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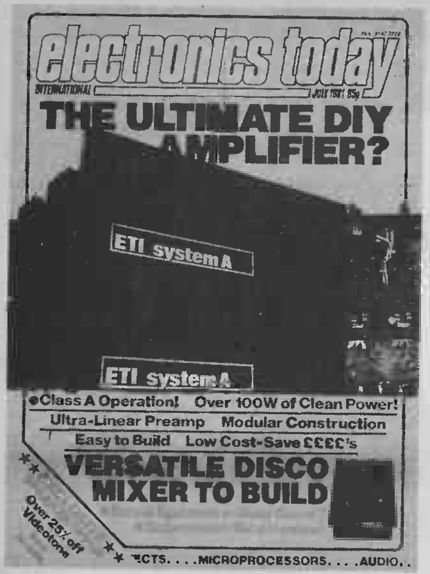

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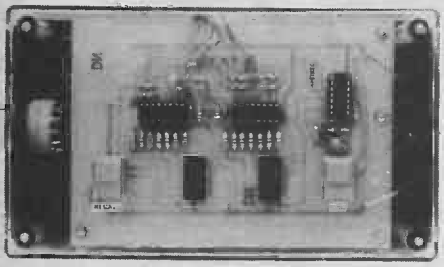

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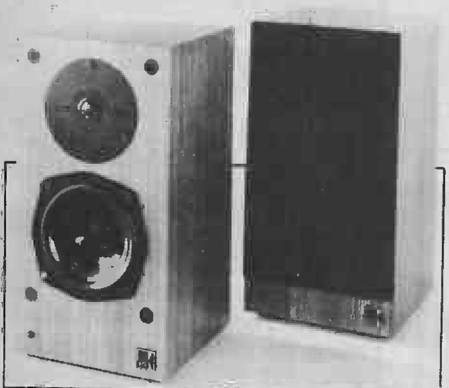

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DIGEST

Healthy Video

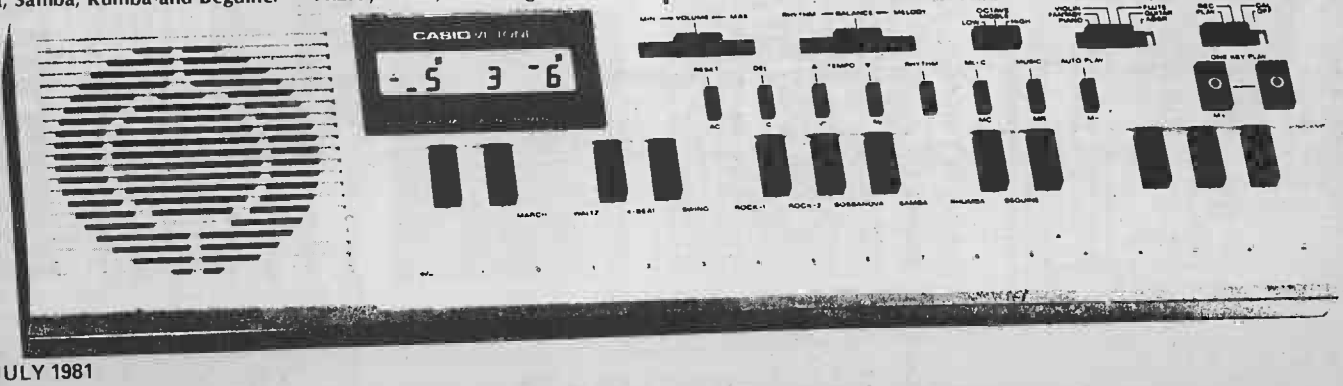
Bib have introduced the Video Recorder Maintenance Kit to their Videophile Edition range. Use of this kit will reduce undue wear on the critical tape contact points in either VHS or Betamax machines. The kit comprises five special head cleaning tools, a bottle of tape head cleaning fluid, a can of Dust-Away air blast, an inspection mirror and an anti-static cleaning cloth. There is also a crosshead screwdriver. Each item is separately replaceable and if the kit is used regularly it will extend the life of the machine and prevent the build-up of abrasive dirt particles as well as ensuring better sound and picture quality from tapes. The Bib Kit can be obtained from all leading video specialists and stores selling video equipment. The recommended retail price is £9.98, including VAT.

Rapid Cat

Not a fast feline but the new catalogue from Rapid Electronics. With a considerably extended range of products, the new edition contains most basic components at competitive prices. The catalogue is available by sending 28p in stamps, or free with orders over £5. Rapid Electronics, Hillcroft House, Station Road, Eynsford, Kent.

Musical Interlude

All you budding musicians out there, hear this! Casio's new VL-Tone can help you to create your own musical masterpieces and, with a bit of practice, you will be able to play compositions which normally take years of practice. You can choose from five preset sounds — Piano, Fantasy (?), Violin, Flute and Guitar and with these you can create over 80 million sound wave variations by using the ADSR selector (Attack, Decay, Sustain and Release). You can also choose a rhythm accompaniment from March, Waltz, 4-Beat, Swing, Rock 1, Rock 2, Bossa Nova, Samba, Rumba and Beguine. You can program in your melody and play it back again at any speed you like. To play manually you have a choice of 29 keys with a note range of 2.5 octaves and this range can be extended three times by using the octave shift switch. The VL-Tone is also programmed with a German folk song so you can practice tempo changes using the tempo control. If you get bored with playing tunes you can also use the unit as a calculator which has both square root and percentage functions. The VL-Tone can either run on mains using a suitable adaptor, or four AA-size manganese dry batteries. You can obtain the VL-Tone from Tempus at the discount price of £35.95. Their address is 'Talk of the Town', 19-21 Fitzroy Street, Cambridge CB1 1EH.



Recharging Radio

The new Battery Saver radio from Fidelity Radio Ltd is all set to change our attitudes towards battery-powered home electronic products, because it recharges *ordinary* batteries! It is powered by a standard PP9 or equivalent and circuitry is incorporated in it which automatically recharges the battery when the radio is plugged into the mains. A battery can be recharged up to four times and this can be carried out whether or not the radio is switched on. Fidelity reckon that this radio can more or less pay for itself over its average lifetime of five years. This assumption is based on each battery lasting three months and the cost of enough batteries to last for five years being about £24. Using the facilities of the Battery Saver means that replacement batteries will only cost £6 over the same period of time. The expected cost of the unit will be around £20 so, in all, this could prove to be a very cheap radio. The unit is a three-waveband battery/mains portable radio operating on Long/Medium and FM wavebands. Push-buttons are used to control on/off, and waveband selection is by rotary control with a slider for volume. A red LED indicator shows when the battery is being recharged. A telescopic aerial is built-in for improved FM reception and there are sockets for earphone and mains supply. It weighs 1.45 kg. and comes complete with mains lead.

Where Eagles Dare

At this time of the year there are many stories in the national tabloids about the stealing of eggs from the nests of rare birds like ospreys, avocets, kites and golden eagles. Now the Royal Society for the Protection of Birds will be keeping an eagle (eugh!) eye on them — particularly at night, when the nests are at their most vulnerable from the unscrupulous activities of the egg collectors. This is thanks to the donation by ITT of special night vision monoculars. The device consists of a small vacuum tube which gathers the small amount of starlight or moonlight available and converts the light image into an electron image, multiplying the electrons thousands of times and then converting the electron image into a bright display, similar to the way a television picture is formed. The electron multiplying device consists of a small sliver of glass composed of over a million tiny cores of optical fibres. As an electron enters one of the cores, it emits a secondary electron each time it bounces off the side of the core. The result is more than 10 million electrons leaving for each one that enters. This pocketscope was first developed for sufferers of retinitis pigmentosa or night blindness. As it is so useful for studying nocturnal animals, ITT have presented a number of sets to the RSPB. The device measures less than 5" and weighs only a few ounces.

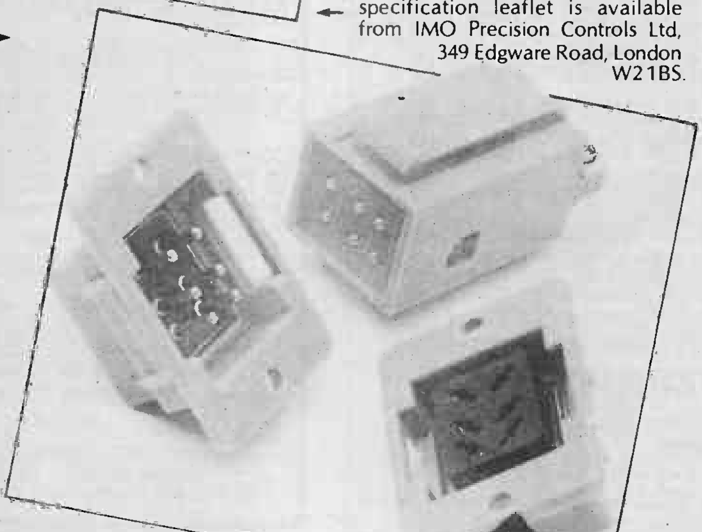
Components

Foto Fixtures

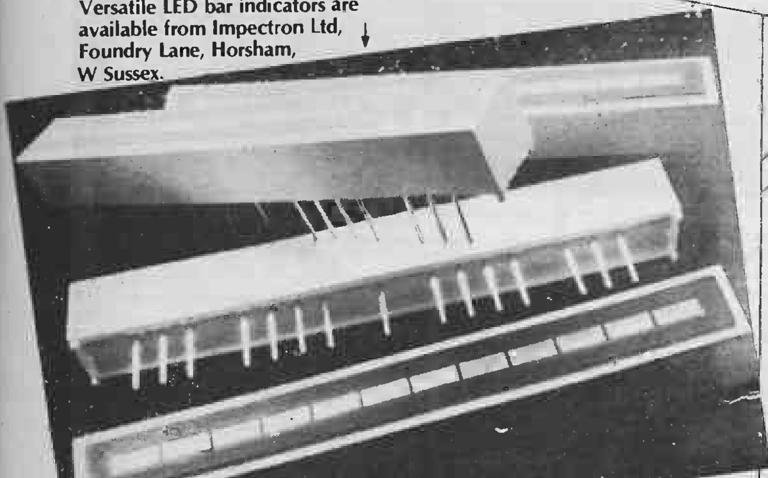
A new range of low-cost, metal enclosed photo-electric switches is now available from IMO Precision Controls Ltd. The Omron E/B photocell range is fully sealed to IP66 by a rugged metal case accepting direct conduit entry. All models operate directly from 110 or 240 V AC 50/60 Hz and include a single changeover output relay related at 3 A/240 V AC. A comprehensive specification leaflet is available from IMO Precision Controls Ltd, 349 Edgware Road, London W2 1BS.

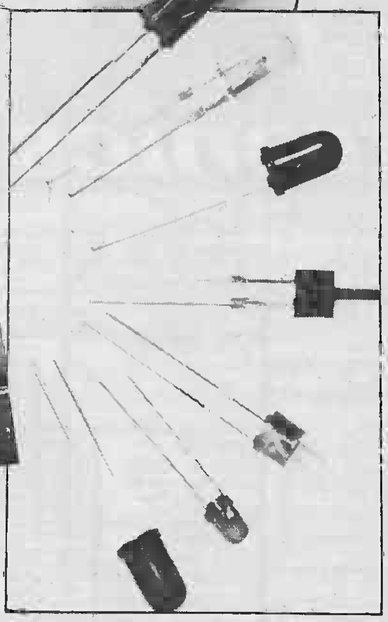
Snappy Connections

A miniature, multi-way connector with an integral snap-lock to prevent accidental disconnection is available from H & T Components. The Series 1300 connector has been designed for use in equipment where vibration or internal movement takes place. Both the plug and socket halves are hooded and the plug incorporates twin barbed locks, one on each side of its moulded plastic case. Disconnection occurs only when a light pressure is applied to the two locking barbs. The characteristics of the Series 1300 include a current rating, per circuit, of 5 A, operating voltage of 400 V AC (max), proof voltage of 1.2 kV and contact resistance of 5 mR (max). Further information from H & T Components, Crowdy's Hill Estate, Kembrey Street, Swindon, Wilts SN2 6BN.



Versatile LED bar indicators are available from Impectron Ltd, Foundry Lane, Horsham, W Sussex.





A wide range of LEDs are now available from Digital · Devices along with details and specifications of individual LEDs, from Val Long, Digital Devices Ltd, 134 London Road, Southborough, Tunbridge Wells, Kent.

CB2B

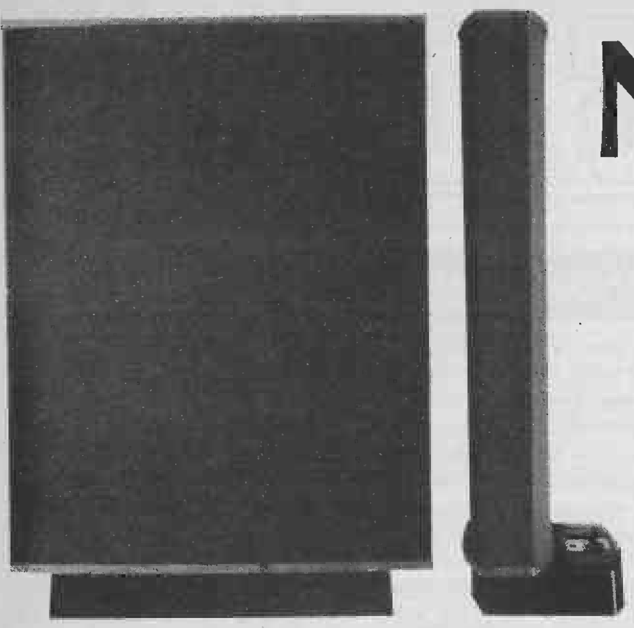
At long last a specification has been published by the Home Office for the legalisation of Citizen's Band radio. Two frequencies will be allocated: 934.025 to 934.975 MHz and 27.60125 to 27.99125 MHz. For the 934 MHz (AM) frequencies the maximum power is 8 W (25 W ERP), 20 channels at 50 kHz channel spacing. Hand-held units are restricted to 3 W PEP. On the 27 MHz (FM) frequencies the maximum power is 4 W (2 W ERP), 40 channels at 10 kHz spacing. Frequency tolerance: ±1.5 kHz. Maximum frequency deviation: ±2.5 kHz. Adjacent channel power: −60 dB to 2 uW, spurious emission less than 50 nW. This new frequency allocation has been chosen to reduce the possibility of harmonic interference to aircraft landing systems etc. Also included in the spec are certain regulations. Antennas higher than 10 m from the ground must be attenuated by 10 dB. The Home Office will also permit modifications to existing equipment providing it complies fully with the specification. For equipment which has not been approved by the Home Office or the Post Office the onus is on the manufacturer to comply with the specs. All equipment must have a small badge etched or permanently affixed to the front of the rig which should be not less than 6 mm in diameter and have the letters 'CB/27/81' not less than 1 mm high in the centre. And that about sums it up!

Talking Time

The Trafalgar Watch Company is trying to ensure that you'll never be late again! They have just announced three new watches — one talks, one sings and the other makes ordinary alarm-type noises. The Tel-Time actually speaks the time and can be programmed to do it in either English, German or French. The expected price is £59. The singing watch has nine different tunes ranging from 'Happy Birthday' and 'Jingle Bells' to 'Scotland the Brave' — all yours for about £29. The third watch is a quartz analogue model combining electronic precision with a traditional watch face and an alarm. This one sells for about £39. Further information can be obtained from the Trafalgar Watch Company Ltd, Trafalgar House, Grenville Place, Hale Lane, London NW7.

Alarming!

Here we have yet another offering from the ubiquitous Clan Casio. The PQ-20 chronograph can either fit flat in your pocket or sit on your desk or bedside table using its integral stand. It has a three-way alarm which can give you a sunrise symphony (Mozart's Symphonie Nr 40 G Moll), a bright buzz, or a combination of both! There is a snooze function which sounds every four minutes for 60 seconds after the preset alarm time. There is also an hourly alarm, or, if the mood takes you, you press a button and hear the symphony any time you like. You can obtain the PQ-20 from Tempus at the discount price of £12.95. The address is 'Talk of the Town', 19-21 Fitzroy Street, Cambridge CB1 1EH.



NEW QUAD SPEAKER

It's finally here — the long-awaited successor to the Quad Electrostatic. M. Burroughs examines the revolutionary concept behind this new arrival on the hi-fi scene.

It's widely accepted that one of the best speakers on the market for the last 25 years has been the Quad Electrostatic. Yet Quad have remained in the unusual position of having a 'range' of one, disdaining the usual practice of hi-fi manufacturers who sell speakers with varying power and quality. The advent of a successor is therefore something special, and Quad have been security-conscious and secretive prior to the launch (although your ever-vigilant ETI spy managed to scoop the world with a sneak photo last month).

What is it that's so special about electrostatic speakers? Although research on them took place long before Rice and Kellogg invented the moving coil loudspeaker, the latter design has dominated the industry, and the market, since the earliest days of hi-fi. Despite the problems inherent in moving coil designs — cabinet resonance, cone stiffness and stored energy, for example — electrostatic speakers are few and far between. Why is this?

Charge Of The Light Membrane

The basic design of an electrostatic loudspeaker involves placing a thin, light, electrically-charged membrane between two electrodes. The electrodes are 'acoustically transparent' (they've got holes in them), and when a signal is fed to the electrodes the changing electric field set up between them causes the membrane to vibrate in sympathy, generating the sound waves. There are many advantages in this arrangement. The membrane is being driven at every point on its surface, so it has no need of stiffness and can be made extremely light. This in turn reduces the stored energy and improves the high frequency characteristics. No cabinet is necessary, so resonance problems disappear. Nevertheless, the design and manufacture of electrostatic speakers is still quite tricky.

Moreover, until now even electrostatic designs have had to achieve the desired dispersion characteristic over the whole frequency range in the same way as moving coil speakers; by using several drive units of similar or varying sizes, with or without crossovers. It is in the solution of this problem that the Quad ESL-63 breaks totally new ground.

The Plane Truth

The concepts behind the ESL-63 were described by Peter Walker in his paper, 'New Developments In Electrostatic Loudspeakers', presented to the 63rd Convention of the Audio Engineering Society in Los Angeles in May 1979. The starting point for the theory is the fact that the ideal loudspeaker would be a small pulsating sphere; this is not practical as a design, especially at reasonable power levels, because impossibly high pressures are required. However, we can achieve the next best thing, as follows. The sounds emitted from the ideal source take the form of expanding spheres of pressure waves. If we take an imaginary plane some 30 cm from this source and perpendicular to the line joining the source and the observer, it can be easily seen that sounds appear at the plane as a series of concentric waves radiating from the centre and getting weaker. The analogy can be made with a stone thrown into a pond, creating expanding ripples which die away with distance.

The next step is to replace the imaginary plane with a real surface, and remove the ideal source. If the plane surface can be made to vibrate so that it reproduces this pressure pattern, an observer will hear the same sounds as he would if he was listening to the ideal source itself.

Although the complete mathematical treatment would be out of place in an article such as this (besides which, I don't understand all of it myself!), the final equation has important implications for the designer. If the outer electrodes of an electrostatic speaker are a distance 2d apart and E is the initial polarising voltage on the membrane when central, then a current I into the speaker terminals will produce a pressure at a distance r from the speaker equal to

$$\frac{E}{d} \cdot \frac{I}{r} \cdot \frac{1}{2\pi c}$$

Instead of the usual complicated equations (which generally have to be evaluated by approximation, even for simple shapes), we have a simple expression for the sound pressure at any point which involves only two electrical quantities and two distances, and is independent of frequency, shape or area. Thus the performance of such a loudspeaker depends entirely on the currents in its electrical circuits, and these are not circuit analogies of mechanical and acoustical systems but real (albeit complex) circuits, which can be tailored as required.

Of course, the final loudspeaker must be of finite area (unless you live in a TARDIS), so the discontinuity at the edge will generate interference waves and a fall in response at low frequencies (due to the missing radiation beyond the boundary). But the equation shows that such acoustical problems can be corrected 'merely' by adjusting the electrical currents. Furthermore, the speaker is only called on to reproduce the sound waves as found some 30 cm from the 'ideal source', so the generation of high pressures is not necessary.

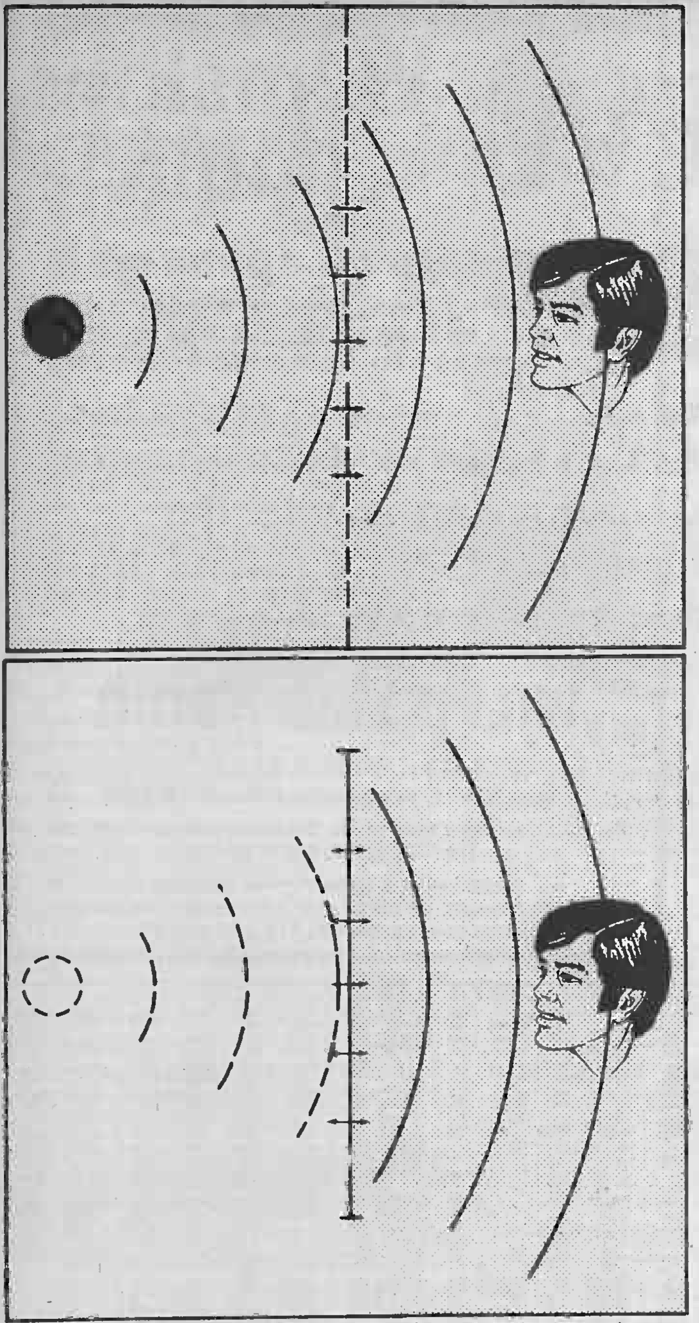


Fig. 1 The principle behind the Quad ESL-63. The air pressure pattern in a plane some 30 cm from an 'ideal' source (top) is reproduced by a membrane (bottom). To an observer the sound is the same in both cases.

Practice Made Perfect

In practice the Quad ESL-63 fulfils the theoretical design requirements by using two sets of concentric annular electrodes. The signal from the amplifier is fed to these electrodes via a series of delay lines, thus producing the 'ripple' pressure pattern. The interference waves mentioned above appear as reflections in the delay lines; modifying the delay line to remove these reflections also eliminates the interference waves. The result is totally homogeneous sound source, phase true and without any of the faults associated with standard speaker designs.

Since the ESL-63 is a dipole source (the wags at Quad have nicknamed it FRED — Full Range Electrostatic Doublet), it has a figure-of-eight sound dispersion pattern and radiates no energy in the plane of its diaphragm. By placing the speakers at an angle to the walls of the room, excitation of both horizontal axial modes is 3 dB less than with an omnidirectional source; the vertical modes are discriminated against. Reflected sound is reduced to give improved localisation of the stereo image.

Two protection circuits are fitted to the speaker; one limits the maximum input voltage to the loudspeaker and the other shorts the signal out if a fault condition is detected (don't use these speakers if your amplifier isn't short-circuit protected!).

Quad seem to have taken rather more effort over the styling of the ESL-63 than with their previous design, and you won't have to worry about the wife refusing to have them in the living room.

Well Developed

Development of this speaker has taken 18 years (that's right, ESL-63 stands for Electrostatic Loudspeaker 1963). Has it been worth the wait? As yet we've not been able to listen to a pair, but the frequency response shown in Fig. 3 is pretty impressive. Quad are certainly giving the ESL-63 the Rolls-Royce treatment; initial launch is to be through 'a small number of Quad dealers selected for their ability to demonstrate the outstanding qualities of the loudspeaker'. Customers will be allocated a pair of serial numbers and an anticipated manufacturing date. At £1,000 a pair (including VAT) the Quad ESL-63 is strictly up-market, but if you enjoy your music and have the money to spend, a trip to your nearest Quad dealer could be well worth the effort.

SPECIFICATIONS

Dimensions	Height 92.5 cm	
	Width 66 cm	
	Depth 27 cm including 15 cm base	
Weight	Nett 18.7 kg	
	Gross 23 kg	
AC Supply	240/200 V/120/100 V/50-60 Hz/5 VA	
Impedance	8 R nominal	
Sensitivity	1.5 ubar/V referred to 1 m	
	(ie 86 dB/2.83 V_{RMS})	
Maximum Input	Continuous input voltage 10 V_{RMS}	
	Programme peak for undistorted output 40 V_{PK}	
	Permitted peak input 55 V_{PK}	
Maximum Output	2 N/m^2 at 2 m on axis	
Directivity Index	125 Hz	5 dB
	500 Hz	6.4 dB
	2 kHz	7.2 dB
	8 kHz	10.6 dB
Axis Band limits	−6 dB at 35 Hz third order	
(Low level)	−6 dB > 20 kHz	

Fig. 2 (Right) Block diagram of the speaker, from the centre outwards; the black rectangles are cross-sections through the annular electrodes. The audio signal is fed directly to the innermost electrode, but passes through successive delay/attenuation circuits on its way to the outermost electrodes.

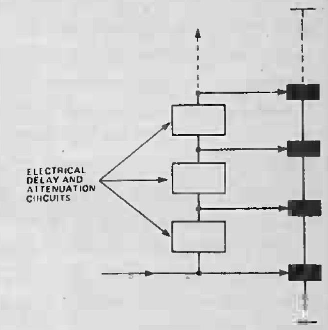
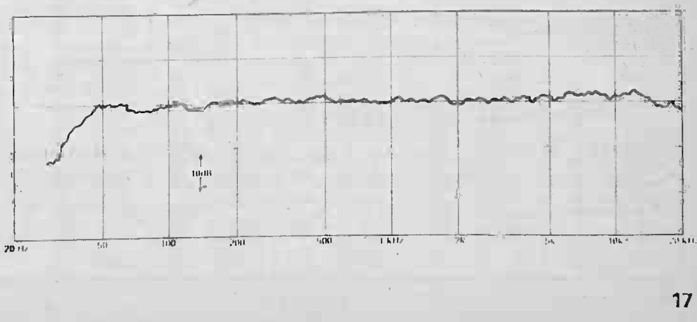


Fig. 3 (Below) The frequency response of the Quad ESL-63; this is pretty flat!



ELECTRONICS IN SUBMARINES

As a weapon, the submarine gives immense power to its controllers as it cruises silently beneath the waves. David Chivers explores its applications, shortcomings and future possibilities as the modern creature of the deep.

The first submarine invented for military purposes was the 'Turtle'. This vessel, the creation of David Bushnell (1742-1824) carried a 150 lb explosive charge for attaching to the keel of an enemy ship. The first submarine attack was made during the American Civil War when the Confederate ship David damaged the USS New Ironside and started a new age of modern warfare. Of course any submersible is of no great military use without suitable weaponry. The earliest weapons were explosive charges projecting from the submarine on a spar. These were laid against the target, and it was hoped that the submarine would be at a safe distance when the explosion occurred. Static explosives used in this way became known as mines while the first automotive torpedo was produced in 1868 by Robert Whitehead.

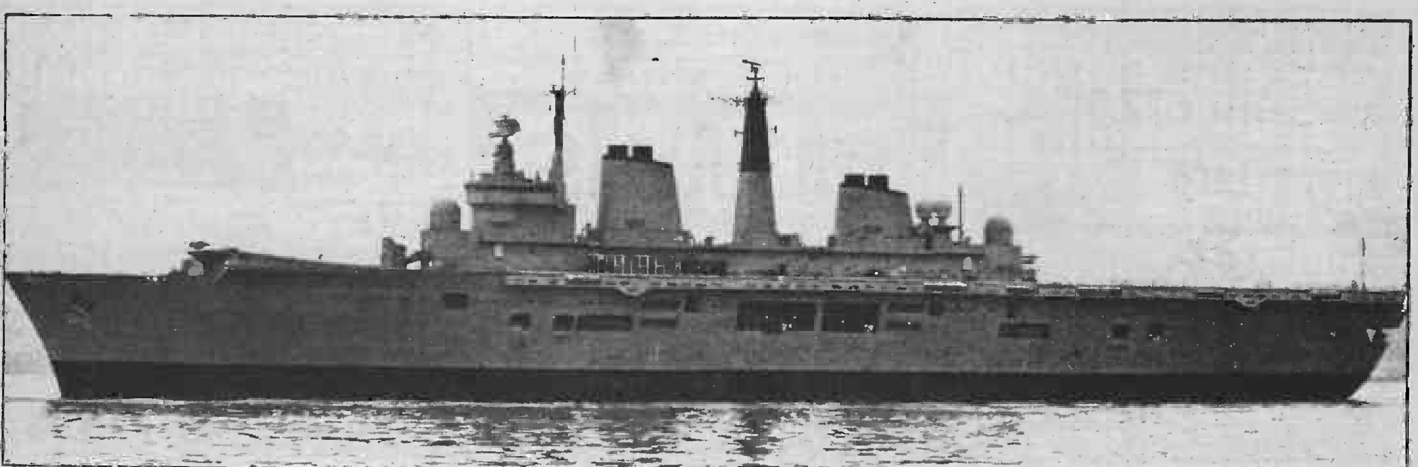
Whitehead's torpedoes were originally fired from ships and the first sinking claimed for the torpedo was in 1878 when two Russian merchantmen sank a Turkish steamer. However, in 1885 torpedo tubes were fitted into a submarine and though remaining a useful weapon in smaller surface vessels, it is in the hands of the sub-mariner that the torpedo comes into its own. By 1914 the British 21" torpedo — still the standard gauge — could cover 6,000 m at 40 knots.

The First World War

It was not the British, but the German torpedo which was to have the most drastic effect in the First World War. Being dependent on her sea lanes for raw materials Britain was — and still is — vulnerable to siege by submarine. Here the submarine was being used not simply as a naval weapon, but as a strategic weapon. The first torpedo attack from a submarine was made by U-15 which fired a torpedo at HMS Monarch. The torpedo either failed to reach its target or failed to detonate. After the outcry in the US over the sinking of the Luisitania on 7th May 1915, with the casualties including 143 US nationals, the Germans called an end to unrestricted submarine warfare. However, in 1917 the German High Command calculated that if submarines were once again allowed to attack all vessels in the North Atlantic, Britain would be forced to surrender within five months. Although the resumption of unrestricted submarine warfare brought the US into the war, losses ran at 800,000 tons per month and Britain came close to collapse — Admiral Jellico gave Britain until July 1917 before food ran out. On 10th May, at the insistence of Lloyd George, ships were being sent in protected convoys and the results — both in reduction of losses, and gains in submarines destroyed — were spectacular. Although eight million tons of shipping had been lost by the end of that year, Britain survived.

Sound Developments

Another development emerged from the experience of the First World War — Asdic, developed by an Allied scientific committee from which it took its name. The principle was simple — a short pulse of high frequency sound is radiated from a sonic transducer. Any large dense object close enough to be detected reflects the pulse which is picked up by an array of hydrophones. The time taken for the pulse to be reflected is an indication of the distance of the object. The development of this simple idea became known later by its American name —



HMS Invincible, an ASW cruiser. The ramp near the bows is for launching Sea Harrier jump jets.

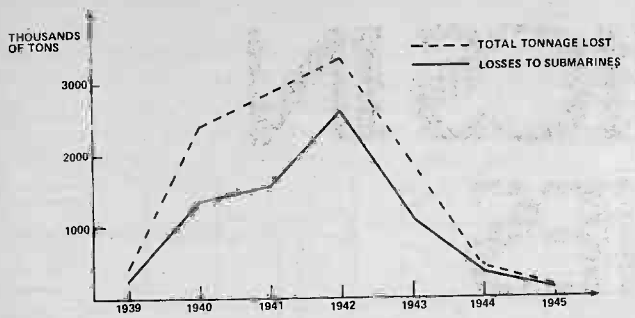


Fig. 1 This graph shows the yearly total losses of British Merchant shipping from 1939 to 1945.

sonar, or sound navigation and ranging, and by the Second World War it was a fast advancing science, though accurate ranging and positioning of a detected object was still not easy.

