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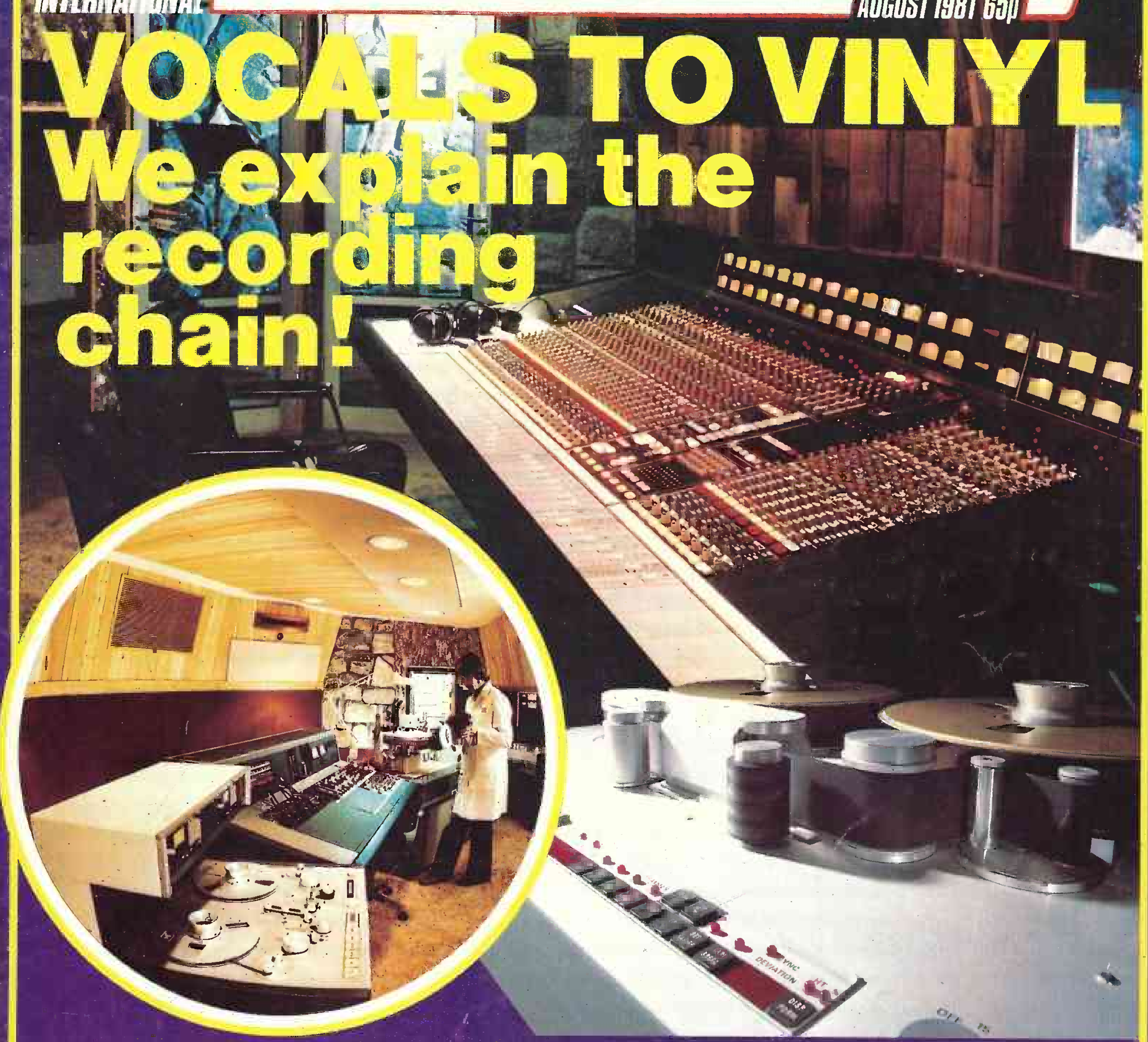
AUGUST 1981 65p

VOCALS TO VINYL

We explain the recording chain!



BUILD YOURSELF A WATCHDOG

It's an alarming experience!

DIY HEARTBEAT MONITOR

DISCO HANDCLAP SYNTHESISER PROJECT

MERIDIAN DETHRONED – A NEW AMP CHAMP?

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

complete with its own power supply arrangement, can then be coupled to the output of the unit so that it activates when relay RLA closes.

If you decide to mount a re-entry switch on the front door of the house (so that you can enter the building without activating the alarm), take care to conceal its wiring. If required, a number of re-entry switches can be wired in parallel so that, for example, the system can be temporarily disabled from either the front door or the main bedroom.

The alarm system is very simple to use. The panic and fire alarm side of the circuit is permanently enabled and can be operated at any time. The anti-burglar section is enabled only when the main key switch is set to the on position. If LED2 lights at the moment of turn-on it means that part of the burglary sensor system is either open or closed when it should not be, possibly due to an open door or a chair resting on a pressure mat, for example. The fault must be rectified before the system is put to full use.

If you leave the house or pass through a protected area after turning the system on, remember to use the re-entry facility before returning to the unit, or you'll sound the alarm and annoy the neighbours.

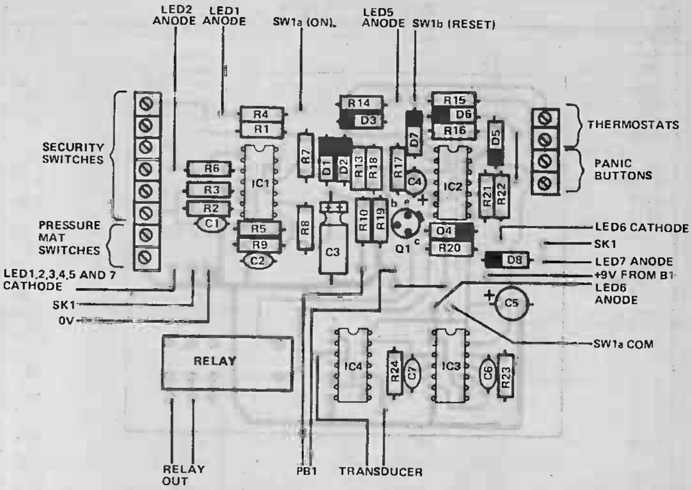
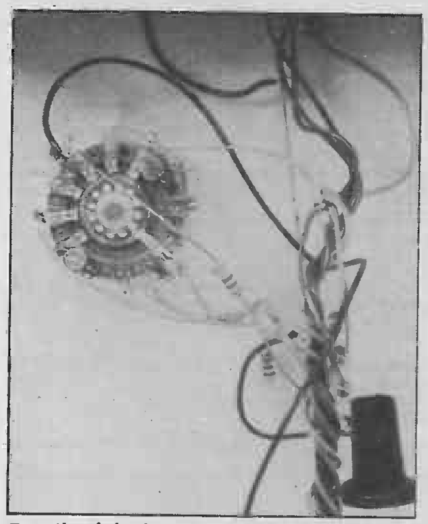


Fig.3 Overlay for the main board. Note that R11,12 and 25 are mounted off-board.



Details of the keyswitch wiring. The off-board resistors can also be seen.

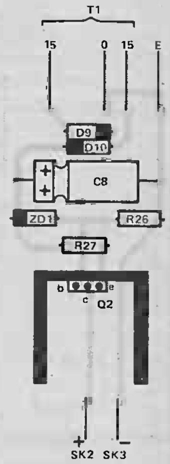


Fig.4 The charger overlay.

ETI

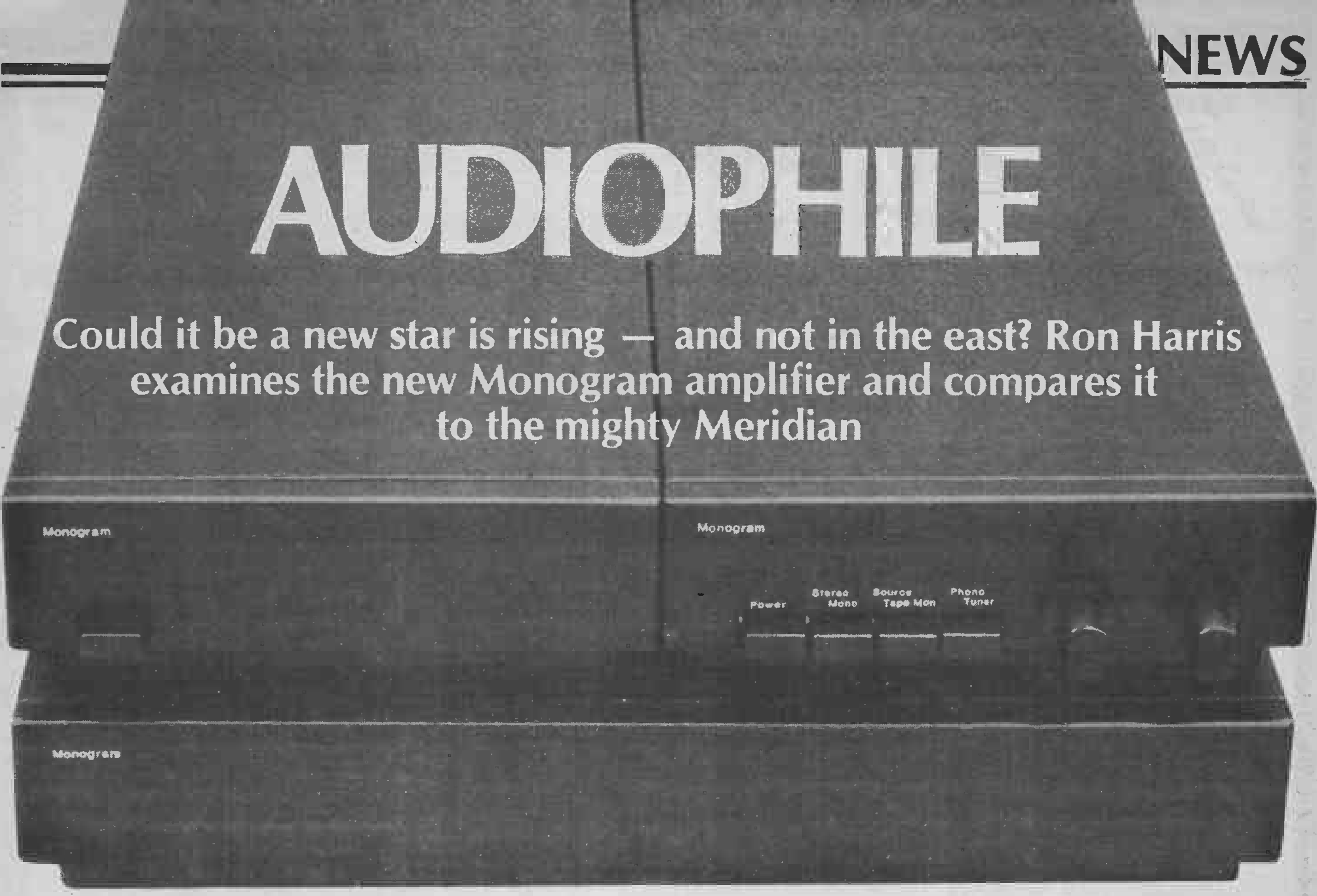
PARTS LIST

Resistors (all ¼ W, 5%)	
R1,13,18,19	10k
R2,23	1M0
R3	56k
R4,6,11,12,14	820R
R5	100k
R7	470k
R8,9	4k7
R10	47R
R15	4M7
R16,24	22k
R17,22,25	1k0
R20	18R
R21	12k
R26	2k2
R27	22R
Capacitors	
C1,2	100n ceramic
C3	220u 16 V axial electrolytic
C4	100u 10 V tantalum
C5	100u 16 V electrolytic (PCB type)
C6	220n polycarbonate
C7	10n polycarbonate
C8	470 40 V axial electrolytic
Semiconductors	
IC1	4093B
IC2,3	4001B
IC4	4011B
Q1	BC109
Q2	TIP32A
D1,4,8,9,10	1N4001
D2,3,5,6,7	1N4148
ZD1	BZY88 2V7
LED1,4,7	0.125" green LED
LED2,3,5,6	0.125" red LED
Miscellaneous	
T1	9-0-9 @ 75 mA
SW1	two-pole six-way wafer key switch
SK1	DC socket and plug
SK2,3	4 mm sockets (and plugs)
TX1	piezo-electric transducer
RLA	6 V DPCO, PCB-mounting

PCB-mounting terminal blocks; Verocase (order code 202-21031G); case for charger unit (order code Samos 002).

AUDIOPHILE

Could it be a new star is rising — and not in the east? Ron Harris examines the new Monogram amplifier and compares it to the mighty Meridian



Tilting at windmills has been a family pastime in the Harris household for some years. Even back in the dim dark days of school I was scoring zero out of ten for a Physics exam paper which *proved* that Einstein was wrong! Anything considered immovable is there for one reason — to be shifted rapidly one inch to the left. Including reputations.

You see I reckon I've found an amplifier that out-performs *anything else* on the market.

Now 'anything' is an all-encompassing kind of word, and by definition must include that which is presently held in reverent awe by the 'fanatical few' at the altar of ultra-fi. In Britain today that means only one name — whisper it softly — Meridian! Hence the windmill syndrome.

It doesn't help the cause any that the unit in question sells for around £200 less than the sacred object.

