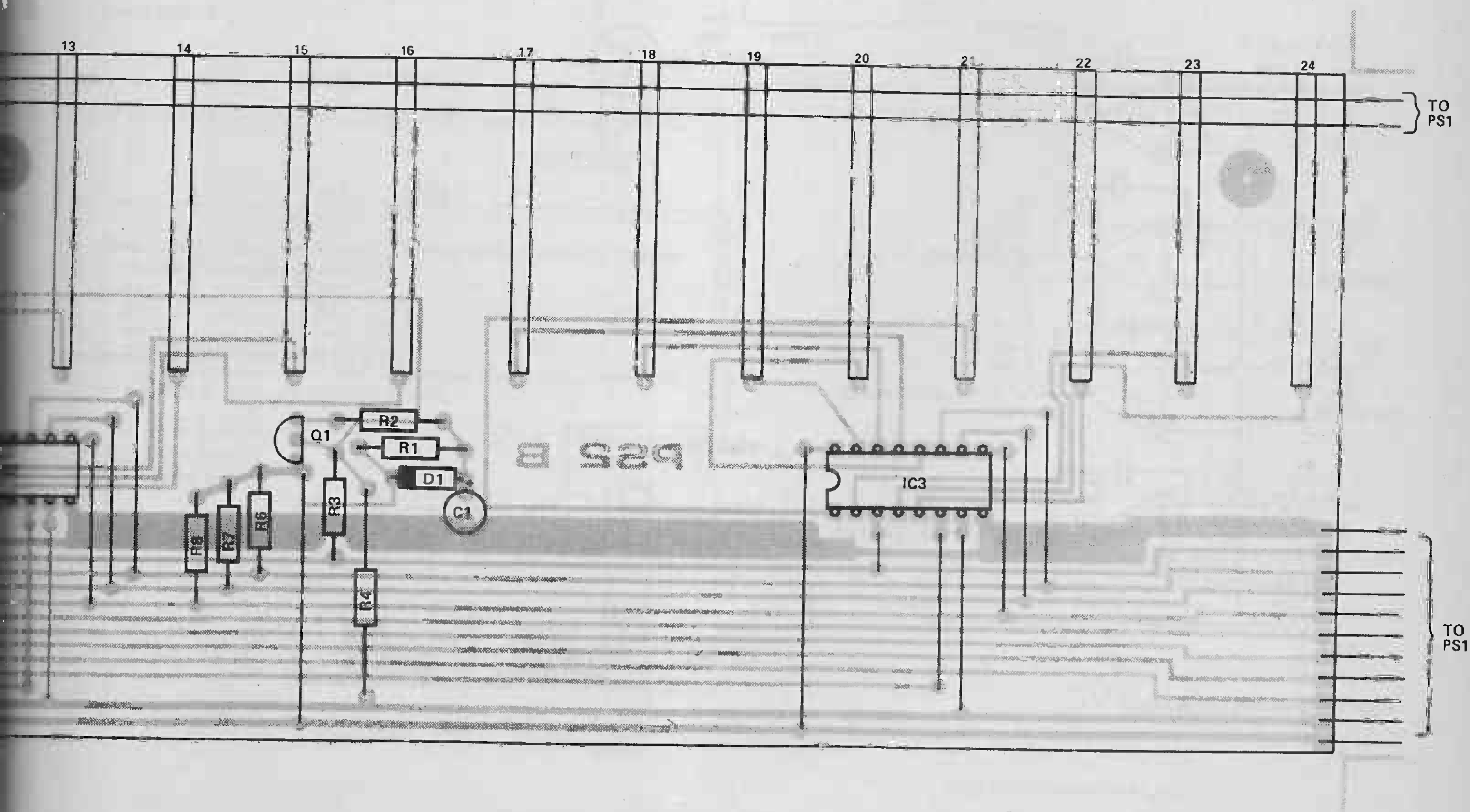
44
45
46
47
48
HIGHEST NOTE
49
IC6
IC7


13
14
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18
19
20
21
22
23
24
TO PS1
Q1
R2
R1
D1
C1
R3
R4
R6
R7
R8
IC3
TO PS1

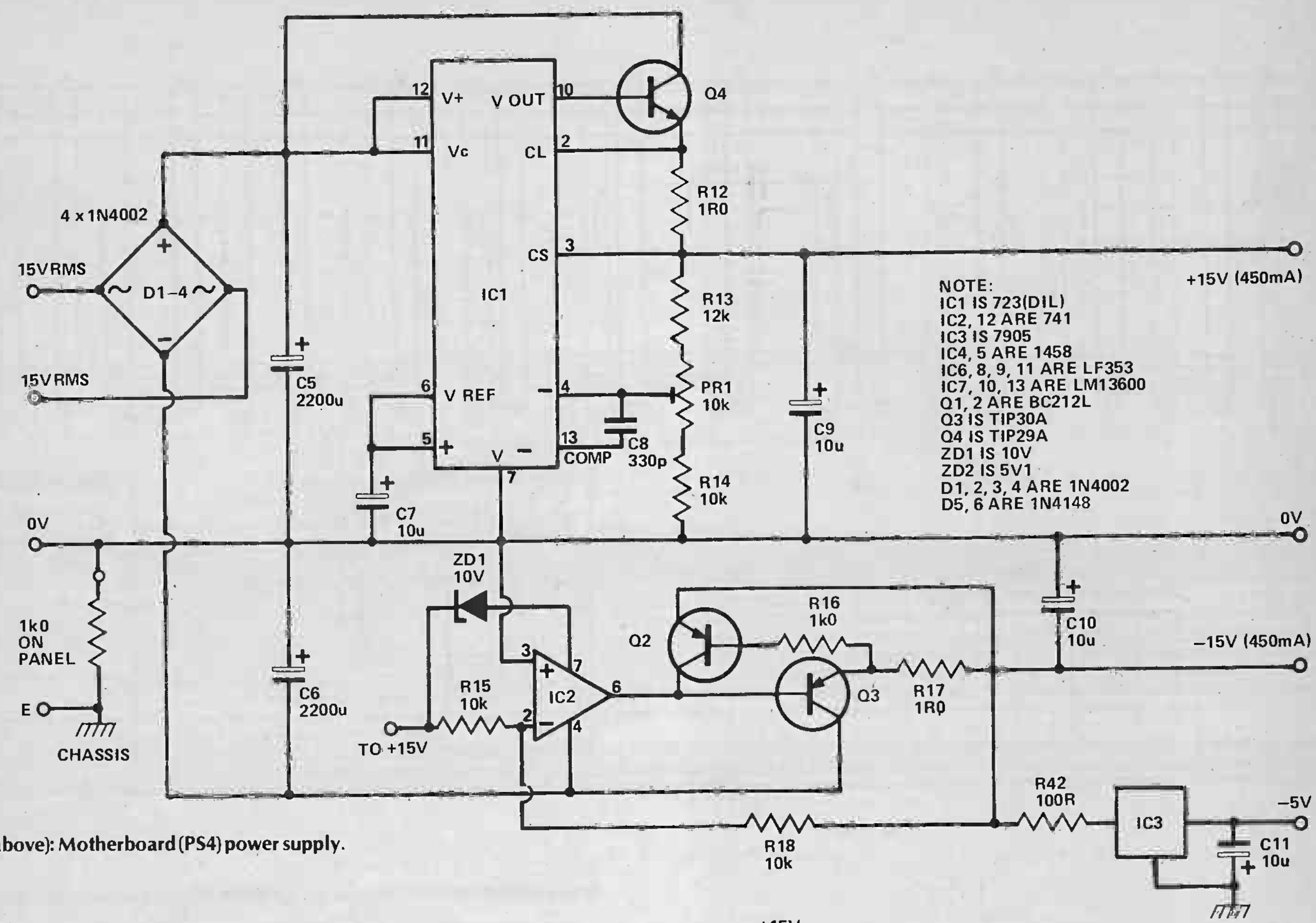
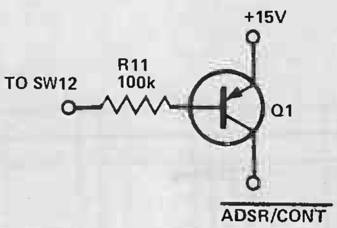