Second Attempt

The lessons so bitterly learned in the First World War had to be relearned in the Second, as Hitler's U-boats tried to cut Britain's Atlantic lifeline. The ensuing struggle was a long one; known as the Battle of the Atlantic it was just as ferocious and decisive to the war as the Battle of Britain or the D-Day invasion. The total losses of merchant shipping from 1939-45 ran to over 11 million tons; 2,627 ships. Of these losses, over 7½ million tons were lost to submarines, and another 800,000 to mines. At first U-boat victories escalated until there was a real possibility that Britain would indeed be starved into surrendering. The Royal Navy learned fast and with improvements in both weaponry and tactics the tide was slowly turned. The refinements of the convoy system were essential to the defeat of the U-boat menace. It was discovered that no matter how many ships there were in a convoy, the number lost to submarines would remain more or less constant. Thus large, well protected convoys were most efficient. In addition, instead of allowing anti-submarine vessels to chase lone sightings far from the convoy, by staying close to the merchant ships the otherwise elusive submarines would be drawn into the zone of detection and could be destroyed.

U-boats Undermined

New technology and new weapons aided the submarine hunters. Once radar was standard on the escorting ships, submarines were forced to abandon their favoured night-time surface attack. Weapons such as the 'hedgehog', a multi-barrelled mortar, could fire bombs over 200 m from the ship, thus obviating the need for a direct overhead run. However, the losses continued to mount, and it was not until effective aircraft cover could be provided over most of the convoy route that U-boat losses began to rise dramatically.

This use of aircraft to hunt and destroy submarines was a great step forward. Carrying high frequency direction finding equipment to locate the source of enemy radio transmissions, radar to locate surfaced vessels and a powerful searchlight for night-hunting, aircraft proved to be effective hunters. When a switch to centimetric radar was made, even the warning receivers carried by U-boats to alert them to 150 cm transmissions became useless. The Battle of the Atlantic was finally won, but at an enormous cost: the lives of over 30,000 men of the Merchant Marine.

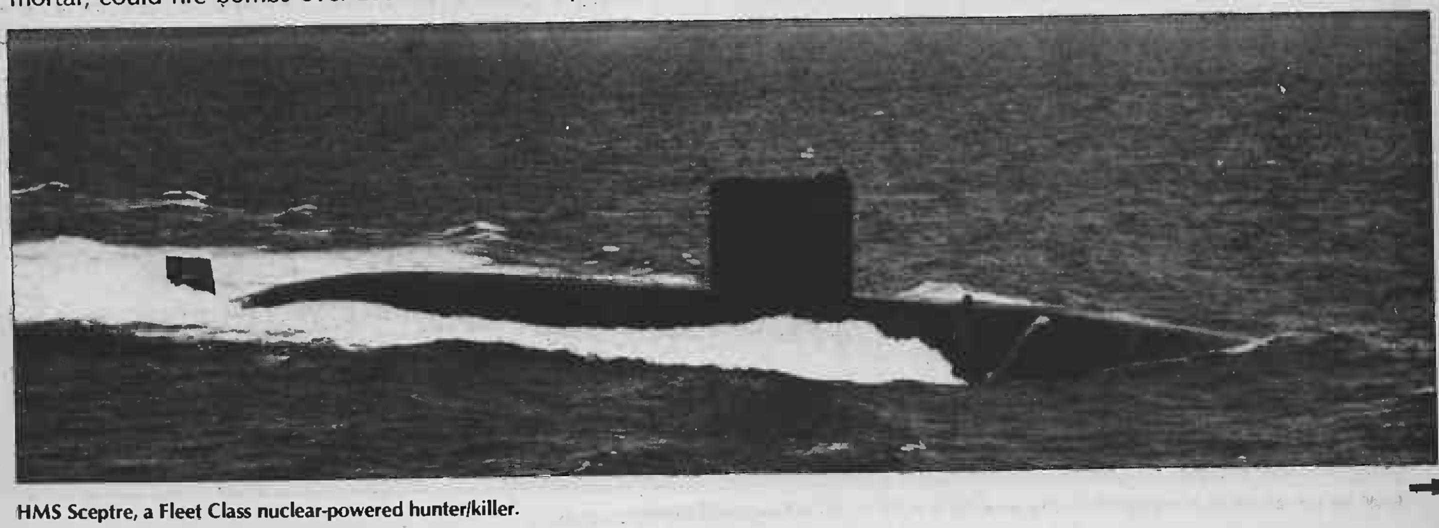
The Nuclear Age

In 1952 the first nuclear-powered submarine was launched. The advent of nuclear power plant and air recycling equipment changed the face of submarining so that the ability to remain submerged could depend only on the endurance of the crew. Today's submarines are larger and faster than their predecessors; they attain higher speeds below water than above it and at 30 knots can outrun many surface vessels. Improvements in streamlining make modern craft quieter and more difficult to detect. The submarine had developed from a ship capable of temporary underwater operation to a true amphibian, designed to operate underwater where it is at its best. As a weapon the submarine may be divided into two classes — the strategic missile submarine, and the patrol and hunter/killer submarine.

The SSBN

Nuclear ballistic missile submarines — SSBNs — are the monarchs of the ocean. Fitted with nuclear power they have no need to refuel for many years. Their ballistic missiles enable them to strike at an enemy many thousands of miles distant. Britain uses four such submarines to house the Polaris missile, the current strategic nuclear deterrent. The Government has chosen the Trident system to replace Polaris. Trident is a three-stage solid fuel rocket with a range of 7,000 km, with a high accuracy due to stellar guidance in the final phases before re-entry into the atmosphere. On re-entry, up to eight independently targetable warheads may be deployed, each with a yield of 100 kilotons.

This formidable weapon would be quite useless as a deterrent if the submarines in which it was housed could be located and destroyed as part of a pre-emptive strike, but both East and West rely on the ability of the SSBN to escape detection and



HMS Sceptre, a Fleet Class nuclear-powered hunter/killer.

have large fleets of these submarines. The latest American vessel is the Ohio class submarine. One of these houses 24 Trident missiles, and its 16,500 ton displacement dwarfs the 7,000 ton Polaris submarine. The Soviet Union is also putting its faith in large submarines; their 'Typhoon' class now under development is likely to displace some 18,000 tons, and will possess a deep diving capability far in excess of today's vessels. The Typhoon may well deploy the new SS-N-17 missile, another solid-fuelled weapon with inertial guidance and post-boost phase stellar sighting.

Tactical Submarines

The SSBN is at the very apex of submarine development, but both NATO and the Warsaw Pact envisage a period of conventional war before a possible nuclear exchange. If NATO were to fight a conventional war in Europe, then Britain's survival would depend more than ever on her sea lanes. At present, stocks of ammunition, oil and spare parts to last for 30 days of conventional war are kept in Europe. However, to keep NATO forces supplied beyond this time and to bring across the Atlantic the heavy equipment of the US reserve forces, vast quantities of materials will have to cross the Atlantic by sea. This massive task involving the movement of some 12 million tons of stores will involve the whole of the US Merchant Marine as well as 500 European merchant vessels. Ship movements would be at the rate of 1,000 per month across the Atlantic; moving the oil alone would require 13 million tons' worth of tankers.

This vast traffic across the Atlantic would be easy prey for submarine packs, and it is a problem taken very seriously by NATO commanders and planners. Nearly all NATO warships have an anti-submarine warfare (ASW) capability, and the enemy they will face will be far more difficult to detect and destroy than the old U-boats.

Anti-Submarine Warfare

The surface ship is the traditional craft from which to hunt submarines, and its primary means is sonar. Modern sonar techniques have advanced considerably over the old Asdic 'ping' signal. Asdic sonar is 'active'; it emits a noise to be reflected and received. Passive sonar merely listens to the noise from the quarry itself. A vessel passing through the water produces mechanical noise — for example, from propulsion systems, engine vibrations and so on — and cavitational noise, the sound produced by small air bubbles against the moving hull.

Both types of sonar have their advantages but the passive sonar is generally the most useful. Passive sonar has a greater range than active (the sound need only travel half the distance) and the emissions of active sonar are likely to be picked up by the target vessel long before contact is made by the hunter. However, passive sonar is of little use against a stationary shut-down vessel, and suffers from difficulties of ranging the target, though to combat this an 'array' of passive sonars can be used to provide cross-referencing and triangulation. Sonar may also give the velocity of a moving object using Doppler shift analysis.

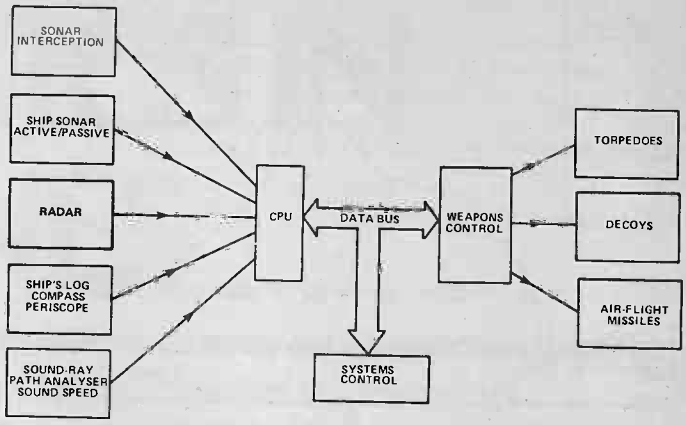


Fig. 2 Block diagram of the fire control system for a modern submarine.

The Submarine In ASW

A submarine is the ideal choice for ASW. Both active and passive sonars work with far greater efficiency and the submarine can position itself after taking account of gradations in temperature, sound-ray analysis, the nature of the sea bed and so on. Some submarines have large arrays of passive hydrophones in a projection on the bows. The new US Ohio class uses a towed array.

The standard method of attack is with a conventional torpedo, though the modern torpedo is anything but ordinary. The new Mk 24 torpedo in service with the Royal Navy has an electric motor and seawater batteries, giving a range of some 20 km at 50 knots. The torpedo is wire-guided, and an on-board computer relays information back to the parent ship. Once close enough it activates its own active or passive sonar guidance whose commands can be overridden if necessary. The US Mk 48 torpedo is similarly advanced and can make repeated attacks on a target if it misses on the first run, or was deflected by a decoy.

HMS Ajax is a Leander Class frigate with the Ikara ASW system and Seacat guided missiles.

Another development is to give the torpedo a mid-range air flight. Fired from a standard torpedo tube the SUBROC missile leaves the water and sustains rocket-powered flight to a position above the target, where the nuclear warhead separates, sinks and detonates over the submerged submarine.

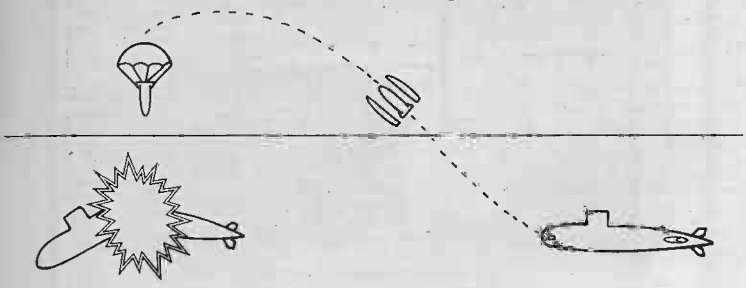


Fig. 3 Submarine attack by SUBROC.

Sting Ray

In 1979, Marconi and the Ministry of Defence signed a contract — worth more than £200 million — to complete the development of the Sting Ray lightweight anti-submarine torpedo and to supply an initial quantity of the weapon. Sting Ray is capable of defeating the submarine threat until well into the 1990s.

It is the brain of Sting Ray, ie the programmable digital computer, which sets it apart from any other guided weapon. This allows Sting Ray, using an electronically steered multi-beam sonar, to search for the target, home in on it and remain on target despite enemy counter-measures or evasive manoeuvres.

Sting Ray is the first underwater weapon to use a programmable digital computer to control its behaviour. It means that the user will be able to react quickly to changes in the threat by updating the software — ie the sets of mathematical equations within the torpedo's brain. It is this development that enables Sting Ray to counter the threat well into the 1990s. Current lightweight torpedoes are fast becoming obsolete in the face of the speeds and depths of Soviet submarines which are being built at the rate of one a month, twice NATO's rate.

Aircraft

A major disadvantage of both surface vessels and submarines in the anti-submarine role is their slow speed. This restricts their ranges of operation so for wide area surveillance the fixed wing aircraft is most useful. Probably the most powerful submarine hunter in the world is the Royal Air Force Nimrod Mk 2. This has a crew of 12 and a vast array of electronic equipment. It can cruise at high speed on four engines, then cut down to two for patrols of up to 12 hours. The Nimrod's major sensors are the powerful EMI Searchwater radar (which can detect a submarine's exposed periscope) and sonobuoys, small autonomous sonar sets dropped into the ocean which transmit data by radio link.

By dropping a pattern of active or passive sonobuoys the aircraft can detect and locate submarines over a wide area. Sonobuoys retain a surface antenna, while dropping a hydrophone array to cover from 18-140 m. The standard British sonobuoy, Jezebel, can operate from one to eight hours depending on the power required and detect frequencies from 10 Hz to 3 kHz. Transmissions are on the NATO standard of 140-176 MHz at 1 W RF output. Another device which can be dropped is the bathythermal buoy, giving a reading of sea temperatures to a depth of 400 m.

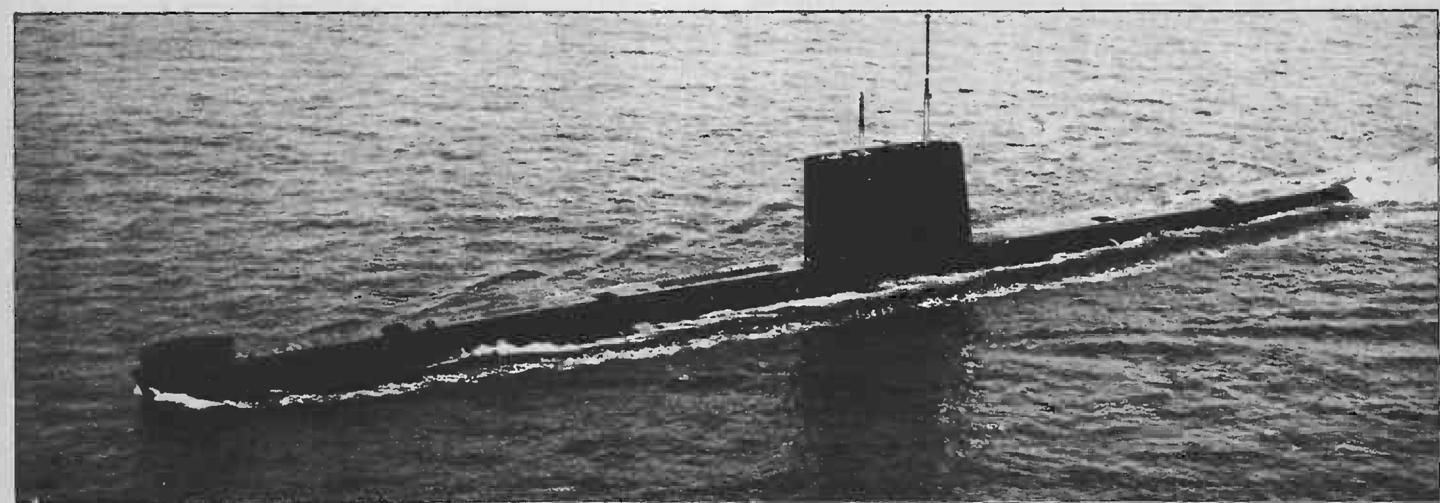
Fixed Surveillance

The final method of submarine detection is the use of fixed sensors. The United States has expended much effort in this direction and is still upgrading the systems. The theory is that by placing a line of fixed sensors across a route to be taken by hostile submarines — such as the Faroe Islands gap — all such movements can be monitored. The United States has a system guarding the Atlantic Seaboard of North America known as SOSUS, or Sound Surveillance System. This is an underwater array of passive hydrophones with data links to a central control.

Summing Up

Although the art of underwater warfare grows ever more sophisticated, the lessons of the past must not be forgotten. Twice in the last hundred years Britain has been on the point of collapse through naval blockades carried out by submarine. On both these occasions the siege was broken by the tactical use of the weaponry available and the introduction of new weapons and detection systems. In any future war Britain will not have the time to learn from mistakes — the Atlantic route will be critical from the onset and a war is likely to last weeks rather than years. In this situation it is essential that successful anti-submarine techniques are advanced to cope with the latest threats.

Our thanks to Marconi for permission to reprint the information on Sting Ray, and to the Ministry of Defence for the photographs.



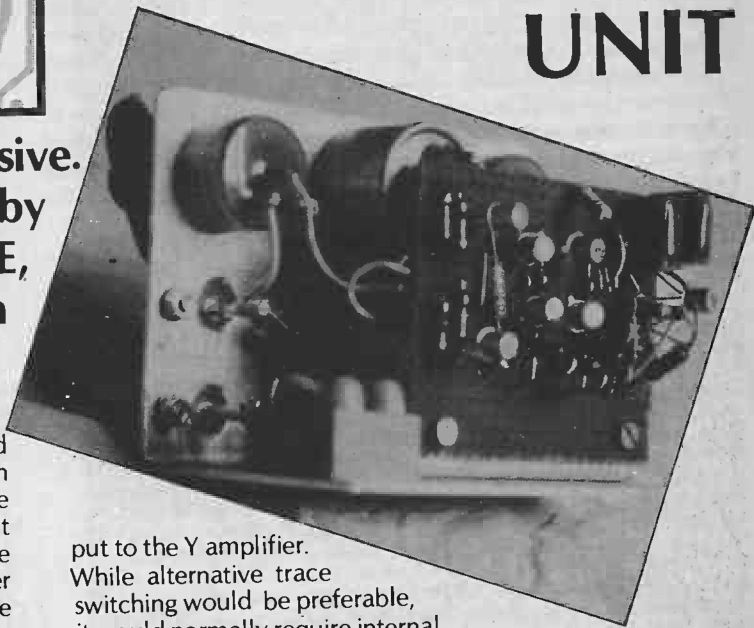
HMS Otter, a diesel electric patrol submarine, showing the hydrophone array on the bow.

ETI

READER'S DESIGNS

SCOPE BEAM SWITCHING UNIT

Oscilloscopes are useful but expensive. This simple accessory, submitted by P. Cairns, RTechEng, MIPRE, MIE, will increase the versatility of even the most basic scope.



This simple but effective beam switching unit may be used in conjunction with the majority of oscilloscopes in general use. Such a device considerably extends the usefulness of an oscilloscope in all audio and LF work as it allows the simultaneous display of two signals on a single trace instrument. For example, both the input and output of a filter network or amplifier may be displayed, allowing immediate assessment of phase difference and comparative visual indication of phase shift when carrying out circuit alterations. In stereo work the simultaneous display of both channel outputs or internal circuits can prove to be of value when assessing balance, distortion, effect of tone networks etc of the system. In many instances, therefore, the use of a beam switching unit may be considered to double the potentiality of the average oscilloscope.

When a double beam oscilloscope is already available, the use of two beam switching units allows a four channel display on one instrument. This can prove of great value in the analysis of complete stereo systems and in biological work.

Chop And Change

Signal chopping involves sampling two (or more) signals and displaying them simultaneously, but separately, on a single beam deflection system. This is achieved by having two identical circuits which are alternately switched on and off at a faster rate than the frequency of the two input signals applied to these circuits. The standard method is to use a square wave with fast rise times, a small part of each signal being alternately displayed on the upper and lower edges of the square wave. If the switching square wave frequency is high compared to the signal frequencies, two clean and separate signals are displayed. The analogy may be made to an AM carrier transmission, the square wave switching being the carrier wave and the two signals applied being the upper and lower sidebands, except in this case these 'sidebands' are independent of one another in frequency, amplitude and modulation depth.

The principal advantage of signal chopping is that it allows the complete switching function to be performed external to the oscilloscope, no internal connections or modifications being required. The output of the unit simply plugs into the Y input. Thus, the oscilloscope can always be restored to its normal single trace function at any time by simply disconnecting the input to the Y amplifier.

While alternative trace switching would be preferable, it would normally require internal modifications and some re-design of the oscilloscope and could prove fatal to the oscilloscope, if not the operator.

The principal disadvantage is that unless a very fast switching rate is used, and thus more complex circuitry, the dual trace presentation is normally limited to signals in the LF and audio range. While higher frequencies can be displayed, the dotting effect due to the switching action becomes more apparent as the signal frequency is increased. With this limitation in mind, the use of a dual trace presentation can prove of considerable use in LF and audio work. The system described allows for the display of signals of different frequencies and waveforms on either channel, a shift control which allows the two traces to be superimposed or interchanged with one another, independent amplitude controls on both channels and DC coupling throughout. This latter function can be of importance in the biological field.

Switch To Transistors

The basic method of operation of this type of beam switching unit is illustrated in Fig. 1. The principle of the circuit is shown in Fig. 2. Switches SW1, 2 are operated together, one switch being open when the other is closed and vice versa. With SW1 closed the signal applied to the Signal 2 input is allowed to pass to the output terminal, the degree of signal attenuation being dependent upon the value of the resistor network components:

$$V_{OUT} \sim V_{SIG2} \times \frac{R2}{R2 + R3 + R4}$$

The input resistance of the oscilloscope is effectively in parallel with R2 and can be taken into consideration if necessary. In most practical circuits the CRO input resistance is so high compared with R2 that it can be ignored.

By using transistor switches in place of SW1 and SW2 and ensuring that they are driven by a fast symmetrical square wave between cut-off and fully bottomed conditions, the two signals applied are effectively 'shared' between the common output terminals.

The essential features of a practical circuit are fast switching, clean waveforms with good rise times, reasonably high input impedance and low output impedance, independent amplitude controls on both signal channels, a shift function allowing signals to be superimposed or interchanged, negligible interaction or interference between channels and a good transient response. Another useful function, often incorporated in commercial units, is a method of allowing the oscilloscope to be externally synchronised or triggered from either channel signal by means of a switched Sync/Trig output. This is particularly useful when signals of differing frequency are under consideration.

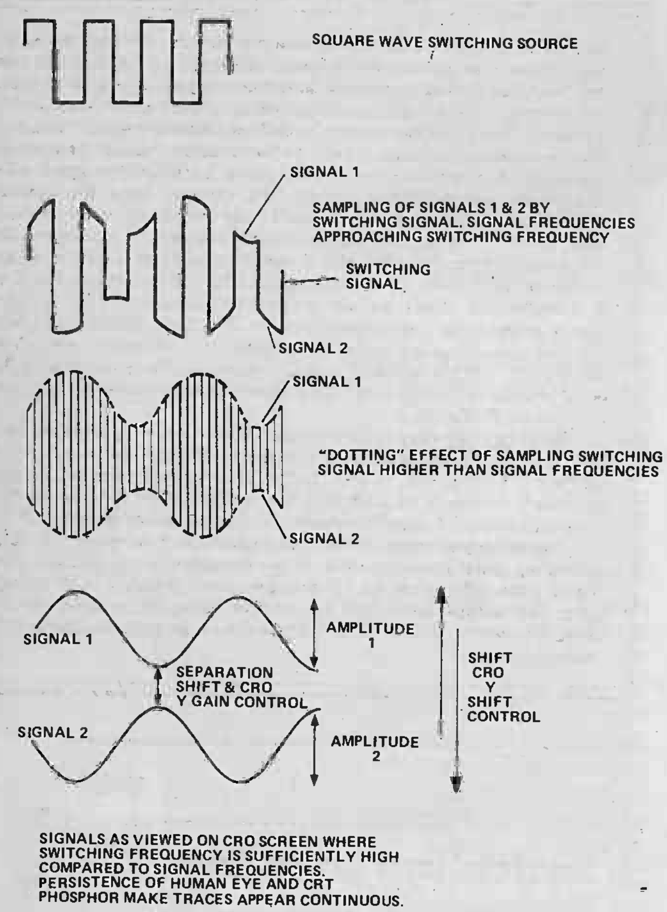


Fig. 1 Waveforms showing the operation of the beam switching unit.

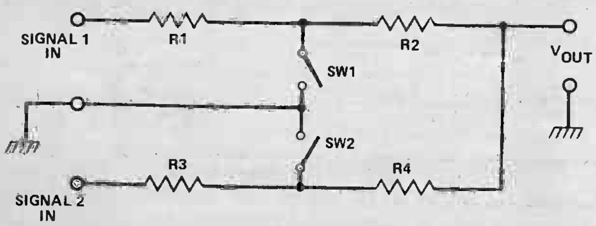
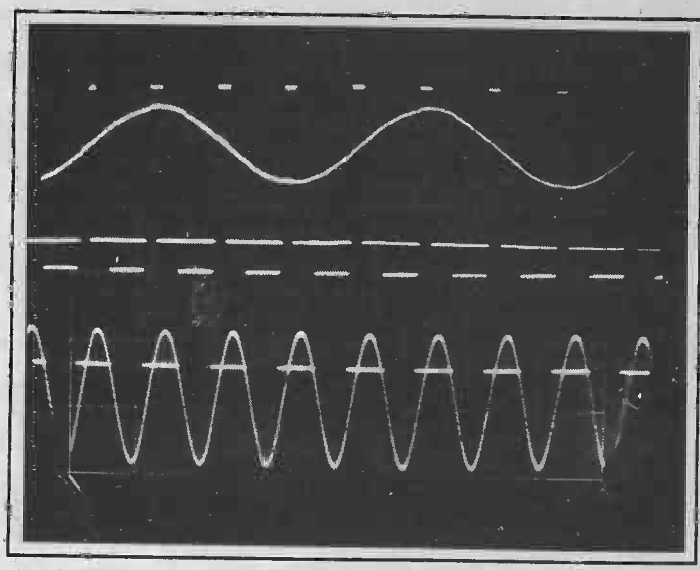


Fig. 2 This simplified circuit shows the principle of operation.



This photograph is of a multi-channel display using two beam switching units feeding into a double beam oscilloscope. The scope is set to a Y amplifier sensitivity of 100 mV/cm, and a sweep rate of 5 ms/cm. From top to bottom the traces are: Channel 1 — 50 Hz sine wave; Channel 2 — 1 ms pulses derived from Channel 3; Channel 3 — 200 Hz square wave; Channel 4 — 200 Hz sine wave.

PARTS LIST

Resistors (all ½W, 5%)

R1	5R6
R2,5,6,7,16	1k0
R3,4	47k
R8,9	220k
R10,11	10k
R12,13	22k
R14,17	100k
R15	680k

Potentiometers

RV1	5k0 linear wirewound
RV2,3	100k linear wirewound

Capacitors

C1,2	470p mica or ceramic
C3,4	100n polyester

Semiconductors

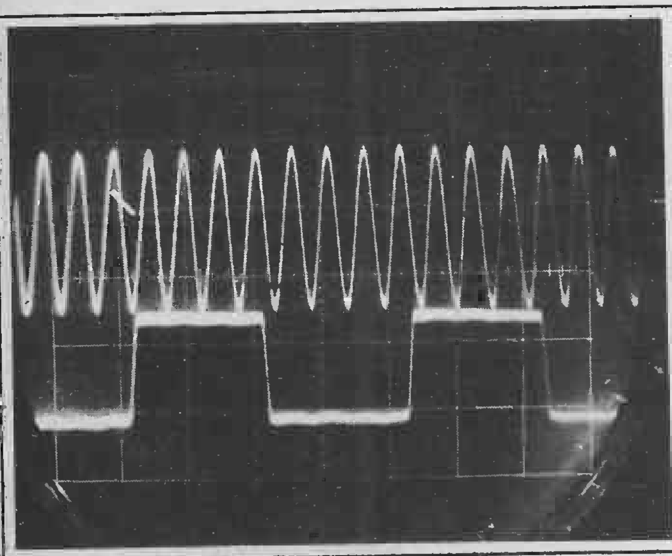
Q1,2,5,6,7	BCY70
Q3,4	BC107
ZD1	BZY88 5V6
ZD2	BZY88 3V3

Miscellaneous

SPDT slide switch, coaxial sockets (4 off), case to suit

SPECIFICATION

Measured input impedance at 1 kHz	30k
Measured output impedance at 1 kHz	7k5
Maximum output (peak-to-peak)	2 V
Sync amp gain at 1 kHz	3 dB (approx)
Shift control allows 100% overlap without interaction	
Channel frequency response	
200 kHz	−3 dB
350 kHz	−6 dB
Supply 9 V DC ±25%	
Drain at 9 V	28 mA
Switching frequency	25 kHz (approx)



Here the output from a single beam switching unit is being displayed on a single beam oscilloscope. The Y amplifier sensitivity is 50 mV/cm and the sweep rate is 5 ms/cm. Channel 1 (at the top) is a 400 Hz sine wave; Channel 2 is a 50 Hz square wave. No trace of 'dotting' is evident in either picture.

Final Design

As may be seen from the specification, all the above features are catered for in the circuit shown in Fig. 3. The device is intended as a beam switching unit and not an oscilloscope preamplifier. In practice, the unit can be applied to Y amplifiers with sensitivities in the 50-250 mV/cm range, though it worked quite well outside this range, depending on the signal drive and channel amplitude setting. An average oscilloscope working sensitivity was found to be in the 100 mV/cm range.

The construction of the unit is quite simple and neither the layout nor the DC supply are at all critical. In practice the unit should offer no problems. If a calibration signal is available from the oscilloscope the two channel inputs may be connected to this source. The unit output is connected to the CRO Y input. The Y amplifier sensitivity can be switched to the 100 mV/cm range (or nearest sensitivity) and RV2, 3 adjusted to give signals of reasonable amplitude. It should be possible to get approximately 2 V peak-to-peak with RV2, 3 at maximum setting before distortion sets in (the Y amplifier sensitivity will have to be adjusted to suit, eg 0.5 V/cm).

The shift control, RV1, should allow the traces to be displayed one above the other or superimposed. There should be no discernible interaction between channels. The CRO Y shift allows both traces to be simultaneously moved up or down.

By connecting the Sync/Trig output to the CRO external Sync/Trig input terminal (some oscilloscopes do not have this feature) and switching to external Sync/Trig with different frequency inputs to both channels it should be possible to select synchronisation to either channel by means of SW1.

The transient response is excellent, there being no discernible overshoot or droop even when displaying square waves and pulses having fast rise times (in the region of 200 ns).

HOW IT WORKS

Supply is from a 9 V battery or external power unit, the necessary voltage levels being obtained by zener diodes ZD1,2. Q1,2 form a conventional multivibrator circuit, antiphase outputs being developed across collector load resistors R2,5. These outputs are DC-coupled to the bases of Q3,4, which are simply emitter follower stages. These provide low impedance output drive to the switching circuit, sharpen up the multivibrator waveform and also serve to isolate the signal switches from the switching source. The outputs from the emitter followers are DC-coupled to the switching circuits Q5, 6 (corresponding to SW1,2 in Fig.2). These circuits are referred to a different DC supply level than the rest of the circuit, the common earth being approximately 3V3 negative to the positive line. This ensures that they are switched hard between the two extreme on/off conditions. Q5,6 are connected in the inverted mode. The out-of-phase switching waveforms are applied to the base inputs, the transistors being alternately switched on and off. The small transient spikes generated at the emitters due to the rapid switching action have no noticeable effect on the final CRT display.

The signal inputs are fed via amplitude control potentiometers RV2,3, R12,13 to the emitters. The common output is taken from each emitter in turn via R10,11. Shift control is achieved by the application of a small change in DC voltage to the emitter of Q5. RV1, connected across the main DC supply, provides this bias via isolation resistor R14.

The two signal inputs are also taken to the Sync amplifier Q7, R8,9 providing isolation while SW1 allows the selection of the Sync/Trig signal from either channel. Q7 provides approximately 3 dB voltage gain, the output developed across R16 being AC-coupled to the Sync/Trig output via C4. R17 provides a discharge path and defines the Sync/Trig output.

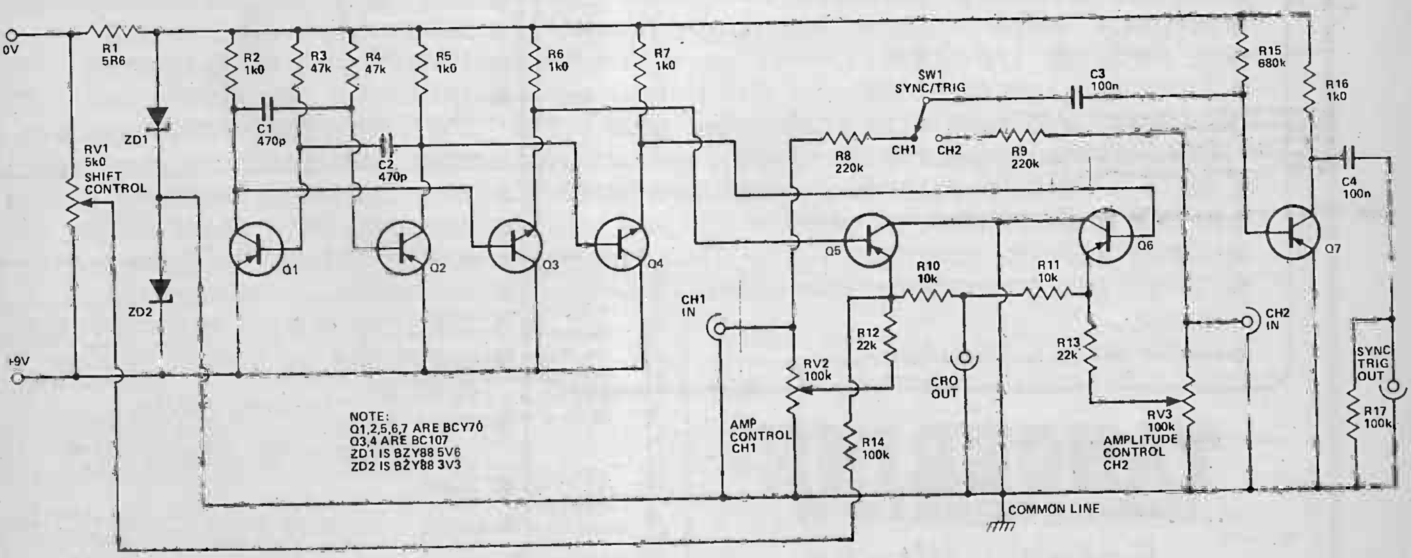


Fig. 3 Diagram showing the final circuit.