The amplifier I refer to is the Monogram 106/109 combination. The '106' part is a high quality preamp, available in either moving-magnet or moving-coil livery, and the '109' tag attaches to a power amplifier (with separate PSU) of well over 100 W RMS per channel. This month's Audiophile considers each in turn and compares the Monogram directly against the Meridian 101/105. The outcome of this you already know, as it was the remarkable difference between the two that prompted the opening — and cast doubt upon the supposed omnipotence of the much recommended Meridian!

Simply Preamplifies?

The 106 has been designed to provide a versatile high quality control unit which is capable of getting the best from a wide range of signal sources. The RIAA is exceptionally accurate — within 0.25% — and the disc input is a three-stage design, using low feedback overall and buffering the cartridge from the equalisation networks. This means that the cartridge will see a non-frequency dependent loading at the input. The value of this (constant) load can be user-programmed using two DIL switches on the PCB.

This in itself is a very useful feature, with some of the new generation of high-output moving coils benefitting in particular. Three values of both resistance and capacitance can be selected. See Table 1 for the figures behind the facts.

Inputs provided for, other than disc, are tape (with monitor) and tuner. No tone controls are present and the volume control has a 'step' action, although it is not an actual step-attenuator.

TABLE 1

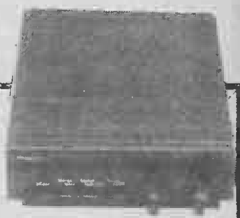
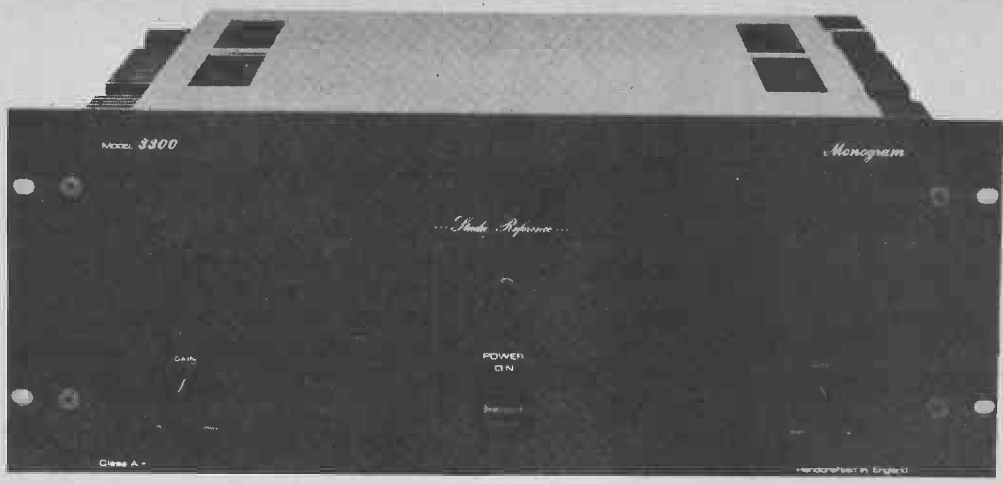


Table 1. Test results: 106 Preamp

Inputs	
Moving magnet:	S/N ratio; better than 80 dB (ref 10 mV) Overload; better than 250 mV (20 Hz-20 kHz) Sensitivity; 3.0 mV Input impedance; switchable:- R- 47k, 100k, 33k C- 120pF, 270pF, 290pF RIAA accuracy; better than 0.2%
Tuner/Tape:	S/N ratio; better than 80 dB (ref 150 mV) Sensitivity; 150 mV Input impedance; 50k
Output level:	7 V maximum
Freq response:	20 Hz-40 kHz (±0.2 dB)
THD (total preamp):	better than 0.01%
Price:	£159 (MM) £179 (MC)

A moving coil module is also available with 40 dB extra gain. This retains the switchable input impedance, with a range of 33R-100R.



Although not strictly part of the review intended, I could not resist the prospect of a chance at the 3300 power amp. Rated at slightly less than twice the power of the 109, it promised to be an interesting beast under test.

As you can see from the test results it did not disappoint. The PSU contained herein is rated in excess of 1 kW in order that constraints upon this part of the design never become a stricture on the amplifier as a whole.

The sound quality produced by the 3300 is little different to that of the 109, as might be expected. If anything I'd give the 109 the slight edge under dynamic conditions, using a domestic system. Maybe the separate supplies, although individually lower rated impart to it a better coherence under hard drive. Otherwise all comments made herein on the 109 apply equally to the 3300.

The unit was primarily designed for studio usage, of course and so would be much more rugged — witness the massive metal work, ¼" jack connectors (or Cannons), fan mounting holes etc. etc.

The normal recommended retail of the beast is around £650, but interested parties should contact Monogram for the present price which is more £200 less than this, since the model is being phased out to allow the company to concentrate on the 106/109 range. At under £400 the 3300 is excellent value indeed for those needing a lot of power into any loading conceivable.

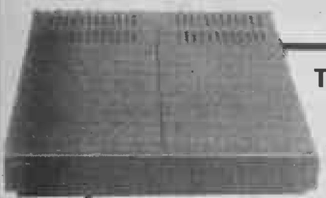


TABLE 2

Table 2. Test results: Monogram 109 and 3300 power amplifiers

	109	3300
Power output: (both driven per channel)	120 W (8R) 274 W (4R) 430 W (2R)*	215 W (8R) 347 W (4R)*
Transient delivery:	>300 W (4R)*	>400 W (4R)*
Half-power bandwidth:	10 Hz-70 kHz	10 Hz-50 kHz
THD:	<0.05% (20 Hz-20 kHz at 120 W)	<0.05% (20 Hz-20 kHz at 200 W)
Damping factor:	>150 (20 Hz-20 kHz) >400 (<1 kHz)	>100 (20 Hz-20 kHz) >400 (<1 kHz)
Hum and noise:	−100 dB (ref 100 W)	−120 dB (ref 200 W)
Input impedance:	10k	15k-22k (gain setting)
Input sensitivity:	0 dB (775 mV)	0 dB (775 mV)
Price:	£340 (including PSU)	£POA (see above).

*Burst power and sustained power into 2R is given in this form because both these amps exceeded my test rig capabilitilies. From current measurements it would appear that the 109 delivers in excess of 450 W into 2R and the 3300 around 560 W. Both figures are the highest I have measured from a domestic amplifier and illustrate an unrivalled ability to provide undistorted and unclipped power into any load.

As the output impedance is less than 600R, long leads to power-amps are no problem, as the load will not affect frequency response.

Overall the standard of construction is very high indeed and a clever PCB layout minimises both interwiring and noise/hum paths within the box. A toroidal transformer is fitted, to aid the hum figures still further.

As you can see from the spec the standard is a high one — and the 106 exceeded that spec on every parameter I measured. In some cases it bettered the limits of my test gear, thus leaving me simply to agree that the figures are 'reasonable'!

Powerful Amps

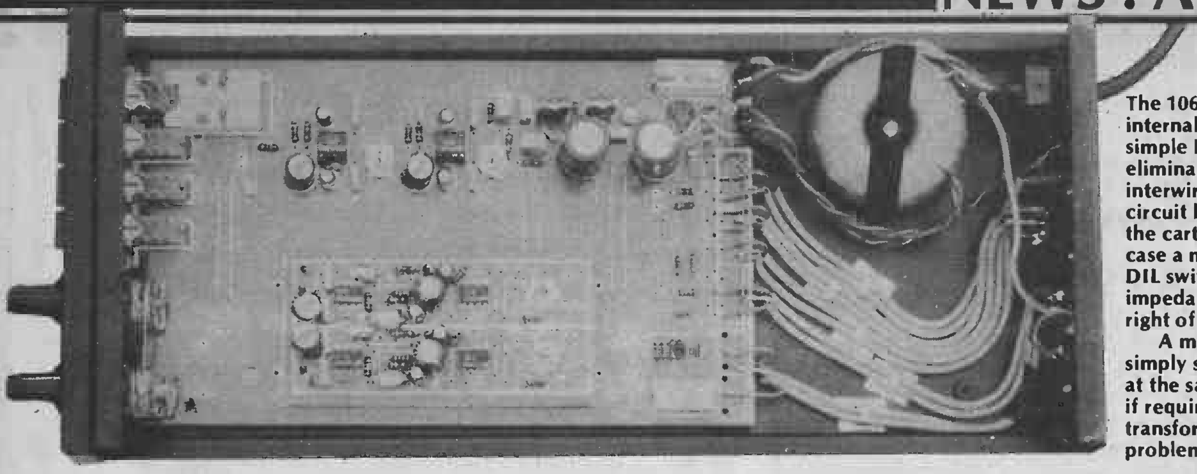
The 109 power amps are separately cased, with the PSUs also residing in their own matching case. The two amp cases share a common front panel and are linked together by the casing system. Sensible power connectors are employed, and good large screw terminals are provided for speaker connections. Little points maybe, but correct connectors indicate that thought has been expended on all aspects of the design.

Input is via phono plugs — and all preamp outputs will be of this variety also. Being a confirmed hater of all things DIN, I find this most encouraging!

The PSU for each channel — they are separate — is rated at more than that required for 100 W output (we will return to this later). Toroidal transformers are again present and the constructional standard is impeccable. It is nice to be able to praise a British company for finish and construction for a change — too often is this the sole preserve of Oriental offerings.

Circuitry is of a mode entitled 'enriched bias' which means that for 90% of the available output, the amplifier runs in pure class A and will only switch to AB at powers in excess of that 90%. In this way class A sound can be obtained without the otherwise necessary hardware.

It is worth noting that Monogram marketed a design using this configuration many years before the Japanese production



The 106 revealed in all its internals! Note the deceptively simple PCB layout which eliminates much of the interwiring usually required. The circuit block at the lower edge is the cartridge amplifier, in this case a moving magnet input. The DIL switches for setting impedance can be seen to the right of this.

A moving-coil board can be simply substituted into the PCB at the same location (centre left) if required. The toroidal transformer minimises hum problems.

machine caught up (the 3300 'Professional Amplifier' rated at over 200 W RMS).

I had a chance to listen to this unit also and will return to it after completing the task in hand, the 106/109.

Peaks, Points And PSUs

The rating of 100 W RMS proved to be very conservative. I measured in excess of 140 W into 8R, and 274 W into 4R. Into 2R I hadn't the equipment to determine the power accurately, but it was over 300 W! This power was *sustainable* power, not some transient figure.

On musical peaks into a real load, the Monogram proved capable of over 250 W across the entire audio spectrum: a performance which guarantees that it will drive any speaker to any level desired! I know of no other power amplifier currently available which can equal the 109 here.

I tested the Meridian 105 in the same manner, only to have it blow fuses repeatedly. It proved capable of peak powers of 200 W into 4R, but only for the shortest possible time period. So short, in fact, that a bass drum or such loses 'punch' into a difficult speaker. Into 2R it would not perform at all well and I gave up when my fuse supply ran out!

This is not meant as a particular indictment of the Meridian, as it itself is better than most other commercial amps. It is mentioned merely to illustrate the superiority of the Monogram circuit. It has the current capability and the Meridian does not.

The speakers used in the test were my own KEF 105 II and the notorious Linn Isobariks. If you think that 2R is an unrealistic impedance into which to test amplifiers, then you should see what the impedance of the Linn gets up (and down) to across the audio band. By comparison my tests were kindness indeed!

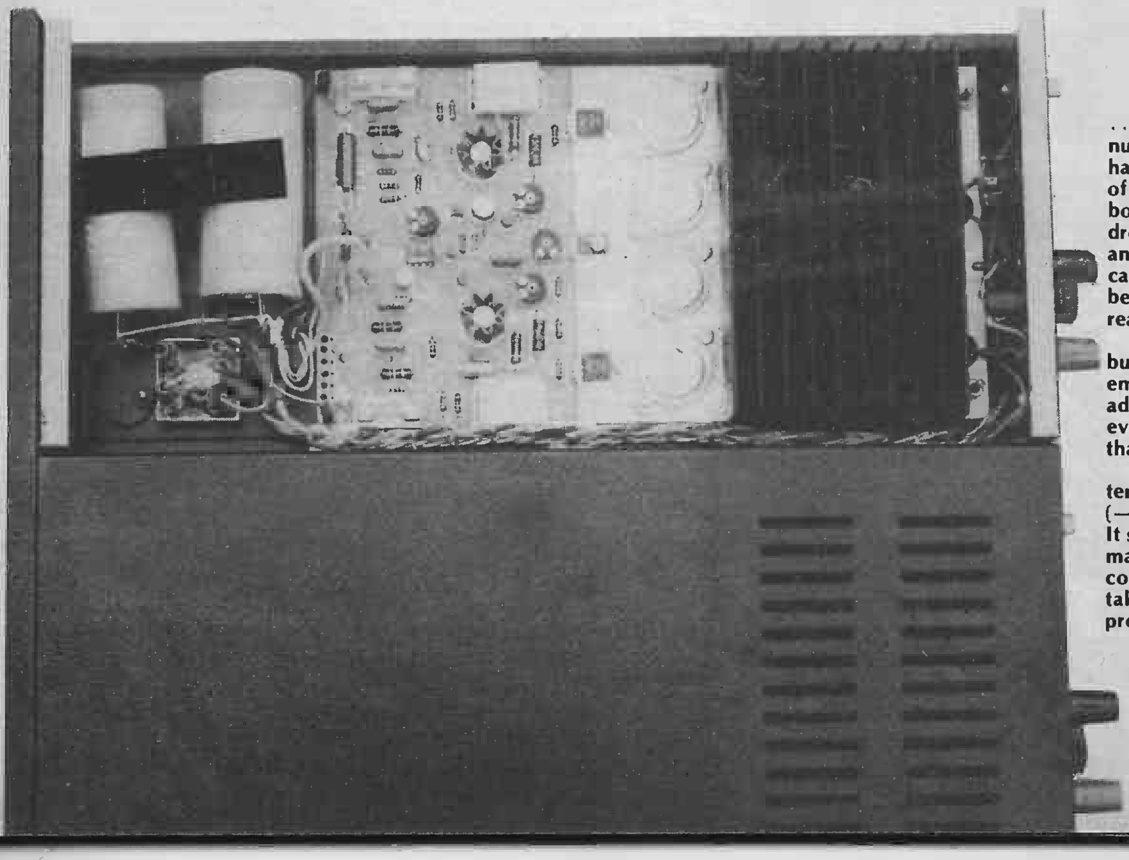
Putting It All Together

And so to some music at last. I'd spent longer than usual at the test bench this month, simply because the 109 refused to admit defeat. With two test loads in ruins *I* gave up and headed for the hi-fi.