Fig.7 (above): Motherboard (PS4) power supply.



POWER SUPPLY

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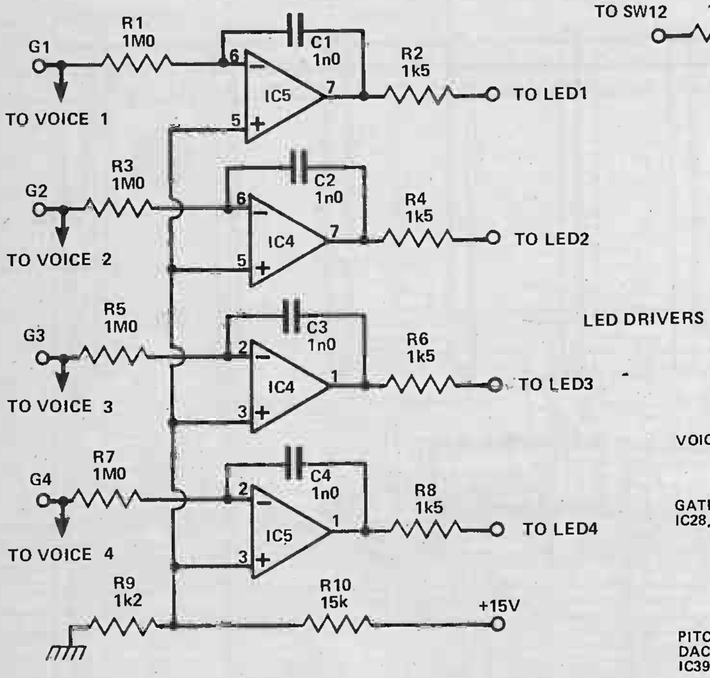
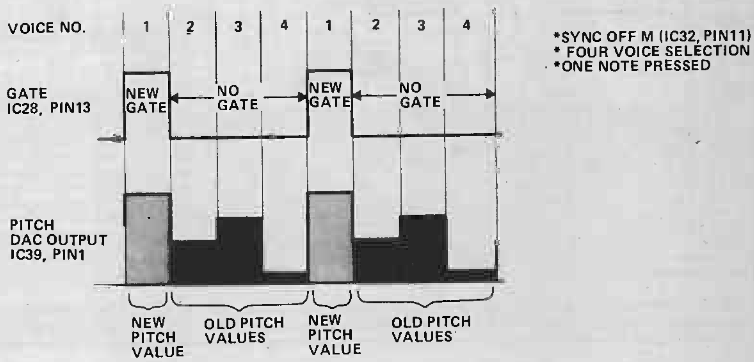
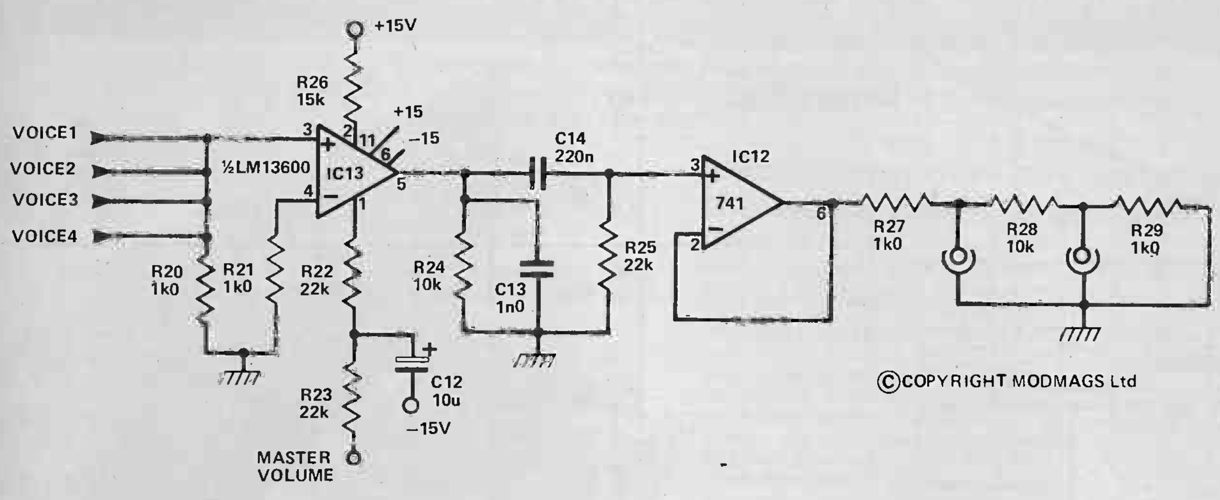


Fig.8 Motherboard LED drivers.

Fig.9 (below) Pitch and gate parameters.