ETI

SOLAR ENERGY

One of the many proposed solutions to the energy crisis has been the direct production of electricity by solar cells. In this article Dr. I. Berkovitch takes a look at the progress in this field.

Direct generation of electricity by solar cells is attractive because they create no material or noise pollution; have no moving parts needing maintenance; can be made in modular form, delivered to site; and do not consume fossil fuels. Then why are they not more widely used? The obvious answer is based on their cost. In remote places on earth, or in satellites, the consideration of first cost is outweighed by other issues. But for 'normal' use they will have to become much cheaper.

Opening the solar electric session at a recent conference[1] T. J. Coutts and R. Hill quoted current estimates of the US Department of Energy as being $7000/kW of generating capacity for solar cells against about $1000/kW for fossil fuel fired generating stations. It is thought that solar cell systems will become competitive at around $2000/kW in about six years because of inflation in the costs of fossil fuel sources. Official estimates are based on the assumption that this reduction in cost will be achieved and that the solar cell industry will be producing 2 GW per year by the year 2000. Eventually they are expected to contribute up to 30% of US electricity and to be supplying a world market of several tens of gigawatts.

An industry on this scale will obviously need large supplies. Table 1 therefore summarises the principal solar cells under development and gives estimates of the mass of rare element needed in each case per gigawatt per micrometre thickness assuming 10% conversion efficiency. After considering these figures against estimates of world reserves, Coutts and Hill have concluded that thin film devices are likely to be essential. Of these, only the amorphous silicon cell and the thin $CdS\text{-}Cu_2S$ or $CdZnS\text{-}Cu_2S$ cells will be free of serious problems of material supply.

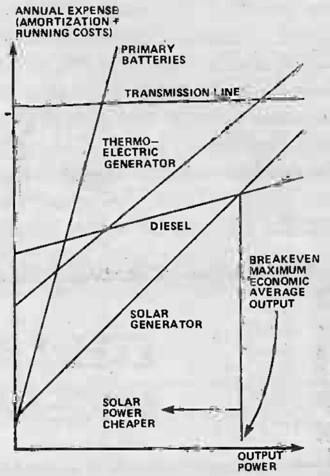


Fig. 1 The comparative costs of providing electrical power at remote locations. Below a certain output power, solar energy is cheaper than any of the alternatives.

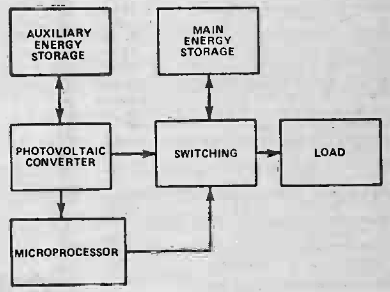


Fig. 2 Block diagram of the microprocessor converter optimiser proposed by El-Hosseiny Taha El-Shirbeeny.

Autonomous Systems Or Connected To Grid?

Where primary solar electric generation is already used in autonomous units, it needs built-in storage for times when sunshine is poor or absent. In areas where there is a grid available, the solar generator can be connected to the grid — eliminating the need for storage since the grid can be drawn on to meet excess load and can accept surplus output. These may be called contributory systems; they have their own problems. P. R. Wolfe estimated that in the Middle East a community of 50 houses could run basic internal and street lighting, a refrigerator, educational TV and a shallow well pump from a 3 kW (peak) solar array. Built as a decentralised system to reduce transmission losses, this system could use existing roof areas. It needs from 13 to 20 sq m per kW and would best be used with DC appliances. Storage adds to the costs and whether this system or connection to a grid is adopted depends on the comparative economics of installation and operation. Current comparisons are shown in Fig. 1.

When a grid is available, and the system connected to it, Wolfe still recommends localised rooftop systems rather than 'unimaginative' major stations covering acres. In the UK an average 50 sq m of area on the roof is reckoned to be able to generate an average of 10 kWh/day, and new buildings could be designed to take solar arrays on the roof. Each individual user will then need to have a power conditioning unit with two functions. It 'conditions' the output from the solar array so that it operates continually at optimum efficiency; it also converts the DC to AC with a suitable waveform for direct connection to the grid. Efficiencies of over 90% are expected for microprocessor-controlled units. These contributory systems are seen as the secondary stage of development of solar electric equipment. Solar costs relative to fuel costs have already fallen fast and they are expected in due course to make contributory systems viable in terms of fuel saving alone.

A microprocessor-based converter optimiser has also been proposed by El-Hosseiny Taha El-Shirbeeny as a minimum-cost solution to meet criteria for reliability, continuity and quiet switching. Figure 2 shows a diagram of his converter system and Fig. 3 is a block diagram of the hardware.

Most of the work is done in three interrupt service routines — the control program, the computation program and the display program. The routine for the control program is shown in Fig. 4. The analogue output voltage and current from the photovoltaic converter are measured, the converter power computed and compared with the optimum limits. The manual mode is then checked. If it is not requested, the optimiser uses the electronic switching to connect the load circuit.

The Solar Power Satellite

The most ambitious of all solar power projects is that of the solar power geo-stationary satellite. To many people the concept seems like a bit of science fiction. Yet all the serious assessments have led to the conclusion that it is technically feasible, economically viable and can be realised in practice. The essence of it is converting solar energy into electricity in space, where much higher insolation levels are available than on earth, then transmitting the energy to earth under control.

First proposed by Dr Peter Glaser in 1968, the Solar Power Satellite (SPS)[2] has since been the subject of major studies, notably by the US National Aeronautics and Space Administration (NASA). The latest evaluation programme has covered

- system definition
- societal assessment
- environmental assessment
- comparative assessment

A baseline reference system has been adopted by NASA to form a common design basis for assessments. This is a 50 sq km array of solar cells placed in geo-stationary orbit, converting solar energy into DC electricity, and then to a microwave signal at 2.45 GHz, directed in a beam to a ground receiving station. There it is absorbed by an array of dipole elements in a receiving and rectifying antenna — known as a rectenna — and converted to low frequency AC (Figs. 5 and 6).

R. M. Shelton and R. A. Henderson summarise the conclusions of a major symposium on the subject in April 1980 as indicating that there are no 'show-stoppers' in any area; SPS is a serious contender for consideration in future energy scenarios, but of course it will need a truly operational and economic space transport system.

British studies have led to the view that there are big opportunities for UK industry in the concept; but it would all have to be on so large a scale that government support would be essential, a national organisation would need to be created, and the activity would have to be part of a European contribution. It was thought that 20 to 30 years would be necessary to realise SPS but we need a shorter-term target. "What is required" write Shelton and Henderson "is a project capable of realisation in 5 - 10 years in which the technologies and capabilities necessary to contribute to SPS can be aquired and developed."

Would the project really be economic in the conditions of UK and Western Europe? P. Q. Collins and R. Tomkins have used the NASA reference design to examine the cost implications taking into account load factors, reliability, system planning and integration, rectenna siting and transmission. They comment that since this is a space project, most of the work on it has been done by the space industry. Their own estimates start by taking the average works cost of electric power generated by nuclear plant (the cheapest) as 0.9p/kWh in 1978/9. The maximum prices that could be paid for microwave 'fuel' to meet this target at 90% and at 70% load factors are shown in Table 2.

What would now be useful would be for the utilities themselves to evaluate the cost of building and operating a rectenna, in order to make their own assessments of the maximum 'fuel' cost for the microwave energy.

In addition to cost there will also be the problem at the grid interface of variation of output. There will be long-term factors including deterioration of solar cells and other components, and shorter-term ones due to such factors as ionospheric scintillations. For shorter-term levelling, proposals examined have included using batteries, flywheels, superconducting magnets where the magnetic field acts as the store, electrolysing water for hydrogen production and storage or even load control schemes where tariffs would reflect the inconvenience of interruptible supply.

Hydrogen production is considered the cheapest form of load levelling if hydrogen were needed for making fuel (according to R.V. Gelsthorpe and R. H. Swansborough); but if the emphasis were on electrical power directly, then batteries would be favoured. Hydroelectric pumped storage is seen as a longer-term levelling option. Conventional generators would be needed in any case for stand-by duty. Overall, there is a general air of optimism that photovoltaic methods will be contributing increasingly in the near future to our power needs.

Fig. 4 Software flowchart for the control program — one of the three interrupt service routines used by this system.

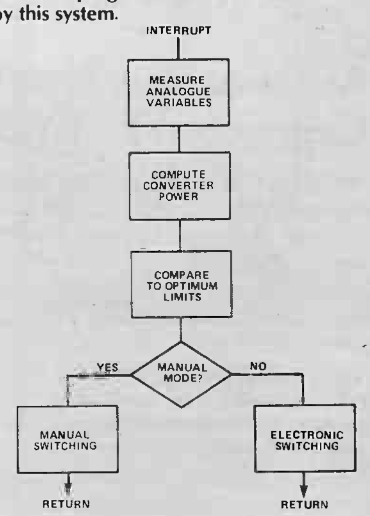
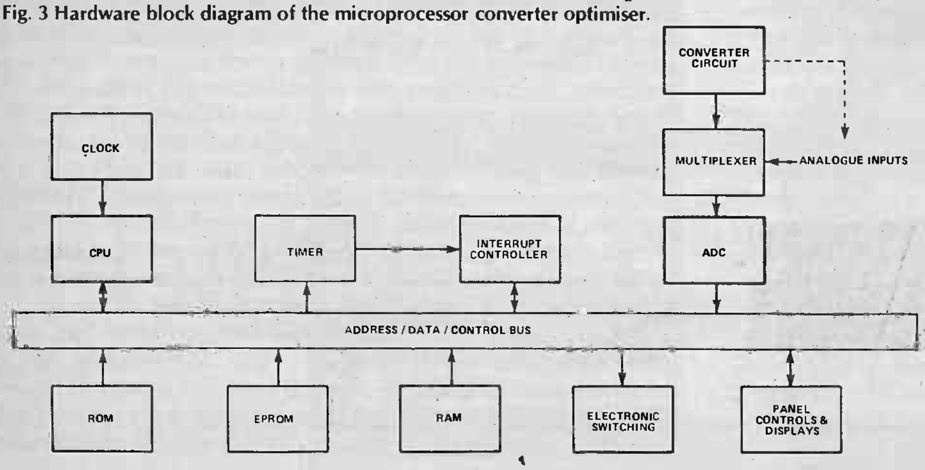


Fig. 3 Hardware block diagram of the microprocessor converter optimiser.



Semiconductor(s) Involved	Type of Cell	Rare Element	Density (Kgm m³)	Maximum Recorded Efficiency	Mass of Rare Element Required per GW** per um Thickness (tonnes)	Mode of Operation
Si	single crystal amorphous thin film		2200	18 6	22	Flat plate or low concentration
CdS/Cu_2S	thin film	Cd	8650	9	67.3	Flat plate
CdS/InP	single crystal InP all thin film	Cd and In	8650 7310	14 5	67.3 57.6	Only feasible with concentration
ITO*/InP	single crystal InP, thin film ITO	In	7310	14 -	57.6	Only feasible with concentration
$CdS/CuInSe_2$	thin film	Cd In Se	8650 7310	6	67.3 24.9 22.5	Flat plate
CdS/CdTe	thin film	Cd Te	8650 6240	6	107.8 33.2	Flat plate
$GaAs/Ga_xAl_{1x}As^+$	single crystal thin film	Ga	6095	23 5	37.1	Very high concentration flat plate

* ITO = Indium Tin Oxide
** Based on assumed 10% efficiency
\+ x taken as 0.2

Table 1. Principal solar cells under development.

References
1. 'Future Energy Concepts'. 3rd International Conference. Organised by the Institution of Electrical Engineers, in association with 17 other institutes and associations of engineers, 1981.
2. Glaser, P. E. 'Power from the sun: its future'. Science Vol. 162, Nov. 22, 1968.

(P. Q. Collins and R. Tomkins)
Maximum rectenna 'fuel' cost for base load operation

Capital Cost	Operating Cost 90% Load Factor	Operating Cost 70% Load Factor	Target Cost	Maximum Fuel Cost 90% Load Factor	Maximum Fuel Cost 70% Load Factor
200 £/kW	.08p/kWh	.10p/kWh	.90p/kWh	.82p/kWh	.80p/kWh
300	.11	.15	.90	.79	.75
400	.15	.20	.90	.75	.70
500	.19	.24	.90	.71	.65
600	.23	.29	.90	.67	.61

Table 2. Operating costs for the rectenna.

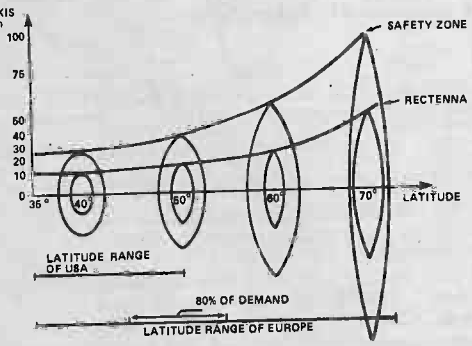


Fig. 5 A major disadvantage of the Solar Power Satellite is the large size of the rectenna. The graph shows the long axis as a function of latitude.

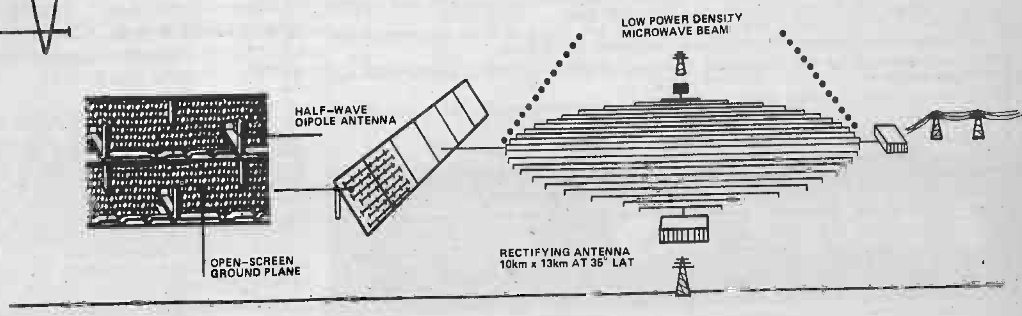


Fig. 6 Structure of the receiving and rectifying antenna — 'rectenna'.

ETI

DISCO MIXER

Attention DJs — are your mixes miserable, your panning pathetic and your octaves uneven? Cure your problems with the DJ 90 stereo mixer. Design and development by Tim Orr.

The DJ 90 is a stereo mixer, having two stereo inputs for magnetic cartridges, one stereo input (AUX) with a flat response and a microphone input. The three music inputs can be mixed together using slider controls and it is also possible to automatically pan, at either fast or slow rates, between the record deck inputs. The spectrum of the music signal is controlled by a five section graphic equaliser with two octave spacing as well as a special beat lift device. A voice-over unit (ducking) has been included as well as an override function for interrupt announcements. The microphone input can also be modulated, at a variable rate, to produce growl effects. A monitor section with a stereo headphone output allows the operator to listen (pre-fade listen) to any of the music inputs. The level of the selected signal path is displayed on an LED PPM.

Voltage controlled amplifiers have been used to control the signal levels in all seven audio paths. They have the ability to produce automatic cross-fades and ducking, as well as reducing crosstalk. The signal is not transmitted anywhere until the control voltage is correct. Therefore it will not crosstalk until it has been faded up, and then it doesn't matter.

Construction

The input board has components on only one side. Stick self-adhesive rubber feet beneath this PCB so that it does not short to the chassis. The panel board has components on *both* sides. First solder in all the small electronic components and then the panel controls. All the switches, sliders, rotary switches and LEDs go on one side of the PCB, and all other components on the other side. Be careful to get the LEDs the correct way round and to space their height so that they just make it through the holes in the panel. Also, solder the LEDs on both sides of the PCB. No other components are soldered each side.

The mixer uses an external power supply of ±12 V, for which a circuit is given. A kit of parts for this PSU is available (see Buylines), or you can substitute any other supply you may already have.

Great care must be exercised to avoid mains hum pick-up — the RIAA curve has maximum sensitivity at 50 Hz. The wiring from the pick-up should avoid all mains transformers and screened cable must be used. It is desirable to reduce to a minimum the unscreened pick-up leads fitted by the deck manufacturer. The record deck metalwork should be earthed to the earth tag on the mixer chassis. All exposed metalwork in the console should also be bonded to earth, of course. The chassis has a lip so that it may be fitted into a cutout in your console and connected to the amplifiers. The amplifier used in the prototype uses two of the 100 W modules of the mixer amplifier published in ETI February 1979. Keep the mains wiring well away from signal-carrying wires and use snubbers of 1uF 600 V in series with 100R to stop switching clicks when decks are switched. This will keep extraneous sounds to an unobtrusive level, but if preferred solid state relays controlled by low voltage switches could be used.

Setting Up

Once construction is completed, the 15 presets must be adjusted as follows:

PR1, 3, 5, 7 These adjust the gain of the two deck preamplifiers. Use the monitor and set them up for an average signal level of −6 to 0 dBm from a record.

PR2, 4, 6, 8, 11, 12 These are used to minimize the THD of the VCAs. Adjust them for zero voltage on the outputs of the 1458s following each LM13600, ie IC3 pin 1, pin 7, IC6 pin 1, pin 7, IC11 pin 1, pin 7 respectively. If a distortion analyser is available then use a 1 kHz sine wave with 0 dBm at the outputs of IC3, 6, 11.

PR9 This sets the gain of the microphone amplifier. Adjust to suit the microphone being used.

PR10 Set the MODE control to S. Adjust PR10 so that the DC voltage at IC8 pin 12 is zero.

PR13, 14, 15 These presets set the OFF level on the three music input sliders. Set all three sliders to OFF, inject music signals at the input sockets and adjust the presets until the VCAs just turn off.

This completes the setting-up procedure and the mixer is now ready for use. →

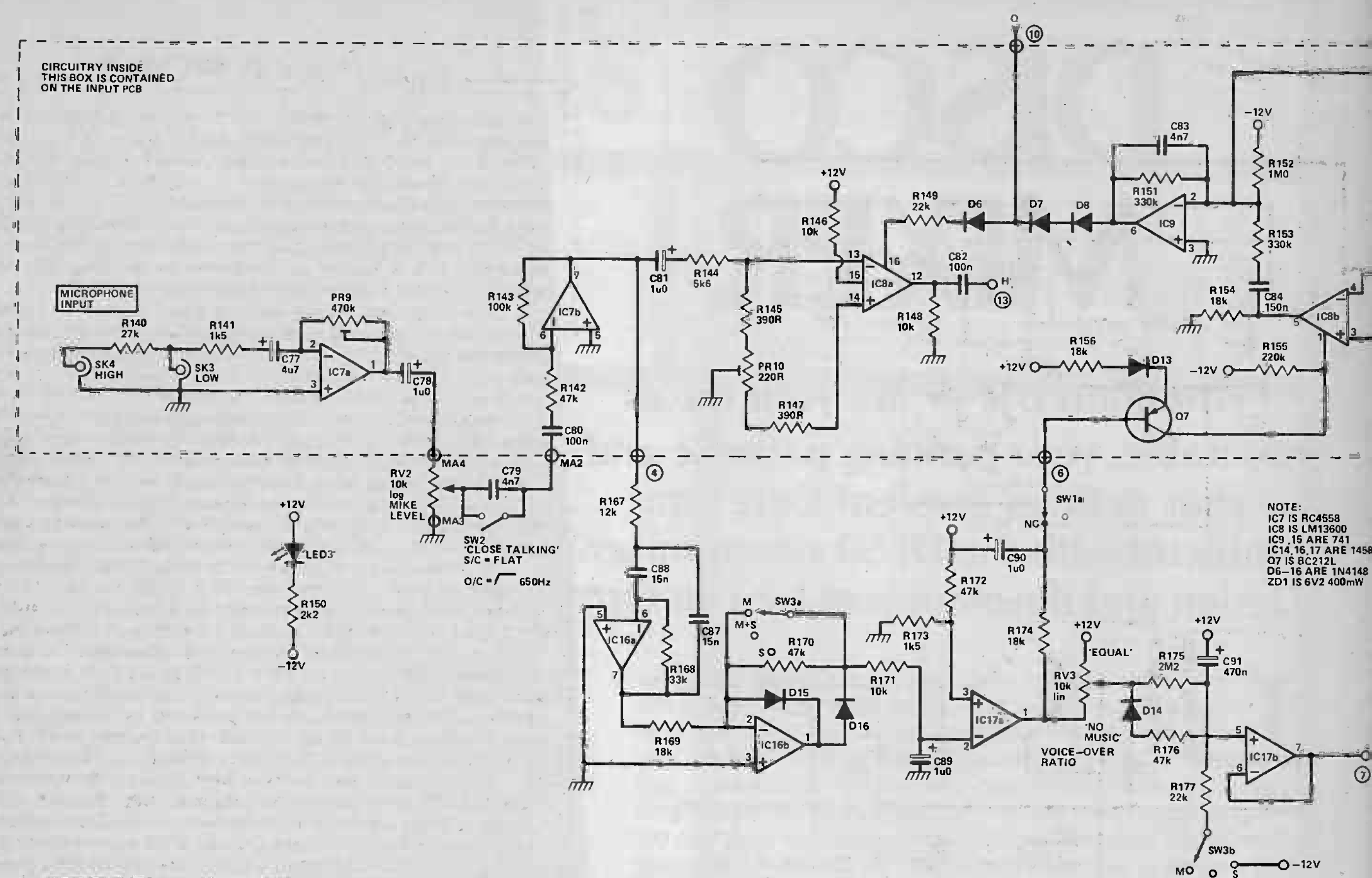


Fig. 1 (Above) Circuit diagram of the microphone preamplifier, voice-over and growl circuitry. The project is built on two PCBs linked by ribbon cable and Molex connectors — circled numbers are ribbon cable terminals and MA2,3,4 are the pins of Molex plug 'A' (see overlays next month).

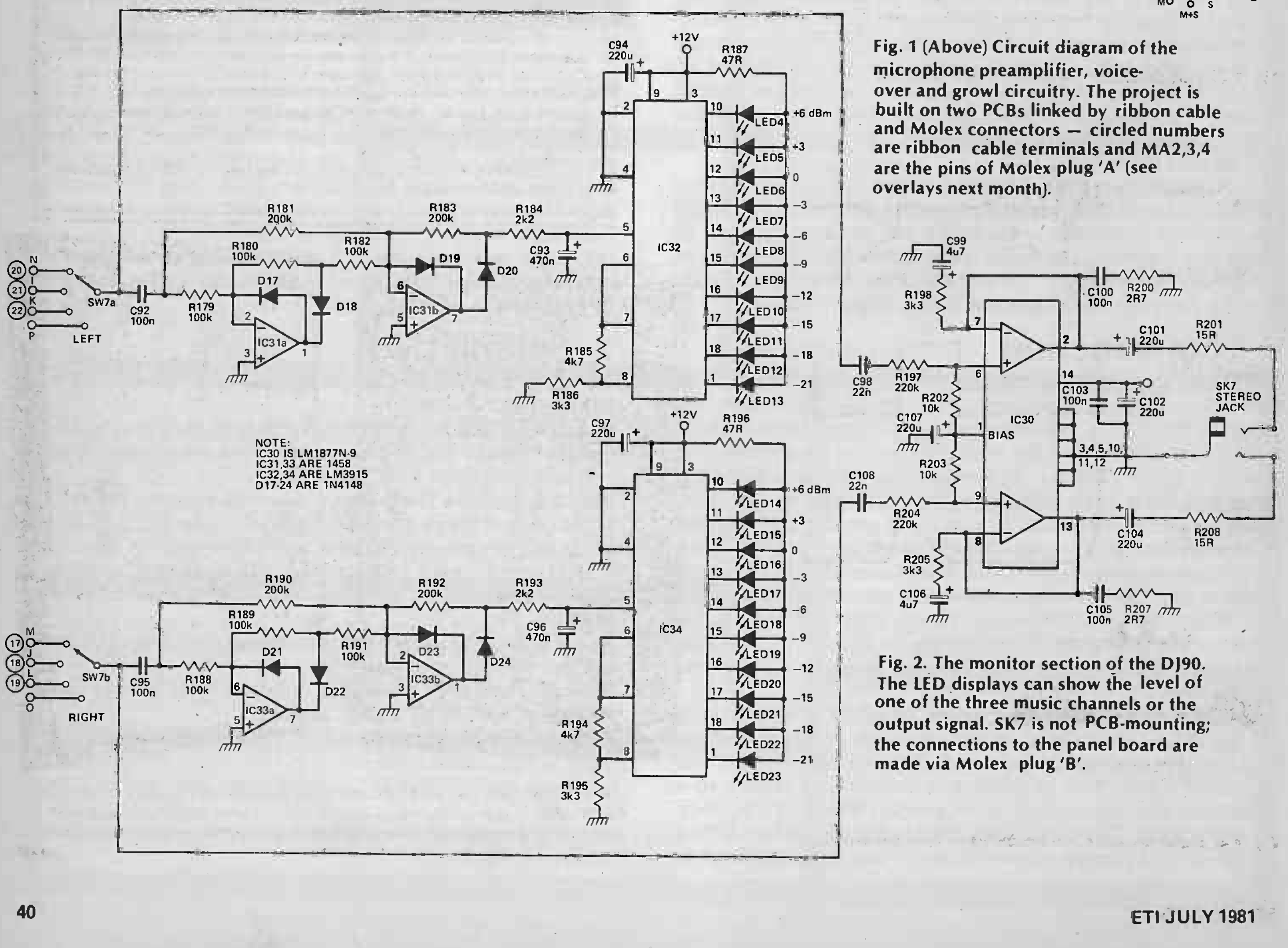
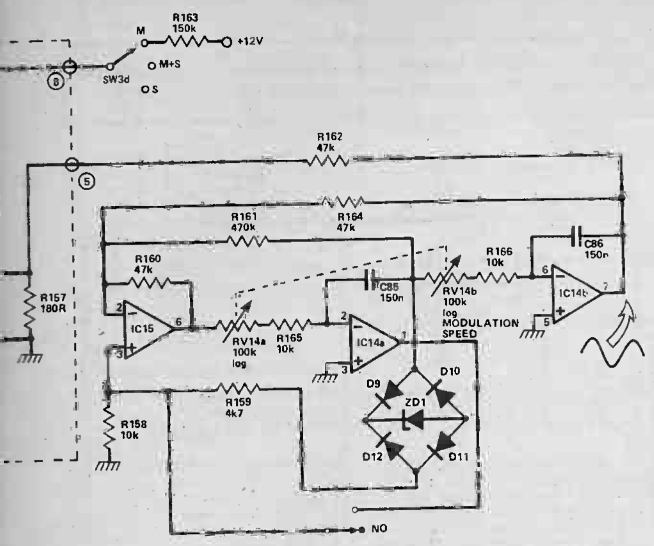


Fig. 2. The monitor section of the DJ90. The LED displays can show the level of one of the three music channels or the output signal. SK7 is not PCB-mounting; the connections to the panel board are made via Molex plug 'B'.



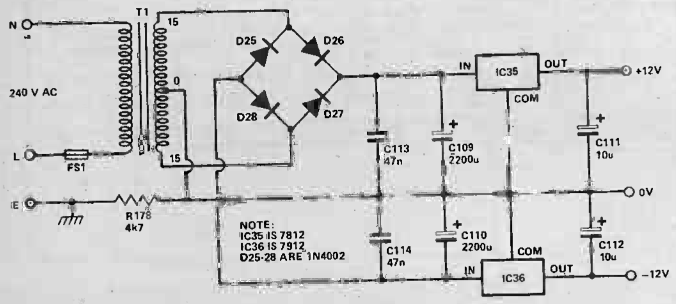


Fig. 3 Circuit diagram of a suitable power supply.

HOW IT WORKS

The microphone preamplifier (IC7) can accept high and low impedance sources. The amplified signal is fed to IC8 and to IC16. IC16 is a low Q band-pass filter (speech bandwidth) and an envelope follower. A comparator (IC17a) detects when the signal level has exceeded a preset threshold set by R172, 173. When this happens the comparator output goes low, generating a waveform at IC17b pin 7 that attenuates the music level in the three music channels. This is known as voice-over or ducking. The level to which the music is attenuated is controlled by RV3.

The comparator also enables the growl function. IC14 and IC15 form a variable frequency sine wave oscillator. The sine wave output passes through a VCA (IC8b) which is turned on when the comparator goes low. This prevents the sine wave breaking through when no speech is present. The sine wave is level-shifted by IC9 and is used to amplitude-modulate the microphone signal using the VCA IC8a. The output of this VCA goes straight to the output mixer of the unit, IC26, as already mentioned. A mode control switch SW3 determines which signal sources are enabled. S selects speech only, M + S selects music with voice-over, and M selects music only. However, in this last mode the override function may be used (SW8). The override slightly attenuates the music channels and turns on the microphone channel.

As all the signal switching is performed with band-limited control voltages, virtually clickless switching is produced. The control voltages are converted into currents by differential transistor pairs Q1,2, Q3,4 and Q5,6. These currents are then used to drive the VCAs. The relationship between voltage and attenuation is ultimately logarithmic. The voice-over control voltage is injected into the transistor pairs via R75, 76. A negative-going voltage attenuates all music channels. Override attenuation is produced by robbing part of the control current down D3,4,5. The individual channel gain is determined by the emitter current. This can be defined by slider settings (RV5, 14, 15) or by the auto-pan control voltages. When in the auto mode, the output of IC29 ramps linearly up and down under the control of SW4. Complementary versions of this waveform are produced at the output of IC28a and IC28b. Thus the magnitude of the emitter currents in Q1, 2 and Q3, 4 are complementary and so as the output of IC29 ramps up and down automatic panning between decks is performed.

The monitor section is a pre-fade listen for all three music channels and a post-fade listen of the output signal. Full wave rectifiers (IC31a, IC33a) and peak envelope followers (IC31b, IC33b) monitor the signal level and provide the drive signal for the LM3915 PPM units (IC32, 33). A dual medium power amplifier (IC30) generates the headphone output. There is no level control on this output and so R201, 208 should be altered to suit the headphones.

The mixer requires a ±12 V supply derived via voltage regulators in a completely standard circuit.

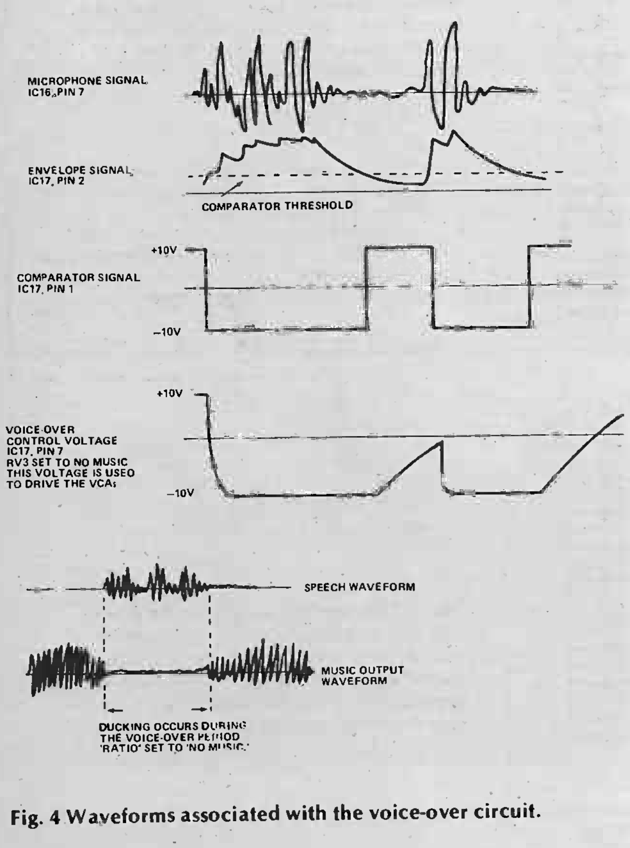
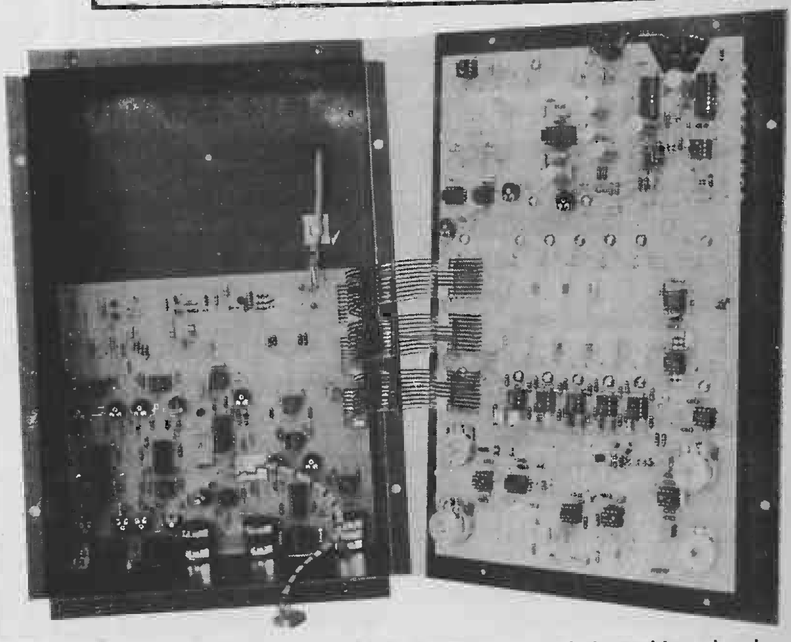


Fig. 4 Waveforms associated with the voice-over circuit.