Initial listening revealed a beautifully detailed and extended sound, with a true, solid bass response and well extended treble. On the KEF 105 the Monogram produced the best bass I've ever heard on a home system, showing outstanding control over the bass cones — as witnessed by the high damping factor.

Mid-range was even more of a revelation. Compared to my usual reference (and the Meridian later borrowed for these tests) it was like taking away a veil from the music. Instruments became easier to differentiate and individual lines easier to follow. Try following a single voice in a massed choir, or an individual hand-clap in audience applause, and you'll see what I mean. Treble was clear and sharp but never hard or fatiguing.

It was at this point that I decided to go in search of a more widely meaningful comparison. My own 'beefed up' Lecson has been tuned to give me what I consider a comparable



. . . and the 109 similarly denuded. As you can see no space has been wasted! The advantage of running low level AC into the box fron the PSU is that line drops become insignificant and any interference picked up on the cables stands a good chance of being rectified to death before reaching the PCB.

Heatsinking may look small, but with the circuit technique employed, it proved more than adequate. Even after a heavy evening's work it remained less than incinerating!

Note also the hefty screw terminal on the rear of the amp (— the end nearest those words!) It still surprises me how many manufacturers employ coupling connectors which simply will not take cable of adequate proportions.

performance to anything I'd heard up to now, but simply pointing out that the Monogram betters this is of little significance to the market place as a whole. Besides which I had the feeling that the Monogram is something rather special and how better to prove it than by comparison to the best?

Comparing Directly

An A-B was set up using the KEF 105 IIs, and a Thorens TD160S with the Ortofon MC30 and Dynavector DV20A II pick-up systems. Right from the start it proved to be a one-horse race. Bass on the Meridian was less extended and nowhere near as controlled. The mid-range was simply less convincing all around, being less detailed and harder. The treble sounded thin and splashy when directly compared.

Reasons? The Monogram delivers power into the load in an almost frequency independent manner. It handles low impedances with ease and has the power reserves to be free of clipping effects at very high levels.

I would strongly suggest, if you are in the amplifier market at present, or were contemplating an upgrade into this league at all, that the Monogram is the one to listen to first.

In case you have any trouble getting a demo, Monogram have said you can give them a ring on 01-573 1566, at 281 Balmoral Drive, Hayes, Middlesex. Tell them ETI sent you!

Follow-on

After last month's loudspeaker reviews, only one of the manufacturers took up my standing offer of a reply to a review. This was Tangent and their letter is given below.

Tangent Acoustics U.K. Ltd.
4 Viking Way, Bar Hill,
Cambridge, CB3 8EL, England.

Modmags Limited
145 Charing Cross Road
LONDON
WC2H 0EE
26th May, 1981

Attn. Mr. R. Harris

Dear Mr. Harris,

Many thanks for the copy proofs of the TM3 review.

As mentioned on the telephone you have certainly captured the design philosophy of the TM3. Tangent do not regard the TM3 as a high dB reproducer, although I must confess we have never encountered the disappearing effect! Maybe some manufacturers have shares in 'aspirin'.

In the TM3 we have designed a speaker capable of producing Hi-Fidelity at a volume level conducive to neighbourly harmony. Not all people who enjoy music live in detached houses on top of hills, or in the flight paths of Heathrow. Market research shows that the person willing to spend £125.00 on a pair of small speakers dwells in a normal house and usually with neighbours at the other side of a wall who may be larger in stature than the owner of the TM3.

As soon as we have the new XLR2s spare we will let you have a pair for future review.

Yours sincerely,
TANGENT ACOUSTICS UK LIMITED

P. HARDCASTLE

ETI

HEATBEAT MONITOR

Are your meditation exercises really working? Take a trip into the world of Zen and the Art of Electronics with this simple, state-of-the-art project. Design and development by Plamen Pazov.

The heartbeat rate is an important factor to be taken into consideration for many disciplines. From fitness tests to controlled relaxation, sleep monitoring and biofeedback, one needs a simple means of checking the cardiac rate. There are many ways of doing this ranging from the simple manual one-minute pulse count to expensive invasive methods and electrocardiography.

The manual method is simple enough: all you need is a clock. But it can only be done during a rest period and gives a mean value over one minute. One of the first points that will strike you if you build this project is the speed at which the heartbeat rate itself can change under internal or external influences. (Try kissing your girlfriend and you will see what I mean!)

At the high end of the market is the electrocardiograph (the doctor's equivalent to our oscilloscopes). It will display and/or record the tiny electrical signals arising from cardiac activity. The graph or display will obviously give much more information than just the beat rate; the more sophisticated monitors can even detect and actuate an alarm at a preset minimum or maximum.

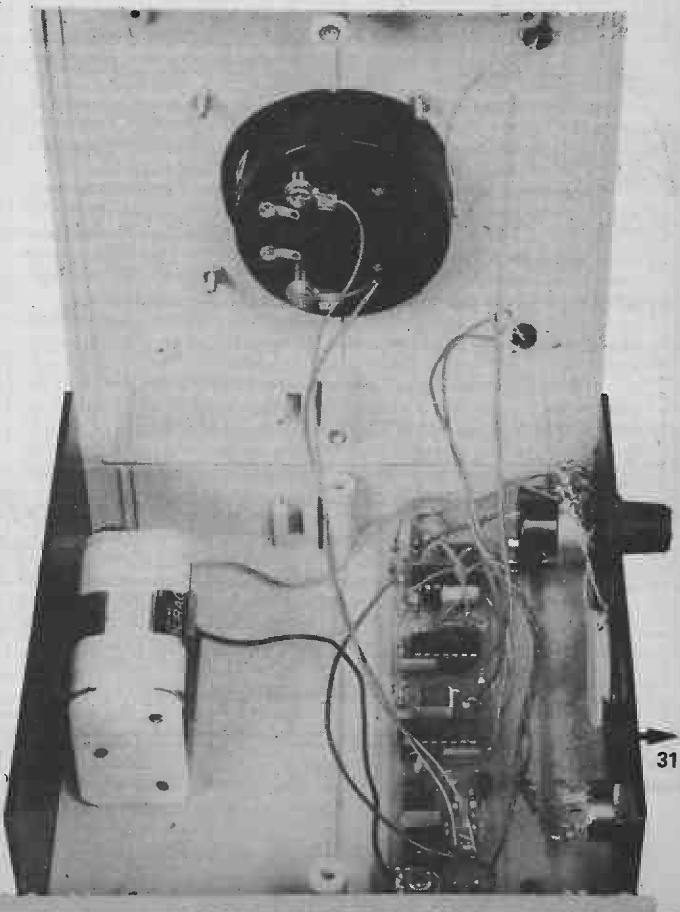
Another current trend, initially used for foetal heartbeat detection and monitoring, is based on an ultrasonic Doppler effect blood movement sensor. In this system an ultrasonic wave of a few hundred kilohertz is acoustically coupled through the skin and propagates through the body. A very small proportion of the initial energy is reflected by the moving bloodstream and picked up by a sensor; the reflected sound has its frequency changed slightly (Doppler effect) and the difference can be detected by beating the outward and return signals. The result is a frequency signal directly proportional to the blood speed. This method is more directional and less sensitive to external factors, but is more difficult to implement.

A much cheaper method, widely used in electronic blood pressure meters, is based on either microphone or pressure sensing, with very limited bandwidth (usually 0.5 to 5 Hz). Unfortunately, this approach is unreliable as it is very sensitive to movement and friction noises.

IR Instruments

There is another cheap method of detecting blood movement, but it is rarely used at present. It is based on the fact that blood absorbs infra-red light. Therefore the skin reflectivity at infra-red wavelengths is inversely proportional to the blood influx, and changes cyclically with each heartbeat. These reflectivity changes can be detected and used to evaluate the heartbeat rate. The big problem is isolating the tiny variations in the reflected light level from the much larger variations due to skin colour, pressure, ambient light and so on. In earlier models, the compensation was manually adjusted and the slightest variation in pressure, position or a passing shadow was enough to unbalance the sensing unit. This in turn resulted in somewhat unreliable instruments and accounts for the restricted use of the technique.

The monitor presented here is based on this sensing method but avoids its problems by the use of an autopolarising circuit, which will happily work over more than three decades yet will adjust itself in only a few seconds to any new conditions. This only uses one cheap package of uncommitted CMOS transistor pairs, the 4007, and a special sensor housing gives protection from ambient light and allows us to use a low power infra-red source. The measuring of the rate itself is made by locking a Phase-Locked-Loop oscillator (4046) onto the incoming frequency and displaying the control voltage of the VCO. A somewhat unconventional low-pass filter in the feedback loop allows an initial lock in three to four cycles and small variations are tracked in only one cycle. This means that an instantaneous frequency read is possible and rate changes from beat to beat can be measured. An audible and visual indication of the detected beats is available, as well as an 'unlocked PLL' indicator.



Multiple triggering is rare, even at high noise levels, due to a hysteresis level detector (4093) and the use of CMOS circuitry throughout means low current consumption (less than 50 mA, inclusive of the light source current) and therefore portability. A stabilised voltage supply (78L05) for certain parts of the circuitry ensures a stable operation of the preamp and reliable frequency reading for a battery voltage as low as 7 V.

Construction

The construction of this project should present no problems if the overlay and photographs are followed — the only precautions are to avoid touching the IC pins in case of damage by static, and to keep the PCB clean in order to minimise leakages around IC2. The sensor assembly is fairly easy to make, using a small piece of scrap aluminium and a Velcro band. The leads to the sensor should be shielded — see the photographs.

To calibrate the unit a reference frequency has to be fed to the input — the simplest method is to place an LED in front of the phototransistor. The reference frequency required is quite low and can be obtained either from a low frequency generator or by dividing down from a higher frequency (or the 50 Hz mains). With the reference at 0.66 Hz, adjust PR2 to give a zero reading on the scale — this corresponds to the minimum reading of 40 beats per minute. Now alter the reference to a suitable upper limit (eg 2 Hz gives an FSD of 120 beats per minute), and adjust PR1 to give FSD on the meter.

Many interesting experiments can be set up with this piece of equipment, although it must be pointed out that non-approved instruments cannot be used for medical purposes. Female readers wishing to participate in experiments on the heartbeat rate of staff members should apply to the editorial department at the usual address.

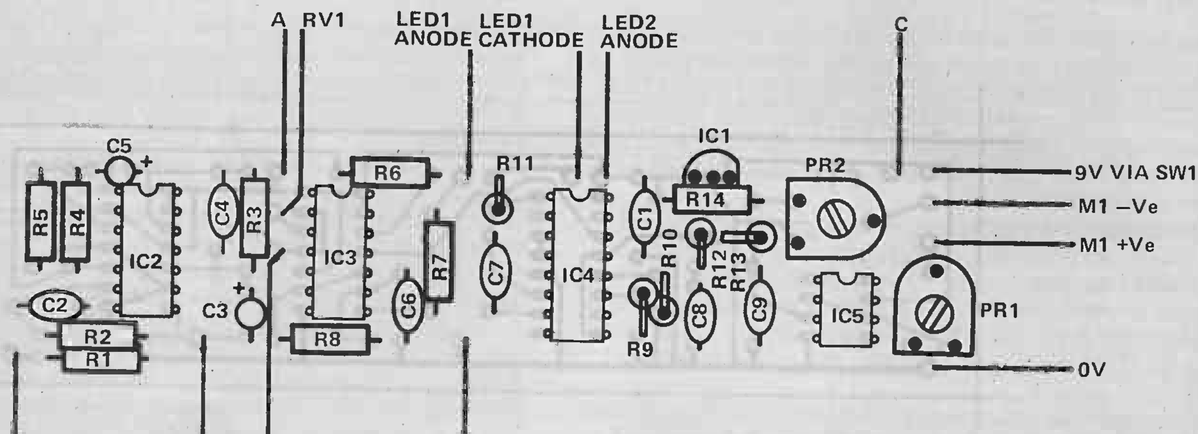


Fig.1 Component overlay for the Heartbeat Monitor. Several of the resistors are mounted vertically. In our unit the wires from points A-D were taken to a DIN socket on the front panel; the sensor was then connected via screened cable in a DIN plug.