PORTAMENTO

	RA	RB	RC	CA	CB
1	R30	R31	R32	C15	C16
2	R33	R34	R35	C17	C18
3	R36	R37	R38	C19	C20
4	R39	R40	R41	C21	C22

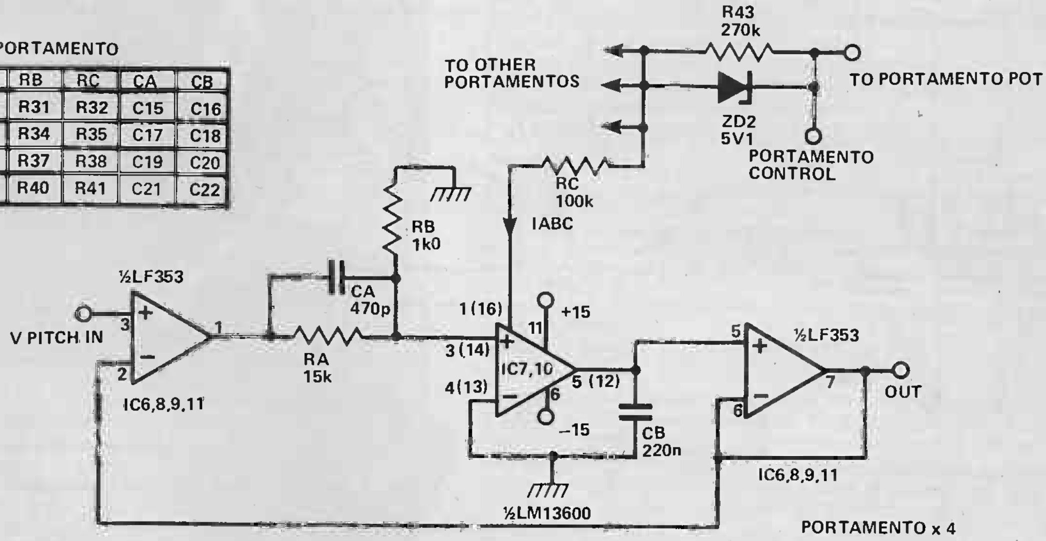
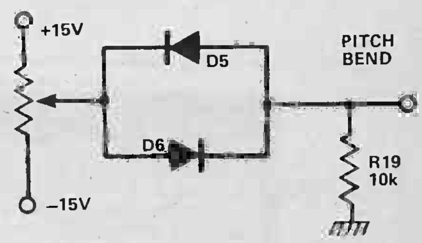


Fig.10 Motherboard portamento, master volume and pitch bend controls.

BUYLINES

Powertran Electronics can supply a complete kit of parts for each option of the Transcendent Polysynth.
1 voice £320
2 voices £368
4 voices £464
4 voice expansion kit £275
All prices are exclusive of VAT. Powertran Electronics, Portway Industrial Estate, Andover, Hants SP10 3NM.



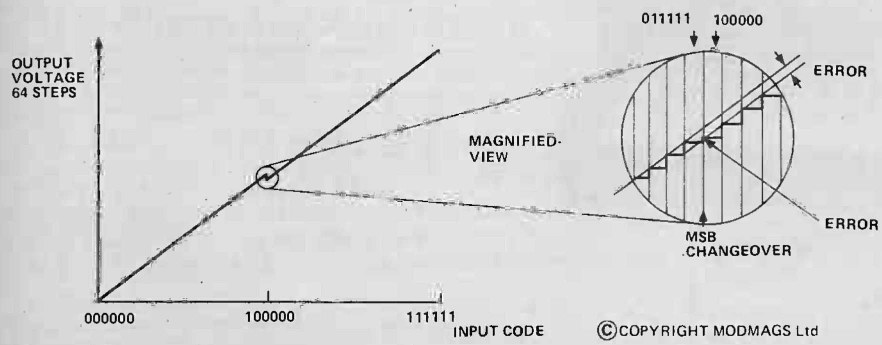


Fig.11 (left) Typical DAC error.

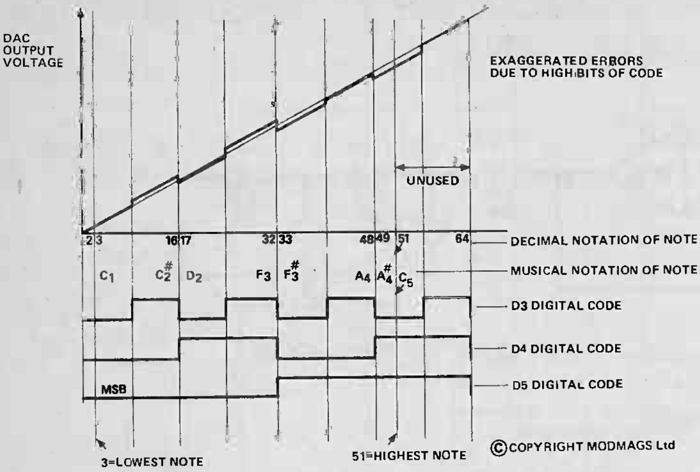


Fig.12 Points of worst error in the DAC

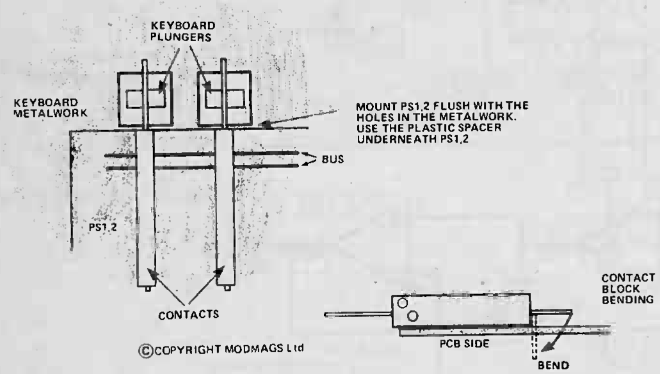
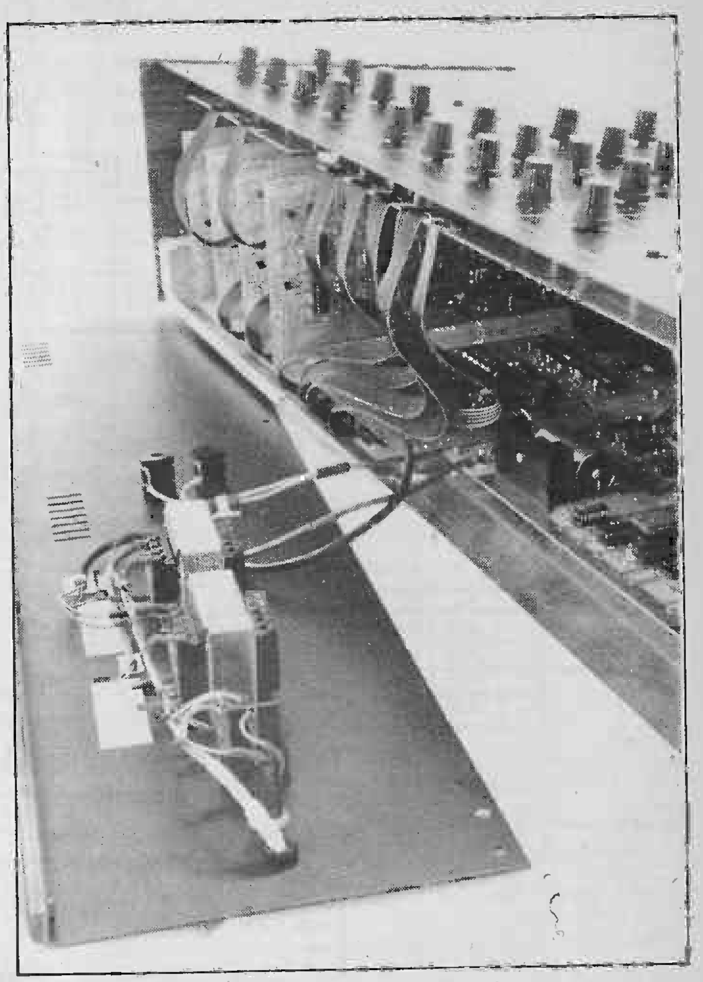


Fig.13 Mechanical assembly of keyboard plungers.