The DJ90 is built on two PCBs, the large double-sided panel board and the smaller, single-sided input board. Interconnections are made with three 10-way ribbon cables and a Molex connector, which can be seen wired to the centre of the input board.

Next month we conclude the DJ90 project with a description of the input and equaliser circuitry, and the overlays for the three boards.

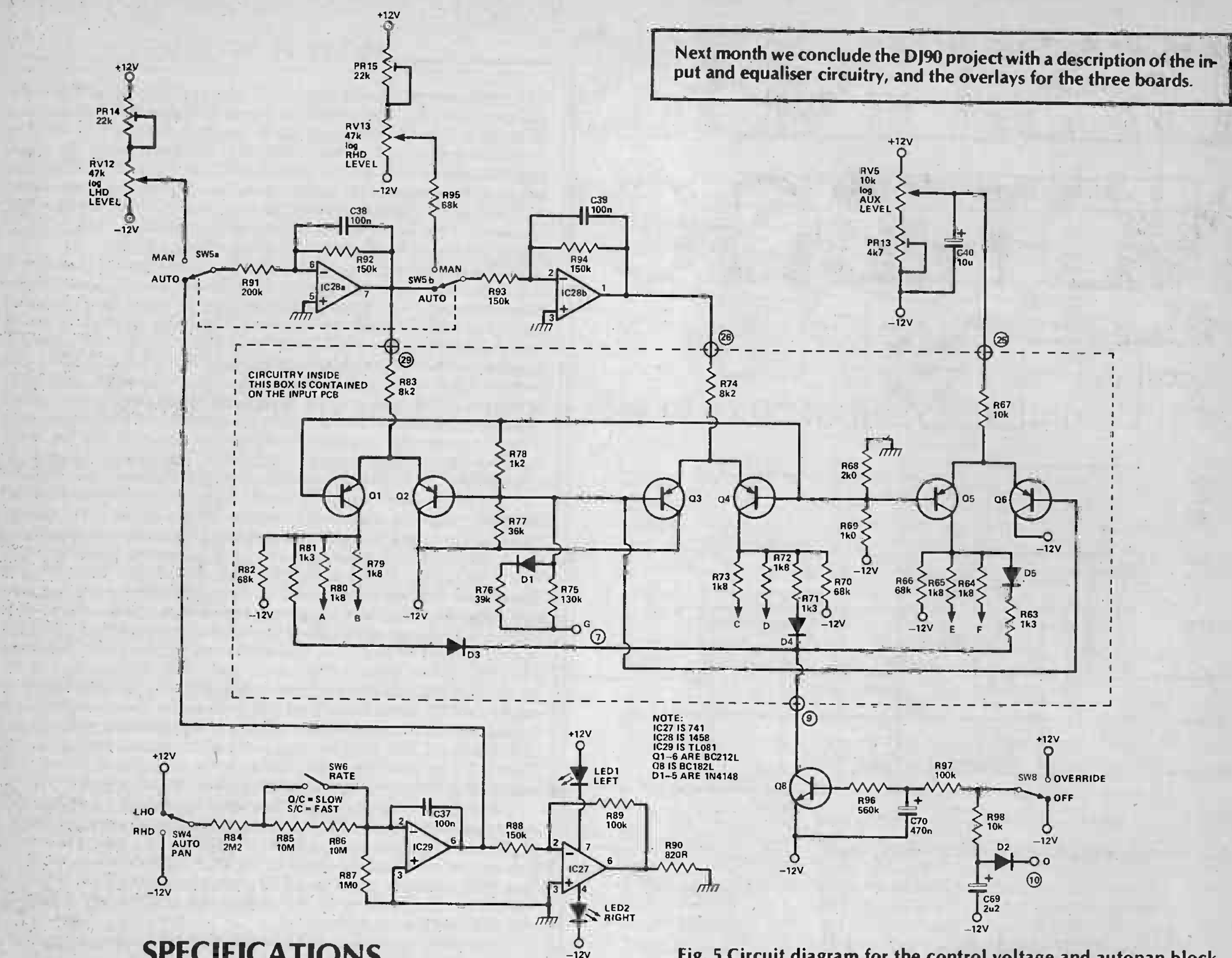


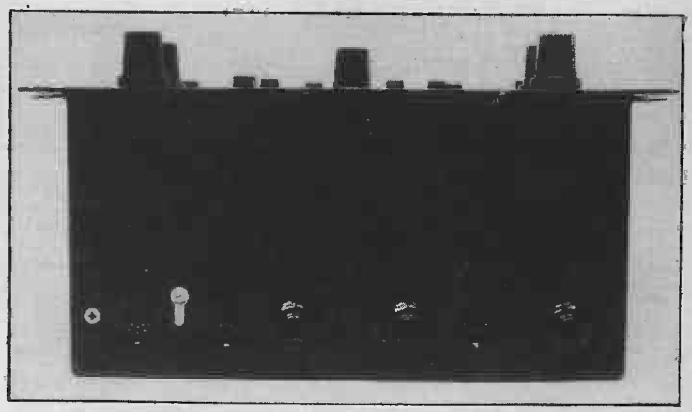
Fig. 5 Circuit diagram for the control voltage and autopan block.

SPECIFICATIONS

Size: 22.5 x 33.0 x 10.5 cm
Deck input impedance: 47k in parallel with 47pF. Deck inputs have RIAA equalisation.
Auxiliary input sensitivity: −6 dBm into 47k.
Microphone channel gain: 56.5 dB into 1k5 (low)
31.0 dB into 28k5 (high)
Normal output level, output slider set at maximum: +6 dBm into 600R.
Equaliser frequencies: 50, 200, 800, 3200, 12800 Hz. ±15 dB.
Beat lift frequency: 90 Hz, 0 to +13 dB
Noise performance:
RIAA preamplifier output noise (input open circuit): −73 dBm
VCA driven by noise from RIAA stage: — 71 dBm (on) and −115 dBm (off)
AUX VCA output: −81 dBm (on) and −115 dBm (off)
IC12 mixer output: −105 dBm all three music channels off.
−81 dBm with AUX channel on.
−71 dBm with one deck on.
−69 dBm with both decks and AUX on.
Microphone equivalent input noise: −117 dBm (ie 1 uV$_{RMS}$)
Minimum slider attenuation on music inputs: 66 dB.
Override attenuation: 6 dB.

BUYLINES

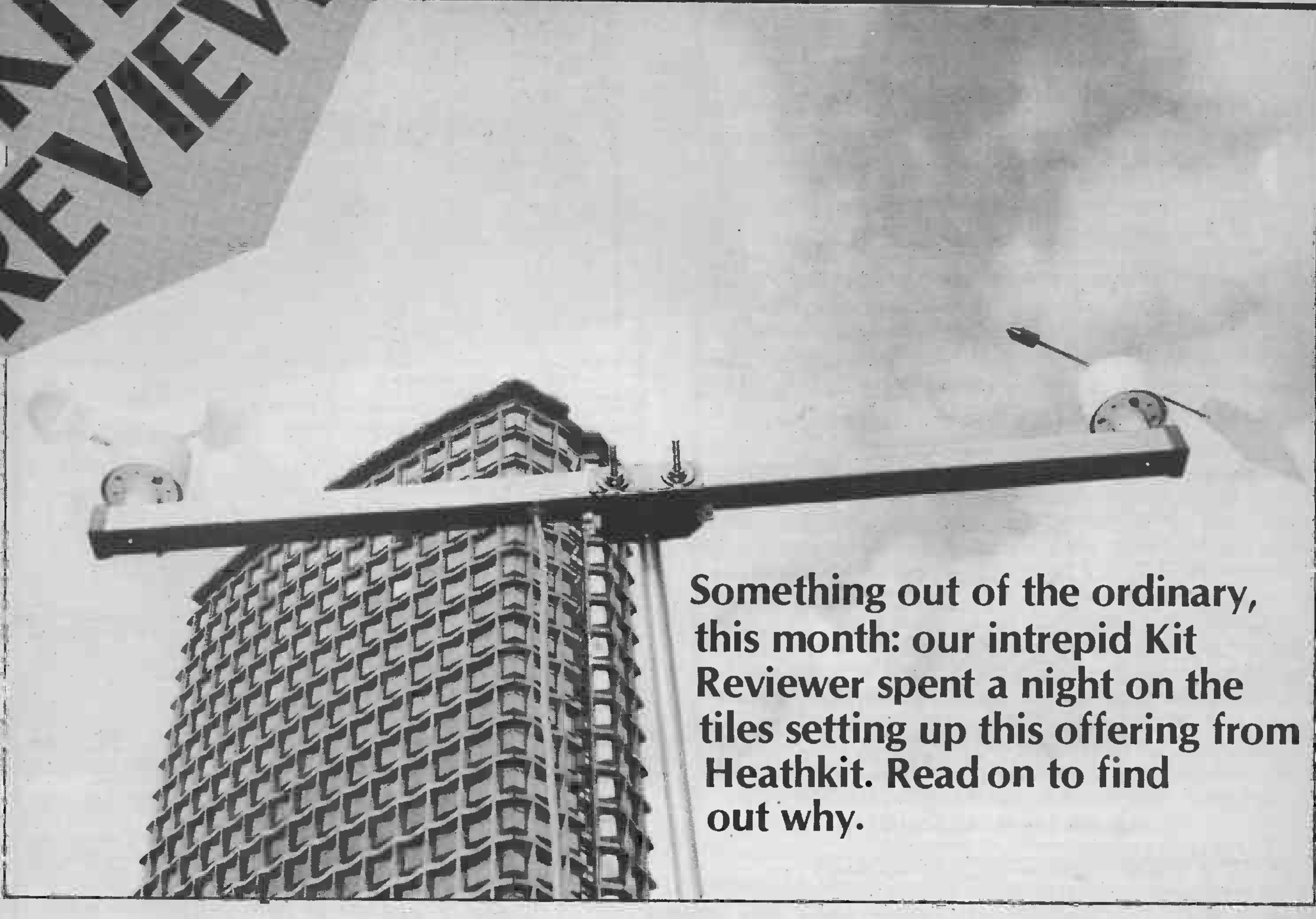
A complete set of parts for this project, including fully finished metalwork, nuts, bolts etc, will be available from Powertran Electronics for £97.50 plus VAT for the mixer and £9.90 plus VAT for the power supply, post free. For delivery by Securicor add £2.50 (VAT inclusive). Powertran also supply the separate parts for the mixer, eg metalwork set, PCB, semiconductors etc. Telephone Andover 64455 or write to Powertran Electronics, Portway Industrial Estate, Andover, Hants SP10 3NM.



The back panel, showing the input and output sockets and the deck earthing tag.

ETI

KIT REVIEW



Something out of the ordinary, this month: our intrepid Kit Reviewer spent a night on the tiles setting up this offering from Heathkit. Read on to find out why.

As I'm writing this in my cosy little office, there's an east wind blowing outside at a speed of 6 mph, with gusts up to 10 mph. The gadget telling me this is the Digital Wind Speed and Direction Indicator (Model ID-1590E), one of the excellent range of Heathkits. The kit is in two parts; a transmitter that is mounted on a TV mast (for example) and carries the sensing circuitry, and a small receiver unit that displays the readings.

Reed All About It

The transmitter consists of a long boom with a small plastic housing at each end. One of these contains eight reed switches, arranged in a circle around the central spindle of the weathervane. As the weathervane rotates, a magnet on the spindle closes one or two of the reed switches, lighting the corresponding lamps on the front panel of the receiver. The eight principal compass points can thus light singly or in pairs to give 16-point resolution. Heath have used a neat trick to halve the number of signal-carrying wires. The lamps and switches are connected in pairs to one wire and are driven by an AC supply; diodes in series (but with opposite polarities) ensure that current in one direction down the wire lights one bulb, current the opposite way lights the other.

The other sender carries three anemometer cups, rotating on a spindle with four magnets attached. As the cups spin the magnets open and close a single reed switch; the pulses from this are counted by circuitry in the receiver to obtain the wind speed.

The receiver itself is a small, well-designed plastic box with a black and mock teak finish. The direction arrows and speed display (seven-segment neon tubes) glow orange against a black background — a nice effect, and it won't look out of place in your living room. There are three display options (any two of miles per hour, kilometres per hour and knots). One option must be chosen during construction — a switch on the back panel selects one or other of your two chosen units. A second switch is provided for calibration purposes. The receiver and transmitter are linked by a length of eight-core cable.

Manual Construction

The kit is presented in the usual impeccable Heathkit fashion, with top-quality components packaged in separate envelopes and bags, all the necessary hardware, and an instruction manual the size of a copy of ETI. The first job, as always, is checking that all the components are present; Heath provide a seven-page checklist for this purpose, with drawings of each component to aid identification. At this point I discovered two things — the eight-core cable is available in several lengths to suit your application, so it must be obtained separately; and the mains cable grommet was missing. A short stroll down the road to the nearby Heathkit shop solved both problems. As far as the hardware goes, I had a few extras in my kit.

The construction notes are superb, and it's fair to say that if you can't manage to build a Heathkit you couldn't build *any* kit. A separate leaflet is provided, giving soldering instructions; these are of a standard that would have even a complete beginner soldering perfect joints. Each step in the main manual is accompanied by detailed diagrams and exploded drawings, and each instruction (these go down to the level of "cut off the excess lead lengths after soldering") has a box so that you can check it off once it's completed. Exact measurements are given for cutting wires, with a rule printed along the bottom of the page.

As far as the receiver was concerned, I encountered no problems. Calibration is simple and requires no special equipment

— with the calibration switch up, the circuitry counts the 50 Hz pulses from the transformer secondary. Two presets have to be adjusted, one for each of your chosen ranges (I decided on miles per hour and knots), until the display agrees with the calibration tables in the manual. Adjustment begins with the presets in mid-position, and in my kit they were supplied dead centred — a nice touch.

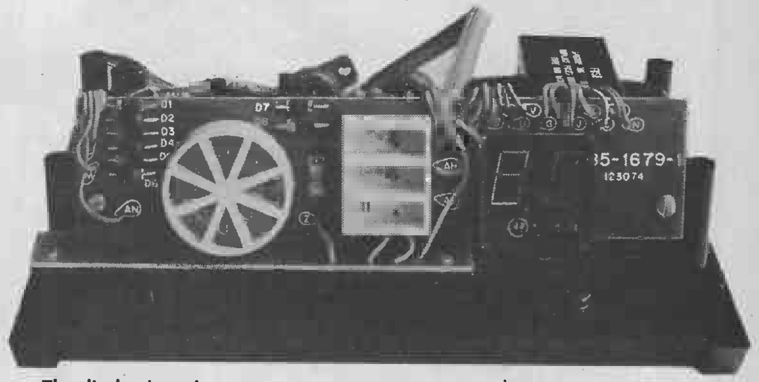
The entry hole for the mains cable is sized for American two-core cable, and I had to take a drill to it so that the grommet (necessary to comply with European regulations) could be fitted.

The receiver, in tasteful teak-and-black, shows that an east wind is blowing with a speed of 9 mph.

Up, Up And Away

The transmitter was also fairly straightforward to put together, but when I came to test the wind direction sender, up to four lamps were lighting instead of the maximum of two. Hmmmmm. Referring to the comprehensive fault-finding charts provided in the manual showed that the magnet position was probably at fault. A few minutes' work with a pair of pliers solved the problem.

After making a few final checks, it was up to the Modmags roof, with its picturesque view of Centre Point, to install the transmitter boom. Position is important; the boom must be away from objects that will cause wind turbulence or shield the sender, and oriented so that the direction display reads correctly. The cable can be led into the building at any convenient point, and the entry hole sealed; the boom itself is weatherproof.



The display board uses seven-segment neon tubes and small light bulbs; the latter are fitted inside the plastic light-shields.

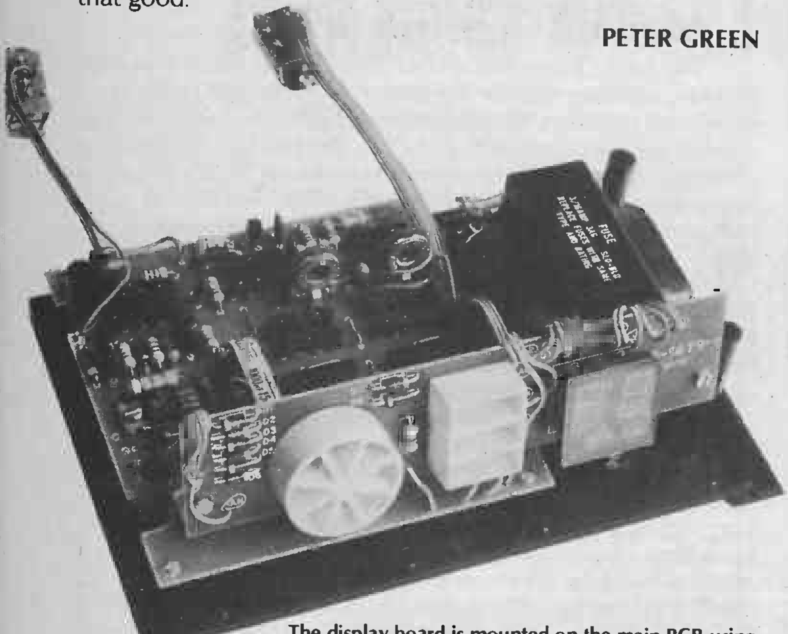
Conclusions

A quick glance at the price in Buylines may cause a sharp intake of breath, as it did with several members of staff here. But you have to bear in mind that you're paying for the knowledge that everything has been done to ensure that your kit works first time; for things like Teflon bearings and high quality screen-printed PCBs; and for sheer professional quality. I can guarantee that if you put the receiver into your house without comment, people will assume you bought it ready-made — it's that good.

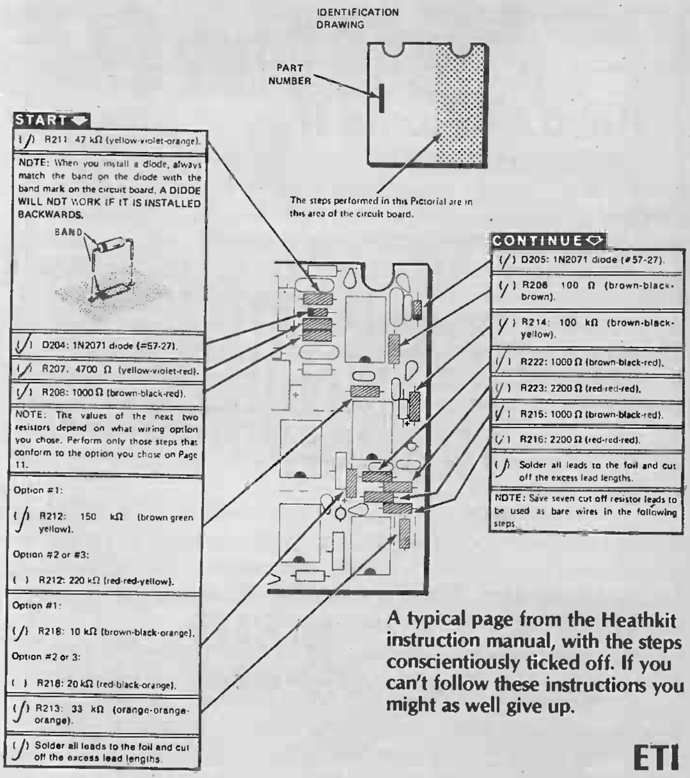
PETER GREEN

BUYLINES

Heathkit Digital Wind Speed and Direction Indicator (ID-1590E) £93.00 including VAT and postage. Eight-core cable (IDA-1290-1 (50 ft)) £11.00. Other lengths available.
Heathkit Electronic Centre, 233 Tottenham Court Road, London W1P 9AE. Head Office and Mail Order Sales Department: Heath Electronics (UK) Ltd, Gloucester GL2 6EE.



The display board is mounted on the main PCB using the angled brackets supplied. Circuit interconnections are made with ribbon cable, and Heath give detailed cutting instructions. The switches are ready to be fastened to the back panel.



START

(/) R211 47 kΩ (yellow-violet-orange).

NOTE: When you install a diode, always match the band on the diode with the band mark on the circuit board. A DIODE WILL NOT WORK IF IT IS INSTALLED BACKWARDS.

BAND

(/) D204: 1N2071 diode (#57-27).

(/) R207. 4700 Ω (yellow-violet-red).

(/) R208: 1000 Ω (brown-black-red).

NOTE: The values of the next two resistors depend on what wiring option you chose. Perform only those steps that conform to the option you chose on Page 11.

Option #1:

(/) R212: 150 kΩ (brown green yellow).

Option #2 or #3:

( ) R212: 220 kΩ (red-red-yellow).

Option #1:

(/) R218: 10 kΩ (brown-black-orange).

Option #2 or 3:

( ) R218: 20 kΩ (red-black-orange).

(/) R213: 33 kΩ (orange-orange-orange).

(/) Solder all leads to the foil and cut off the excess lead lengths.

IDENTIFICATION DRAWING

PART NUMBER

The steps performed in this Pictorial are in this area of the circuit board.

CONTINUE

(/) D205: 1N2071 diode (#57-27).

(/) R206 100 Ω (brown-black-brown).

(/) R214: 100 kΩ (brown-black-yellow).

(/) R222: 1000 Ω (brown-black-red).

(/) R223: 2200 Ω (red-red-red).

(/) R215: 1000 Ω (brown-black-red).

(/) R216: 2200 Ω (red-red-red).

(/) Solder all leads to the foil and cut off the excess lead lengths.

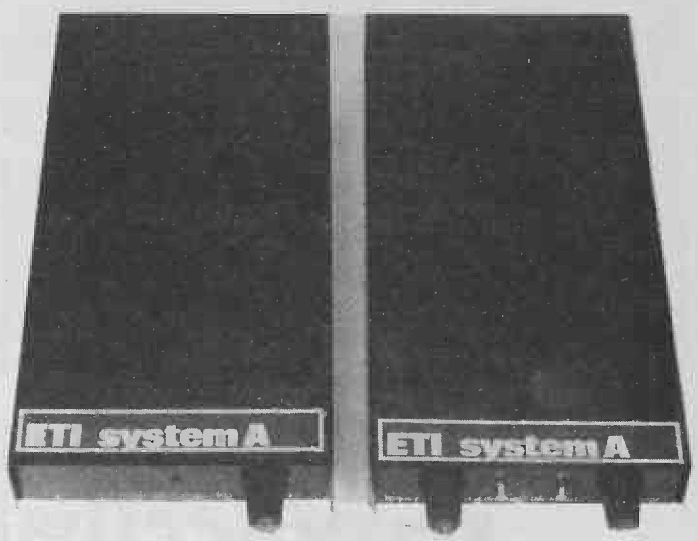
NOTE: Save seven cut off resistor leads to be used as bare wires in the following steps.

A typical page from the Heathkit instruction manual, with the steps conscientiously ticked off. If you can't follow these instructions you might as well give up.

ETI

SYSTEM A AUDIO AMPLIFIER

Look no further. This superb amplifier is quite simply the best. Designed to out-perform even commercial equipment, the System A combines ease of construction with Class A quality. Design and development by Stan Curtis.



The initial design brief for this amplifier — 'no compromise' signal reproduction but at the lowest possible cost — proved to be deceptively difficult! The first preamp design eliminated all switches and controls to leave a pick-up input socket, an output socket and a volume control, but such a layout would be far too spartan for even the most serious audio enthusiast. The minimum input requirements were thought to be pick-up, tuner and tape, with tape recorder/monitor output. A stereo-mono switch is unnecessary for serious listening, as are all the other controls that came to mind (except volume and balance!).

The next choice was between discrete or integrated circuits. Despite the obvious benefits and inherent simplicity of IC-based circuitry, I decided upon good old-fashioned transistor stages. Why? Several reasons:

1. If labour costs are disregarded (which they are in this case) the discrete transistor version costs less.
2. Discrete stages can be more easily optimised for a particular design requirement, and give a lower component cost and higher sound quality.
3. There is a purely emotional feeling that when using audio ICs, the designer hasn't really contributed very much to the final design!

(In fact the final circuits are, in effect, discrete component operational amplifiers, so something of the IC design philosophy has obviously rubbed off.)

Pick-up An Input

Provision has been made for the preamp to be used with virtually any available pick-up cartridge, through the use of plug-in input circuit boards. Two input circuit boards have been designed although both use the same printed circuit layout. One is for moving-magnet cartridges and the other for moving-coil cartridges. The gain of both these modules can be varied to suit different cartridges by the change in value of a single resistor. Input loading (both resistive and capacitive) can be changed by the substitution of alternative components and, as a source of guidance, a comprehensive table has been produced showing the requirements for the majority of pick-ups currently on sale.

The whole of the preamp design is extremely flexible, permitting alterations to ensure compatibility with other equipment. The basic version has a nominal 775 mV output level and a 75 R output impedance.

PSUing Quality

The power supply is built into a separate case to achieve better screening as well as increasing the versatility of the system. This new 'Audiophile' system is conceived as a modular 'building block' concept offering a variety of facilities. The power supply is capable of powering several preamplifiers but will also be used to power a matching parametric equaliser unit and two other blocks still under development. Their basic designs will follow the existing format and will be published in the months to come.

As for the preamplifier, work goes on to take advantage of its ability to accept alternative input modules, and the design of new modules will be published periodically to enable constructors to update their models.

Outward Bound

Provision has been made on the main PCB for the fitting of an output coupling capacitor (C15). Normally this shouldn't be necessary and the two pads should be joined by a wire link to couple the output directly to the power amplifier. A very small number of power amplifiers are totally DC-coupled, so any DC voltage on their input terminals would result in an unacceptable DC offset across the loudspeaker. In such a situation the capacitor should be fitted. Its value can be selected to suit the input impedance of the power amplifier; a value of 3u3, 35 V (tantalum) is acceptable with a 10k input impedance and 470n with a 50k input impedance. The capacitor polarity should be aligned to correspond to the residual DC offset at the output of the preamplifier.

Construction

Although no metalwork plans have been provided it will be seen that the prototypes have been housed in a simple, compact, and functional case consisting of an aluminium chassis and a substantial steel cover. (Arrangements have been made for supplies of these cases to be made available to ETI readers — see Buylines.)

TABLE 1. SPECIFICATION

PREAMP			
Rated output level:	775 mV (0 dBm)		
Maximum output level: (20 Hz to 20 kHz)	7V8		
Total harmonic distortion (including noise)			
Auxiliary input, 775 mV output	20 Hz	0.01%	
	1 kHz	0.01%	
	20 kHz	0.01%	
Pick-up input, 1V5 output	20 Hz	0.02%	
	1 kHz	0.02%	
	20 kHz	0.02%	
Pick-up input overload (ref rated input at 1 kHz)			
	Moving Magnet	Moving Coil	
20 Hz	43 dB	44 dB	
1 kHz	43 dB	40 dB	
20 Hz	43 dB	32 dB	
Input sensitivity (ref 775 mV output at 1 kHz)			
Auxiliary	65 mV		
Pick-up (moving magnet)	2.3 mV		
Pick-up (moving coil)	550 mV		
Noise level, 'A' weighted (ref 775 mV output at 1 kHz)			
Auxiliary	−90 dBA		
Pick-up (moving magnet)	−80 dBA		
Pick-up (moving coil)	−76 dBA		
Channel separation, pick-up input (unused channel loaded)			
	1 kHz	62 dB	
	20 kHz	69 dB	
RIAA equalisation accuracy: (20 Hz to 20 kHz)	±0.2 dB		
Frequency response: (auxiliary input)	±0.5 dB, 5 Hz to 35 kHz		

The above figures are for the standard version. The performance of the alternatives will vary in terms of sensitivity etc.

POWER AMP	
Biasing mode:	Class A
Rated power:	60 W RMS into 8R, 20 Hz to 20 kHz
Transient delivery:	150 W into 8R

Harmonic and intermodulation distortion: less than 0.06% at rated power output (20 Hz to 20 kHz), decreasing monotonically with decrease in power. Distortion is virtually unmeasurable at small signal levels.

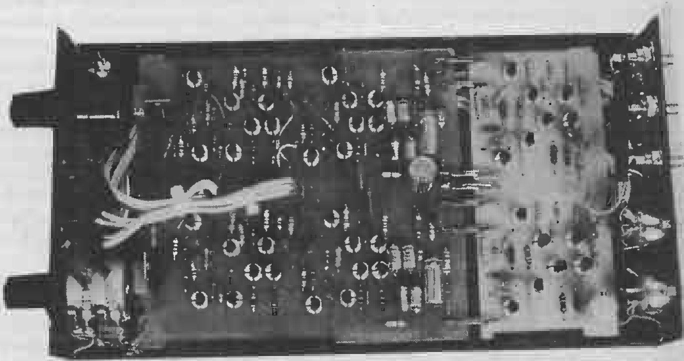
Frequency response: (ref 0 dB at 1 kHz)	10 Hz −1 dB 120 kHz −6 db
Power bandwidth:	5 Hz to 60 kHz
Hum and Noise:	100 dB below 24 V RMS output (CCIR)
Sensitivity:	700 mV RMS for 60 W into 8R

Negative feedback: the open loop gain is reduced by 22 dB by the application of overall negative feedback.
Transient intermodulation distortion: zero

The preamplifier circuitry has been constructed on two printed circuit boards which plug together using high quality gold-plated connectors. The construction of these boards should present no difficulties if the layout is followed correctly. There is a certain amount of wiring using screened cable and it is essential that this be done neatly and correctly. A wiring diagram has been given which shows the loom in detail and this arrangement should be followed fairly closely. The ends of all screened cables should be sleeved to avoid the danger of stray strands shorting out the signal. Particular attention is drawn to the earth connections which are always a problem with stereo amplifiers. The arrangement as drawn works. Others might not! You may wonder why this wiring has not been incorporated on the PCB. This could have been done for ease of assembly but only at the cost of the loss of isolation between the various signal and supply paths. In this context it is interesting that one of the world's best regarded preamps, the ultra-expensive Levinson, using several hundred dollars' worth of military grade, PTFE-insulated screened cable in the pursuit of signal isolation. However, our budget model uses common-or-garden screened cable to do the same thing! The use of this cable plus some care in layout results in a quite respectable figure for stereo separation at high frequencies.

It is recommended that the phono sockets for the pick-up inputs be gold-plated. These are expensive and difficult to obtain but, for optimum results to be obtained, they must be used. I have undertaken a lot of research into the effects of signal connections and have found that, while in theory both the gold-plated and nickel-plated contacts give equally good connections, in practice and over a period of time the gold-plating will prove its worth. I will say no more because a full summary of the problems associated with connectors would fill an article of its own.

Most of the transistors used are uncritical and the recommended types can often be substituted for, provided that due regard is paid to voltage ratings and so on. However, the 2N4401 first stage transistors are notably quieter than many alternative 'low noise' types (BC109 etc) and these should be fitted. The input transistors (Q1 and Q2) used for the moving-coil stage (module A-MC) are medium-power devices selected from the BC160 family. They are tested for low noise under the specified operating conditions. Transistors of this type could be fitted on a 'pot-luck' basis but this may lead to disappointment, frustration, and a need for a new nozzle on your solder sucker! Alternatively the correct pre-tested transistors can be used and a supply of these has been made available (see Buylines).



Inside the prototype preamplifier. Construction is on two boards, the main preamp module A-PR and the smaller input module. The latter is connected to the main board and the phono input by gold-plated connectors (see Buylines). This enables different input modules to be easily exchanged to match different cartridges. If you're *certain* you'll only ever be using one cartridge, you could dispense with the connectors and solder wire links instead.

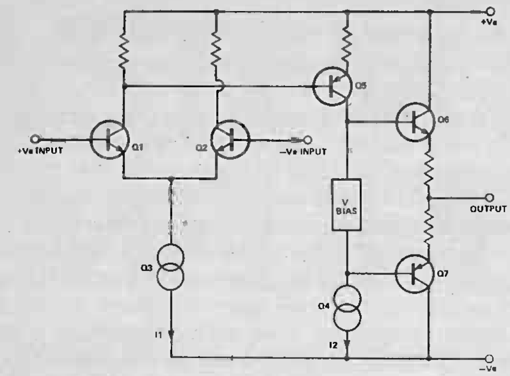


Fig. 1 Simplified diagram of one gain stage.

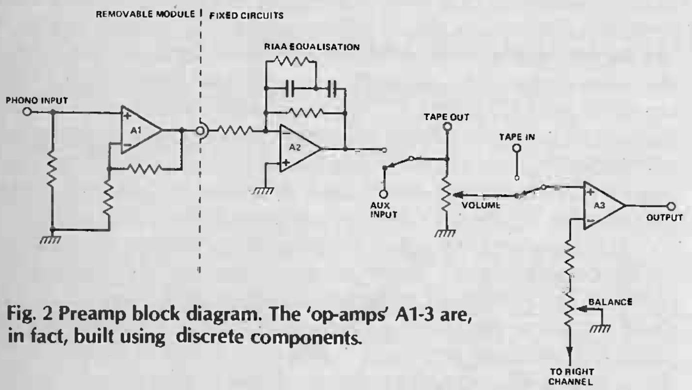


Fig. 2 Preamp block diagram. The 'op-amps' A1-3 are, in fact, built using discrete components.