HOW IT WORKS

The circuit consists of a signal extractor and conditioner, followed by a very low frequency meter.

Infra-red light is emitted from the LED in OCS1 and a small fraction is reflected back to the phototransistor from the bloodstream. The photocurrent from the sensor is sunk by IC2a, an enhancement mode MOS transistor which acts as an adjustable current source with a gate voltage of about 3 V. The R2/C2 network fixes the rate at which current sinking can alter: any variations with a higher frequency are passed on to the next stage via C3.

Transistors IC2b,c form an inverting pair polarised to half supply voltage by R3 — the gain of this stage is fixed by the ratio of C3 to C4. C5 transmits the signal to the next stage formed by inverting pair IC2d,e, polarised by R4 and offset by R5 so as to switch over only on the positive peaks of the signal. This is a high gain stage as the signal is fed directly to the inverter input.

All the previous stages are working at very high impedances as the photocurrent is of the order of nanoamperes. All the MOS transistors are working linearly in this circuit.

Any noise in the signal is removed by IC3a, a Schmitt NAND gate, which provides a clean digital impulse for the frequency meter. The output of IC3a also drives LED1, giving a visual indication of the beat rate, and gates on audio oscillator IC3b. Gates IC3c,d effectively double the driving voltage to the audio transducer.

The frequency measurement is made by locking IC4, a 4046B phase-locked-loop, onto the heartbeat frequency and displaying the control voltage of the PLL VCO on an analogue meter. This is just about possible if C7 is a good quality capacitor and with high resistances at R9 and R10. The linearity is very good, and the design of the low-pass network means that the instantaneous frequency of each beat can be displayed. A smoothed voltage from C9 is buffered by IC5 in a voltage follower configuration and both the maximum and minimum frequencies can be calibrated using PR1 and PR2 respectively.

BUYLINES

The only unusual component used in this project is OCS1, a TIL139 optically-coupled switch used as the sensor. This device is available from Marshall's and Watford Electronics. The PB-2720 is available from Ambit International.

You should be able to get Velcro from the haberdashery department of any large department store.

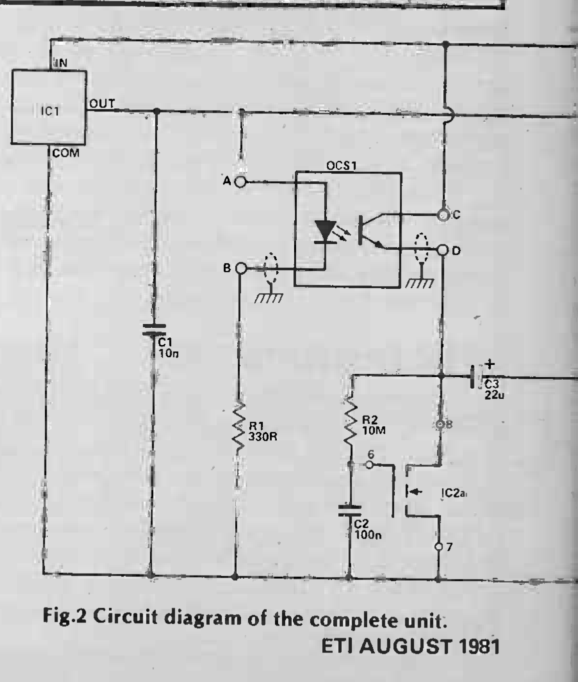
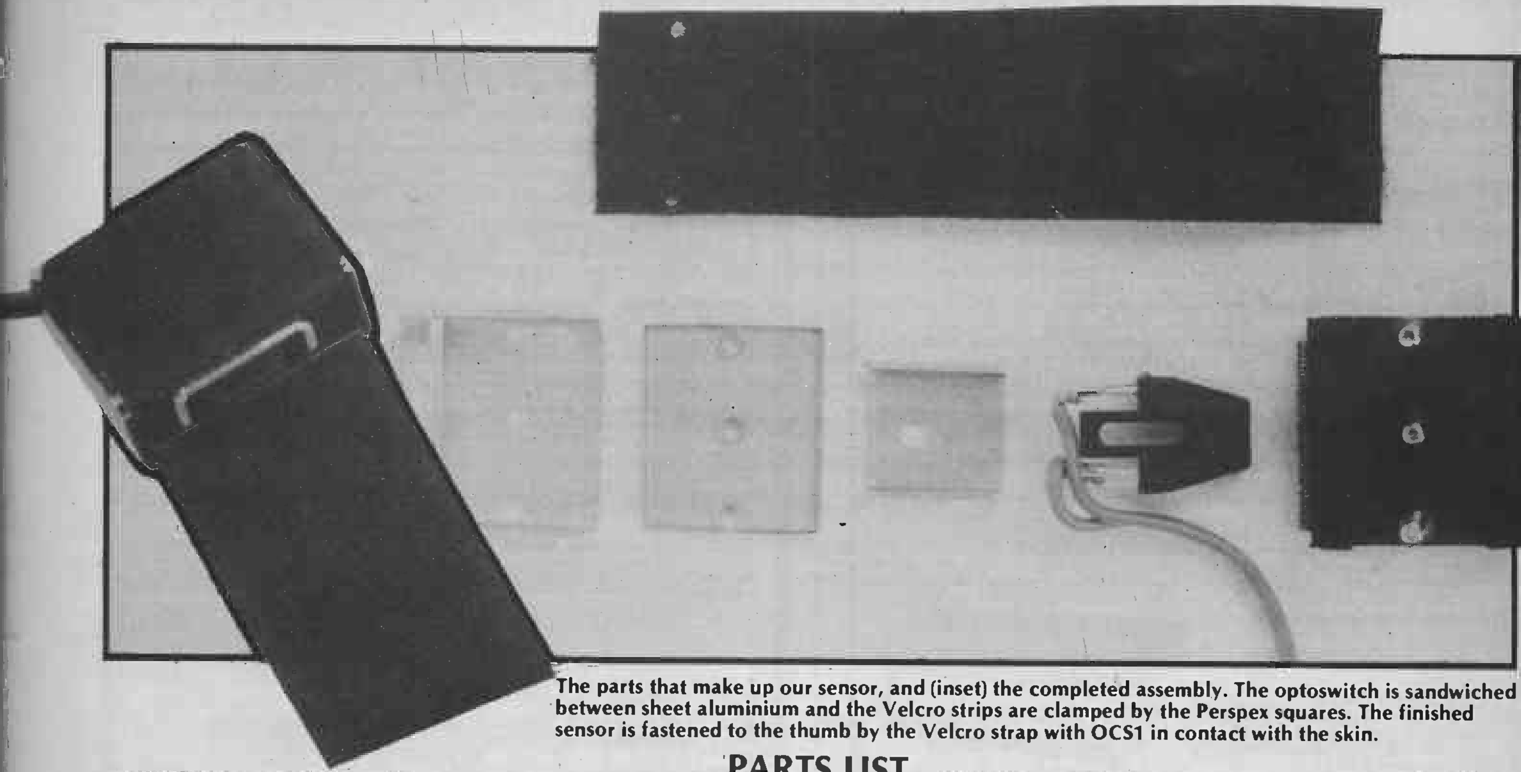


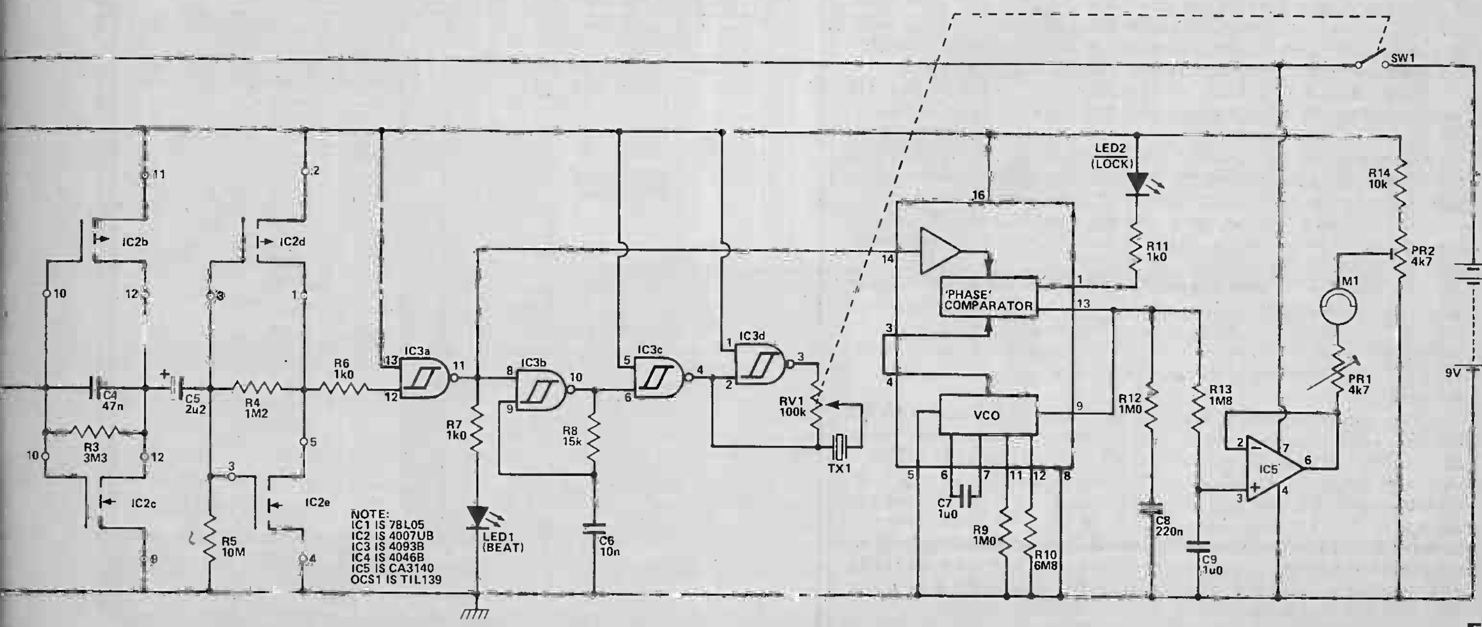
Fig.2 Circuit diagram of the complete unit.



The parts that make up our sensor, and (inset) the completed assembly. The optoswitch is sandwiched between sheet aluminium and the Velcro strips are clamped by the Perspex squares. The finished sensor is fastened to the thumb by the Velcro strap with OCS1 in contact with the skin.

PARTS LIST

Resistors (all ¼ W, 5%)		C3	22u 16 V tantalum
R1	330R	C4	47n polyester
R2, 5	10M	C5	2u2 35 V tantalum
R3	3M3	C7,9	1u0 polycarbonate
R4	1M2	C8	220n polycarbonate
R6,7,11	1k0		
R8	15k	**Semiconductors**	
R9,12	1M0	IC1	78L05
R10	6M8	IC2	4007UB
R13	1M8	IC3	4093B
R14	10k	IC4	4046B
		IC5	CA3140
Potentiometers		OCS1	TIL139
RV1	100k linear with integral switch		
PR1,2	4k7 miniature horizontal preset	**Miscellaneous**	
Capacitors		TX1	PB2720
C1,6	10n polyester	M1	1 mA FSD moving coil meter
C2	100n polyester	Battery holder (six HP7)	



ASTROLOGUE SPECIAL REPORT

In Part 2 of our Special Report, Ian Graham gives a technical run-down of Surrey University's amateur research satellite.

Last time I looked at the UOSAT project objectives and developments to date. Now let's see what it is designed to do. Due for launch in September by NASA (although the launch may be brought forward to July), UOSAT is the first satellite designed to transmit data (including pictures of the Earth's surface) in a form that can be displayed on a domestic television. It will carry a voice synthesiser with a 150-word vocabulary to speak the telemetry, experimental data and spacecraft operations in English. Most standard amateur narrow band VHF receivers will be able to listen in on 145.825 MHz with a simple fixed pair of crossed dipole aerials, ie you won't have to track the satellite across the sky.

Solar Science

The spacecraft also carries a number of scientific experiments including a series of beacons transmitting at different frequencies, two particle counters to provide information on solar activity and auroral events and a magnetometer (identical to those used on the Voyager missions to Jupiter and Saturn) to measure the Earth's magnetic field. Together, these experiments are designed to study how, for example, solar activity affects the transmission of radio signals through the ionosphere.

Control is provided by a powerful on-board computer based on the RCA 1802 microprocessor. The computer's instructions will be loaded from the Surrey University Command Station.

Say Cheese

An Earth-pointing camera will photograph an area the size of Scotland (300 miles square). The image will be formed on a solid state charge coupled device (CCD) and stored in the on-board computer until transmission to ground. Each picture will take three to four minutes to transmit. Normally you need very expensive detection, decoding and display equipment to produce a weather picture. However, UOSAT's signal will be transmitted in such a way that it can be received simply and displayed on a domestic television. The pictures will have a resolution of about 2 km with enhanced land features and land/sea boundaries. Experimental data in graphical form will also be available in the same way.

Power

Four solar panels made by Solarex (USA) will provide 27 W each when fully illuminated. Total average power will be around 17 W, which is supplied to the 14 V, 6 Ah Ni-Cd battery (90% efficient). Regulated power is supplied at +10 V, −10 V and +5 V (overall efficiency 87%). The average usable power for spacecraft electronics is around 11W5.