The power supply is mounted on the rear panel.

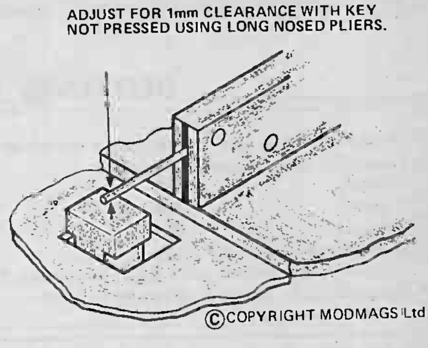
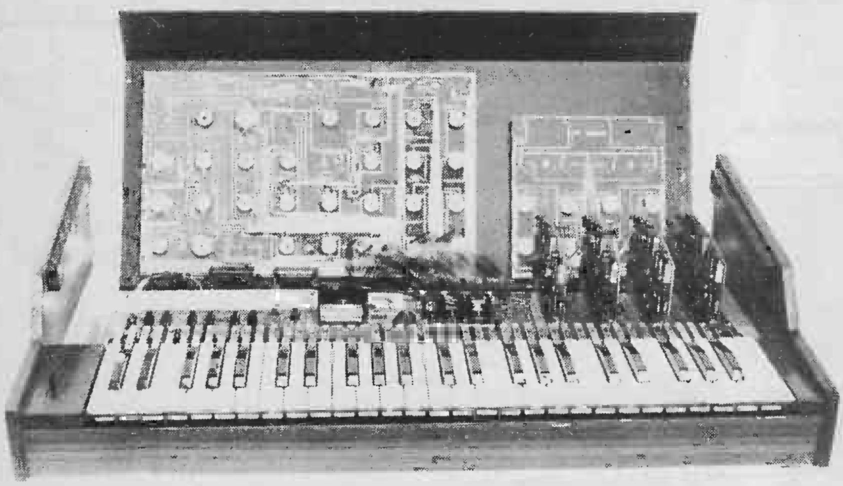


Fig.14 Adjustment of 1mm gap above keyboard plunger.



The top panel can be raised to show control board PS5 and PS6.

Next month we conclude the Polysynth project with details of the control and voice boards. (PS5, 6 and 7).

ETI

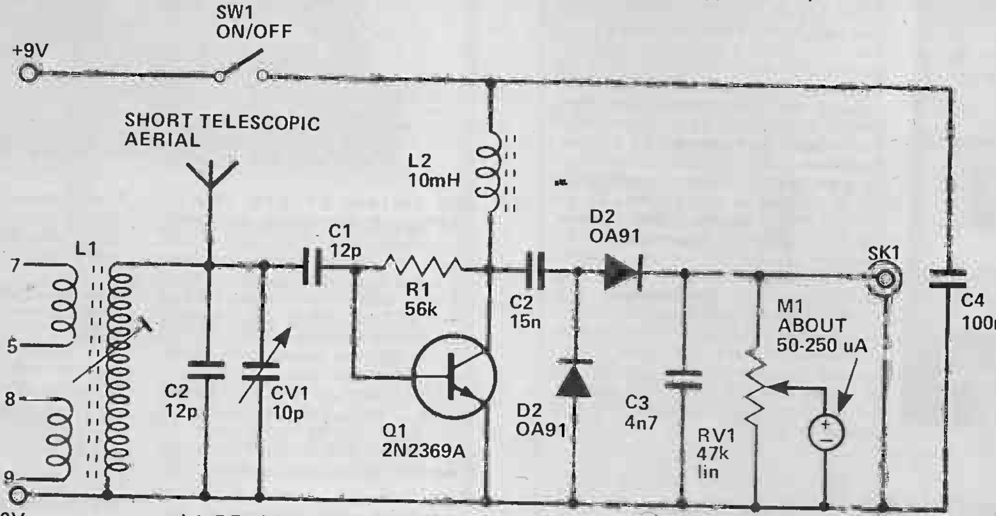
27 MHz Radio Control Monitor

A radio control monitor such as that described here is invaluable when setting up a radio control transmitter. This circuit has good sensitivity and it is unnecessary to have the monitor in very close proximity to the transmitter in order to obtain a signal of adequate strength even when it is used with a low power transmitter.

The signal picked up by the telescopic aerial is coupled directly into the tuned circuit. The core of L1 is adjusted so that CV1 is able to tune the monitor to any frequency within the 27 MHz radio control band. The setting of this core is not too critical, since the unit covers somewhat more than the entire band.

Q1 is used as a common emitter amplifier with L2 as its collector load and R1 to provide base biasing. C1 couples the signal in the tuned circuit to the input of the amplifier. The value of C1 is chosen to give optimum signal transfer. The high impedance signal in the tuned circuit is matched to the low input impedance of Q1 by a sort of capacitive divider action (the input capacitance of Q1 forming the other section of the capacitive divider).

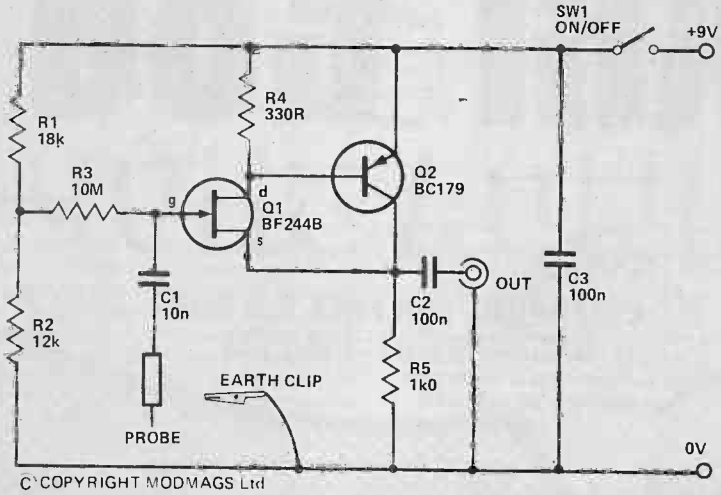
C2 couples the output of Q1 to a straightforward rectifier and smoothing circuit. This produces a positive bias voltage which is roughly proportional to the strength of the received signal. This bias voltage is used to drive M1, which gives a comparative indication of received signal strengths. An inexpensive meter is perfectly suitable for use in the unit due to the arbitrary scaling. RV1 can be used to reduce the sensitivity of the unit, if necessary. The modulation signal of the transmitter (if it is an AM type) can be monitored using a crystal earphone connected to SK1. Current consumption is approximately 10 mA.