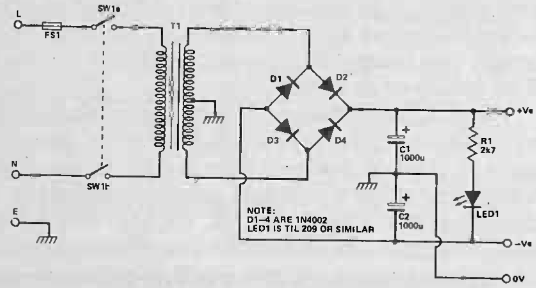
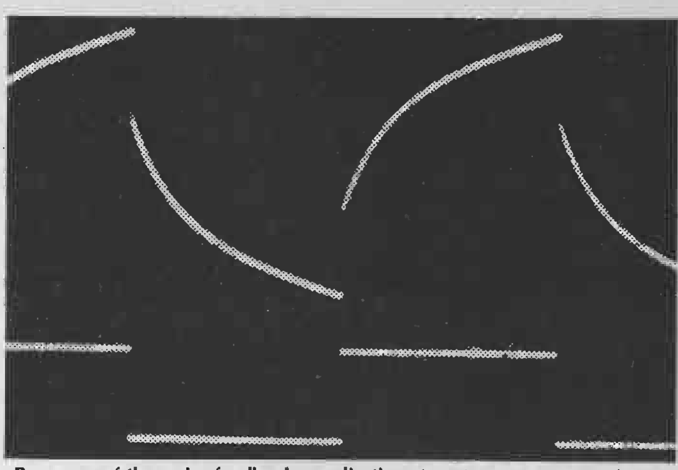


Fig. 4 Circuit diagram of the A-PSU preamplifier power supply.



Response of the series feedback equalisation stage to a square wave input signal.

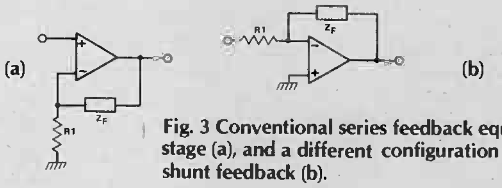


Fig. 3 Conventional series feedback equalisation stage (a), and a different configuration using shunt feedback (b).

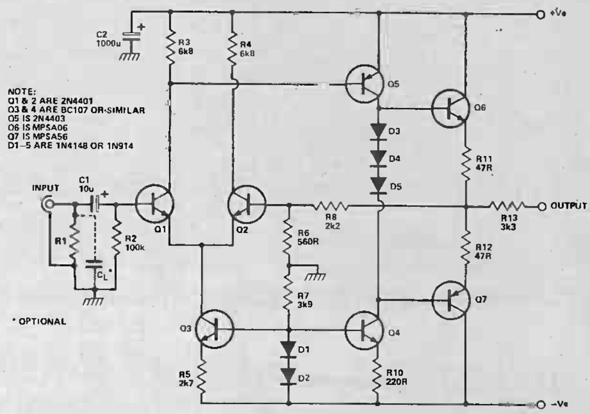


Fig. 5 Circuit diagram of the A-MM moving magnet module.

Each stage of the preamplifier uses a virtually identical discrete component operational amplifier. This op-amp is shown in simplified form in Fig. 1. The input stage is a long-tailed pair composed of transistors Q1 and Q2 whose collector current is determined by a constant-current source (Q3) and works out at about 100 uA for each transistor. This current has been chosen to give a low noise figure for this stage. The second stage is a voltage amplifier (Q5) which drives a constant current load (Q4) to set the standing current of this stage at about 2 mA. The four series diodes bias on the complementary output stage (Q6, Q7) to give a quiescent current of 8 mA. This value of standing current ensures that all the amplifier stages continue to operate in the linear Class A region even when driving low impedance loads.

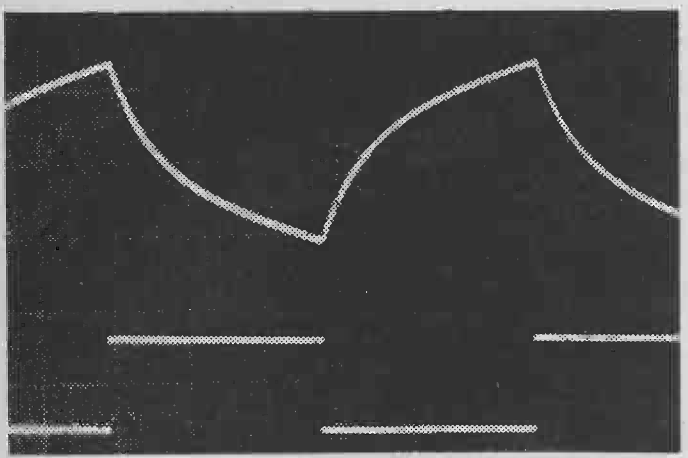
The moving-coil stage is virtually identical to the other op-amps except for the use of some different component values. Whereas the other stages are optimised for low noise when driven from medium impedance signal sources, the moving-coil cartridge can represent an almost pure resistance of between 2 and 10 R. To achieve a better noise figure medium-power transistors are used in the input stage, and each is operated at a collector current of slightly over 1 mA.

The three stages are arranged as shown in the system block diagram (Fig. 2). The first stage can be either a moving-magnet or a moving-coil stage. Whichever is chosen, the gain and input loading are optimised to suit the pick-up cartridge in use. This stage has a flat frequency response and no feedback equalisation. It does, therefore, buffer the cartridge from the equalisation stage and so ensures that the cartridge loading is not frequency-dependent.

The second stage is the equalisation stage with the RIAA network wired in a shunt feedback arrangement. This stage has a voltage gain of 20 dB (x10) at 1 kHz and brings the signal level up to a nominal 50 mV before the switching circuits. After the volume control comes the third stage (A3) which is wired as a simple 20 dB (x10) line amplifier. However, the feedback resistor is wired to ground through a potentiometer which acts as a balance control, giving a gain variation of 11 dB on this stage.

Shunt Feedback

The purpose of the equalisation stage is to provide a fixed degree of frequency de-emphasis exactly complementing the RIAA specified pre-emphasis applied when a record is cut. Although the equalisation is normally specified over the band 20 Hz to 20 kHz it was assumed that the response curve would be continued outside of the audio band. Most important, the replay response above 20 kHz should continue to reduce with frequency until at some infinitely high frequency the output is zero. This requirement is disregarded by most audio engineers who concentrate primarily on the audio band performance, but the music signal reproduced from a disc contains transients whose frequency content can lie outside the arbitary audio band. (Question: why 20 Hz to 20 kHz? Answer: because it has always been so!) The conventional series feedback stage of Fig. 3a is unable to provide an accurate transfer of these high frequencies. This is because the gain does not drop towards zero with increasing frequency but towards unity. The voltage gain of this stage is equal to $1 + (Z_F \div R1)$; so even if Z_F is made infinitesimally small the minimum gain cannot be less than unity. The same is not true of a shunt feedback equalisation stage such as the one shown in Fig. 3b. Here the voltage gain is equal to $Z_F/R1$, so that as Z_F continues to reduce so the gain continues to drop until finally the minimum gain is determined by the signal leakage through the stage. The accompanying photos show the reproduction of a square wave through the two types of equalisation stage and it will be clearly



The response of the same stage when wired for shunt feedback.

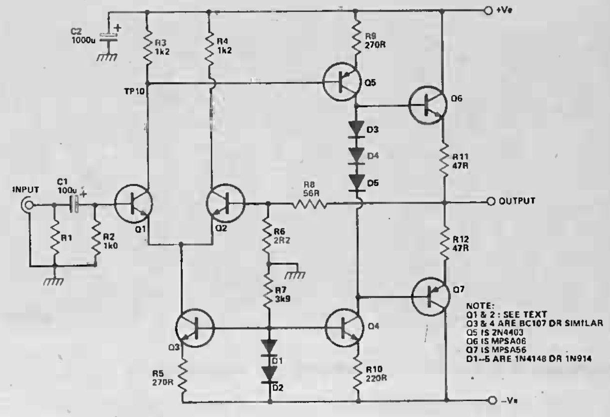


Fig. 6 Circuit diagram of the A-MC moving coil module.

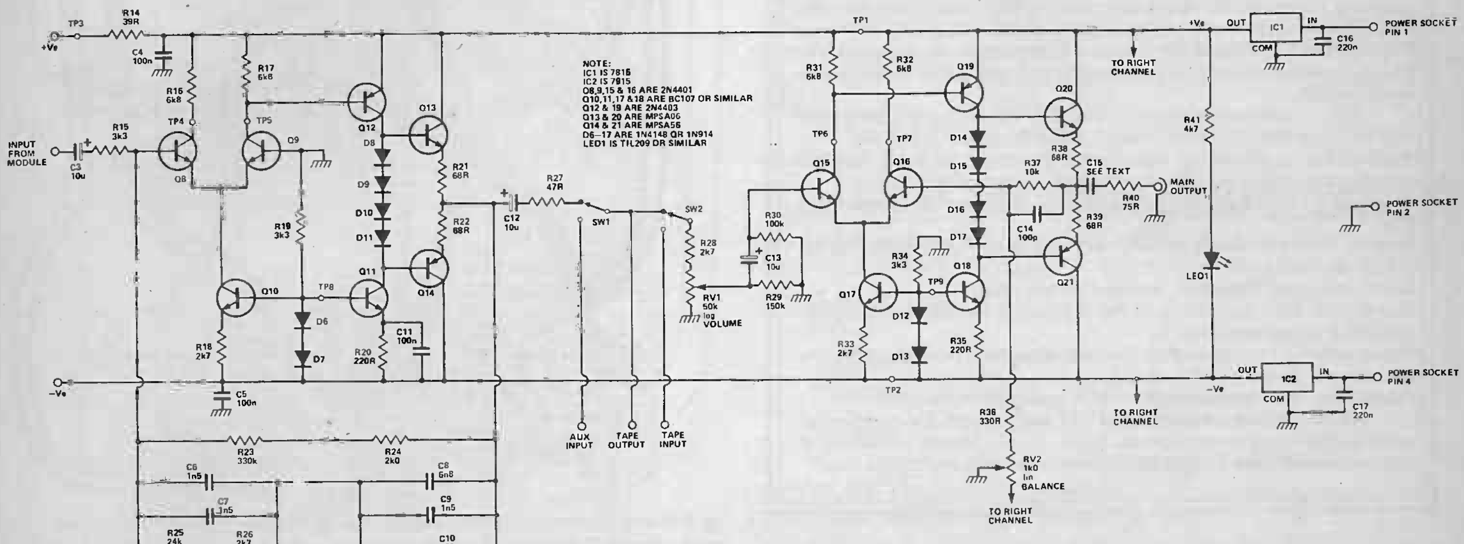


Fig. 7 Circuit diagram of the A-PR main preamplifier module.

HOW IT WORKS

seen that the series feedback arrangement imparts a degree of treble boost to the signal.

So why isn't the shunt feedback system commonly used in commercial preamplifiers? The answer is noise; to be exact, the noise generated by the series input resistor R1. Both input configurations use a nominally 47k resistor to load the cartridge, but in the series arrangement it is 'shorted-out' by the (approximately) 200R resistance of the cartridge. However, with the shunt arrangement this 47k resistor remains in series with the signal path and hence contributes a lot of Johnson (thermal) noise. It has been calculated that the maximum theoretical signal-to-noise ratios of the two stages (measured over the band 20 Hz to 20 kHz and RIAA equalised) are:

Shunt feedback 58.5 dB

Series feedback 72
Both ref. 2 mV at 1 kHZ

This difference is enough, in our world of specmanship, to have consigned the shunt feedback stage to the dustbin for many years.

However, to get the best of both worlds I have gone back to the system I used many years ago at Cambridge Audio. This is the use of a linear series feedback input stage followed by a shunt feedback equalisation stage. The equalisation stage can now work under far easier conditions as the signal has some initial preamplification. Furthermore the input resistor (R1) now no longer needs to be 47k but can be a lower value chosen to set the stage gain. In this case it has been set at 3k3 and so its noise contribution is quite low.

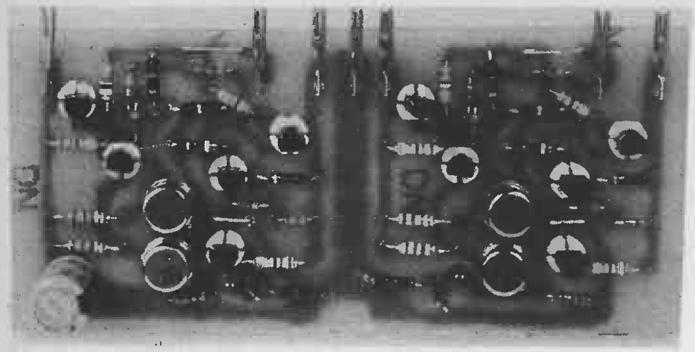
Now we have an input arrangement which buffers the cartridge from the equalisation stage (and so makes the input loading indepenent of the equalisation), continues the RIAA equalisation curve at high frequencies, and achieves the low noise figures typical of the conventional series feedback arrangements. Just as important, the shunt feedback sounds different (and in my opinion better), and that is the deciding factor. A revealing experiment is to wire one preamplifier in shunt and one in series feedback and (having equalised their gains) to listen to each in turn reproducing the 'off-record' noise. It will then be apparent that some preamplifiers emphasise such noises more than others.

Power Supply

The power supply circuitry is kept simple and consists of two integrated circuit regulators (IC1, IC2) which give a low ripple ±15 V supply to the circuits. The positive rail is further decoupled at the pick-up stage by resistor-capacitor filters (R14, C2). The negative rail is adequately decoupled for this stage as the long-tailed pair (Q1, Q2) is fed through a current source, but the positive rail is connected directly to the collectors of this stage and so some additional decoupling is required. The decoupling capacitor needs to be of quite a high value to maintain a low impedance supply. If this value is reduced the low frequency distortion can become excessive.

The supply indicator LED is wired across both supply rails so that the absence of either one will cause the LED to go off.

The power supply module is also simple. The incoming mains supply is fused and switched and fed to a toroidal transformer. The centre-tapped secondary feeds a bridge-rectifier to produce a split rail supply across the two reservoir capacitors (C1, C2). The off-load voltage at this point should be a nominal ±21 V. Again the supply indicator LED is wired across both rails as a monitor.



This photograph shows the A-MC moving coil input module.

BUYLINES

Most of the components specified are readily available from the usual suppliers except for the connectors and the low noise transistors. The board-to-board gold-plated connectors (horizontal, 45°) are type 434-172, and the vertical input-to-board connectors are type 434-188. These are available from RS Components Ltd, and can be ordered via a local stockist.

Kits of parts for the System A amplifier are available from Jelgate Ltd, 215 High Street, Offord Cluny, Cambs. Prices are as follows:
Preamp Kit 1 containing two chassis (preamp and PSU), toroidal transformer, and all the chassis-mounting components; £28.
Preamp Kit 2 containing the A-PR and A-PSU PCBs and all components; £26.
Preamp Kit 3 containing A-MM/A-MC PCB and components; £12 for either version.
Set of four input transistors, selected for low noise; £2.
Power Amp Kit 1 containing all the metalwork, heatsinks and chassis-mounting components; £105.
Power Amp Kit 2 containing transformer, capacitors, power supply components and power transistors; £65.
Power Amp Kit 3 containing A-PA PCB and components; £23.

All these prices are exclusive of VAT and carriage. The cases are all ready-painted and screen-printed. Items can be bought separately; a comprehensive price list can be obtained from Jelgate.

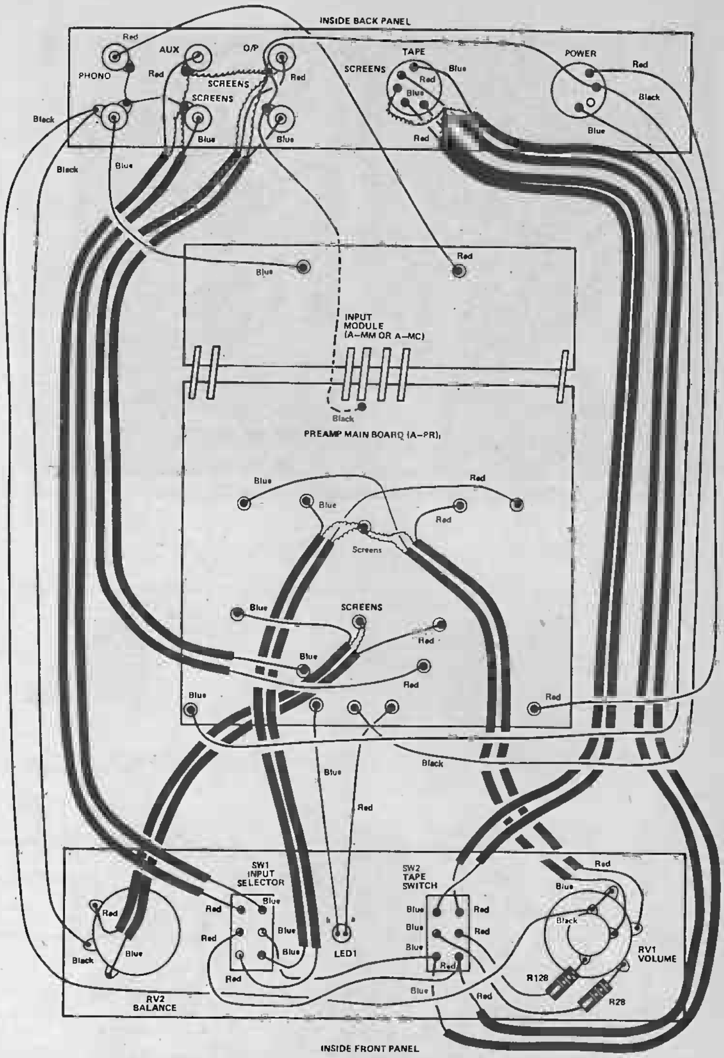


Fig. 8 Wiring diagram of the preamp. No wiring from the preamp main board crosses the input module; all cables are taken towards the front panel and back down either side of the case to the rear panel. See photos.

PARTS LIST

INPUT MODULE A-MM

Components are listed for one channel only — add 100 for other channel.

Resistors (all ¼ W, 5%)

R1	100k (see text)
R2	100k
R3,4	6k8
R5	2k7
R6	560R
R7	3k9
R8	2k2
R10	220R
R11, 12	47R
R13	3k3

R9 is not used

Capacitors

C1	10u 35 V tantalum
C2	1000u 16 V electrolytic (PCB type)

Semiconductors

Q1,2	2N4401
Q3,4	BC107 or similar
Q5	2N4403
Q6	MPSA06
Q7	MPSA56
D1-5	1N4148 or 1N914

Miscellaneous
Connectors, PCB.

INPUT MODULE A-MC

Components are listed for one channel only — add 100 for other channel.

Resistors (all ¼ W, 5% except where stated)

R1	100k (see text)
R2	1k0
R3,4	1k2
R5,9	270R
R6	2R2 2% metal film
R7	3k9
R8	56R
R10	220R
R11, 12	47R

Capacitors

C1	100u 6V3 tantalum
C2	1000u 16 V electrolytic (PCB type)

Semiconductors

Q1,2	BSS15 (specially tested — see text)
Q3,4	BC107 or similar
Q5	2N4403
Q6	MPSA06
Q7	MPSA56
D1-5	1N4148 or 1N914

Miscellaneous
Connectors, PCB.

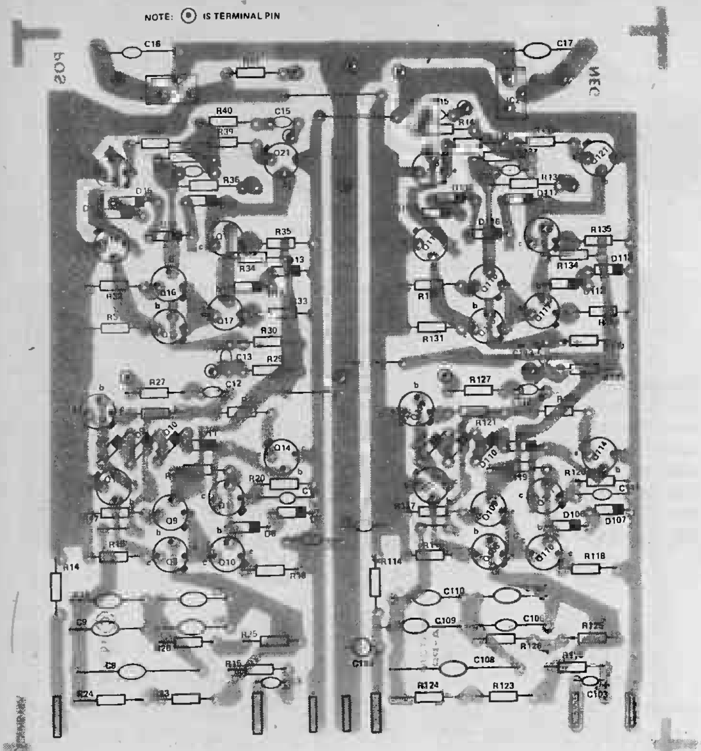


Fig. 9 The A-PR overlay. For off-board connections see Fig. 8.

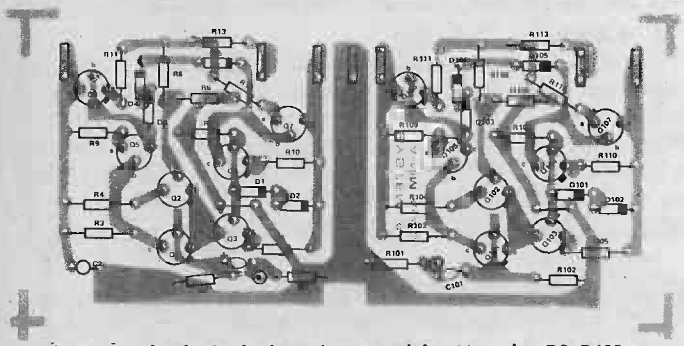


Fig. 10 Overlay for both phono input modules. Note that R9, R109 are replaced by wire links in the A-MM module.

Testing

The power supply should, because of its simplicity, present few difficulties. Before any connection to the mains supply, a visual inspection should be made to check the wiring, the polarities of the capacitors and rectifier, and not least the wiring of the mains switch. It never ceases to amaze me just how often mains switches are wired to short across the supply at switch-on. So take a little care and save a few bob!

With all checks completed, the fuse is fitted and a meter wired between the positive and negative output lines. The mains supply can be connected and for a 240 V nominal supply the meter should read 21 V (±2 V). Then measure the supplies to 0 V to check that they are equal and that the LED is illuminated.

The preamplifier is fairly straightforward to test, albeit rather repetitive. The two power supply regulators are protected against excessive currents (eg shorts) and over-temperature, so they are unlikely to come to any grief providing they are correctly inserted into the PCB. Each of the amplifier stages on the main board can be isolated from the power supplies by the removal of wire links and, of course, the input module can be unplugged, so in the event of a fault the offending stage can be isolated.

Before connecting up the power supply it is a good idea to give the PCBs one final visual check, paying particular attention to transistor types, diode and capacitor polarities, and solder bridges on the PCB tracks. Now connect the power supply and monitor the supply lines. They should measure ±15 V (±0V6) and the LED should light up. The controls should now be set as follows;

Input:	PU
Tape:	OFF
Balance:	Central
Volume:	Minimum

Now measure the DC voltage between earth and the junction of the two emitter resistors in the output stage of each amplifier. This voltage should be zero, but can be ±2 V without any significant effect on the workings of the

PREAMP MODULE A-PR
Components are listed for one channel only — add 100 for other channel.

Resistors (all ¼ W, 5% except where stated)

R14	39R
R15,19,34	3k3
R16,17,31,32	6k8
R18,33,28	2k7
R20,35	220R
R21,22,38,39	68R
R23	330k 2% metal oxide
R24	2k0 2% metal oxide
R25	24k 2% metal oxide
R26	2k7 2% metal oxide
R27	47R
R29	150k
R30	100k
R36	330R
R37	10k
R40	75R
R41	4k7

Potentiometers

RV1	50k logarithmic
RV2	1k0 linear (preferably wirewound)

Capacitors

C3,12,13	10u 35 V tantalum
C4,5,11	100n 63 V ceramic disc
C6,7,9	1n5 2% polystyrene
C8	6n8 2% polystyrene
C10	560p 2% polystyrene
C14	100p ceramic
C15	see text

Semiconductors

IC1	7815
IC2	7915
Q8,9,15,16	2N4401
Q10,11,17,18	BC107 or similar
Q12,19	2N4403
Q13,20	MPSA06
Q14,21	MPSA56
D6-17	1N4148 or 1N914
LED1	TIL209 or similar

Miscellaneous

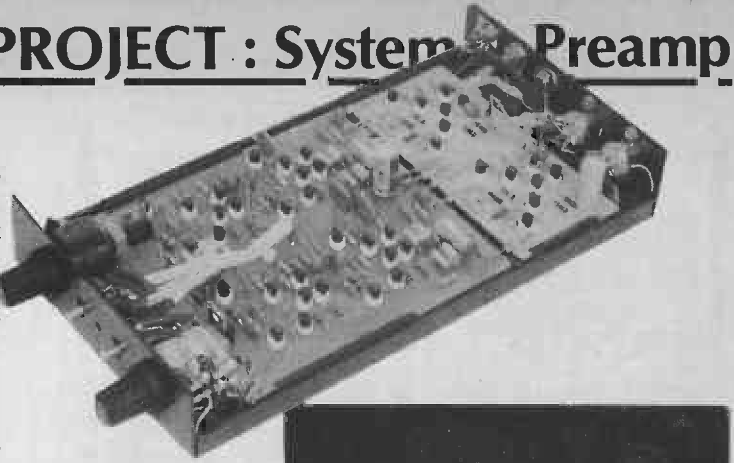
SW1,2	DPDT slide switch

Connectors, PCB, phono sockets, DIN sockets, Veropins, screened cable, case, knobs to suit.

preamplifier (although the blocking capacitor will be necessary). That completes the DC tests. The preamplifier will now almost certainly work but if you have test equipment available it would be a good idea to test each channel with an audio signal and to centralise the balance control.

The total current drawn from the negative supply is about 120 mA for the moving-coil version and 115 mA for the moving-magnet version; and about 15 mA less from the positive supply.

As an aid to fault-finding a list of test-voltages has been provided which can be used in conjunction with the main circuit diagram.



The cases supplied for the System A project ensure a professional appearance, and can be obtained from Jelgate Ltd (see Buylines). The modules shown here are the prototypes.

Table 2. Voltages measured between test points and ground with Avometer Model 8. These voltages should be taken only as a guide.

TP1	+15V	TP6	+14V3
TP2	−15V	TP7	+13V6
TP3	+14V5	TP8	−13V8
TP4	+13V6	TP9	−13V8
TP5	+14V3	TP10	+13V

ADC	
ZLM	E
XLM-II	G
XLM-III	E
VLM-II	E
AKG	
P7E	G
P8E &	
P8E-S	E
AUDIO-TECHNICA	
AT-10	F
AT-11E	F
AT-12E	F
AT-13EA	E
AT-25	B
AT-30	C
Signet MkIIIE	C
Signet TK5E	F
Signet TK7E	E
AZAK	
DC2100K	C
CORAL	
MC81	C
777E	C
777Ex	C
DECCA	
Blue	E
London	F
DENNON	
DL103C	A
DL103S	A
DL103D	A
ELITE	
MC555	B
EE1500	G
EMPIRE	
500 D	E
2000 1E	G
2000 E4	G
600 LAC	E
2000 X	E
EMT	
XSD15	A
GOLDRING	
G 900 IGC	E
MAYWARE	
MC3L	E
MC2C	C
MICRO ACOUSTICS	
2002-e	E
NAGAOKA	
JT-R11	B
ORTOFON	
MC 30	C
SHURE	
75-ED	H
M75EJ	H
M97HE	H
V15-IV	G
SONUS	
BLUE	E
GOLD BLUE	E
SONY	
XL35	E
XL 55	A
STANTON	
680	E
881	E
681 EEE	E
SUPEX	
SD 9015	E
900E	C
TECHNICS	
EPC-300MC	D
ULTIMO	
10X II	E
20A	G
DV 20C	A
DV KARAT	A

Table 3. Cartridge matching table.

Variations On A Theme

Alterations can be made to the input modules to suit a wide range of cartridges. The recommended changes are given below; Table 3 lists most cartridges and the matching module. If your cartridge doesn't appear, write to us with an SAE and we will tell you which variant is suitable.

Moving-coil Cartridges

The gain of the A-MC input module can be varied by changing resistor R6. This resistor has a value of 2R2 to give a sensitivity of 550 uV on the standard version. Changing R6 to 0R6 (eg two 1R2 resistors in parallel) will increase the sensitivity to about 150 uV. The input loading can be varied by changing resistor R1 fom the standard value of 100R to any other value. The four recommended alternatives are:

A	550 uV sensitivity,	R1 = 1k0
B	150 uV sensitivity,	R1 = 100R
C	550 uV sensitivity,	R1 = 100R
D	150 uV sensitivity,	R1 = 1k0

Moving-magnet Cartridges

Again, the input loading of module A-MM can be changed by using an alternative value for resistor R1. An input capacitor C_L can also be wired across R1 to lower the input impedance at high frequencies and so 'equalise' the output from some cartridges. The gain of the standard version is set by R13 and gives a sensitivity of 2.3 mV. Reducing R13 increases the sensitivity and vice-versa. The four recommended alternatives are:

E	Standard version
F	R13 = 8k2
G	C_L = 180pF
H	R13 = 8k2 and C_L = 180pF

Table 3 assumes that the cartridges are mounted in tone-arms which have a total cable capacitance of about 100pF and below.

Next month we present the System A Class A power amplifier. This article will also include the overlay for the A-PSU module which had not been finalised when this issue went to press.

ETI

MICROBASICS

Beginning in BASIC but hard up for cash? The new ZX81 from Sinclair brings you home computing for less than £50. Peter Freebrey has been running his fingers over its keyboard.

I'm sure that most people interested in electronics have some interest in computers. Not only are we bombarded on all sides with their applications, but advertisements for many types of microcomputer appear in most electronic magazines. We see regular articles on what makes them 'tick' and their smaller relations pop up regularly in such projects as musical doorbells and model train controllers.

The elementary articles on how a computer works are usually fairly easily understood by the electronics enthusiast. He probably understands AND, OR, NAND and NOR gates, he can build flip-flops and appreciates the significance of timing and clock pulses. He should also have a basic understanding of the binary system of counting; he might not be able to tell you straight off what 27 is in binary 1s and 0s but he can pretty quickly work it out!

So far, so good — but what about the next step? "What does a computer do?" — I have heard this plaintive cry so often from someone new to computing. He already knows the simple answer in the back of his mind — it adds and subtracts numbers according to a preset system. He even knows that this preset system is called a program. What he is really asking is something like this . . . "I know a computer can add and subtract numbers very quickly, but how do I make it *do* something to my advantage? I know it can be useful in working out my pay packet but what else can it *do* and *how*?"

Language Barrier

Some are lucky and have a friend with a talent for explaining this surprisingly difficult hurdle. Some are not so lucky and feel they have to accept that, although they could probably make a very simple calculating machine/computer, they wouldn't know what the heck to do with it when it's finished.

This type of problem *can* be solved quite simply — but you need access to a computer. The 'hands on' approach is by far the quickest, surest and most painless way to learn what a program is and what it can do. What you will be learning first is how to communicate with the computer: how to tell it what to do and when to do it. In effect you will have to learn a new language. Do I hear groans from those who only scraped through French, Latin or German at school? Well — I refuse to believe that any reader of this magazine is unable to learn 50 or 60 new words. Even 20 to 30 will enable you to write some pretty complex programs.

Down To BASICs

There are many arguments raging as to what computer 'language' should be used for different purposes. Although the BASIC language has its opponents, most of the readily available microcomputers use some variant of BASIC. It is easy to understand and many of the keywords (commands) are self-explanatory. PRINT means just that — print something onto the screen; SIN X is the sine of the angle X; GOTO X is go to the program line X and carry on from there. There are a number of less obvious keywords. Some are just abbreviations of words . . . INT (X) is the integer part of number X; SQR X is the square root of the number X.

How you then use all these commands depends on what you want the computer to do. To get the computer to do something other than work as a simple calculator you have to *program* it. This means you must give it a set of step-by-step commands which it will store in its memory, so that when you tell it to RUN its program it will faithfully follow the steps you have told it to take. This is where the 'hands on' experience counts. You can practice by writing simple programs — perhaps at first just getting the computer to print on its screen all the square roots for numbers 1 to 100. That doesn't seem too useful, does it? But, as you learn how to write simple programs you will also familiarise yourself with what is happening and what a computer can do — this is only the beginning.

One of the reasons that many of us have not been able to get this 'hands on' experience is simply lack of money. Until you try you don't know whether a home computer is going to be a useful tool, a cunning games opponent or just an interesting toy. So it is not surprising that the slightly baffled beginner is loath to invest perhaps several hundred pounds to find out. Even if you have the opportunity of 'playing' with one at an exhibition or maybe at a friend's house, you rarely have enough time to fully or even partially appreciate the possibilities of what it can do. The Sinclair ZX80 changed that and the ZX81 will take this change even further; for less than £70 (£50 if you build the kit) you can get that precious 'hands on' experience.

Eighty-Oneupmanship

Sinclair Research Ltd of Cambridge have for many years tantalised the electronics fraternity with a continuing selection of 'state-of-the-art' goodies. In recent years Sinclair seem to have been riding on the crest of the technological wave, offering tiny calculators, watches, televisions and now computers. Often these are launched at prices few others can match, together with a specification that must have other manufacturers shaking in their shoes and thinking of industrial 'bugs'. The ZX80 came to us at a price that enabled many people to own their own personal computer. They are now happily writing and running programs that a year ago they would never have believed possible. It would seem that the ZX80 also had an appeal to those who in the course of their working hours use computers extensively. These people, although interested in understanding the workings of computers, have in the past not felt it worth the financial commitment to own their own personal computer. The ZX80 changed that. Now we are offered the ZX81, cheaper still than the ZX80 *and* with an improved specification.