Telecommand

Two spacecraft control modes are available with 64 two-state commands:

(a) direct real-time control of spacecraft functions by ground command stations

(b) indirect, stored program control by the on-board computer following a command list loaded in advance from the ground.

Valid ground commands will override simultaneous commands from the on-board computer. There is also a high rate uplink for program loading.

Telemetry

Sixty analogue telemetry channels and 45 digital status points are available for transmission by VHF and UHF data beacons in the following formats: 1200, 600, 300, 110 and 75 baud ASCII, 45.5 baud RTTY (Baudot), 10 or 20 wpm Morse code and synthesised voice. Any pair of the above formats can be available via the VHF and UHF data beacons simultaneously. The 1200 baud telemetry option also has a Channel Dwell facility.

UOSAT's PCBs have been designed with the aid of a £25,000 computer, which reduces the design time for each board from about three weeks to three days. It has been loaned by Racal-Redac Ltd.

Data Beacons

Two VHF/UHF beacons provide primary engineering and experiment datalinks to the outside world using simple ground receivers.

Frequency	145.825 MHz	435.025 MHz
Modulation	NBFM/CW	NBFM/CW
Data format	AFSK(NBFM)	AFSK(NBFM)
Power output	450 mW	400 mW
Efficiency (total DC/RF)	45%	40%

Table 1. General data beacon (left) and engineering data beacon (right) specification.

The spacecraft microcomputer is based on the RCA 1802 MPU. Software is resident in dynamic RAM loaded from the ground via the telecommand link and can be altered from the ground during flight.

Experiments

Propagation studies: 7.001, 14.001, 21.001 and 20.001 MHz phase-referenced beacons are provided for ionospheric experiments. Two microwave beacons (2.401 and 10.470 GHz) will encourage SHF propagation studies.

Two particle radiation counters (one for electrons at > 40 keV and the other for protons at > 2 MeV) will aid in solar activity and auroral studies. A three-axis, wide range flux gate magnetometer will examine the fine structure of the Earth's magnetic field.

Edu K Shunal Experiments

An Earth-pointing CCD two-dimensional array will provide land and sea image data, transmitted by the general data beacon using minimum shift AFSK at 1200 bps line synchronous.

The image format is 256 X 256 pixels with 16 grey levels and is stored in an on-board 0.25 megabit memory. The camera optics will cover a 500 X 500 km area with a resolution of around 2 km. Land features and land/sea boundaries will be enhanced. This system will also be used to display telemetry and experiment data in graphical form.

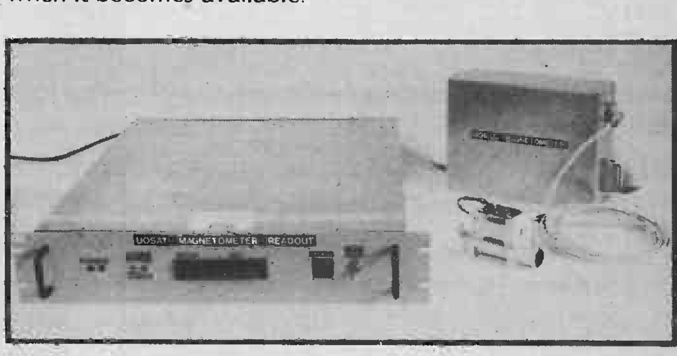
The data can also be spoken by the voice synthesiser in NBFM on either side of the data beacons.

The simple crossed dipole aerial necessary for reception of the spacecraft transmissions (top left). The storage and display electronics is shown top right. The 1500 km weather picture (bottom) received from a weather satellite was used to develop the system.

Systems Experiments

The Z-facet of the spacecraft will be kept aligned towards the centre of the Earth by passive spacecraft stabilisation using gravity gradient forces and active attitude control using a two axis magnetorquer acting against the earth's magnetic field.

Watch out for future Astrologue reports on the progress of UOSAT and a hardware review of the ground station receiver when it becomes available.



The magnetometer head (lower right) which will be carried on a 50 ft boom on UOSAT is identical to those carried on the Voyager missions to Jupiter and Saturn. The supporting electronics (top right) are mounted inside the spacecraft. The ground station equipment (left) will receive data on the Earth's magnetic field.

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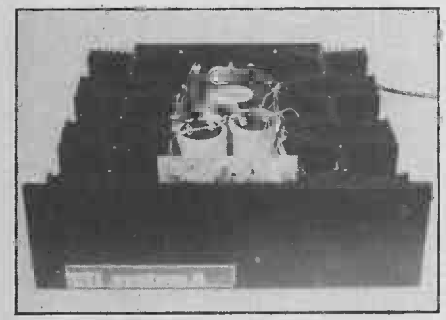

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SYSTEM A AUDIO AMPLIFIER

Get out the Bullworkers, part two of the System A series describes the Class A power amplifier. Big is beautiful! Design and development by Stand Curtis.

There is one amplifier configuration that is universally accepted as the ideal for audio use: Class A operation. Many early amplifiers operated in Class A, but as output powers rose above 10 W the problems of heat dissipation and power supply design caused most manufacturers to turn to the simpler, more efficient Class B arrangements and to put up with the resulting drop in perceived output quality.

The ETI System A applies the unchallenged excellence of Class A operation to the design of a reference amplifier free of the aberrations of commercially available models. Class A biasing is recognised as the ideal operating mode for an amplifier, offering the uncompromising accuracy demanded by dedicated audiophiles. The superiority of this amplifier depends on the output devices being constantly operated in their linear region, above cut-off and below saturation. Such operation results in the smoothest transfer function and the widest bandwidth.

The System A amplifier has a clarity and a tonal response that produces a superior perspective of depth with a sense of reality: instruments appear in precise position out of a silent background. The musical 'naturalness' of this amplifier is due to its lack of the constrictions of commercially desirable circuitry and the single-minded approach to a no-compromise sound quality.

Why Class A

The amplifier has an excellent technical performance even when operated in the conventional, but less desirable AB mode. With an open-loop (ie no overall negative feedback) distortion of around 0.1% (1 kHz) and a frequency response stretching well outside the audio band, the use of the large amounts of negative feedback (found in most commercial competitors) is completely unnecessary. However, extensive correlation between measurements and subjective performance using a wide variety of amplifier types led to the conclusion that Class A biasing is the optimum for audio amplifier performance.

When biased to Class A, the transistors are always turned on, always ready to respond instantaneously to an input signal; Class B and AB output stages require a microsecond or more to turn on. Thus Class A operation permits cleaner operation under the high-current slewing conditions that occur when transient audio signals are fed into difficult loads.

The continuous operation of the output stage in the linear collector region results in a more desirable distribution of distortion harmonics than is possible in Class B or AB, because the non-linearities in the transfer curve are smoother and free of the abrupt transitions of Class B and AB. The gradual non-linearities resulting from Class A operation produce distortions of low orders; primarily second and third harmonics. These lower order harmonics tend to be far less offensive to the ear than high order harmonics, being far more musical in nature (they are predominant in the harmonic spectra of most musical instruments). Higher order harmonics tend to 'harden' the overall sound. Such is the linearity of the ETI Class A Amplifier that a mere 22 dB of gain reduction is made in the form of negative feedback.

Each amplifier is a completely separate self-contained mono unit. The use of mono amplifiers, while costly in terms of components, provides the maximum stereo signal separation under dynamic operation with complete freedom from cross-modulation effects, giving an improvement in subjective depth and accurate instrument imaging.

A glance at the photographs will also explain why each amplifier is made as a mono-block. A stereo version would be just too heavy, unwieldy, and hernia-inducing for even the most dedicated audio fanatic (but if you know different . . .). Ideally each power amplifier can be located next to its respective loudspeaker and connected to it by very thick but short leads, thereby avoiding the losses associated with loudspeaker cables (30 A cable is suitable).

Protection — A Racket?

This Class A power amplifier is totally free of the usual protection circuits with their unavoidable colourations, distortions, and current-limiting characteristics. Instead we use an output stage having an exceptional power capability for an amplifier of such a low rating. With its substantial heatsinking this amplifier is capable of sustained operation with difficult loads.

The ETI System A amplifier maintains complete control over the driven loudspeaker throughout its operating cycle. The true Class A operation avoids the inherent phase irregularity and inadequate current-sinking ability of comparable Class B and AB designs. The provision of an extremely low-impedance power supply gives the Class A amp a short-term current delivery and, equally important, current-sinking capability far in excess of any known Class AB power amplifier of similar rated output power.

Amp Of Substance

The output stage is quite substantial, using a total of six 250 W power transistors. Fairly 'old-fashioned' power transistors have been used (the MJ4502/802 family) in preference to some of the higher performance devices now available. They have been chosen because the die used to mount the semiconductor junction is of a large area; the device is quite rugged and can handle high currents. The short-term current capability of the output stage is, in fact, of the order of 90 A, somewhat in excess of the current capability of the wiring!

The power supply is equally substantial, using a 500 VA toroidal mains tranformer and two massive computer grade reservoir capacitors. These components are expensive but essential. The rest of the construction is equally massive with a steel chassis supporting six very large heatsinks. However, construction is straightforward provided that the builder has strong arm muscles, and circuit alignment simple — there are but two adjustments — quiescent current and DC offset voltage nulling.

Construction

The constructional layout shown in the drawings and photographs should be followed as closely as possible. (With such high currents flowing down the cable forms, problems can easily occur if too many changes are made.) The heatsinks and the power supply components are assembled onto the baseplate and wired up in accordance with the wiring diagram. The recommended wire types and gauges should be adhered to.

Close-up of fuse wiring on back panel.

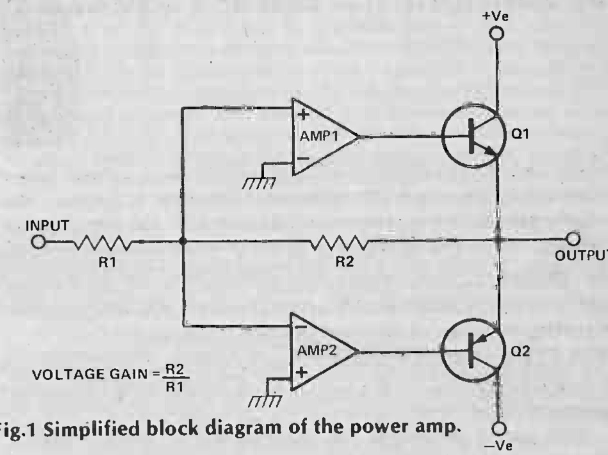


Fig.1 Simplified block diagram of the power amp.

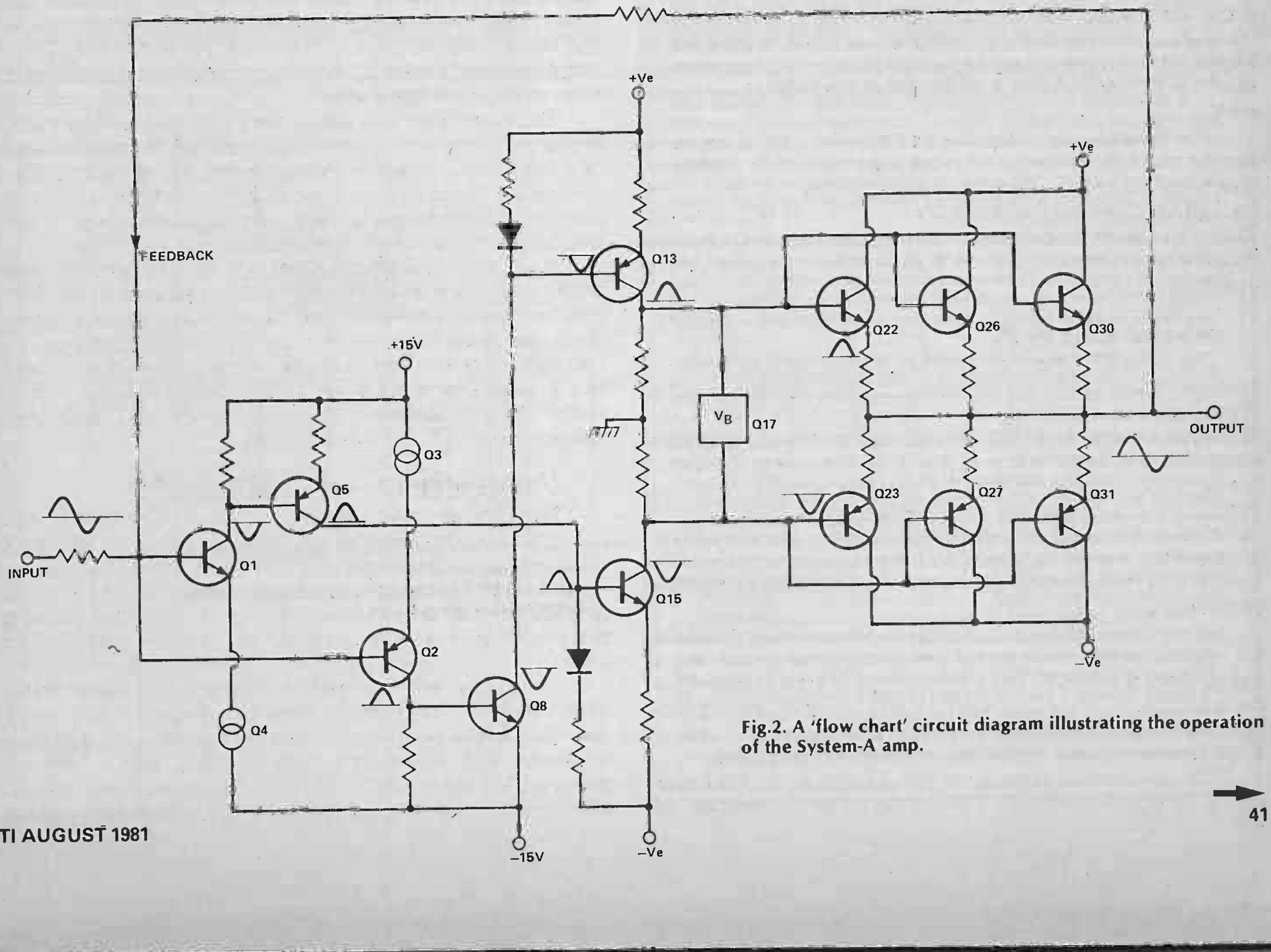


Fig.2. A 'flow chart' circuit diagram illustrating the operation of the System-A amp.