Active 'Scope Probe

Most oscilloscope probes that provide a tenfold boost in input impedance (usually from 1M0 to 10M) are passive devices and consequently give a corresponding tenfold reduction in voltage gain. This active probe circuit gives an input impedance of approximately 10M shunted by only about 5 to 10pF and has a voltage gain of almost exactly unity. It will not have any significant effect on the bandwidth of the equipment when used with an oscilloscope whose bandwidth is 10 MHz or less and which is AC coupled (the probe is mainly intended for use when investigating high impedance audio or RF signals). Signal levels of up to about 3 V RMS can be handled without clipping, or about 6 V RMS if the supply voltage is raised to 18 V.

The obvious choice for a circuit of this type would at first seem to be a FET used in the source follower mode as this can provide a high input impedance, low input capacitance, low output impedance and nominally unity voltage gain. The problem with such a circuit is that the true voltage gain is only about 0.9 to 0.95, which would obviously have an adverse effect on the accuracy of any form of calibration fitted to the 'scope. Therefore, the circuit finally devised uses Q1 as a common source amplifier, which directly drives common emitter amplier Q2. There is a 100% negative feedback loop from the collector of Q2 to the source of Q1 and this gives the circuit almost precisely unity voltage gain. This arrangement retains the attributes of a source follower stage and gives good results in other respects as well. Gate biasing for Q1 is provided by R1-3 while DC blocking at the input and output is provided by C1 and C2 respectively. Current consumption of the unit is about 5 mA.



FOUR INPUT MIXER

Four into one will go with this mini-mixer. Why not build one for your band today?

Design and development by John Fitzgerald.

If you are in a band and your PA has a range of inputs you may wonder why you need a mixer. Well one of the advantages is that when your bass player thinks he is not loud enough to be heard over the rest of you, it's up to someone else to turn him up! Otherwise, you can so easily get that snowball effect where one musician turns up the volume only to be followed by all the rest and so on until every sentient being within earshot suffers premature deafness (pardon?).

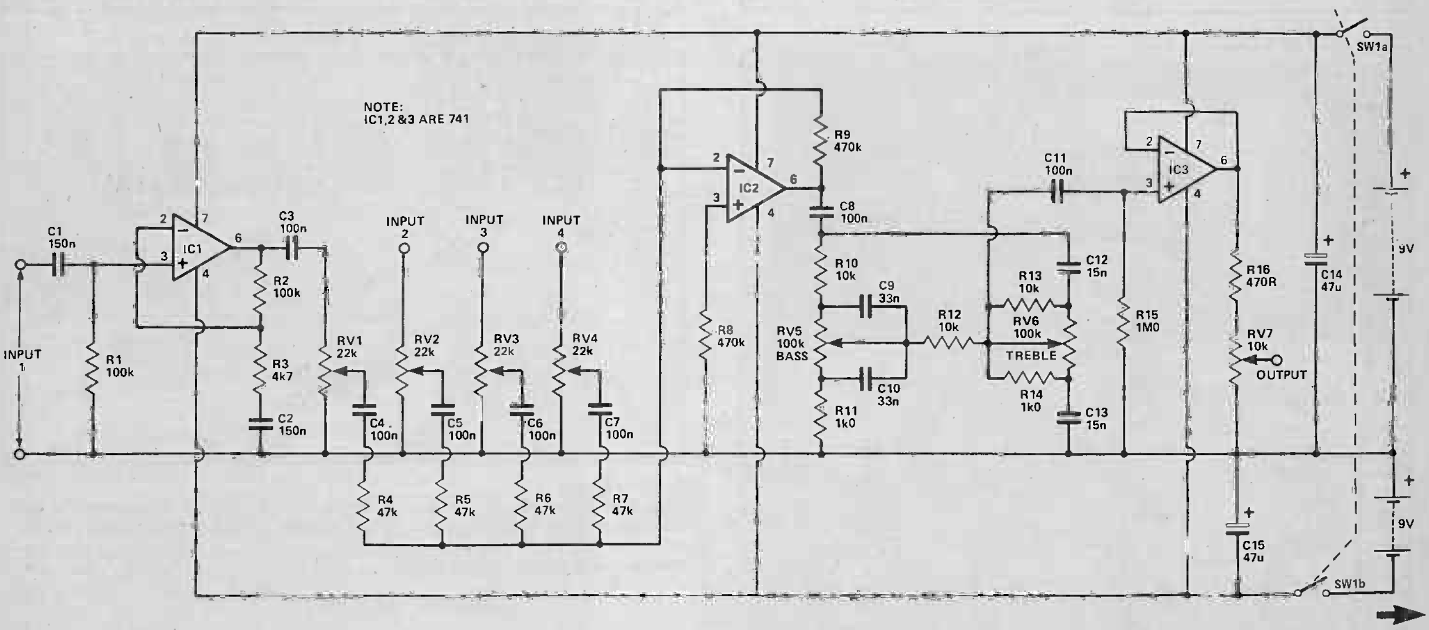
Many bands these days seem to have started on nothing and are maintained on a shoe string. If yours falls into this category, then this is the project for you. No-one is going to claim it's hi-fi but it is cheap. Using 741 op-amps will produce plenty of power versus cost.

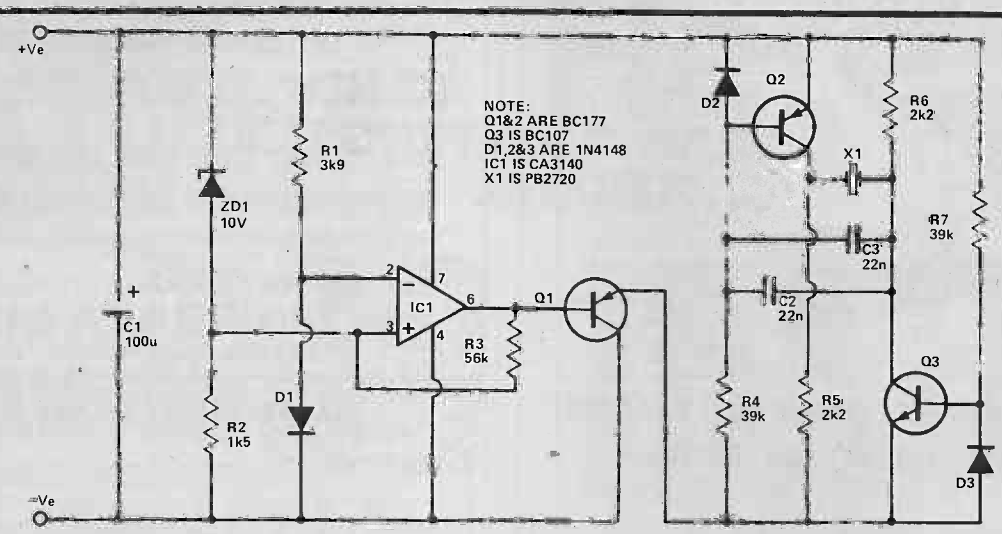
HOW IT WORKS

The heart of the circuit is IC2, an op-amp connected as a conventional 'virtual earth' summing amplifier. This stage mixes the input signals and also has a gain of 10 to compensate for the insertion loss of the passive tone control network. There are three high level inputs and one low level which is input to IC1. This stage provides non-inverting amplification with a gain of about 20. Independent volume controls are provided for each input and the signals are mixed in IC2 before being passed to the tone control network. This provides a 40 dB range of control with an insertion loss at midband of − 20 dB.