Let me say right away that the ZX81 represents incredible value for money and although it is a budget machine, the facilities it offers may be compared favourably with other microcomputers costing many times as much. It has a smaller memory than most others, but with the 1K quoted a great deal can be achieved and should you require more in the future, £50 will buy you a 16K add-on memory pack. This would be quite enough for most needs and still makes the total package cheaper than its competitors.

Touching Moment

The ZX81 does all that it is claimed to do but to keep it at this price it does not use the normal typewriter keyboard; instead it has a calculator 'touch type' keyboard. If you are new to using a computer this should present no real problem as you are probably used to the key spacing of a pocket calculator and the ZX81 will not seem all that different. To conserve space and cost, many of the keys are multi-function — some giving up to six different results! This is a bit of a shock at first, but you soon adapt to it.

The ZX81 comes complete with a 212-page instruction manual. The main bulk of this tome is for the novice, explaining clearly and concisely all the statements and commands at your disposal. Most of the chapters are essentially a 'hands on' teaching program in ZX81 BASIC, with a number of useful exercises to extend the knowledge gained in the preceding text. Chapters 26, 27 and 28 are for the more advanced reader, introducing the concepts of machine code, the organisation of the memory and explaining what and where are the system variables. There are three appendices: Appendix A lists the character set (all the characters that may be displayed on the screen) together with the numerical and hexadecimal code that the ZX81 assigns to each character. Appendix B lists the Report (Error) codes; if you tell the ZX81 via a program instruction to do something that it cannot do, a letter or a number is displayed on the screen to inform you what the error is and at what program line number it occurred. Appendix C gives a short résumé of the ZX81 and its individual characteristics for those who already understand BASIC.

The Index is useful and comprehensive, not only giving page references but also the key sequences required for all keywords and characters.

The ZX81 is much smarter in appearance than its predecessor, the ZX80. Although the basic computer contains only 1K of memory, the small add-on memory pack shown here simply plugs into the back to provide an additional 16K.

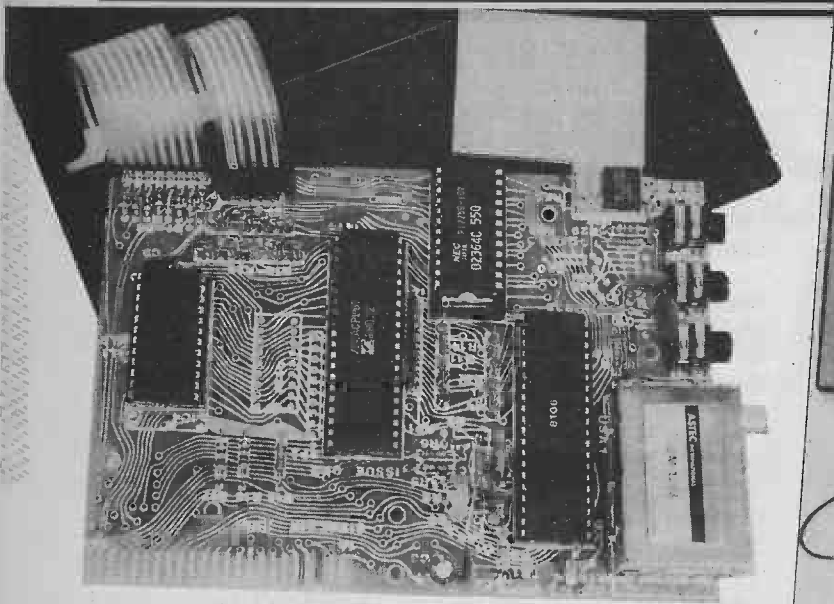
My Enemy Flicker

The ZX80 was criticised for its screen 'flicker' every time a key was depressed and although one got used to this idiosyncrasy it remained slightly disturbing. The ZX81 has two modes of operation: SLOW and FAST. In SLOW mode this screen flicker has been eliminated. As the name implies, operation of a program takes longer in SLOW mode, and it is recommended that FAST mode be used when a long calculation is being performed where reference to the screen is not needed until a result is to be displayed. These two commands may be written into a program so that you may choose when to speed up or slow down the operation of the computer, depending on the display requirements. The ZX81 can be linked to a cassette recorder to store any programs that you have written. Unlike its predecessor, the ZX81 will store a 'named' program and when subsequently commanded to LOAD a 'named' program from tape into its memory will search through a tape full of programs until it finds the specific program requested.

The ZX81 uses a 'one touch' keyword entry technique. When you have entered a program line number the next requirement *must* be a command of some description. The ZX81 recognises this necessity and the next single key pressed will result in the related command being entered and appearing on the screen in front of you. If you have entered a line number and then press key 'P' you will actually enter PRINT. Press 'G' instead of 'P' and you get GOTO. Many keys are multi-function, some capable of generating up to three keywords, a letter, that letter reversed and also a graphics character!

Point Made

Whereas the ZX80 was limited to integer calculations, the ZX81 has a fully floating point notation and this has meant the addition to its repertoire of such as logs and trig functions. Together with its ability to PLOT pixels (quarter square graphics), this means that you can draw sine waves (among other things!) across your TV screen to your heart's content!



Another advantage of the ZX81 is the option of adding a printer to your system. The ZX printer (available summer/autumn) offers full alphanumerics across 32 columns. It also has the facility to COPY, which will print out exactly what is displayed on the TV screen at any given time.

All in all, I can only recommend the ZX81 to any intending purchaser — it is very good value for money. It does not have the facilities one would require for a business machine, but anyone wanting to learn *personally* what a computer can do, without initially spending a fair amount of money, should seriously consider one. After all, the ZX81 does not cost much more than the cassette recorder that you would need to buy for some other makes of home computer.

All the necessary leads are supplied, but the power supply is only available separately from the kit (above).

The ZX81 without its clothes on (top left). Kit builders need have no worries about complexity — there are only four ICs and a handful of other components. The spare IC position is for an alternative RAM chip. The touch keypad isn't part of the main PCB and if you're clever you might be able to replace it with a conventional one.

SYSTEM/MULTIFUNCTION COMMANDS	
BREAK	Halts run and returns to command mode.
CLEAR	Deletes all variables, freeing the space they occupied.
CLS	Clears screen — clears display file.
CONT	Continues STOPped program.
COPY	Sends copy of display to printer.
EDIT	Returns current line (indicated in program list) to bottom of screen for editing.
FAST	Starts fast mode. Display file is displayed only at end of program, while INPUT data is being typed in or during PAUSE.
FUNCTION	Returns alternative keyword set.
GRAPHICS	Returns alternative character set.
LIST	Lists specified line(s) of program on screen.
LLIST	Like LIST but using printer instead of screen.
LOAD	Looks for specified program on tape — loads it and its variables.
LPRINT	Like PRINT but using printer instead of screen.
NEW	Clears memory and awaits new program.
NEWLINE	Enters command as statement.
PAUSE	Stops computing and outputs the display file to the screen for specified time, or until another key is pressed.
RUBOUT	Deletes character to left of cursor.
RUN	Executes current program.
SAVE	Writes specified program to cassette interface. Records program and variables.
SLOW	Puts computer into compute and display mode, in which the display file is displayed continuously.
STOP	Stops a program that is RUNning.
STATEMENTS	
ABS	Returns absolute value of specified variable.
ACS	Returns arcosine (in radians).
AND,OR,NOT	Comparative tests.
ASN	Returns arcsine (in radians).
AT	Defines position of next PRINT statement (in screen lines/columns).
ATN	Returns arctangent (in radians).
CHR$	Gives character whose code is specified.
CODE	Gives code of the first character in specified string.
COS	Returns cosine of angle (in radians).
DIM	Dimensions array size.
EXP	Returns exponential of specified number.
FOR	Used in conjunction with TO and NEXT to execute a defined loop.
GOSUB	Jumps to defined subroutine.
GOTO	Jumps to specified program line number.
IF	Conditional test, used in conjunction with THEN followed by specified statement.
INKEY$	Reads keyboard, result is character of next key pressed.
INPUT	Assigns value of keyboard entry to specified variable.
INT	Returns integer part of number (rounded down).
LEN	Returns length of specified string.
LET	Assigns specified value to specified variable.
LN	Returns natural logarithm of a number.
PEEK	Returns value of specified memory location.
PI	π (3.14159265. . . .).
PLOT	Blacks in pixel at specified co-ordinates.
POKE	Places assigned value in specified memory location.
REM	No effect on program, allows inclusion of text for comments.
RETURN	Returns to main program from subroutine.
RND	Returns random number, ≥ 0 and <1.
SCROLL	Scrolls the display file up one line.
SGN	Returns sign of number.
SIN	Returns sine of angle (in radians).
SQR	Gives square root of number.
STEP	Used with FOR. . . NEXT loops, defining increment between loops.
STR$	Returns string representation of number.
TAB	Defines column in which PRINT statement will begin.
TAN	Returns tangent of angle (in radians).
UNPLOT	Like PLOT, but blanks out a pixel instead of blacking in.
USR	Calls machine code subroutine.
VAL	Evaluates a string as a numerical expression.

ETI

DESIGNER'S NOTEBOOK

Piezo-electric 'buzzers' such as the PB-2720 are super-efficient and inexpensive sound generators, easily driven by simple CMOS circuitry. In this month's Notebook, Ray Marston shows how to use them.

There is a frequent requirement in instrumentation designs, for example, for some form of alarm or 'fault condition' indicator, perhaps to warn of a short-circuit or overload condition in a power supply or an overspeed condition, loss of oil pressure and so on in a car or truck. If you ever need to design such an alarm, you have the options of using either a visual (lamp or LED) or an acoustic type of output indicator.

The major snag with purely visual indicators is that they are only effective if you happen to be looking at them when they activate. Clearly, acoustic indicators are the most effective types of 'attention grabbers', but in the past they tended to be rather expensive to implement both in terms of money and in power consumption and physical bulk.

The recent introduction of small, inexpensive and highly efficient piezo-electric acoustic transducers such as the Toko PB-2720 (available from Ambit International at about 44p each) has totally changed this situation, however, and it is now possible to build effective acoustic indicators at costs that are very low.

PB-2720 Basics

The PB-2720 piezo-electric transducer is a super-efficient electric-to-acoustic power converter. It consists of a metal plate bonded to a thin slice of piezo-electric ceramic and is housed in a small plastic-moulded resonant chamber.

If you apply an AC signal across the two input terminals of the PB-2720, you get a corresponding audible output. Figure 1 shows the frequency characteristics of the device when it is fed with a 1V5 RMS input and the output level is measured at a range of 10 cm. Note that a good output level is available across a wide frequency band but this peaks at about 4.5 kHz, at which point an output sound level of roughly 85 dB is obtained at a range of 10 cm from a 1V5 RMS input. If you are not familiar with acoustic terminology, 85 dB is typical of the subjective sound level of a noisy office or busy street.

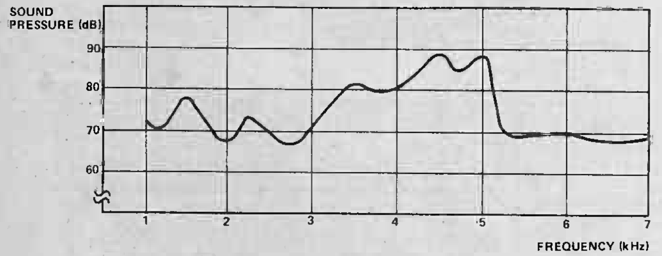


Fig. 1 Frequency characteristics of the PB-2720 'buzzer' with an input of 1.5 V RMS. The sound pressure is measured at 10 cm.

The really impressive feature of the PB-2720 is its high level of power conversion efficiency and consequent low power input requirement for a given power output. Figure 2 shows the input voltage characteristics of the device in terms of current consumption and generated sound pressure. Note here, for example, that a 10 V RMS input at 4.8 kHz causes a current consumption of only 3 mA but results in 100 dB of output, while at 1.65 kHz the input consumes only 1 mA (10 mW) for 87 dB of output. Very impressive.

The explanation for these apparently miraculously low levels of power consumption is very simple. Conventional electromagnetic speaker-type transducers have incredibly low conversion efficiency levels, ranging from a mere 0.1% for hi-fi speakers to 2% for 'cheapo' types. The PB-2720, by contrast, is a piezo-electric device and has an efficiency level of about 50%. Thus, for a given output level it needs an input power of only 1/500th to 1/25th of conventional sound generators.

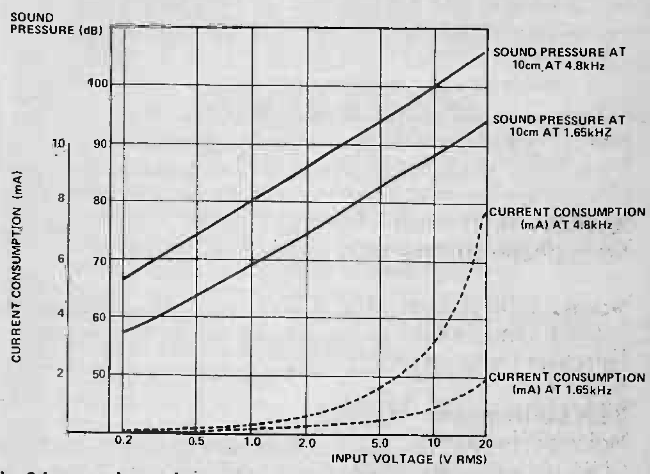


Fig. 2 Input voltage characteristics of the PB-2720 in terms of current consumption and generated sound pressure.

Driving The PB-2720

The PB-2720 is a very easy device to drive. Being ceramic, its input terminals appear to the outside world as a simple capacitor with a static value of about 20nF and a DC resistance of near-infinity: if you drive it with a pure sine wave, you simply find that its impedance decreases as frequency increases.

The most effective and cheapest way to drive the device is to feed it with square waves, but in this case the driver must be able to source and sink currents with equal ease and must have a current-limited (short-circuit proof) output. CMOS drivers fit this bill perfectly.

Figures 3 and 4 show two very inexpensive ways of driving the PB-2720 from a gated 4011B CMOS oscillator; both circuits generate a continuous-tone signal when they are enabled, are gated on by a high (logic 1) input signal, and can use any supply in the range 3 to 18 V.

The Fig. 3 circuit calls for little explanation. IC1a-IC1b are wired as a gated 2 kHz astable, and IC1c is used to give single-ended buffered drive to the PB-2720. The circuit can be gated on electronically, or by PB1. The signal reaching the PB-2720 is thus an approximate square wave with a peak-to-peak amplitude roughly equal to the supply voltage: consequently, the RMS voltage across the load is roughly equal to 50% of the supply voltage.

The Fig. 4 circuit is rather more difficult to understand. IC1c and IC1d are series-connected and used to give a 'bridge' drive to the transducer, in which antiphase signals are fed to the two sides of the PB-2720. The consequence of this cunning drive technique is that the load (the PB-2720) actually sees a square wave drive voltage that has a peak-to-peak value equal to twice the supply voltage and thus gives four times more acoustic power than the Fig. 3 circuit. The effective RMS voltage across the load of the Fig. 4 circuit is equal to the supply voltage. Mystified?

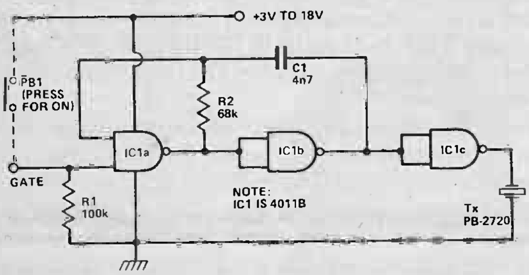


Fig. 3 This basic buzzer circuit is gated by a high (logic 1) input and generates a 2 kHz continuous tone. The PB-2720 drive is single-ended. Sound output (at 10 cm) is about 82 dB from a 10 V supply.

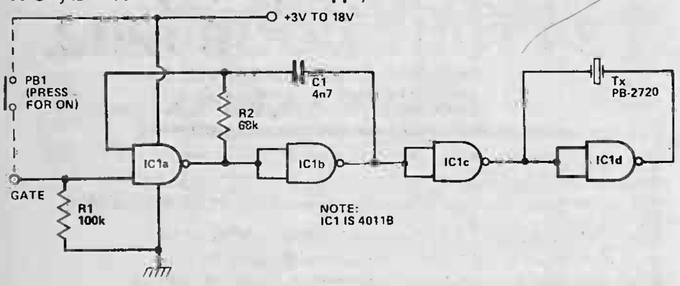


Fig. 4 This version of the basic buzzer circuit uses bridge drive to the PB-2720 and produces an output that is four times louder than the Fig. 3 circuit.

Points Of View

The solution to the action of the bridge-driven circuit of Fig. 4 can be understood with the aid of Fig. 5, which shows the waveforms applied to the load from a bridge circuit when it is fed with a 10 V peak-to-peak square wave input signal. The important thing to grasp when looking at this diagram is the basic concept of reference points. You and I are accustomed to thinking in terms of the common or ground line as being the 'zero voltage' reference point. Thus, when we look at point A in Fig. 6 we see a square wave signal that alternates between 0 V and +10 V. Similarly, when we look at point B we again see a 10 V peak-to-peak signal, but in this case it is in antiphase to the A signal (shifted by 180°).

Now the load in the Fig. 5 circuit (irrespective of whether it is a simple resistor or a PB-2720) sees drive voltages purely with reference to one arbitary side of itself. With this concept in mind, let's look at the drive voltage as seen by the load (the third waveform, the true voltage *across* the load), which assumes that the load is always seeing point A as its 'zero reference' point.

In this case, during period '1' of the drive signal, point B is 10 V positive to point A and is thus seen as being at '+10 V'. In period '2', point B is 10 V negative to point A, and is thus seen as being at '−10 V'. Similarly, through periods '3' to '6' point B is seen as alternating through +10 V, −10 V, +10 V and −10 V.

Thus the load in a 10 V bridge-driven circuit sees a voltage of 20 V peak-to-peak, or twice the single-ended input voltage. Since doubling the drive voltage results in a doubling of the drive current, and power is equal to the V.I product, the bridge-driven circuit will produce four times more power than the single-ended circuit. If you don't believe it, check it with a 'scope, but don't forget to reference your 'common' terminal to one side of the load.

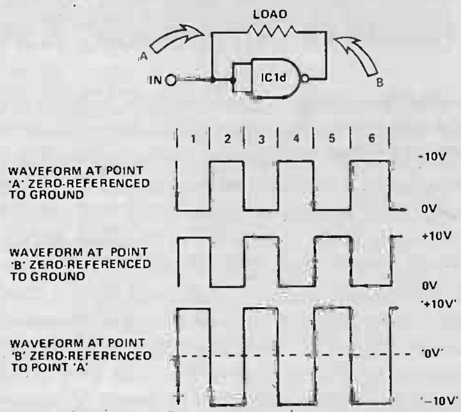
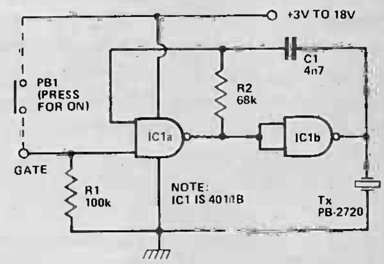


Fig. 5 Waveforms applied to the load from a bridge circuit when it is fed with a 10 V peak-to-peak square wave input signal. Note that the first two waveforms are zero-referenced to ground, but the third waveform is zero-referenced to point A.

Sound Practice

Gated CMOS oscillators/drivers can be used in a variety of ways to produce useful alarm sounds from the PB-2720. A few variations are shown in Figs. 6 to 9. If you are not bothered about waveform degradation and need to use the minimum possible number of gates, you can, for example, drive the PB-2720 directly from the output of the CMOS astable, as shown in Fig. 6. Alternatively, if you want the alarm to be gated on by a low (logic 0) input, simply substitute a 4001B for the 4011B, as shown in the bridge-driven circuit of Fig. 7.

Fig. 6 Direct-output version of the gated 2 kHz buzzer circuit.



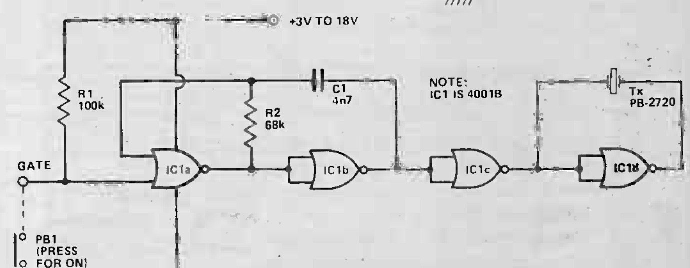


Fig. 7 A continuous-tone 2 kHz buzzer with bridge drive, gated on by a low (logic 0) input.

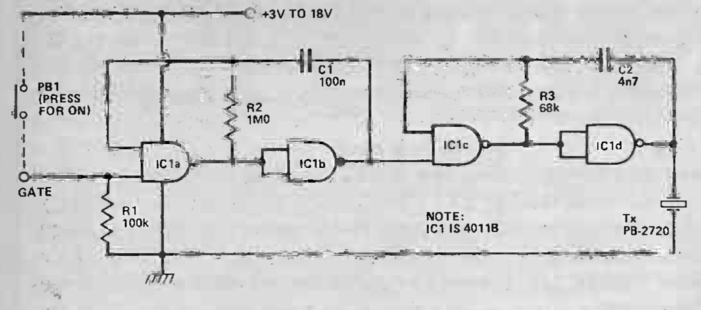


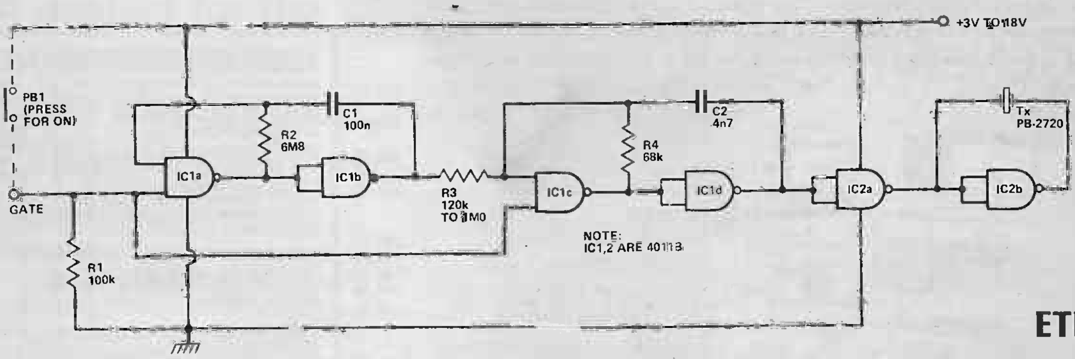
Fig. 8 A gated pulsed-tone alarm, gated by a high input, with direct-drive output.

Figure 8 shows how you can use a single 4011B to make a pulsed-tone (bleep-bleep) alarm circuit with direct drive to the PB-2720. Here, IC1a-IC1b are wired as a gated 6 Hz astable which is used to gate the IC1c-IC1d 2 kHz astable on and off. The circuit is gated on by a high input; if you want low-input gating, simply swap the 4011B for a 4001B and transpose the positions of PB1 and R1.

Figure 9 shows a warble-tone version of the gated alarm. Here, low-frequency astable IC1a-IC1b is used to modulate the frequency of the IC1c-IC1d astable; the depth of frequency modulation depends on the value used for R3.

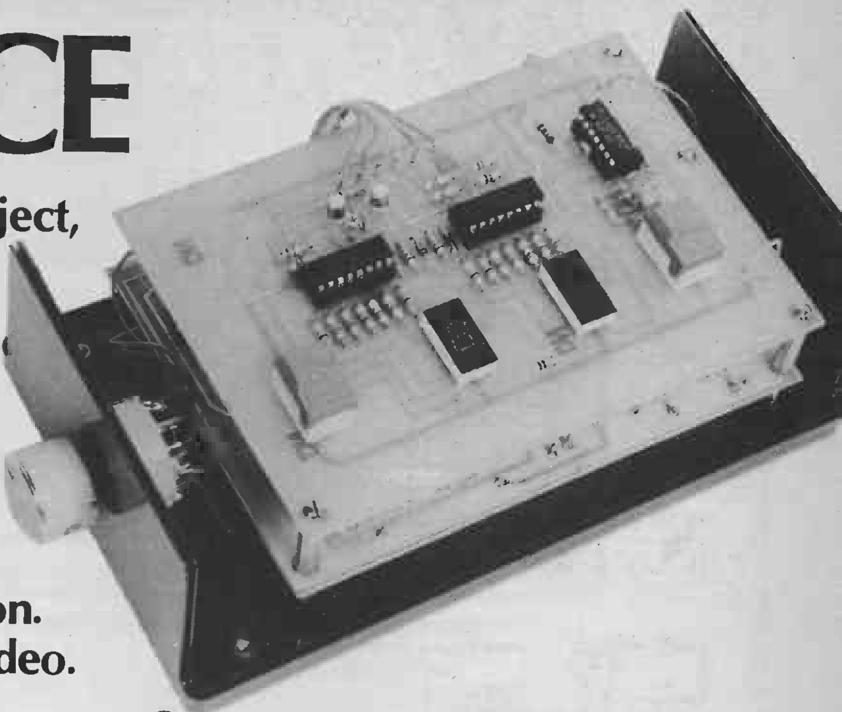
There are plenty of other gated CMOS generator circuits that can be used to drive the PB-2720. The generators can be gated by a wide variety of sensor circuits, so that the alarms are automatically activated by excesses of light, temperature, voltage or current, and so on; lots of suitable circuits can be found in past issues of ETI.

Fig. 9 This gated warble-tone alarm sounds like a British police car siren (dee-dah) and has a bridge-driven output.



ETI

SUPERDICE



The ultimate electronic dice project, this unit gives a digital readout in terms of a single die (1-6), the sum of two dice (2-12), and dice or cards with ranges of 0-9, 1-20, 1-36 or any other range that you care to choose. A project for the dedicated fantasy or war-games enthusiast. Design by Ray Marston. Development by Steve Ramsahadeo.

If you are into long games of chance and/or skill, such as fantasy- or war-games, in which vast quantities of random numbers are selected by rolling dice or pulling cards, you'll be fully aware of the disadvantages of the conventional dice and card systems. Both systems are time-consuming and tedious to use, errors can be made in summing the die numbers, and disputes about numbers can arise if the dice or cards are replaced before they've been seen by all other players.

The ETI Superdice project is designed to overcome all these problems. It's a digital readout unit that, at the touch of a button, generates an output to represent a random number of a single die (1-6), or the correctly weighted sum of two dice (2-12), special dice or cards with ranges of 0-9, 1-20, 1-36 or (with slight circuit modifications) any other number range that you care to choose. The circuit is battery-powered and has all sorts of neat features, such as leading-zero suppression and time-controlled auto-blanking to conserve power, and spin blanking and number recall to enhance the game action and avoid play disputes.

In use, the number range is first selected by SW1 and a random number in that range is then called up by pressing SPIN button PB1: the display blanks while PB1 is pressed, but displays the selected number as soon as PB1 is released. If PB1 is not used again by the end of a 5 s (approximately) period, the display automatically blanks again, to conserve power. If any dispute arises concerning the last number that was displayed, it can be re-displayed by pressing RECALL button PB2.

The most important feature of our Superdice project is its ability to display the equivalent of the sum of two independently generated die numbers, with correctly weighted odds in all cases. The probability of generating these 2-12 numbers is shown below:

2 : 1 in 36	6 : 1 in 7.2	10 : 1 in 12
3 : 1 in 18	7 : 1 in 6	11 : 1 in 18
4 : 1 in 12	8 : 1 in 7.2	12 : 1 in 36
5 : 1 in 9	9 : 1 in 9	

In our circuit, we generate these numbers by using two independently-clocked 1-6 random-number generators and then summing their total before feeding the resulting number to the display system: hence the apparent complexity of our circuit.

Construction

The Superdice project is constructed on a pair of stacked PCBs. The two seven-segment LED displays and the SPIN and RECALL buttons are mounted, together with IC11, IC12, IC13 and associated components, on the upper PCB, which is a simple single-sided board. All of the remaining circuitry (IC1 to IC10) except SW1 and SW2 is mounted on the lower PCB, which is a double-sided affair and uses a large number of Veropins for the through-board connections.

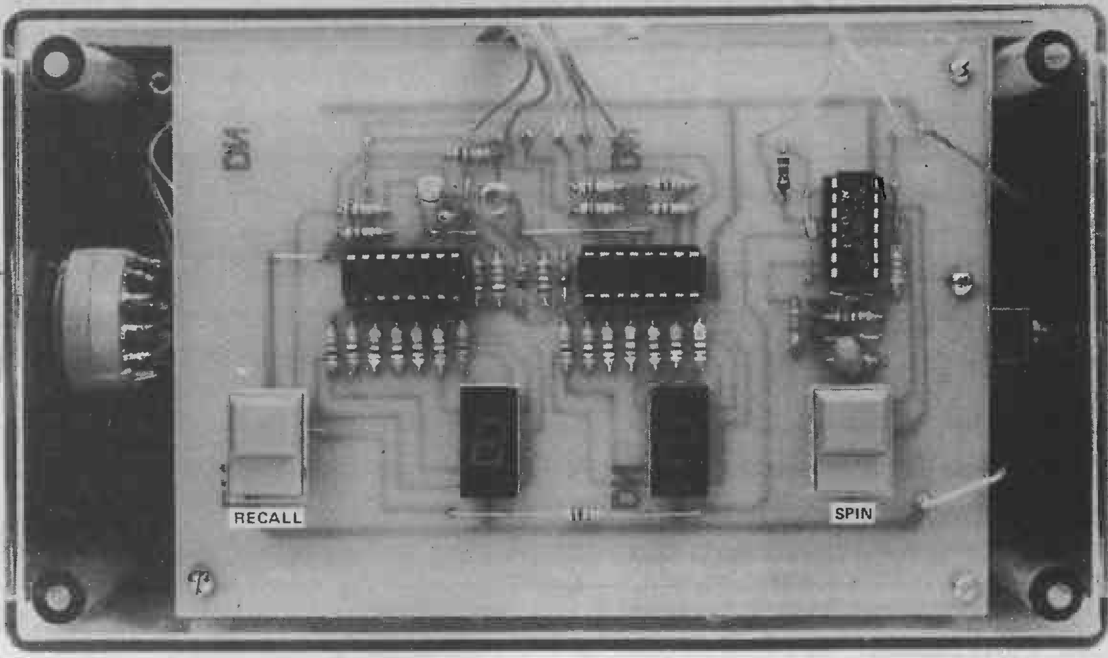
The actual construction of the unit should present few problems. If you are not capable of producing the PCBs yourself (the double-sided one may be a problem), you can buy ready-etched boards for this project (see Buylines).

Start the construction by building the single-sided display board. You can test the completed board by simply connecting it to a 6 V supply, pressing PB1 and checking that the units indicator displays a '0' and the tens display is blanked once PB1 is released, that the display auto-blanks after roughly 5 s, and that the display can be recalled using PB2.

Next, proceed with the construction on the double-sided PCB, taking special care to ensure that the two sides are joined at all the indicated points by Veropins pushed through the board and soldered on both sides of the tracks. When construction is complete, fit the ICs into place in their sockets IN NUMERICAL SEQUENCE, starting with IC1 and ending with IC10.

At this stage you can temporarily interconnect the two PCBs and the two ganged halves of RANGE switch SW1, and then give the unit a full functional check. Check that the unit gives the ranges already described in the introduction. Carry out a long check on the 2-12 (dice) range and ensure, by recording the results of a couple of hundred 'spins', that the odds approximately tally with the list already given, with '7' being the most probable result of a spin and a '2' or '12' the least likely.

Finally, you can complete the construction by bolting the two PCBs together using suitable spacers and fitting the assembly into a suitable case, together with SW1 and SW2 and the battery pack. On our prototype we used 30 mm spacers to fix the assembly to the case bottom and give adequate clearance for the battery pack, and 22 mm spacers between the PCBs to give the correct height to the upper board so that PB1 and PB2 pass through holes cut in the case top.



The case we used (see Buylines) is supplied with a transparent top, so the only holes you need to cut are the ones for the push-buttons.

Odd Odds

The 'spin' ranges of this project can easily be altered to give ranges other than those already described, provided that the range starts with a 0 or 1 and ends with a value below 99, and is required to represent a single die or card. On these ranges, SW1b must not be connected to IC10. The procedure for selecting a range is as follows.

The low end of the range (0 or 1) is determined by SW1b. The presence of a connection between SW1b and the input of IC9c causes the range to start with a '0'; no connection causes the range to start with a '1'.

The top end of the range is determined by ANDing the decoded outputs of IC4 and IC3 with one of the IC5 gates so that they feed appropriate reset or preset-enable pulses to IC1-IC2. The decoded outputs of IC4-IC3 must have a value that is equal to the desired highest number plus one. Thus, to give a top range of 64, the decoded '6' of IC4 must be ANDed with the decoded '5' of IC3. The decoder connections of IC3 and IC4 are as follows:

'0' = pin 3	'4' = pin 1	'8' = pin 9
'1' = pin 14	'5' = pin 6	'9' = pin 5
'2' = pin 2	'6' = pin 7	
'3' = pin 15	'7' = pin 4	

HOW IT WORKS

The basic operating principle of the circuit is fairly simple. Whenever SPIN button PB1 is pressed, two restricted-range BCD counters (IC1-IC2 and IC8) are independently clocked (by gated astables IC11b and IC11a respectively) at a fast rate for the duration of the PB1 closure. Since the clock rates are high, it is not possible to predict the number of clock pulses that will be fed to the counters during the period of the manual switch closure, so the resulting generated BCD numbers are 'random'.