HOW IT WORKS

This amplifier is basically simple, as can be seen from the block diagram (Fig.1). Conventional complementary emitter-followers are driven by two separate voltage amplifiers arranged such that one handles the positive-going signals and the other the negative-going signals. A moderate amount of overall negative shunt feedback is then applied to stabilise the gain. To maintain a balanced and symmetrical treatment of the signal the performance of each 'sub-amplifier' should be the same. Furthermore these amplifiers have been designed to operate independently, without the need for the balancing signal currents from their 'mirror image' halves required in many so-called balanced amplifiers. The simplified circuit (Fig.2) shows that each sub-amplifier consists of two voltage-gain stages. This stage is of a novel arrangement previously used in a Meridian amplifier and subsequently in amplifiers by Lecson and Syntec. In the redesigned form here, the first stage consists of a complementary two-stage common emitter (Q1, Q5) whose gain is about x 2.3. The second stage is a current mirror stage (Q13) which drives the voltage across a load resistor tied to 0 V. The gain of this stage is about x 200. Thus the overall open loop voltage gain is of the order of x 460 and so, as the closed loop gain is x 26.7, the reduction due to negative feedback is x 17.2 or about 24 dB.

Looking now at the final circuit (Fig.3) it will be seen that the input amplifiers are powered from ± 15 V supply rails derived from resistor-zener regulators (R14-ZD1, and R15-ZD2). The current through the first stage (Q1) is held constant, at about 0.36 mA by a floating regulator stage (Q3, Q4) which also provides temperature compensation. The gain of this stage is set by emitter resistor R4 which provides some local negative feedback. The second stage (Q5) is loaded by two series cascode transistors (Q6, Q7), the first having its base tied to ground and the second having its base tied to the − 15 V rail. Thus the maximum collector voltage swing on Q5 is greatly reduced, so reducing the effect of the base-collector capacitance (Miller effect) which would reduce this stage's high frequency bandwidth. In summary, the presence of Q6 and Q7 improves the bandwidth and linearity. The load on Q7 is one half (Q12) of the current mirror and can be visualised as a resistor in series with a forward-biased diode. The second half of the current mirror is a common-emitter stage (Q15, Q16), a simple voltage amplifier except that its collector current equals (or 'mirrors') the collector current of the other half (Q12). This stage is made up of two transistors in parallel which share the current. This arrangement was found to improve the linearity of the stage. The other 'sub-amplifier' (Q2 to Q14) works in exactly the same way but with opposite polarity.

The output stage uses the conventional Darlington emitter follower arrangement, but with three parallel pairs of driver and output transistors. A transistor (Q17) is wired across the bases of the pre-driver transistors (Q18, Q19), providing a bias voltage to set the standing current in the output stage. Q17 is mounted on the heatsink with the aim of keeping this current constant regardless of temperature. Preset resistor PR2 is used to set the value of this current.

It will be seen that both the current mirror stages are driven from power supply rails that are different from those feeding the output stage. The same supply could be used but the signal in the current mirrors would clip well before the output stage, reducing the available output power. In fact the supplies to the current mirrors are made sufficiently high that these stages are still operating in their linear regions when the output stage clips.

The output DC offset voltage is set to zero by preset PR1 in the input stage. In theory there should be no DC offset at the output but, because of component tolerances and consequent mismatching, there always is. PR1 is arranged to make the current in the first stage of one 'sub-amplifier' either higher or lower than in the other and so null out any residual offset.

A simple low-pass filter is created by an R-C network at the input (R2, C2) to reduce the bandwidth of the signal below that of the open loop amplifier and thereby eliminate the generation of any transient intermodulation distortion.

The power supply has to deliver two split rails. The main supply to the output stage is nominally ±40 V at 4 A, derived from the main transformer windings and rectified by bridge rectifier BR1. This rectifier can get very hot so it is bolted onto the chassis. The secondary supply is a low-current ± 50 V to power the voltage amplifier stages. The output from the extra windings is rectified by BR2 and fed to smoothing capacitors C12 and C13. These capacitors are not wired between supply and ground but between the two supplies; this layout reduces their voltage rating.

The mains supply is fed to the transformer via an on-off switch, a fuse, and a thermal cut-out switch. Two neon indicator lamps are used. LP1 is connected between live and neutral and is the 'power' indicator; LP2 is connected across the thermal cut-out. If this cut-out opens the full supply voltage is applied across LP2 which then illuminates as an 'overtemperature' indicator. (This indicator has never operated yet in the prototypes.) Care should be taken to adequately sleeve and insulate all mains wiring and terminals to ensure safe and reliable operation.

Any bare wire ends should be sleeved using silicone rubber sleeving. This may seem an extravagance but your opinion will change shortly after a short-circuit wipes out £18 worth of transistors! A substantial soldering iron will be needed to solder together the power supply components. The use of a low-power iron will usually result in a selection of dry joints on these connections.

The coil L1 is wound onto the body of R40. This is not a critical procedure — about 17 to 20 turns of enamelled copper wire should do nicely. The gauge can be anything you have to hand, from 20 to 26 swg. Use some lacquer or epoxy to hold the wire in place on the resistor, scrape the enamel off the ends of the wire and solder them close to the resistor. The whole thing can now be soldered in place on the board.

Particular care should be taken in mounting the power transistors. Good quality insulating washers and bushes should be used and a generous smearing of thermal paste is essential. These transistors should be bolted to the heatsinks very tightly to ensure good thermal contact at all temperatures.

Assembly of the printed circuit board is straightforward enough using the component overlay as a guide. As usual, particular care should be taken to confirm the polarity and alignment of all capacitors, diodes and transistors; and to avoid putting mechanical strain on any of the components. After assembly the board should be checked on the copper side for dry joints and solder bridges. Such defects on power amps usually result in an expensive bang, so don't skip this admittedly tedious chore.

One final point regarding construction. Once the amplifier has been completed and tested, it should be switched on and allowed to reach its normal operating temperature (about 20 minutes). The amplifier should then be switched off and all the screws tightened up. Differences in thermal coefficients of expansion can result in some of the screws becoming slightly loose, particularly those holding the heatsinks to the top and bottom covers.

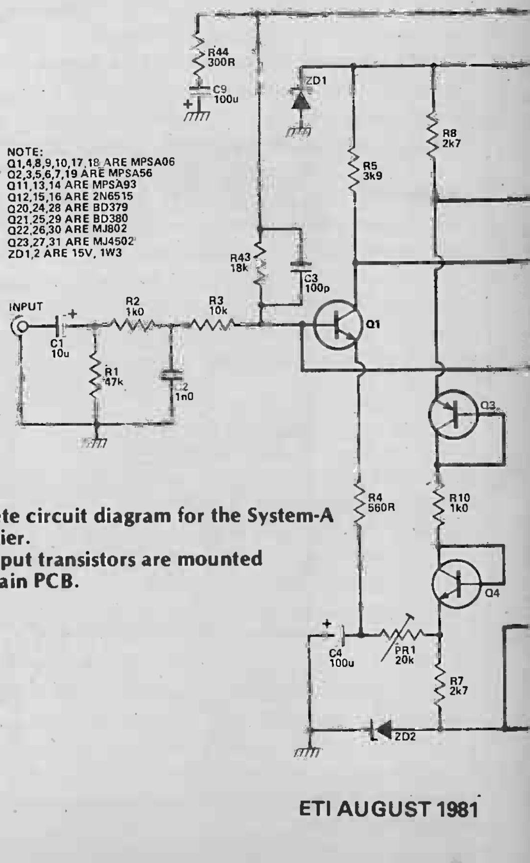
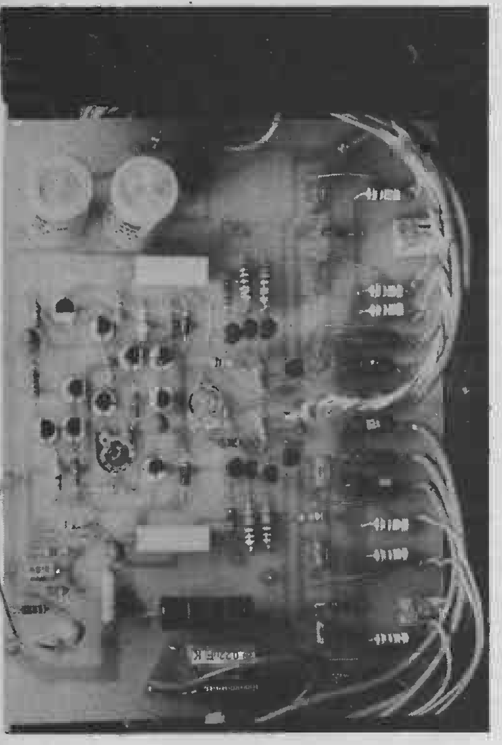


Figure 3. The complete circuit diagram for the System-A class-A power amplifier.
Note that the output transistors are mounted remotely from the main PCB.



Left: a detail shot showing how the PCB is wired into the case.