This output from the tone control is AC coupled to unity-gain voltage follower IC3. AC coupling avoids the problem of wiper 'track noise', which would result from the irregular capacitor charge and discharge currents. R16 is inserted in the output of IC3 to isolate the op-amp from the large capacitive load presented by a long length of screened cable. Potentiometer RV7 provides overall volume control and capacitors C14 and C15 decouple the power supply lines. Two 9 V batteries provide power for the unit and current consumption will be just a few milliamps.

Fig.1. Circuit diagram.





Car Battery Alarm

This circuit is intended for use with a 12 V car or boat battery and sounds an alarm if the battery voltage drops below about 10 V. An audible alarm is used in preference to the more usual indicator light because an audible signal is far less likely to be missed.

IC1 and its associated circuitry form a voltage detector circuit; IC1 is used here as a voltage comparator rather than an operational amplifier. R1 and D1 provide a stabilised potential of about 0V7 to the inverting input of IC1. The voltage supplied to the non-inverting input is equal to the supply voltage minus the nominal 10 V dropped through zener diode ZD1. Thus the potential at the non-inverting input will normally be about 2 V or more and since this is higher than the voltage fed to the other input the output assumes the high state. Therefore Q1, an emitter follower buffer stage, becomes cut off and no power is fed to the audio alarm circuit at its output.

If the supply voltage falls below about 10V7, then the voltage fed to the non-inverting input falls below the reference level at the other input and IC1's output goes low. Q1 is then switched on and power is connected to the alarm circuit fed from its output. Due to imperfections in the performance of practical zener diodes, the actual voltage at which the alarm is triggered is a little lower than the theoretical one and is typically a fraction over 10 V (or 11 V if preferred, by using an 11 V component in the ZD1 position). R3 is used to introduce a small amount of positive feedback which ensures that the unit switches cleanly from one state to the other and prevents erratic operation.

The alarm generator circuit uses Q2 and Q3 as an astable multivibrator and these drive ceramic resonator X1 with the anti-phase signals at their collectors. This gives the necessary large voltage swing to X1 and produces a reasonably loud output at about 2 kHz or so.

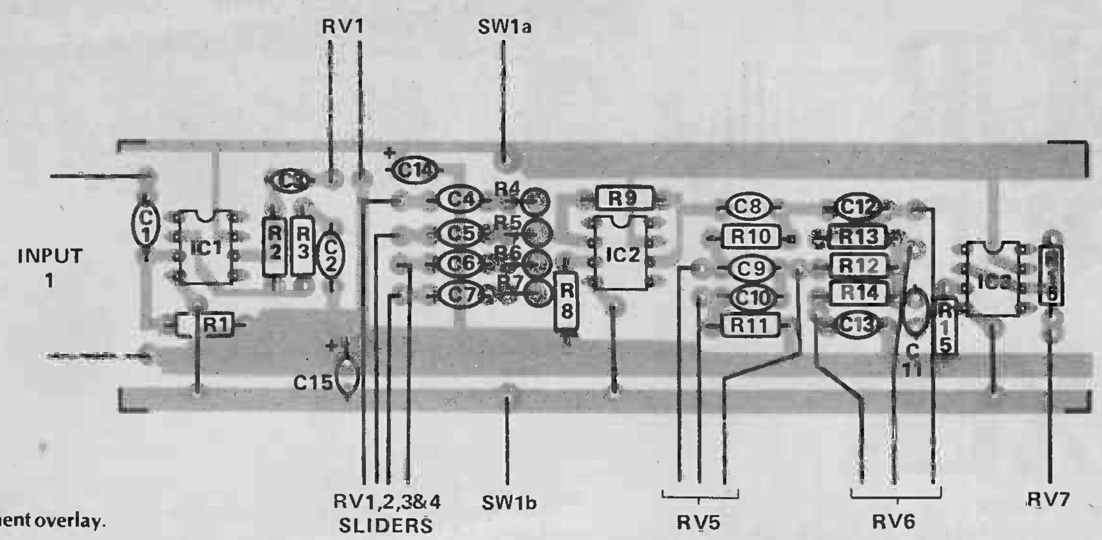


Fig.2. Component overlay.

PARTS LIST

Resistors all ¼ W 5%	
R1,2	100k
R3	4k7
R4,5,6,7	47k
R8,9	470k
R10,12,13	10k
R11,14	1k0
R15	1M0
R16	470R
Potentiometers	
RV1,2,3,4	22k logarithmic
RV5,6	100k logarithmic
RV7	10k logarithmic
Capacitors	
C1,2	150n polycarbonate
C3,11	100n ceramic
C4,5,6,7,8	100n polycarbonate
C9,10	33n polycarbonate
C12,13	15n polycarbonate
C14,15	47u 16 V tantalum
Semiconductors	
IC1,2,3	741
Miscellaneous	
Case, connectors, batteries, DPDT switch, etc.	

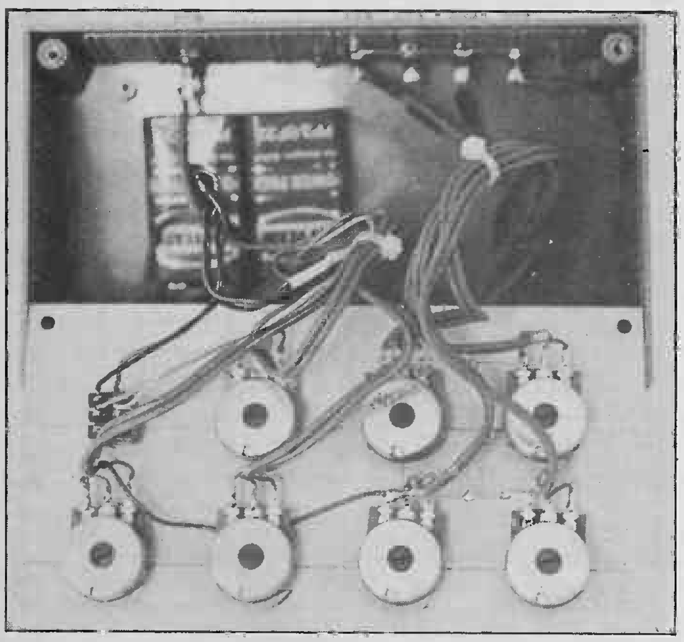
Construction

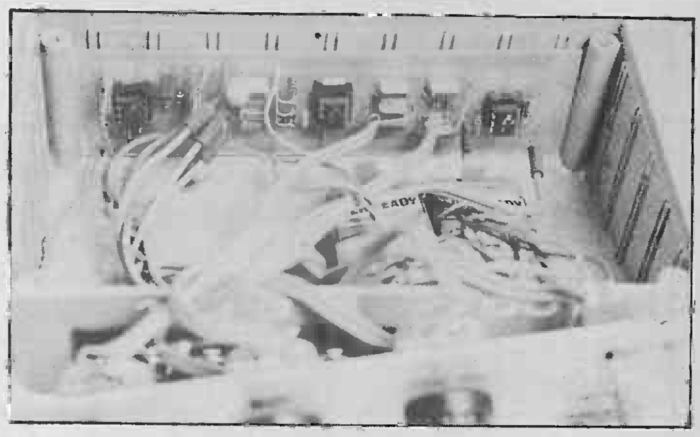
We built our unit into a small plastic case and mounted all the pots on the aluminium top cover, making the required connections with screened cable. Take care not to produce an 'earth loop' when connecting up. The remaining components can be mounted on our PCB; only three wire links are required to supply the op-amps' negative supply. We used polycarbonate capacitors mostly, taking advantage of their small size and good characteristics, though polyester types can be substituted. There are no special precautions to take. Just make sure you put the ICs in the right way up and get the polarity of the two tantalum capacitors right.