When the unit is used in the dice (2-12) mode, these counters are both set to the 1-6 counting mode and their outputs are summed (via the closed 'switches' of IC10) in BCD-adders IC6-IC7. The results of the additions are presented to seven-segment displays DISP1 and 2 via drivers IC12 and IC13, thereby ensuring that the displayed numbers are correctly weighted. On all other ranges, the output of the IC8 counter is effectively disabled by the open-circuit switches of IC10, and the output of the IC1-IC2 counter is added to zero before being fed to the display system; in these cases, the IC1-IC2 counter range may be 1-6, 0-9, 1-20 or 1-36, depending on the setting of SW1.

The display circuitry features leading-zero suppression, spin blanking and time-controlled auto-blanking, and has a manual RECALL facility. The operation of these facilities can easily be understood. Display-drivers IC12 and IC13 are supplied with a blanking facility (BL, pin 4), the action being such that the displays are enabled only when a logic 1 (high) bias is applied to pin 4. In our circuit, the blanking signal of IC13 is obtained from the leading BCD inputs (pins 1 and 7) of the chip via diode OR gate D1-D2 and can only be present when the BCD number is other than zero; the circuit thus has leading-zero suppression.

The displays of IC12 and IC13 can be blanked by driving Q1-Q2 on, or enabled by turning Q1-Q2 off: thus, a blanked number can be recalled by closing RECALL switch PB2 which turns Q1-Q2 off. IC11c-IC11d are used to generate the spin-blanking and time-controlled auto-blanking signal. When PB1 is closed, C4 charges via D3, applying a high signal to one side of IC11d, but the output of IC11c goes low, applying a low signal to the other side of IC11d. Consequently, the output of IC11d goes high and turns Q1-Q2 on, blanking the display for the duration of the PB1 closure. When PB1 is released, the output of IC11c goes high and feeds a high signal to one side of IC11d, while the other side of IC11d is held temporarily high by the charge of C4; IC11d output goes low, turning Q1-Q2 off and enabling the display. After a delay of about 5 s, however, the C4 voltage decays to such a level that the output of IC11d switches high and drives Q1-Q2 on, thereby giving time-controlled auto-blanking.

The only parts of the circuit that require further explanation are the BCD counter stages (IC1-IC2 and IC8). These are 4029B presettable up/down counters, wired in the BCD up mode in our circuit. A feature of these chips is that they have four BCD jam inputs (J1 to J4), and the jam code can be loaded into the counter by feeding a logic 1 signal to the pin 1 preset enable (PE) terminal of the IC. This feature makes it possible to reset the counter to any desired number (rather than simply to zero) at the end of each count cycle.

A jam code of 0001 is wired to IC8. IC9a-IC9b detect the arrival of each '7' state on the BCD output and activate the PE terminal, so that the circuit repeatedly counts from 1 to 6.

Similarly, on the IC1-IC2 counter, a jam code of 0000 0000 or 0000 0001 (depending on the setting of SW1b) is wired to the circuit. BCD-to-decimal decoders IC3 and IC4 and AND gates IC5a-IC5d are used to detect the desired reset count states (7, 7, 10, 21 or 37) of the counters and active the PE terminals of IC1-IC2 at appropriate times, to give the five ranges of 1-6, 2-12, 0-9, 1-20 and 1-36.

Note that the use of IC3-IC4 for decoding (rather than the use of dedicated logic decoders) enables the user to easily change the PE values (a simple re-wiring job) to give any desired count range up to 98.

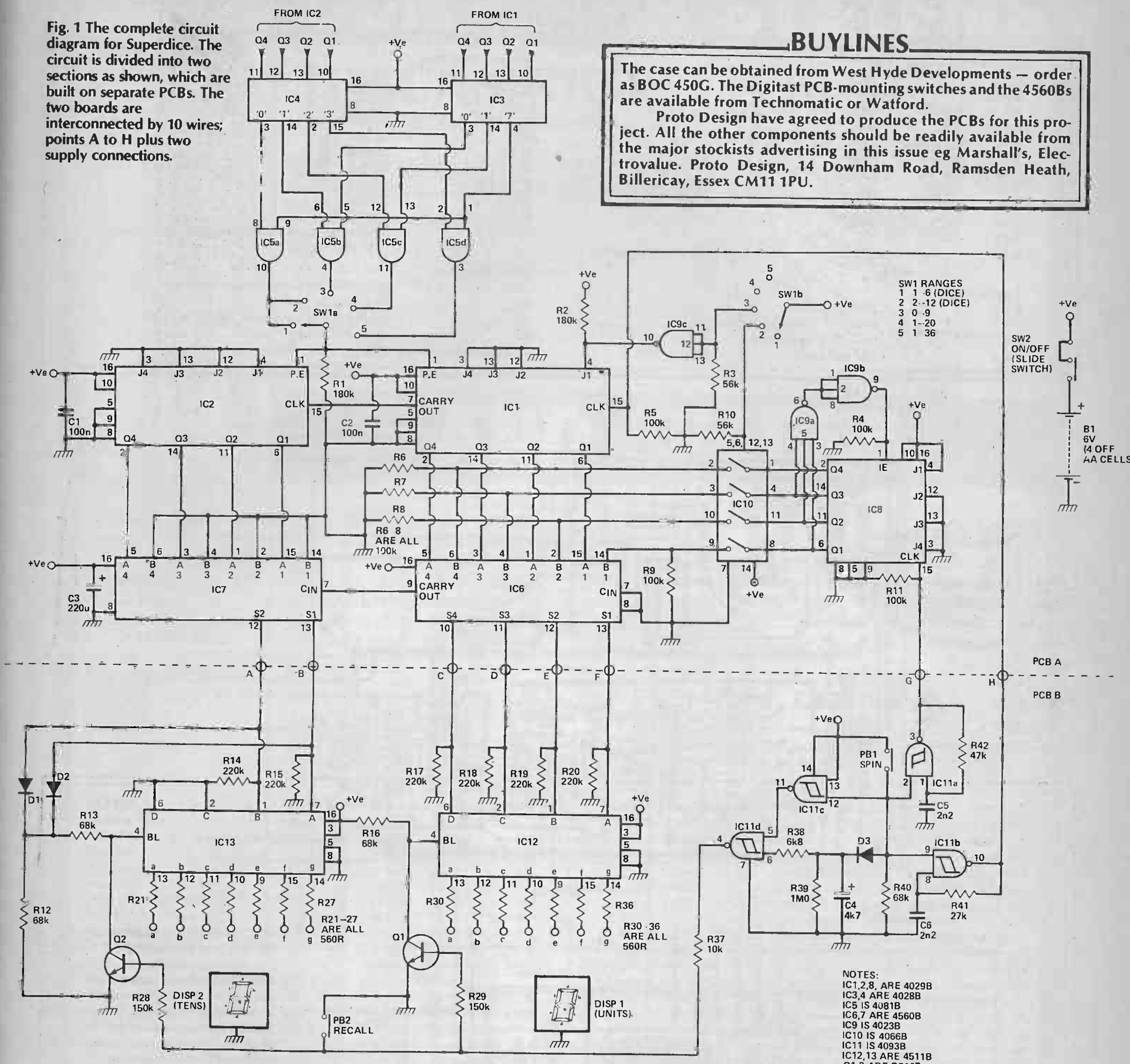
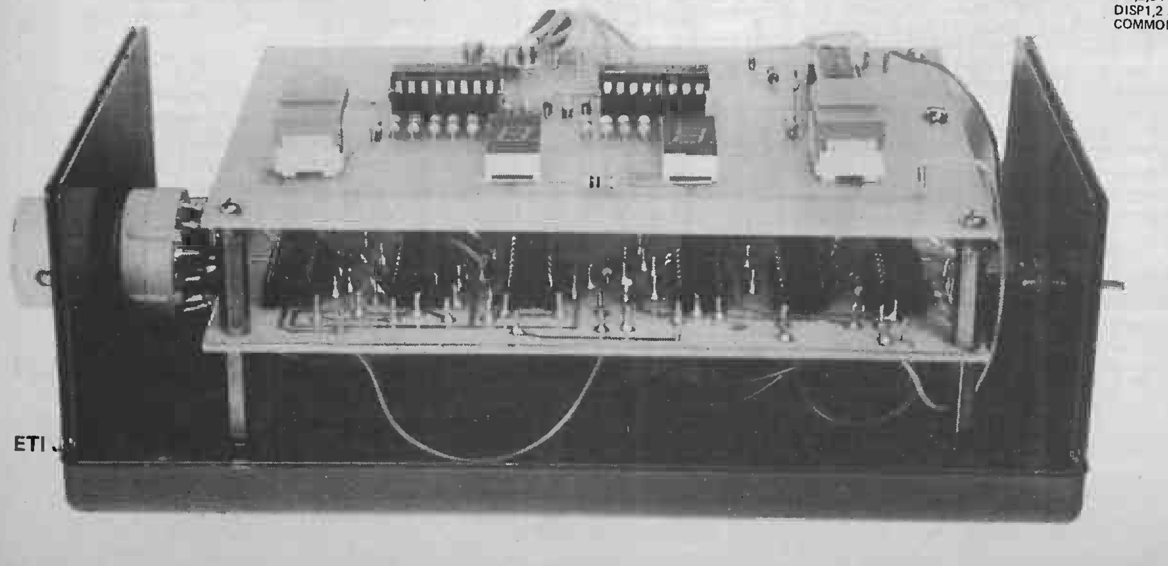


Fig. 1 The complete circuit diagram for Superdice. The circuit is divided into two sections as shown, which are built on separate PCBs. The two boards are interconnected by 10 wires; points A to H plus two supply connections.

BUYLINES

The case can be obtained from West Hyde Developments — order as BOC 450G. The Digitast PCB-mounting switches and the 4560Bs are available from Technomatic or Watford.

Proto Design have agreed to produce the PCBs for this project. All the other components should be readily available from the major stockists advertising in this issue eg Marshall's, Electrovalue. Proto Design, 14 Downham Road, Ramsden Heath, Billericay, Essex CM11 1PU.



This side view shows how the two boards are stacked inside the case, using tapped hexagonal spacers. The battery holder fits in the space under the lower board, and the height of the upper board just allows the push-buttons to pass through the cut-outs.

PARTS LIST

Resistors (all ¼ W, 5%)

R1,2	180k
R3,10	56k
R4,5,6,7,8,9,11	100k
R12,13,16,40	68k
R14,15,17,18,19,20	220k
R21-27, 30-36	560R
R28,29	150k
R37	10k
R38	6k8
R39	1M0
R41	27k
R42	47k

Capacitors

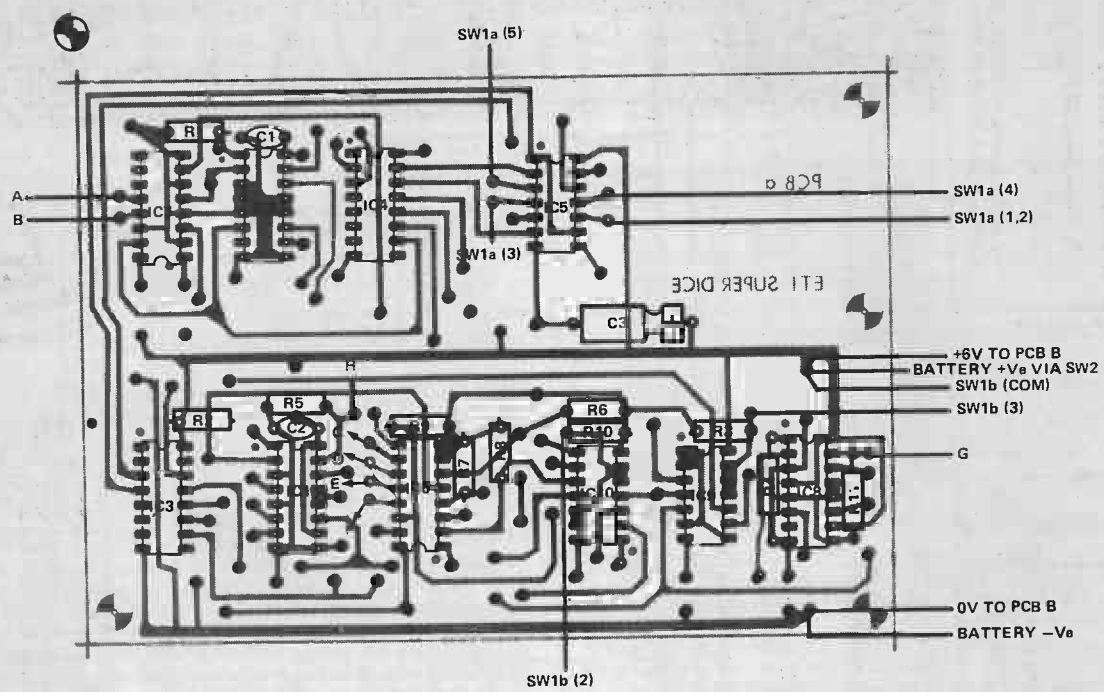
C1,2	100n ceramic
C3	220u 16 V axial electrolytic
C4	4u7 35 V tantalum
C5,6	2n2 ceramic

Semiconductors

IC1,2,8	4029B
IC3,4	4028B
IC5	4081B
IC6,7	4560B
IC9	4023B
IC10	4066B
IC11	4093B
IC12,13	4511B
Q1,2	BC107
D1-3	1N4148
DISP1,2	seven-segment common cathode display, 0.3" character

Miscellaneous

SW1	two-pole five-way rotary switch
SW2	SPDT slide switch
PB1,2	Digitast push-buttons (see Buylines)
B1	four HP7 batteries
Case (see Buylines), PCB (see Buylines).	



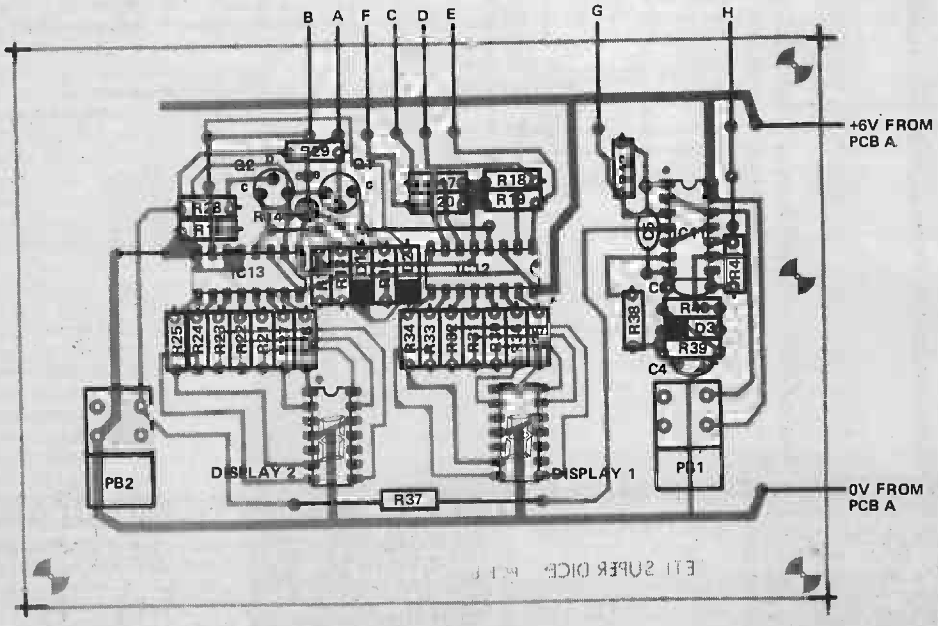


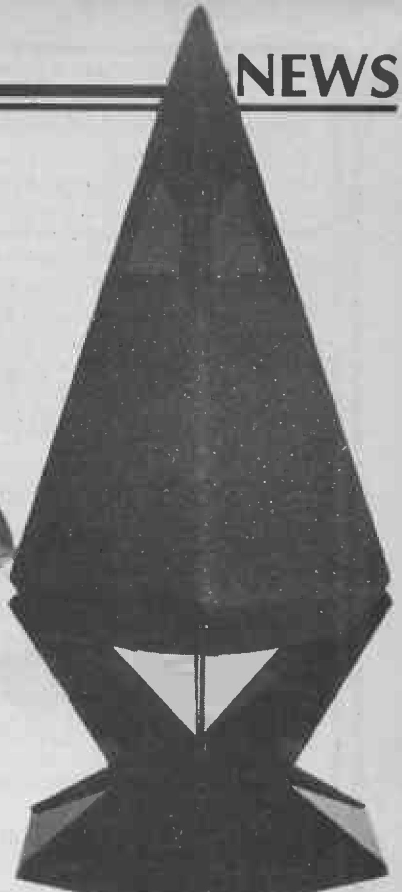
Fig. 2 (Above) Component overlay for PCBa of the Superdice. This is the double-sided board, although only the 'non-component side' tracks are shown. The right-hand side of R2 isn't floating, it solders to the topside track. The two sides of the board must be connected by Veropins (soldered both sides) where indicated by black dots.

Fig. 3 Component overlay for PCBb, the single-sided display board. Make sure you get the right size displays to fit the foil pattern.

ETI

AUDIOPHILE

Not Quad speakers but quadruple speakers are under test by Ron Harris this month. There's also a report on this year's High Fidelity Show.



Exhibition time in downtown Hammersmith. The Cunard Hotel once more hosted the High Fidelity Show, this time in its 1981 guise and, like all good lemmings, yours truly trod the well-worn path to its portals.

As usual the volume level was high (as were most of the exhibitors by 3 pm), and the size of the rooms was the limiting factor on the demonstrations. About the only company totally unaffected by the 'sonic sardines' syndrome was Stax, whose new Lambda headphones were providing some excellent reproduction amid the echoes and hollowed eyes. The Lambda had an open and detailed sound, and a very well extended bass response. I would have liked to listen at greater length, but the swell of sound in between tracks beckoned me on, deeper into the hallowed halls . . .

Wandering around those dimly lit corridors is a little like playing an 'adventure' game, only this time for real. You never know what's going to leap at you from around the next bend, or whether opening that locked door will unleash agony or ecstasy. Can be nerve racking at times. No wonder the bars are always full at hi-fi shows.

KEF were previewing their new Coda II, Caprice II and Carlton II range, which will be released in late summer — and very nice they sounded too. The Coda will cost somewhere around £80-£90 and will prove a major addition to the small speaker market. The Carlton is a development of the 104 and is very good indeed on all counts.

Above left: the new Goldring G900IGC cartridge. Van den Hul for all?
Right: the menacing visage of the Black Diamond omni-monitor. A new speaker concept at last?

Carver and Gale had combined for a demonstration of the active tri-amped Gale loudspeakers, using the Carver cubes (magnetic field amps). Altogether a very interesting and entertaining sound, especially in the mid-range. Released soon, and Audiophile hopes then to lay hands on a cube at long last.

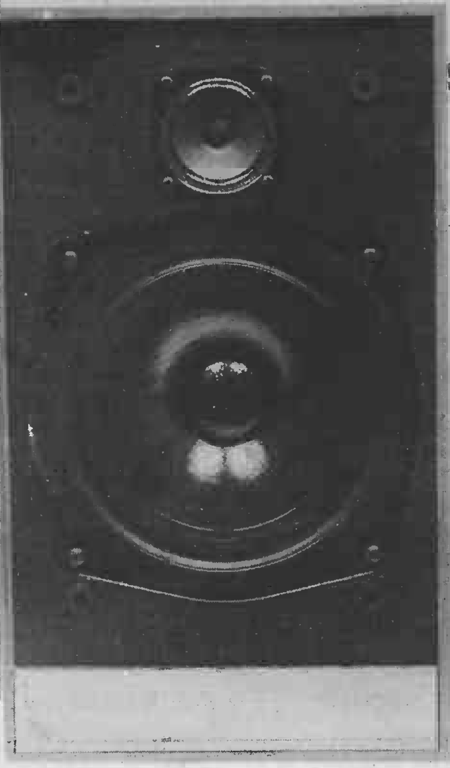
A complete departure from the norm is the new Black Diamond omni-monitor. Hands up all those who remember Sonab speakers? Well it's nothing to do with those at all.

As you can see from the photo, styling is unconventional, but follows 'air-flow engineering principles' to create its sound. Initial impressions were of a well balanced sound with a slight edge to the top end. Good value.

Pick-up On Tapes

Shure Electronics are branching out in the UK and were showing a range of three hi-fi cassette decks, all well endowed with 'facilities'. Even the bottom of the range boasts fluorescent metering and auto-play. The name on the machines is Alpage and the top-end AL-300 has 'Dolby' and 'bias adjust' pots and

Above: the Alpage AL300 cassette deck. New from Shure Electronics it is one of a range of decks with much to offer, including Dolby optimisation.



CARNIVAL 2

Dimensions: 15¾" x 9½" x 5¾"
Drive units: two — 208 mm bass and 12.5 mm paper tweeters
Impedance (max/min): 19/7 (nominal 8 R)
Power requirements: 10-75 W
Max SPL obtainable (1 m on axis): 100 dBA
Frequency response: 80-17 kHz ± 3 dB
Sensitivity (W/dB) (1 m on axis): 96 dB for 6 W
Typical selling price: £115

Finished to a very high standard and with an impeccable pedigree, the Carnivals come well equipped to succeed. The enclosure is very shallow and the surprising drive unit line-up — a large bass unit and paper cone tweeter (when all around is domed?) — is well constructed.

Mordaunt Short are not as well known as their product deserves. The 'Signifier' is one of the best sounding 'large' enclosures around for home usage, in my opinion, but how often does it appear in print?

It may well be that MS, in common with many excellent producers of equipment, are suffering from lack of exposure due to the 'tunnel vision' which has afflicted much of the hi-fi specialist press of late.

Still, I digress from my subject — and lay myself open to accusations of unsweet fruit in bunches.

Initial impressions were of an excellent overall sound balance, with good imaging and well defined bass response. Extended tests showed the treble to be a little hard and slightly prone to sibilance with some material. Not a serious worry though.

Bass information is well presented, but bass extension is naturally limited. The Carnival does a good job of coping without the last octave known to man and is a good all-round performer. Shelf and 'near-wall' usage is to be preferred where possible.

Mid-range information is especially well handled and at high levels nothing untoward happens to the sound balance. (Very high sound levels are possible before stress becomes evident.)

Overall then, a very good speaker indeed and one with a lot to offer the purchaser. It has the rare quality in a speaker of sounding equally at home with any kind of programme material.

At the price the Carnival 2 cannot be considered less than excellent value and is confidently recommended.

LASER 80

Dimensions: 22¼" x 12" x 9¾"
Drive units: three — 250 mm bass/100 mm mid/19 mm dome tweeter
Impedance (max/min): 6 R nominal (3 R minimum)
Power requirements: 10-100 W
Max SPL obtainable (1 m on axis): 98 dBA
Frequency response: 50-20 kHz ± 3 dB
Sensitivity (W/dB) (1 m on axis): 88 dB for 1 W
Typical selling price: £99

Being by far the biggest of the speakers herein, something a little 'deeper' was expected from the Laser 80 and it did not disappoint!

The Laser range is Wharfedale's new 'value for money' line, designed to offer an honest performance for a given price. On this showing they have succeeded admirably.

From the moment they were switched on, the Laser 80s impressed me with their 'larger-than-life' approach. They are not neutral, not were they designed to be, I suspect. The sound balance is full and forward and very detailed. For the size of enclosure the extension to bass response is truly remarkable. This has been obtained at the expense of a rise around 100 Hz and this *can* lead to a slight 'boom' if the enclosures are not sited properly.

Treble is a little exaggerated and maybe emphasised surface noise over a more neutral speaker. They handled power with a nice ease and were especially impressive on high level rock music.
pleasurable and entertaining to listen to.

As an analogy, take the 105 II as being the 'Times' of loudspeakers — accurate, authoritive and dry. The Laser 80 is the 'Sun'! Bright, lively and designed to entertain. Listening to it you might know that it is not as accurate as others, but it is *fun*!

An excellent product and one which will find a wide audience at the price.

oscillators on the front panel. All are 'metal' decks and produced a very creditable sound. More in the near future on these.

Goldring were showing their new attempt to bring the delights of IGC detail and style to the masses. The new 910 and 920 IGC cartridges are developed from the well-beloved G900 IGC model and should do well further down the market place, where pound-notes are thinner on the ground and gems further apart.

All in all an interesting show, but not one of the best. Trade and public support was down this year on last and I felt it suffered in consequence. Maybe it's time for a move?

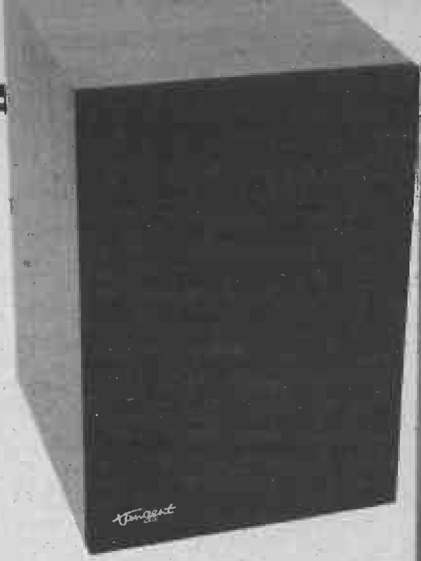
●

Speakers Four All?

Small speaker design has come a long way in the last few years, indeed it is probably the area in which the greatest advances have been made in the art of late.

The LS3 5A design remains as the reference point in this area — or does it? The BBC design is now somewhat long in dental equipment and the contenders for the customer's pound are many and vociferous. Nor is it any longer true that 'small' means 'budget'. Witness the fact that the KEF 101 costs nearly £200 a pair.

The four loudspeakers under scrutiny here all qualify for the label 'small' — some more than others. The idea is to cover a worthwhile range of examples in both size and price. Chosen to



TANGENT TM3

Dimensions: 14½" x 10" x 11"
Drive units: two — 210 mm bass and 19 mm dome (KEF units)
Impedance (max/min): 8 R nominal
Power requirements: 10-75 W
Max SPL obtainable (1 m on axis): 98 dBA (effectively 94 dBA; see text)
Frequency response: 60-18 kHz ± 3 dB
Sensitivity (W/dB) (1 m on axis): —
Typical selling price: £120

The TM3 is an unusual shape for a speaker. It's too deep to fit most shelves, but too small to stand-mount. I ended up having to drag two stools in from elsewhere in the living quarters to get them clear of the floor and away from the back walls, proximity to which immediately induced a nasty attack of the bass booms.

Construction and finish are to a good standard and KEF drivers are employed. The bass unit is comparatively large for the size of enclosure.

Listening commenced with classical material on the TM3 and was well portrayed. The sound stood nicely away from the boxes and the image was clear and firm. Treble appeared slightly recessed and bass material was sufficiently defined to rate a 'good' score. Overall a fine performance considering the size. String instruments in particular were outstanding, the attack being exceptional.

Large-scale works and high-level rock did not fare so well though, and when driven hard the TM3 tends to retreat into its enclosure! The sound stage broke up and the final result was most disconcerting. Just when I was beginning to think the TM3 a very fine speaker indeed, it has to go and run away!

Further listening under varied conditions could not completely dispel this impression, although played at anything less than 'heavy' levels, the TM3 performed well. I tried amplifiers of varying power outputs and characteristics, but to no avail.

This characteristic — it's almost cowardice! — must be viewed as a limiting factor on the TM3's market, but should not lead you to discount it completely. The speaker has a finely detailed performance and good overall balance, albeit slightly mid-range prominent. Highly recommended for its detail and quality sound, then, but with a reservation for rock music freaks prone to sonic suicide. Very good value for money.

fulfil this aim were the KEF 101, Tangent TM3, Mordaunt Short Carnival 2 and Wharfedale Laser 80. In price, from just under £100 a pair, to within a gnat's eyelash of £200. In size, from the 18¾" x 10½" x 9½" of the Laser 80, down to the diminutive 13" x 7" x 7" of the KEF 101.

Each model is considered on its own merits, rather than in a group comparison which would be unrealistic and unfair in this context, as the intended audience is different in each case, I feel. Comparative comments will be offered where appropriate, but I'm making no attempt to pick a 'best' and thereby consign the rest to a subjective Hades. Speakers *must* be listened to *at length* before being purchased and reviewers such as I can do no more than to point out a well engineered example of the art and recommend an audition. Or at least we *shouldn't* do anyway.

Reference At Source

The units were auditioned over a period of time, against a known reference — the KEF 105 II — in order that their absolute performance could be judged accurately. Throughout the tests the cartridge used was an Ortofon MC30, with T-30 step-up transformer. Radio source material was obtained from the Sony ST-J75 FM tuner.

Amplification varied from Lecson through Meridian to Monogram. The reason for this will be made clear in next month's Audiophile, when the latter firm's new pre/power combinations are reviewed.

Recorded material used included the DG Tchaikovsky 4th and 5th, Rickie Lee Jones, The Wall, Sky, Verdi's Requiem and a quick burst or two of direct cut Wagner.

KEF 101

Dimensions: 13" x 7" x 7"
Drive units: two — 110 mm bass and 25 mm dome
Impedance (max/min): 35/7 (nominal 8 R)
Power requirements: 50-100 W
Max SPL obtainable (1 m on axis): 96 dBA
Frequency response: 80-20 kHz ± 3 dB
Rapid bass roll off below 60 Hz (fourth order)
Sensitivity (W/dB) (1 m on axis): 81 dB for 1 W
Typical selling price: £190

The 101 is easily the best presented speaker I've seen. The veneers on the enclosure are superb, the speaker is solid and amazingly heavy for its size. The back-up literature is comprehensive and professional. It even specifies wire gauges for the leads (other than the supplied set) and the maximum allowable length for each.

The whole production inspires confidence and belief from the start. Somehow it seems impossible that this can be anything else but the best small loudspeaker you've ever heard. At first listen, however, the KEFs were a surprise. The bass is rolled off very rapidly indeed — and there's 400uF in series to make sure of that — and this robs the 101 of any 'weight' it might otherwise have had. The mid-range and top end are accurate and smooth, with a curiously warm balance. Single vocals were outstanding but complex material tended to become confused at high level.

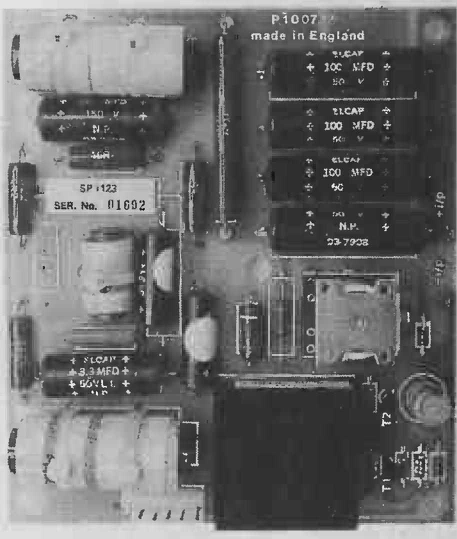
Comparing the KEF 101 with the LS3 5A speaker proved to be very interesting and illuminating. The speakers have totally different sound balances. The KEF is much smoother and less lively, setting the sound back from the listener. The LS3 5A, by comparison, is very forward and explicit.

Careful listening shows that both produce remarkable detail and a creditable sound. Which you personally prefer will depend, I suspect, on what type of material you listen to.

On playing the Floyd's 'The Wall' through the 101s I was staggered at the power handling — they took 200 W of program with not so much as a crackle! Since the 101 is equipped with the KEF SSTOP protection system, it is highly unlikely that anyone could ever damage the speaker, short of hanging it on the National Grid. Mid-range remained good with plenty of detail, but that 'lack of bottom-end' impression robbed them of authority.

Overall a very creditable design, with a refined and clear sound quality. It is one that would appeal strongly to anyone involved with, say, folk or chamber music, where large amounts of bass energy are not required.

Alternatively, if fitted into a system using a separate bass enclosure and amplifier, the 101 could offer a formidable alternative to larger monitors.



Above: the crossover PCB taken from the 101 loudspeaker. Note the line of 100u capacitors to tailor the bass response. These are 'low inductance' components — no capacitor is *pure* capacitance! — in order that filter characteristics can be carefully controlled.

At the lower edge of the board are the SSTOP protection circuits which make the KEF practically invulnerable to overload.

(A photo such as this should serve to dissuade the "suck-it-and-see" home designers who think crossovers can be designed with a little roll of enamelled wire and a lot of faith!)

Conclusions

As I mentioned earlier, I'm not going to derive a 'winner' or three losers. I feel all four speakers have a lot to offer in their own way. The KEF and the Tangent are highly individual speakers, I think, while the Wharfedale offers incredible value for money and will appeal widely. Its approach is very much 'larger-than-life' but totally unpretentious. Ideal for a first system, or indeed any low-cost (rock-orientated?) set-up.

As a contrast the Carnival is as classical an approach to all-round accuracy as you'll find anywhere. Somewhat lacking in bass weight, naturally, but good for the size.

So that's it. Four for all. I hope there is sufficient in the way of hints herein to provide you with a starting point if you're in this market at present.

More than that we cannot do.