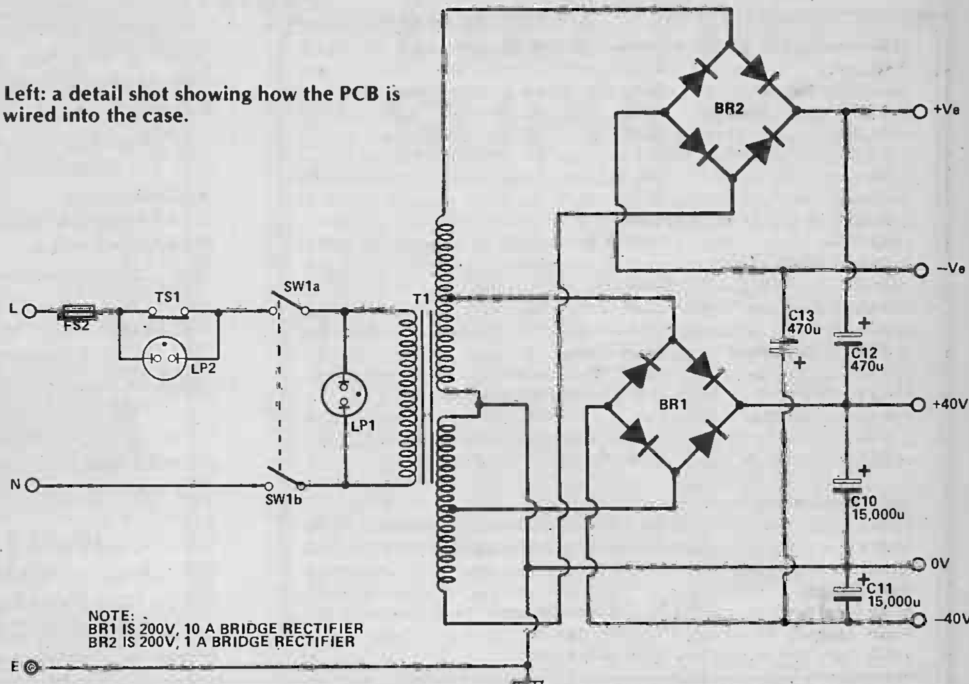


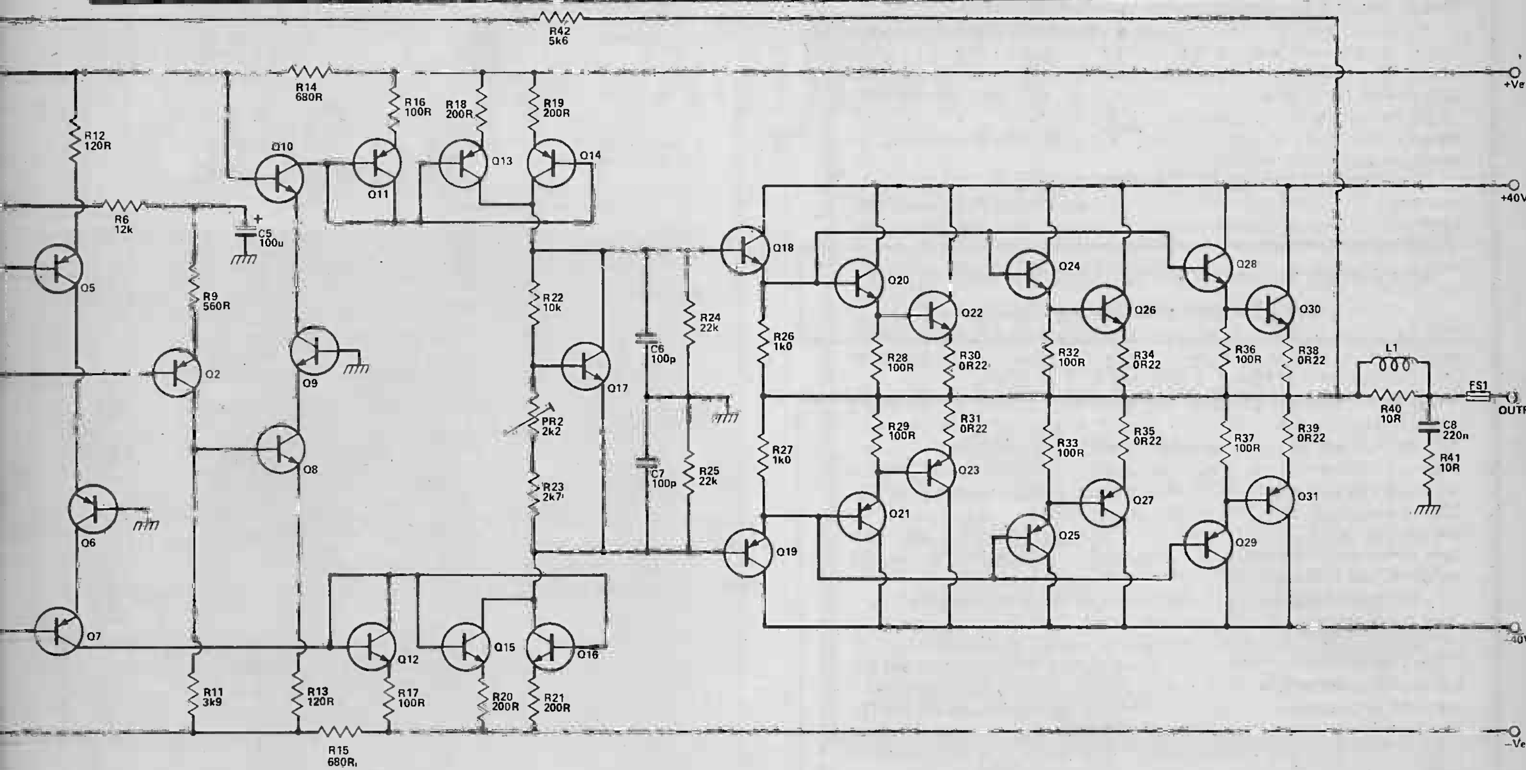
Figure 4. The PSU circuit used to drive the power amps.

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.



PARTS LIST

Resistors (all ¼W, 5% except where stated)		C4,5,9	100u 6V3 tantalum
R1	47k	C6,7	100p miniature ceramic
R2,10,26,27	1k0	C8	220n polycarbonate
R3,22	10k	C10,11	15,000u 50 V electrolytic (Sprague type 36D)
R4,9	560R		
R5,11	3k9	C12,13	470u 63 V electrolytic (PCB type)
R6	12k		
R7,8,23	2k7	**Semiconductors**	
R12,13	120R	Q1,4,8,9,10,17,18	MPSA06
R14,15	680R 4 W	Q2,3,5,6,7,19	MPSA56
R16,17,28,29,32,33,		Q11,13,14	MPSA93
36,37	100R	Q12,15,16	2N6515
R18,19,20,21	200R	Q20,24,28	BD379
R24,25	22k	Q21,25,29	BD380
R30,31,34,35,38,39	0R22 2W5	Q22,26,30	MJ802
R40	10R 1 W	Q23,27,31	MJ4502
R41	10R 2 W (not wirewound)	ZD1,2	15 V, 1W3
R42	5k6		
R43	18k		
R44	300R	**Miscellaneous**	
		SW1	DPST mains switch
Potentiometers		TS1	Thermal cut-out switch
PR1	20k miniature horizontal preset	LP1	Red neon
PR2	2k2 miniature horizontal preset	LP2	Orange neon
		FS1	1¼" 5 A—10 A (to suit loudspeaker)
Capacitors		FS2	20 mm 3.15 A
C1	10u 35 V tantalum	Toroidal transformer, 1¼" chassis-mounting holder, 20 mm panel-mounting holder, phono input socket, loudspeaker screw-terminals, chassis and heatsinks, mounting hardware.	
C2	1n0 polystyrene		
C3	100p polystyrene		

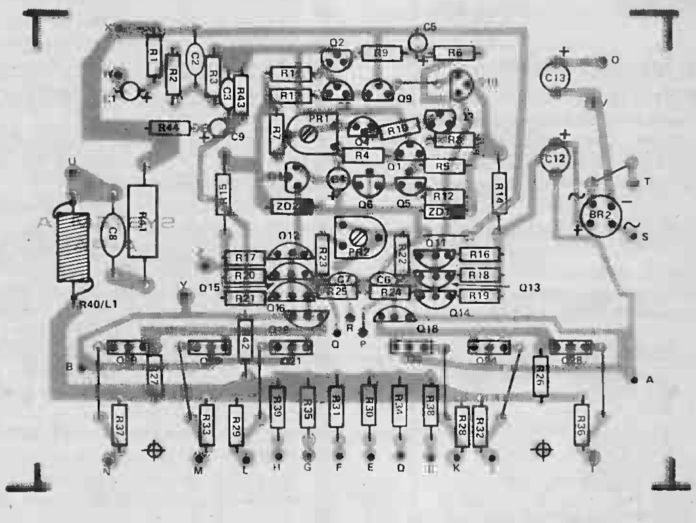
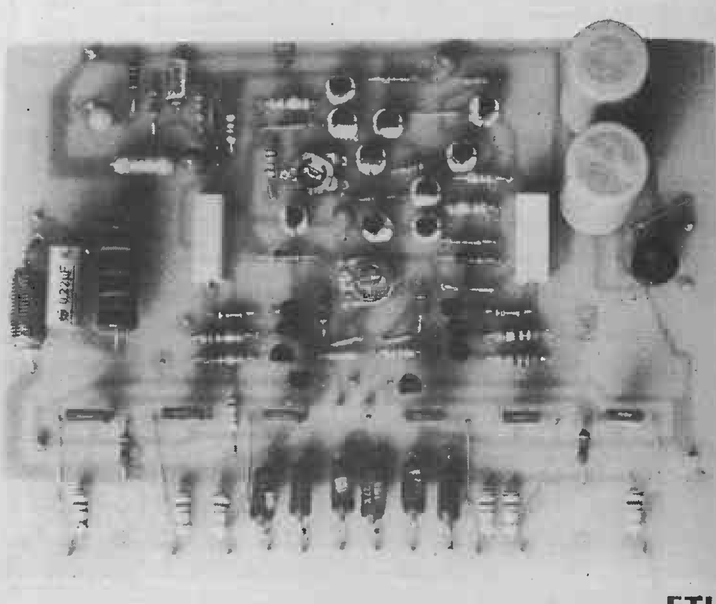


Figure 5. Component overlay for the power-amp PCB.

Next month: we conclude the amplifier project with the PSU and interwiring details.



PIN CONNECTIONS

A	+40 V	N	Q31 base
B	−40 V	O	Wire link to pin Y (underside of PCB)
C	Q30 emitter	P	Q17 collector
D	Q26 emitter	Q	Q17 emitter
E	Q22 emitter	R	Q17 base
F	Q23 emitter	S	Transformer
G	Q27 emitter	T	Transformer
H	Q31 emitter	U	Output
I	Q30 base	V	Wire link to pin Z (underside of PCB)
J	Q26 base	W	Input
K	Q22 base	X	Ground
L	Q23 base		
M	Q27 base		

THE RECORDING CHAIN

Readers of ETI must be familiar with the methods of getting music off a record (you're not? Shame on you), but how does it get on there in the first place? Ron Keeley takes a look at life on the other side of the stylus.

The process of making records is probably as well known, in broad outline, as the ubiquitous black objects which result from it, but what is not apparent from simple descriptions of the 'recording chain' is how some hundreds of thousands of variables can be manipulated to create, in effect, an aural work of art. Making records has become a creative process to which songwriter, arranger, musician, recording engineer and producer all contribute. An arranger/producer can transform a simple pop song into a work as complicated as anything Beethoven ever wrote (outraged classicists should write to the Editor, please), or a writer can create 'impossible' arrangements such as an acoustic guitar solo played over a thundering heavy-metal rhythm track (outraged headbangers, ditto).

Traditionally, the objective in making a record was to capture a musical performance and, in fact, this is how most traditional music — classical and jazz — is still recorded; faithfully transcribed. However, advances in recording technology, specifically the development since 1967 of 8, 16, 24 and even 32 track recorder/reproducers, now make possible much more than simple transcription. Classical music is, and will remain, an exception, but in general a recording is to the original score or performance what a painting is to the subject. It may be a photographic representation, an impressionist interpretation, or even a surrealist nightmare!

Chained Melody

The recording chain is usually regarded as a series of separate, linked stages proceeding more or less smoothly from recording session through to pressing plant. The 'programme chain' model is more extensive, beginning at the planning stage and ending with the marketing operation that transports the 'product' to the High Street stores, and induces us to buy it. Either model is improved if the 'linkages' are regarded as 'interfaces' between various operations, because it is at these points that musicians, engineers and producers are able to exercise creative control.

The first 'interface' is that between a musician and his instrument and, although it is not usually considered as part of any 'chain' it is one of the most important in terms of the musical quality of the recording. It is, in effect, an acoustic/acoustic interface, resulting from the interaction between the sound of an instrument and the acoustic characteristics of the recording venue.

Damp Course

First, room acoustics affect the tonality of instruments. Pipe organs in cathedral spaces are an extreme case, where the naturally long reverberation time enhances and even seems to →

amplify the pipe tones, but violins and brass also need 'space' to develop proper tonality, and even amplified guitar will sound thicker and richer in an acoustically 'live' environment.

Second, a musician needs to hear both his own instrument and those of the others in the group; this is not always easy in studios where the total sound pressure level attenuation over 10 feet can be as much as 56 dB! For the recording engineer, however, this very heavy damping is necessary — some would say essential — if he is to make clean recordings. Too much acoustic reverberation produces a muddy, bass-heavy sound from individual instruments and will cause 'spill' (acoustic crosstalk between a sound source and adjacent microphones), reducing the separation between tracks of the recording and probably resulting in a blurred stereo perspective and/or further loss of clarity.

Critically damped acoustics allow the engineer maximum creative freedom and control. Reverberation and tonality can be easily adjusted using the sophisticated electronics of the mixing console and 'outboard' signal processers, whereas he has very limited influence over the acoustic characteristics of the studio itself. In fact there is nothing 'magic' about acoustically 'dead' rooms; many producers and musicians will avoid them at any cost. The ultimate criteria is 'the sound', so that one studio will develop a reputation as a good rock'n'roll studio while another will be popular for disco or advertising jingles.

In recent years, the design and construction of recording studios has itself become an industry, geared to producing the 'perfect' recording environment. Inevitably this involves a compromise between the conflicting requirements of engineers and musicians. A practical solution is to provide more easily

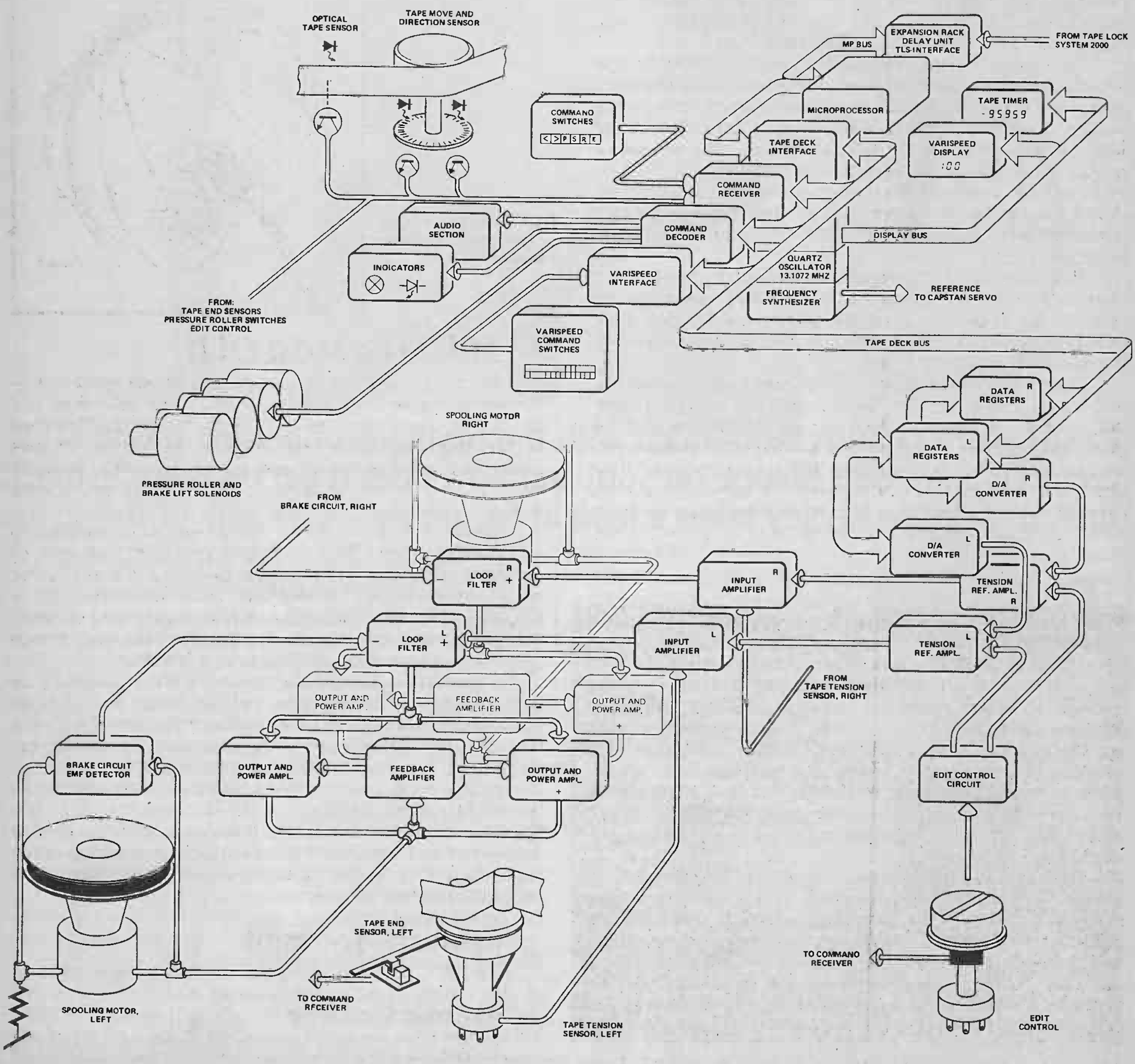


Fig.1 Block diagram showing the spooling motor controls and sensors of a multitrack recorder/reporter (Studer A800).

variable acoustics, either by means of movable wall fixtures, curtains, ceiling louvres, screens and so on, and/or to provide separate 'live' and 'dead' areas.