Remember, if you ever reverse-bias a tantalum cap by more than about 3 V it's dead certain that you'll have blown it up and it'll no longer be a capacitor; more a low-value resistor with the inevitable effect on circuit operation. No problems with this project, though. Simply build it, fix it and mix it!

BUYLINES

All the components for this project should be readily available. In case of difficulty, the polycarbonate capacitors are available from Watford Electronics or Electrovalue.





The PCB (above) tucks neatly away at one end of the case. Make sure the case is deep enough to allow clearance of the batteries and potentiometer bodies.

Front panel control layout (left).

ETI

BANDPASS AND BEYOND

Here's some more from Mr. Orr. Our circuit specialist looks at the development of bandpass design, switched capacitor techniques and some new ICs.

Many machines such as spectrum analysers and vocoders use analysing filter banks, these are often quarter octave devices extending over six octaves. If quarter octave filtering is to be successful then the bandpass filter responses must be very sharp, having almost flat tops and a fast roll-off slope at either side. A poor slope would mean that the filter bank would not be able to resolve incoming signals; for example a sinewave might give a high output in several of the channels. Also, a very peaky response would give large interfilter 'dips' in the overall response (Fig.1). An approximation to the square ideal response can be obtained by using multiple tuned filters. The response of a single pole bandpass filter is shown in Fig.2.

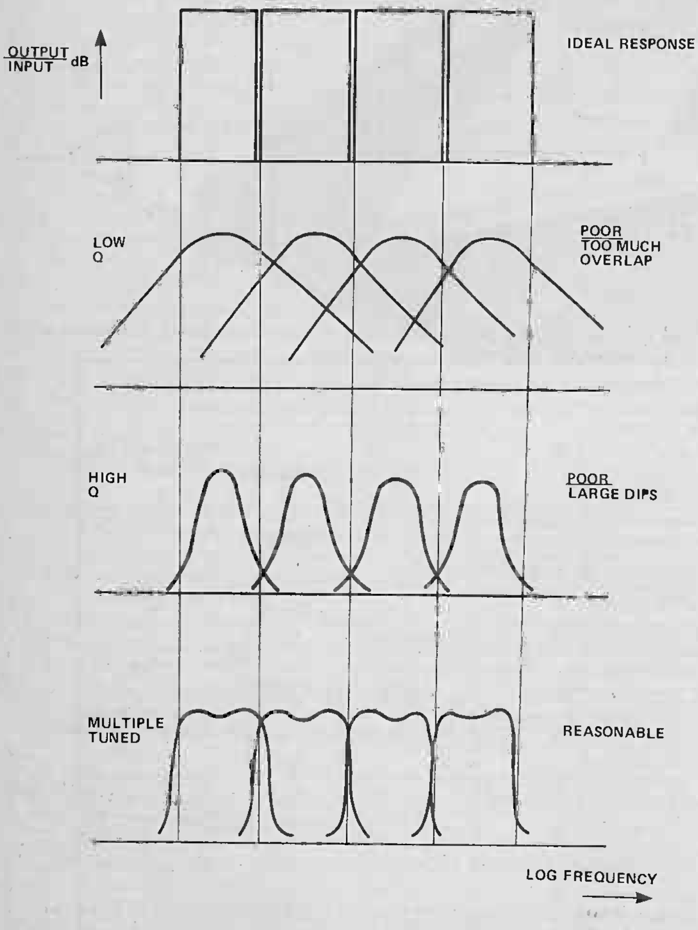


Fig.1. Various responses of an analysing filter bank.

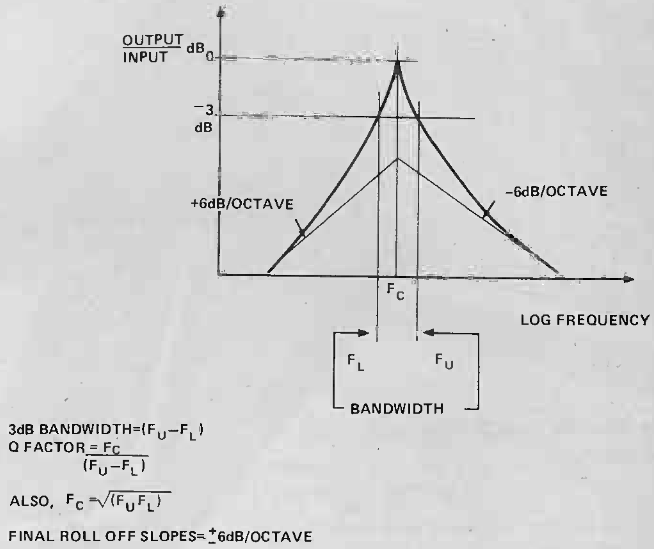


Fig.2. Single pole bandpass response.

By Design

When designing bandpass filters it is important to decide what type of filters to use. Figure 3 shows two bandpass responses. Response A is a peaky filter, whereas B has a flat top to it, but still has the same roll off as A. One sensible design solution would be to use bandpass filters for A, and a highpass/lowpass structure for B. A rule of thumb for making this decision is to calculate the fractional bandwidth:-

$$\frac{F_U - F_L}{F_U \times F_L}$$

If this is greater than unity then use lowpass/highpass filters, if it is less than unity use multiple tuned bandpass filters. Some standard bandpass filter designs are shown in Fig.4. The multiple feedback circuit requires only one op-amp, but is limited to low Q operation (less than 5) and the centre frequency and Q are interactive.