Plea For Choice

As I'm a hoarder of all things printed (I've got copies of 'Hi-Fi Sound' right back to 1970) I ask for your assistance in completing a collection of 'Hi-Fi Choice' publications. I have been relieved of my copy of numbers 3, 4, 6, 7 and 8 by some unmentionable worm from the lowest pit of existence. If I ever find him, his kneecaps are coming off so quick he'll be walking backwards and forwards simultaneously.

My plea is, therefore, that if anyone out there has a copy of the aforementioned issues that are no longer in daily usage — how about selling them to me? I know they're all out of date now, but that's not the point...

Samaritans please write to 'Audiophile' here at ETI, from whence a speedy reply shall wing its way to you.

ETI

SMART BATTERY CHARGER

This 'smart' unit gives a fault warning if your battery is defective in any way: if the battery is OK, the unit will power it up. Design by Ray Marston. Development by Plamen Pazov.

Charging a car battery with a conventional charger unit can be a time-wasting task. Once the owner has connected the battery to the charger, he has to occasionally check the state of the battery with a hydrometer and switch the charger off manually when the battery reaches the fully charged state, to avoid the risk of overcharging and possible plate buckling. Once every couple of years you'll find, after a lot of time-wasting, that the battery state is so deteriorated that it is beyond redemption.

ETI's new battery charger circuit (an update of our April '80 effort) overcomes these time-wasting snags. When you first connect it to the battery, the unit automatically checks that there are no obvious signs of cell damage or destructive corruption of the electrolytic solution. If any fault is evident a fault LED will illuminate and the circuit will refuse to apply a charge current to the battery.

If the battery is sound, the unit will charge it up in the conventional way but will continuously monitor its charge state and, when the battery reaches the fully charged state, will automatically switch to the 'trickle charge' mode (indicated by an LED) in which the charge is maintained without risk of battery damage. The unit thus gives a 'fit and forget' type of battery charging action.

Our charger is designed to charge 12 V batteries only. The unit can either be built as a stand-alone project, complete with transformer and case, or can simply be added to an existing charger to update a conventional design.

Construction And Use

Construction of the unit should present few problems. If you decide to build the complete stand-alone project, assemble the PCB components exactly as shown on the overlay noting that the three LEDs and the bridge rectifier are mounted off-board; then complete the interwiring to the meter, bridge rectifier, transformer, LEDs, and so on, and box the unit. Note in our prototype that the transformer and bridge rectifier are bolted to the metal panel at the rear of the case, which thus acts as a heat sink.

The 5 A meter is an optional item: if you decide not to fit it, simply take the positive output of the bridge rectifier directly to the PCB positive terminal. On our prototype, we use a standard moving coil meter, shunted to 5 A FSD, as the current monitor: a cheaper alternative would be a moving iron meter, which *may* be available from some car accessory shops.

If you decide to use our 'smart' charger circuit to update an existing 12 V battery charger unit, simply wire our PCB to the output of the charger (taking care to observe polarities) and shift the positive crocodile lead to the PCB output. Whichever version of the unit you use, be sure to use a reasonably heavy gauge of wire for the interconnections.

When construction is complete, turn PR1 slider to mid position and give the unit a functional check as follows.

(1) Check that, with no battery connected, LEDs 1 and 2 illuminate.

(2) Short the output terminals together with a 5 A fuse; check that FAULT LED3 illuminates and that negligible current flows through the output terminals.

(3) Connect a sound but partially discharged 12 V car battery in place and check that LEDs 2 and 3 turn off and that a charge current (typically 2 and 4 A) flows to the battery. Rotate PR1 slider and check that CHARGED LED2 can be turned on and the charge current cut off us- →

ing the pot.

(4) Rotate PR1 slider fully towards R7 and charge the battery up using the normal hydrometer technique. When the battery reaches full charge, carefully adjust PR1 so that LED2 just starts to turn on and the charge current falls to a trickle level of a few hundred milliamps.

If PR1 is correctly set, you'll find that on subsequent charges LED2 will first start to flicker as the full charge level is reached. The LED will subsequently turn on at reduced brightness or will alternately cut on and off as the fully charged state is maintained. PR1 should require no further adjustment throughout the life of the charger.

HOW IT WORKS

In a conventional battery charger the unsmoothed full-wave rectified output of a 17 V transformer is fed to the battery. The battery is charged by a pulsed current at a rate determined by the differential voltage between the battery and the charger and by the total series resistance of the circuit (the effective resistance of the transformer, rectifier and battery). A flat battery has a low terminal voltage and typically draws an initial charging current of about 4 A, falling to about 2 A as the terminal voltage rises to the full charge value. The total series resistance of the circuit is sufficient to limit the charge current to a safe value.

You'll notice from the above description that the battery terminal voltage rises as the battery charges up and can thus be used to give an indication of the state of charge of the battery. In our charger circuit, power is fed to the battery via silicon-controlled rectifier SCR1, which in turn is controlled by voltage-sensing circuitry designed around IC1-IC2 and Q2. If the battery is connected when its off-load voltage is below 10 V (indicating damaged plates or defective electrolyte), Q2 turns on and the SCR drive is disabled, and the battery receives no charge. If the voltage is above 10 V but below the fully charged value, Q2 turns off and the SCR turns on early in each full-wave rectified mains half-cycle via R2 and D1, so the battery charges in the normal way. Once the battery voltage reaches the fully charged value, Q2 again turns on, disabling the SCR, and the charging process is then complete. Details of the voltage-sensing circuit are as follows.

IC1 and IC2 are wired as a voltage-window comparator, with their outputs fed to Q2 via indicator LEDs 2 and 3 and via the D4-D5 OR gate. Reference voltages are fed to these two ICs from ZD2 via the R8-PR1-R9-R10 potential divider network, with 5 V being fed to pin 3 of IC2 and roughly 7 V fed to pin 2 of IC1 via PR1. The battery voltage is halved by R3-R4, integrated by R5-C2-D3, and fed to pin 3 of IC1 and pin 2 of IC2 via safety resistors R11 and R12.

Thus, if the battery voltage is below 10 V, the pin 2 voltage of IC2 will be below that of pin 3, and IC2 output will go high, driving FAULT LED3 on and disabling the SCR via Q2. Similarly, if the battery voltage is greater than 14 V nominal, pin 3 of IC1 will be above that of pin 2, and IC1 output will go high, driving CHARGED LED2 on and again disabling the SCR via Q2. Finally, if the battery voltage is in the range 10 to 14 V, the outputs of IC1 and IC2 will both be low, Q2 will be cut off, and the SCR will gate on in each half-cycle via R2-D1 and apply charge current to the battery.

In practice, the terminal voltage of the battery depends on both the battery state and the magnitude of the charging current and decreases when the charging current is removed. Consequently, the circuit does not abruptly stop providing a charging current when the battery reaches full charge, but goes into a skip-cycling mode, progressively reducing the mean charging current to a low trickle value. This action automatically maintains the battery in a fully charged (but not over-charged) state.

The correct setting of preset pot PR1 is established initially by charging the battery up in the conventional (hydrometer) manner until it reaches the fully charged state. PR1 is then carefully set so that the charger goes into the skip-cycling or trickle charge mode under this condition. The PR1 setting is then valid for all subsequent automatic recharging actions. Current monitor meter M1 is an entirely optional component in this circuit.

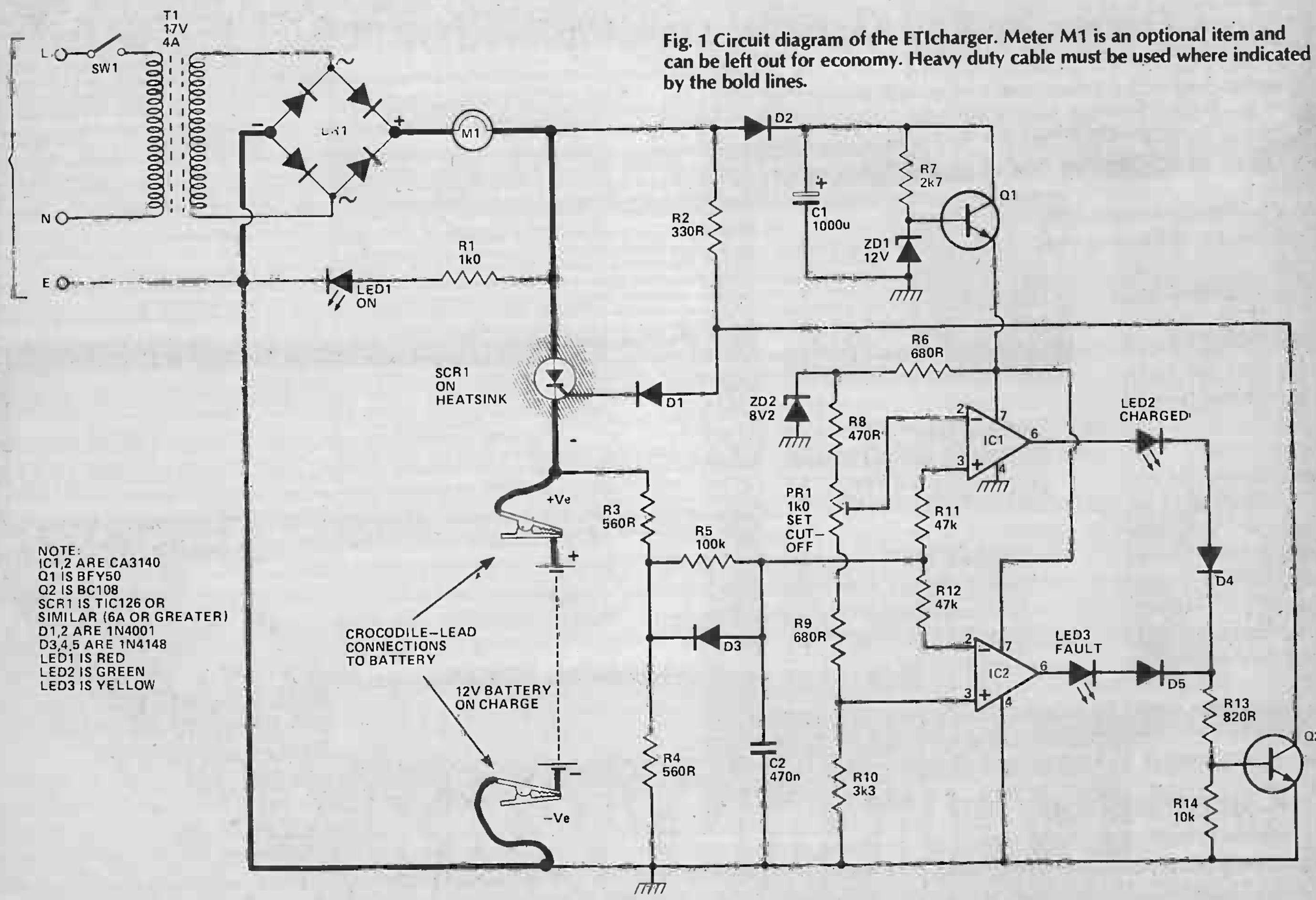
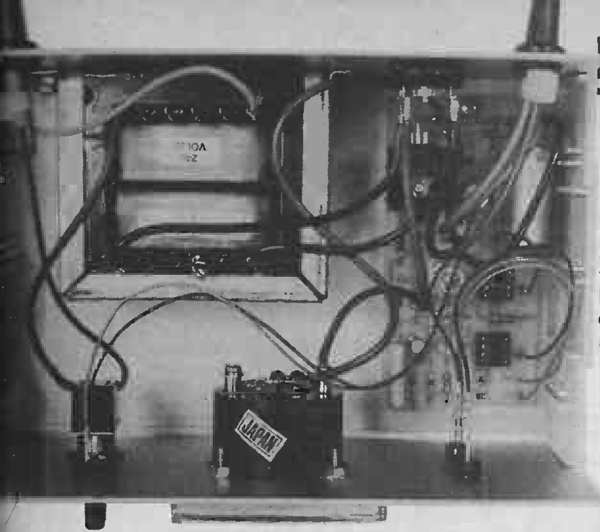


Fig. 1 Circuit diagram of the ETIcharger. Meter M1 is an optional item and can be left out for economy. Heavy duty cable must be used where indicated by the bold lines.



The completed project in its case. The metal back panel acts as a heatsink for the bridge rectifier, but the SCR has its own heatsink mounted on the PCB.

PARTS LIST

Resistors (all ¼ W, 5%)	
R1	1k0
R2	330R
R3,4	560R
R5	100k
R6,9	680R
R7	2k7
R8	470R
R10	3k3
R11,12	47k
R13	820R
R14	10k
Capacitors	
C1	1000u 40 V axial electrolytic
C2	470n polyester
Semiconductors	
IC1,2	CA3140
Q1	BFY50
Q2	BC108
SCR1	TIC126
BR1	50 V, 6 A bridge
D1,2	1N4001
D3,4,5	1N4148
Miscellaneous	
Transformer (Repanco CT4), Verocase ref 202-21039	

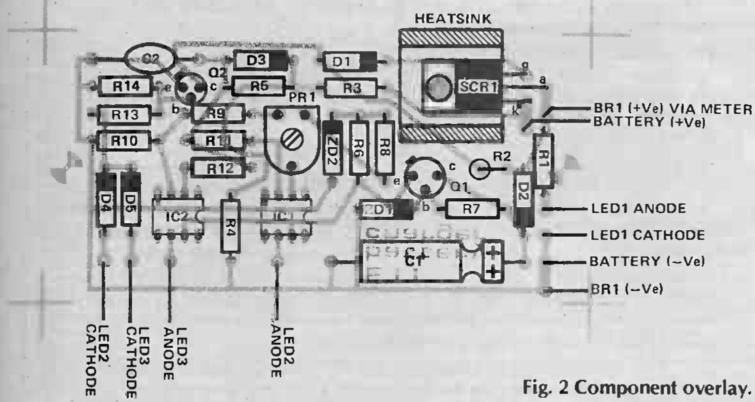


Fig. 2 Component overlay.

BUYLINES

The transformer and C1 are available from Electrovalue. All the other components except meter M1 (see text) are readily available items and should present no problems.

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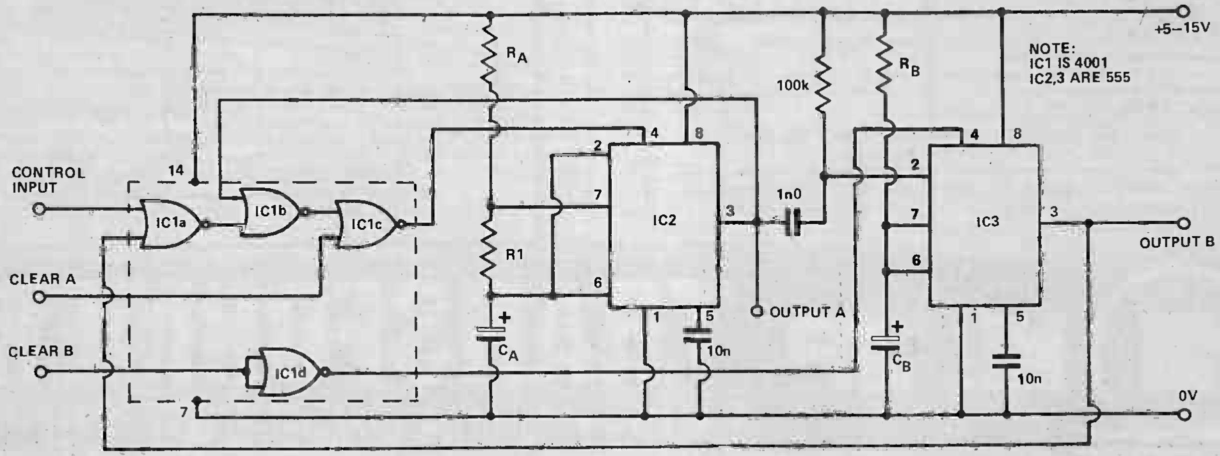
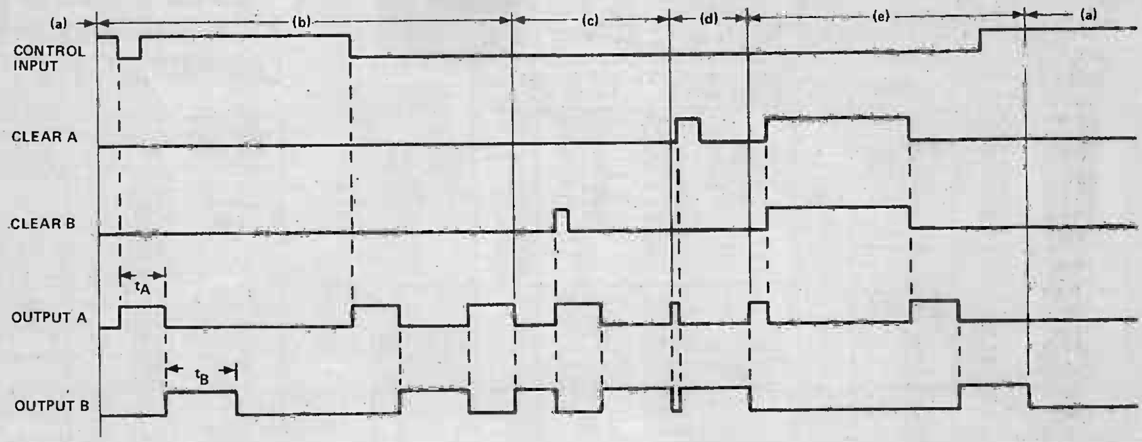
TECH TIPS





Double-pulse Oscillator

C. Shackleton, South Africa

This circuit has an active cycle which produces two output pulses: pulse A when the cycle is initiated, followed immediately by pulse B. A control input is used to initiate the cycle, and is isolated for the remainder of the cycle. Two clear inputs enable either output pulse to be terminated at any time.

Operation may best be understood by studying the waveforms, bearing in mind IC2 is connected as an astable while IC3 is connected as a monostable (this configuration prevents any possibility of latch-up). If both clear inputs are kept low, the following holds:

(i) If IC2 and IC3 are both in the 'rest' condition, and the control input is high (a), nothing happens.

(ii) If the control input is low (b), IC2 begins to oscillate (output A rises), the control input is isolated, and IC2 reset is kept high. After a time

$$t_A = 1.1\,(R_A + R_1)C_A,$$

output A falls, triggering IC3 (output B rises), the control input is again isolated, and IC2 remains reset. After a time

$$t_B = 1.1\,R_B C_B,$$

output B falls and the circuit is back to its 'rest' condition.

(iii) Whenever Clear A goes high, (d) and (e), IC2 resets immediately and remains reset for the duration of the clear pulse; nothing else is affected. Clear B acts on IC3 in the same way; (c) and (e).

The flexibility of this circuit indicates that it can be used in a variety of applications. The prototype controlled the concentration of algae in a tank of mussels. A light-operated system caused the control input to go low when the concentration fell. Algae were then dropped into the tank for time t_A; time t_B allowed the algae to disperse before the cycle could repeat.

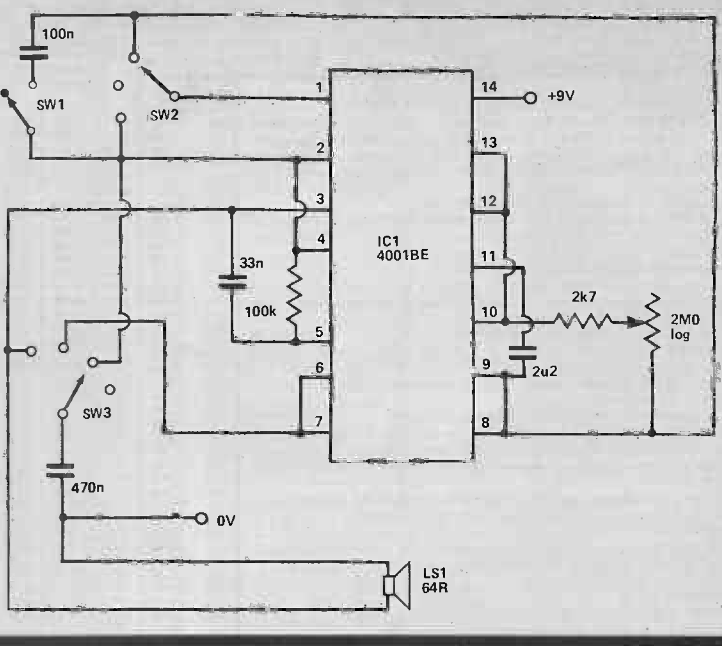
Tech-Tips is an ideas forum and is not aimed at the beginner. We regret we cannot answer queries on these items. ETI is prepared to consider circuits or ideas submitted by readers for this page. All items used will be paid for. Drawings should be as clear as possible and the text should preferably be 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.

Simple Sound Effects

A.G. Smith, Derby

This circuit will generate 24 different sound effects including two-tone sirens, rising tones, seagulls etc. It operates from a 9 V battery and uses only one CMOS IC. Most of the components are not critical, but the speaker must have an impedance of 64-100R. Note that the negative supply from the battery does not go to the negative supply pin on the IC, which must be the buffered version of the 4001.

Altering the 33n capacitor or 100k resistor changes the basic frequency, and the 2M0 pot adjusts the speed of the rise and fall of the tones. A PP6 battery was used to drive the circuit and has been in regular operation for six months without replacement.

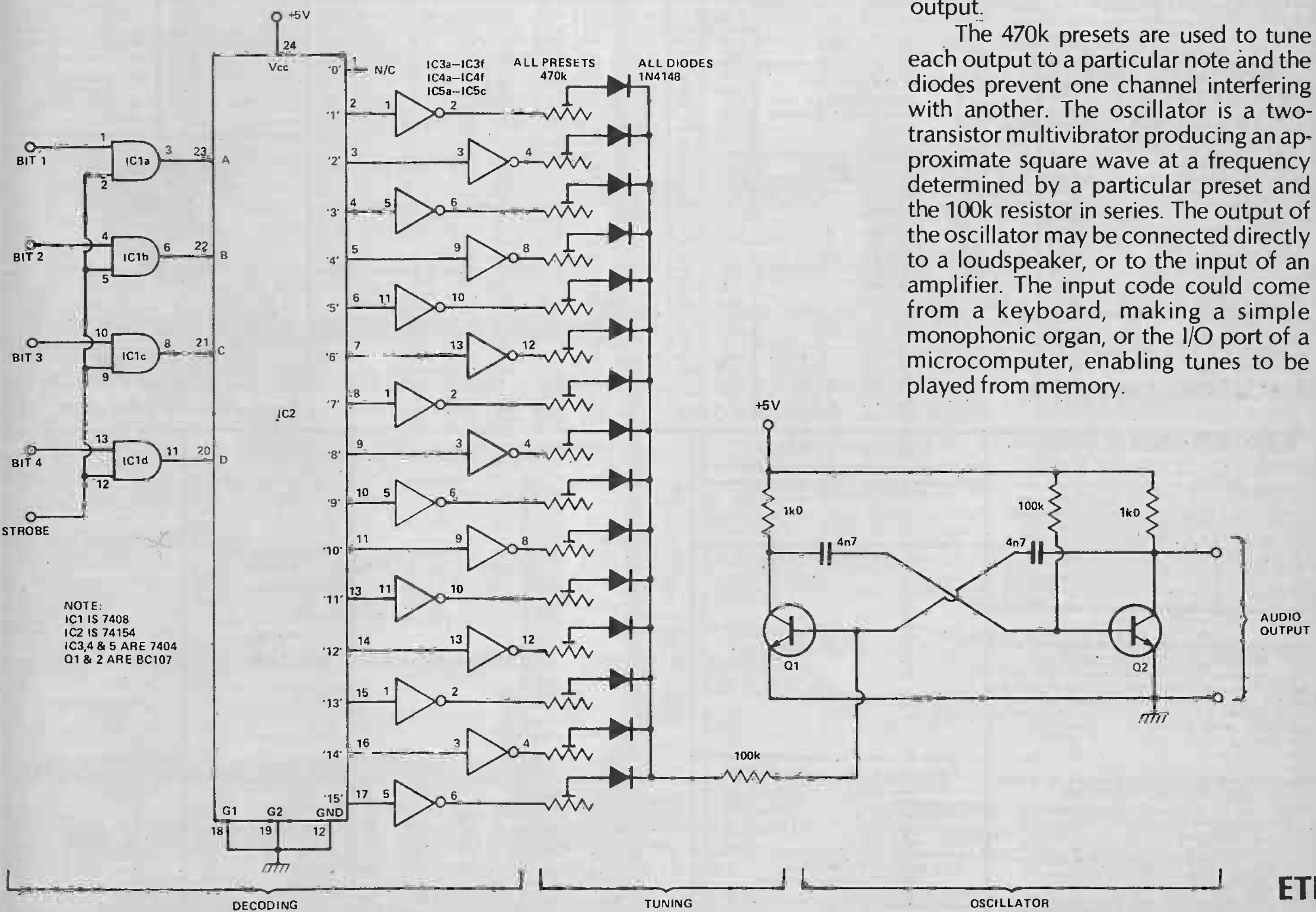


Computer Organ

A. Brown, Ayr

A four-bit word can generate musical notes using this simple circuit. The 7408 only allows the four-bit binary code from the keyboard or I/O port to reach the address pins of the 74154 while the strobe is high. On the keyboard used the strobe stayed high until the key was released, but a latch could be used to hold the data if only a pulsed strobe is available. The 74154 is a 1-of-16 decoder with active-low outputs, ie one output is low depending on the binary code present on the input pins, the rest stay high. The 7404 inverters were used so that a voltage could be sourced to the oscillator, instead of trying to sink one. The '0' output is not connected, to allow an input code which produces no sound output.

The 470k presets are used to tune each output to a particular note and the diodes prevent one channel interfering with another. The oscillator is a two-transistor multivibrator producing an approximate square wave at a frequency determined by a particular preset and the 100k resistor in series. The output of the oscillator may be connected directly to a loudspeaker, or to the input of an amplifier. The input code could come from a keyboard, making a simple monophonic organ, or the I/O port of a microcomputer, enabling tunes to be played from memory.



SPACE INVASION MODIFICATIONS

Space Invasion grows up! Here are the changes to convert it into your very own personal computer.

Back in the November '80 ETI we featured the Space Invasion TV game from Tangerine Computers. The design as published was effectively a microcomputer dedicated to a single task — running the Space Invasion program. With the simple additions and alterations given here, the project becomes a fully-fledged personal computer, equivalent to the Microtan 65 and able to use all of the expansion boards in the Tangerine range.

Before discussing the modifications, we ought to mention a few alterations to the original circuit. Pin 12 of IC11b is shown connected to earth. It should be connected to IC15 pin 34. All new boards being supplied by Tangerine have this modification to the track pattern, so check your board before making any changes. IC20 pins 20 and 21 should be swapped over, and the label adjacent to IC7c was ringed in error — it is diode D2, not IC position D2. The unmarked pin on IC9c is pin 11.

All Change

There are only a handful of extra components required to complete the conversion. An additional character generator enables the screen to display the lower case alphabet. Connection to other boards in the Tangerine system is made via the multiway socket that comes ready-soldered to the PCB — the bus signals need a high driving capability and the tri-state buffers for the address bus are located on the CPU card. (Buffers for the data bus are on the TANEX card.) If you feel content with only 1K of RAM and machine code programming, the address buffers won't be necessary. The remaining components provide various timing and control signals.

Soft Option

Even with these changes, the board will still only play Space Invasion unless you change the software. This is simple — remove the EPROM (IC20), return it to the protective foam from whence it came, and replace it with either the 1K TANBUG ROM or the 2K XBUG ROM, from Tangerine. The former is OK if you are happy with the limitations of the basic board, but if you plan to expand at some later date into a larger system with more I/O, the extra facilities of XBUG make it a better buy.

If you don't already have the Hex keypad or ASCII keyboard, you'll need one or the other to allow you to enter your own programs.

Spreading Out

The first step in expanding the system is to purchase the TANEX card. This is connected to the CPU card by plugging both boards into a motherboard, and provides a cassette interface, 16 I/O lines, two 16 bit counter timers, the aforementioned data bus buffers and an additional 1K of RAM. This board also takes over the address decoding of the memory map for the system, which is incompatible with the simpler, hardwired memory map of the CPU card. If TANEX is to be used, it is necessary to cut the three wire links on the CPU card (LINK RAM, LINK ROM and LINK I/O).

Plugging extra chips into the TANEX board will give you up to 8K of RAM, 16 more I/O lines, two more counter timers, serial I/O and 10K BASIC in ROM. Once you've got the BASIC you'll need an ASCII keyboard, and when you start writing huge programs, TANRAM will come in handy (up to 40K of extra memory).

And what if, after all this, you still feel a yearning to play Space Invasion again? No problem — either swap the ROM chips back again if you only have the CPU card, or insert the Invasion ROM in position E2 on TANEX and proceed as instructed in the original article.

Prices for the unusual chips are as follows; the TANBUG ROM (ask Tangerine for TANBUG1, issue 2 board) costs £20.05 including VAT and postage, XBUG is £20.25, and the DM8678CAE is £8. All these prices include VAT and postage. For details of the other boards in the Tangerine range, get in touch with them at Forehill Works, Forehill, Ely, Cambs.

HOW IT WORKS

The basic Space Invasion unit requires only a few small modifications to the hardware and software to become a useful personal computer. The software is easily dealt with — by removing the ROM chip, IC20, and inserting a TANBUG ROM in this socket, we replace the fixed games program with a general purpose monitor.

The existing circuit already has upper case alphanumerics and graphics options; IC32 is a character generator for lower case alphabet.

If the system is to be expanded by connecting additional boards in the Tangerine range, then tri-state buffers are required. IC33, 34, 35 take care of this, receiving their control signal from the TANEX expansion board.

The remaining components provide additional decoupling and timing signals which are required by the complete system.

PARTS LIST

Resistors (all ¼W, 5%)	
R15,17	10k
R16	1k0
Capacitors	
C17	1n0
C18	47n
Semiconductors	
IC20	TANBUG ROM (or XBUG)
IC29,30,31	74LS74
IC32	DM8678CAE
IC33,34,35	74LS367
Q2,3	BC184

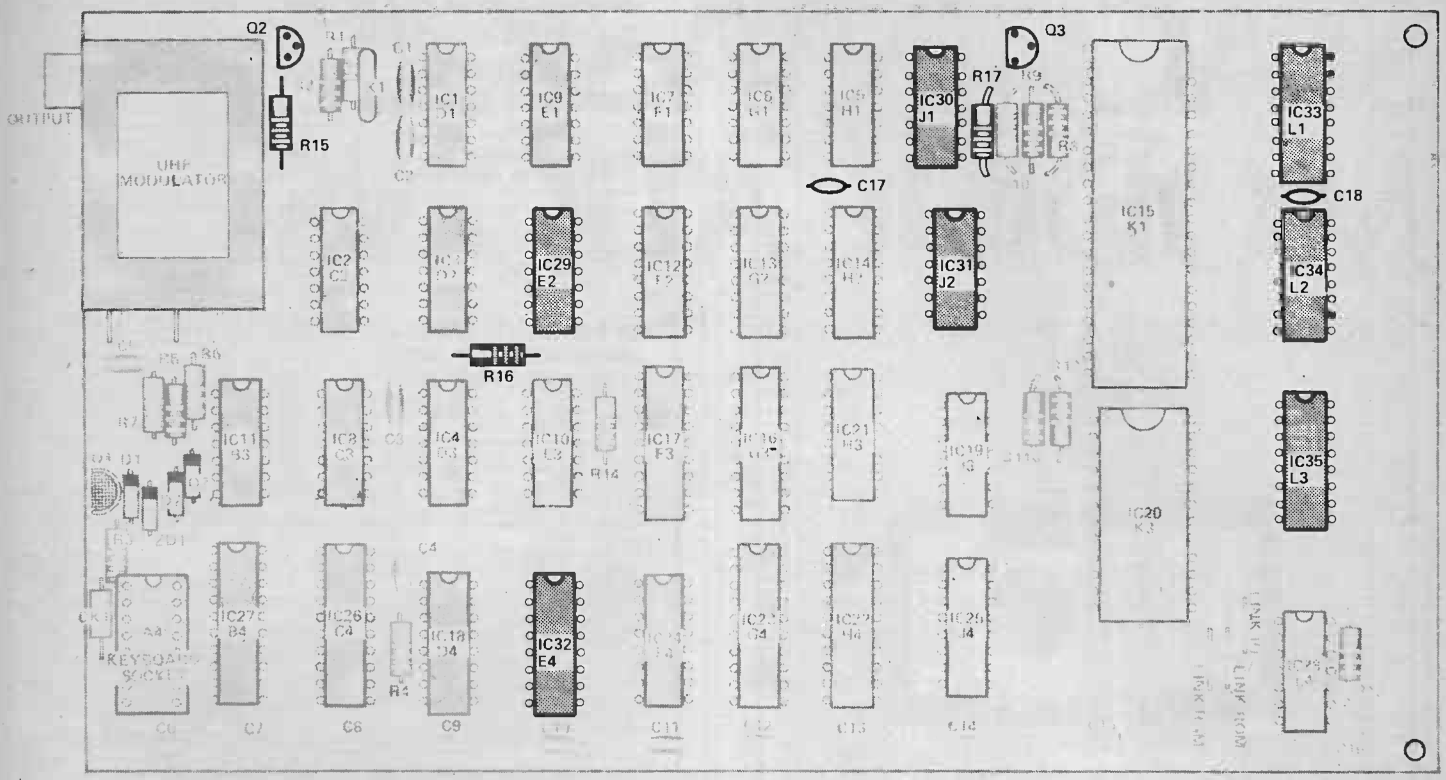


Fig. 1 The additional components are shown in black on the original (fainter) overlay. Tangerine use letter/number combinations to identify the IC sockets and these are indicated on each IC.

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