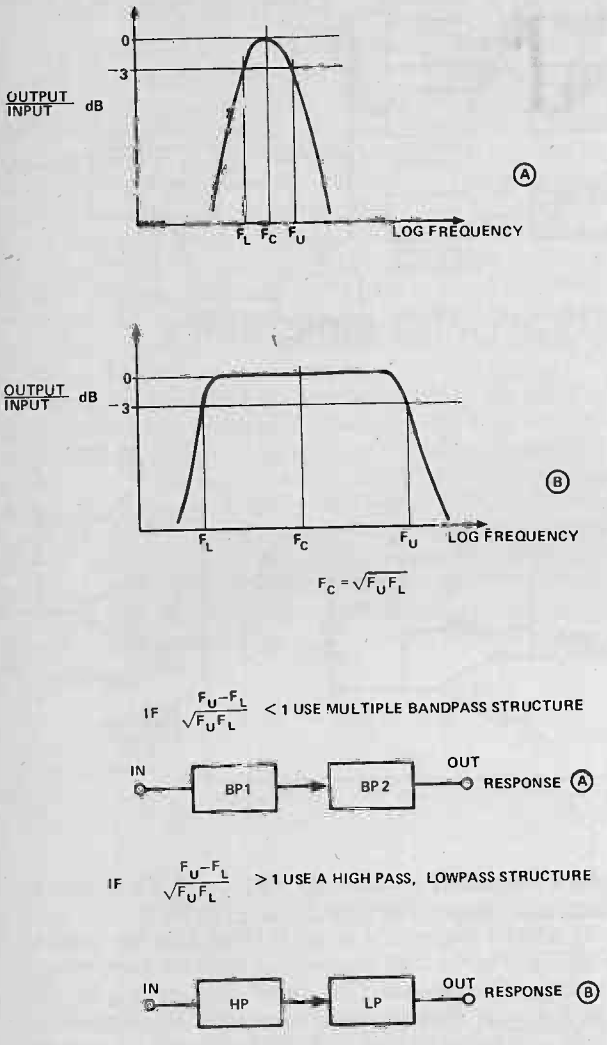


Fig.3. Two bandpass responses and how to realise them.

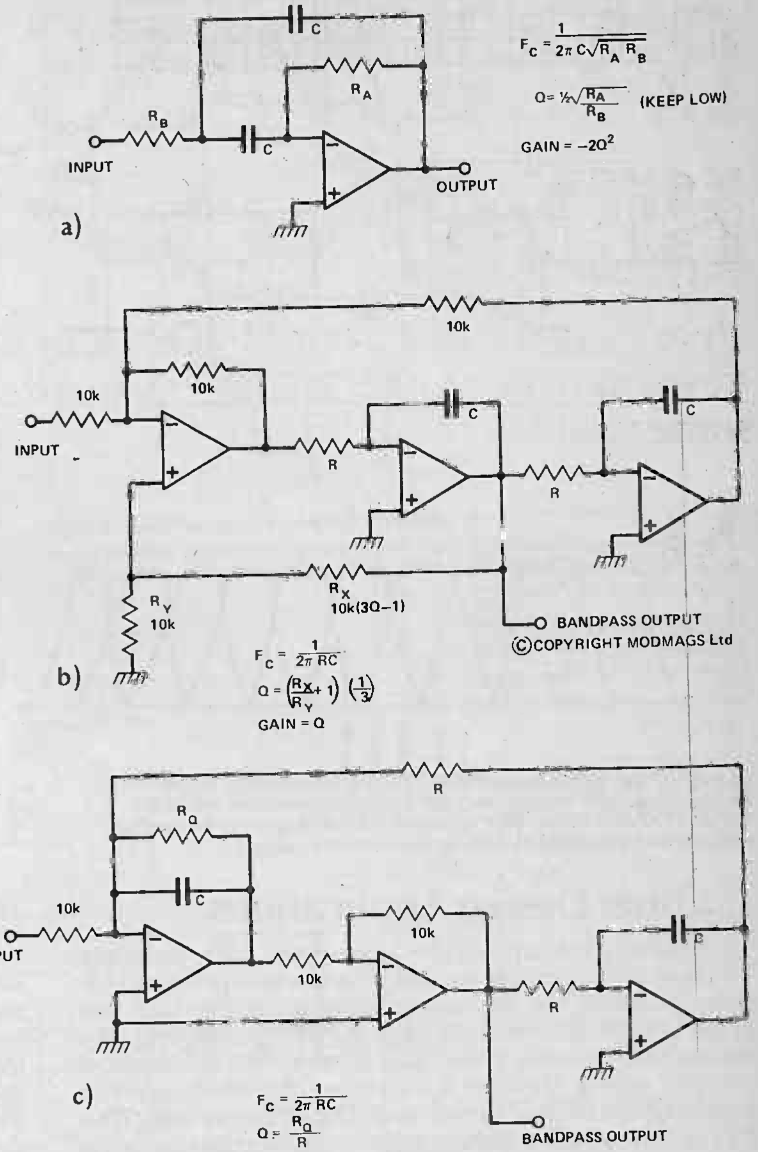


Fig.4. Standard bandpass filter designs. a) Multiple feedback. b) State variable. c) Biquadratic.

The state variable design can produce high Q factors of the order of several hundred. Tuning is performed by changing the R and/or C components. The Q factor is independently variable and is invariant with changes in frequency.

The biquadratic design is similar to the state variable filter. Tuning is performed by changing the R and/or C components and the Q factor is determined by the ratio of Rq to R. As it is tuned to operate at higher frequencies the Q factor will increase linearly in proportion to that frequency.

A voltage controlled biquadratic filter is shown in Fig.5. This employs the relatively new CA3280, which is a dual improved performance version of the CA3080. As the Q factor is a function of frequency, the useful operating range is about 20 to 1.

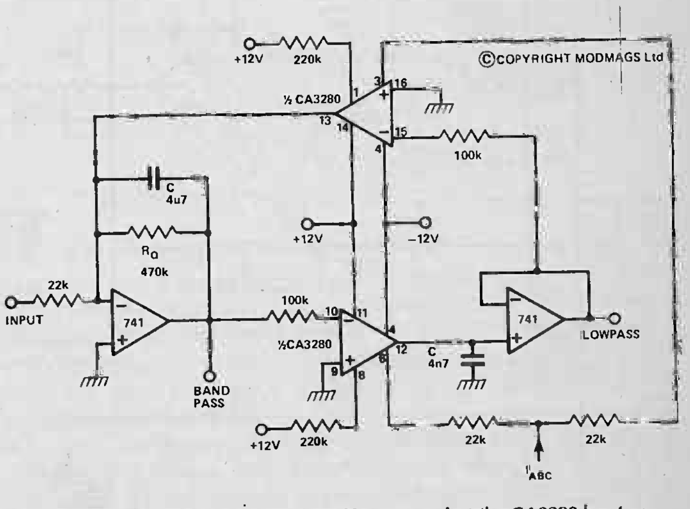


Fig.5. A voltage-controlled biquadratic filter. Note that the CA3280 has two +12 V supplies (pins 14 and 11) but only one −12 V supply (pin 4). (We don't know why either.)

A simple analysing octave filter bank is shown in Fig.6. This is implemented using double tuned filters with Q factors of five. The component values for two channels are shown in the table of Fig.6a. Note that some compromises will have to be made in order to implement the design using low cost E24 resistors. For example, a 255k resistor could be made using two 510k resistors in parallel. The filter bank is converted into a spectrum analyser by adding an envelope follower to each channel and then multiplexing the envelope voltages into an XY display (Fig.6c).