Pro-Session

The second 'interface' concerns the relationship between the musical performance and the way in which it is recorded; it influences both the musician's performance, and the amount of control the producer/engineer has over the recording. Before the widespread use of multitrack recorders all musicians in a group or orchestra were recorded as an ensemble; however, it is now possible to record each musician independently of the rest of the band.

The variables in this interface, then, involve choosing how far to go, one way or t'other. Ensemble recording is preferable when maximum interaction between the musicians is necessary, and is invariably used for recording classical music and most jazz. At the other extreme, each instrument may be recorded separately; this is 'cleaner' and allows the engineer/producer more creative control, but may result in dry, clinical performances of minimal musical value.

This method is popular with the recording industry because it is quick (only professional, seasoned session musicians need apply), and maximises studio time. However, a typical recording session of a group consisting, say, of bass, drums, rhythm and lead guitars, one or two keyboards and a vocalist, will take a middle road. The usual practice is to first record a 'backing track' of two or more of the rhythm instruments, usually bass, drums and a guitar, plus a 'guide vocal' which provides cues for both the rhythm section and the instrumental soloists who will be recorded later.

This approach is a compromise that works most of the time; although the rhythm section are tied down by the requirement to provide a steady, predictable backing track (which, many would agree, is their main task in a group anyway), the soloists are reasonably free.

Another consideration is that simul-sync recording techniques reflect on the way musicians relate to their instruments. Cue systems, which provide 'foldback' so that the musicians can hear both their own and the other instruments, are necessary even when recording the backing tracks; yet they seem to be the last consideration when a studio is outfitted, even though an inadequate cue system can cause a substandard performance from even the most accomplished musician.

A professional mixing desk at Maison Rouge, the studio owned by Jethro Tull.

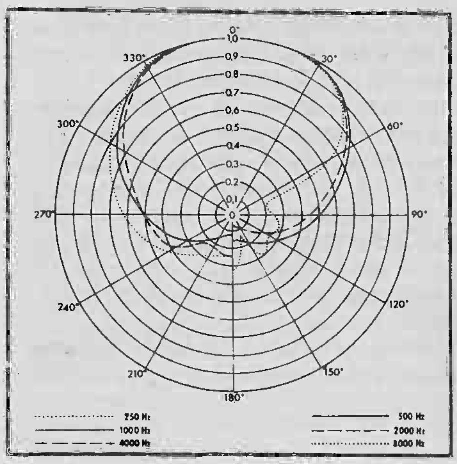


Fig.2 Polar diagram showing cardioid pattern directional characteristics.

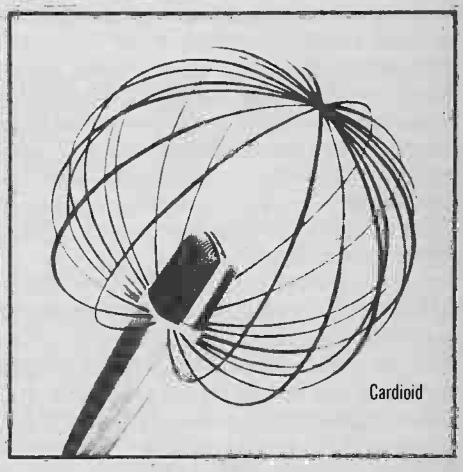


Fig.3 Three-dimensional spatial representation of a cardioid pattern.

Mike The Most Of It

The third 'interface' is acoustic/mechanical/electrical — otherwise known as a microphone. The selection and positioning of microphones is one of the most important steps in the recording chain, and requires an almost intuitive understanding of acoustics and electro-acoustics. Basically there are two distinct techniques: 'close miking' and 'distant miking'.

Ideally, mikes should be placed so as to provide an adequate level to the mixer input or to pick up at least some of the room ambience, enhancing the tonality of the instrument while simultaneously maintaining good separation. It's a juggling act in which perfection is always possible but rarely achieved, and is possibly the main reason why microphone placement is sometimes referred to as a 'black art'!

Generally unidirectional mikes with a cardioid or hypercardioid polar response will be used, as the more directional mikes can be placed further away from the source for better tone, while still rejecting spill from other instruments and most of the acoustic reverb, if there is any.

A certain amount of 'pre-equalisation' or tone control can be had by taking advantage of the fact that the frequency response of a unidirectional mike is not uniform with respect to direction; if spill is not a consideration then an omnidirectional type, which often seems to produce warmer, richer tones, may be used on certain instruments.

Console Yourself

The next 'interface' is electrical/magnetic, and is made up of the mixing console and recorder electronics plus any 'outboard' equipment that may be patched into line. The mixer is the single most important item of equipment for exercising creative control, and a skilled engineer will literally 'play' it as if it were itself a musical instrument.

The mixing console is the heart of the recording operation and, with the arrival of computer automation that will remember hundreds of control settings, it is well on the way to becoming the brain, too. A multichannel studio mixing console contain circuits for dozens of different functions, such as signal level matching, switching and routing, pan controls and so on, but the most important circuits for shaping and controlling sounds are the tone control elements, usually three- or four-band parametric equalisers.

Equal Rights

Although it is kept to a minimum during recording, equalisation is always necessary even in classical recordings. Because professional microphones are extremely sensitive devices, they will respond to mere whisps of sound — sounds that a human listener would not be aware of because human hearing is a matter of psychoacoustics. We automatically filter out any extraneous bits of noise that don't convey information; listening to the band we tend to hear only the music, and not the hiss from the guitar amplifier, the slight flap in the bass drum head or the wheeze of the singer's breath. Unfortunately microphones are not fitted with such sophisticated filters and, when amplified, these noises can be painfully obvious. Equalisation — EQ for short — is used to remove these odds and ends and also to compensate for deficiencies which cannot otherwise be fixed.

Bass guitar, for example, presents particular difficulties. Miked bass is usually too boomy and lacks clarity. The solution is to take a 'split', off the bass itself or off a preamplifier, directly into the mixer. This DI (direct injection) bass will be clear and pure but will lack forcefulness ('balls', in the trade) so the two inputs are mixed together, with sufficient EQ to round out the tones and to produce an even response across the strings (the bottom two strings give a higher output than the top two).

External signal processers are used for a number of purposes. Compressor/limiters are used on vocals to maintain a constant level out to the recorder, and on bass guitar to tighten up the sound. Extra equalisation might be required to remove the annoying ringing tone that snare drums inevitably produce, while noise gates, which turn 'on' when the input crosses a preset threshold level, may be used to minimise spill across the drum microphones. A de-esser may be required to remove sibilance on the harmony vocal track, or one of the many noise reduction systems available may be patched between the mixer outputs and the multitrack inputs to give up to 30 dB improvement in the dynamic range on tape. Much pop music, though, will not exceed the 65 dB range of most tapes and so noise reduction is not always used.

Inside the Maison Rouge studio, with its abundance of Shure microphones.

All Together Now

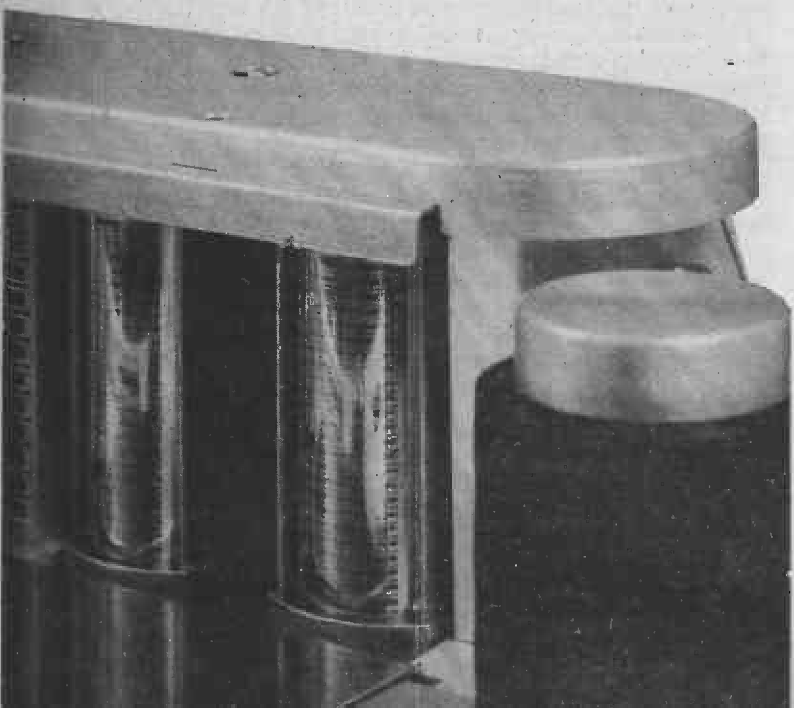
The next stage in the recording chain is the mixdown operation in which the separately recorded parts of the performance are blended into a whole. The interface here is magnetic/electrical/mechanical/acoustic, and is one of the most important in terms of the quality of the final product. This complicated system, which has also been operating during the recording stages, is formed from the tape playback circuits, the mixer, the studio monitoring system and those most valuable appendages, the ears of the engineer and producer.

A mixdown session is essentially an assembly operation in which the separate parts are fitted together, adjusted and polished until the song has the aural perfection of a shining, precision machine — hopefully. The level of each track must be balanced against all the others, and assigned to its proper place in the stereo image. At the same time the producer will refine 'the sound', perhaps adding a touch of EQ to blend the bass guitar more closely with the bass drum, or to improve the separation on the rhythm guitar track, where the drums have spilled over a little.

The importance of 'clean tracks' becomes apparent when the producer asks for a touch of echo on the guitar break; any spill onto the track, or lack of clarity due to excessive reverberation, will simply be accentuated, further muddying the sound. No effect should be used just for the sake of it; however a touch of phasing on the hi-hat cymbal can be very effective. Electronic reverb, added to the main vocal, say, can make it recede into the background as if the singer were a long way off, or double-tracking can be used to create a second vocal line loosely synched with the original.

Monitors Matter

A mix session, especially if 24 or more tracks are involved, can take more time than the entire recording session, and at all times those in the control room are being blasted by high level sounds from the studio monitor speakers. The introduction of noise reduction systems and other advances in recording technology has led to dramatic increases in the dynamic range of recorded music. This in turn, has created a demand for monitoring systems which can reproduce virtually the entire



The 24 track headblocks of the Studer A80/VU recorder. This professional unit records on 2" wide tape, and features electronic tape transport logic, solid state switching of motors, a servo-controlled capstan motor and amplifier channels which are completely self-contained units.

range of the original performance, and can track even the most subtle level changes as, for example, an orchestra builds to a crescendo. Monitoring must therefore be carried out at very high volume, sometimes peaking at over 100 dB SPL, and at these levels the monitoring system must be very good indeed. Even the slightest distortion will become intolerable, in time, and listening fatigue may cause a tired engineer to compensate for purely imaginary flaws in the sound.

In recent years, as studio design has become a specialist subject, so have studio monitoring systems, and particular attention is now paid to the control room acoustics. The latest designs, called 'live end-dead end' or 'single-pass', call for the speakers to be mounted in a 'live' area that has a moderate reverb time, and directed so that the sound passes by the engineer's position to be absorbed by a 'dead' space behind the console.

The Reel Thing

The final stage of studio operations is the tape-transfer, interfaced once more by the mixer. Now the engineer must actually 'fly' the console, duplicating exactly the adjustments and switch punches that were determined during the long hours of assembly, adjustment and polishing. The result of this is a 'production master' tape, which is then sent for final processing at the mastering studio.

Master Minder

Record mastering is the operation in which the production master is 'cut' onto a lacquer-coated aluminium master disc, the

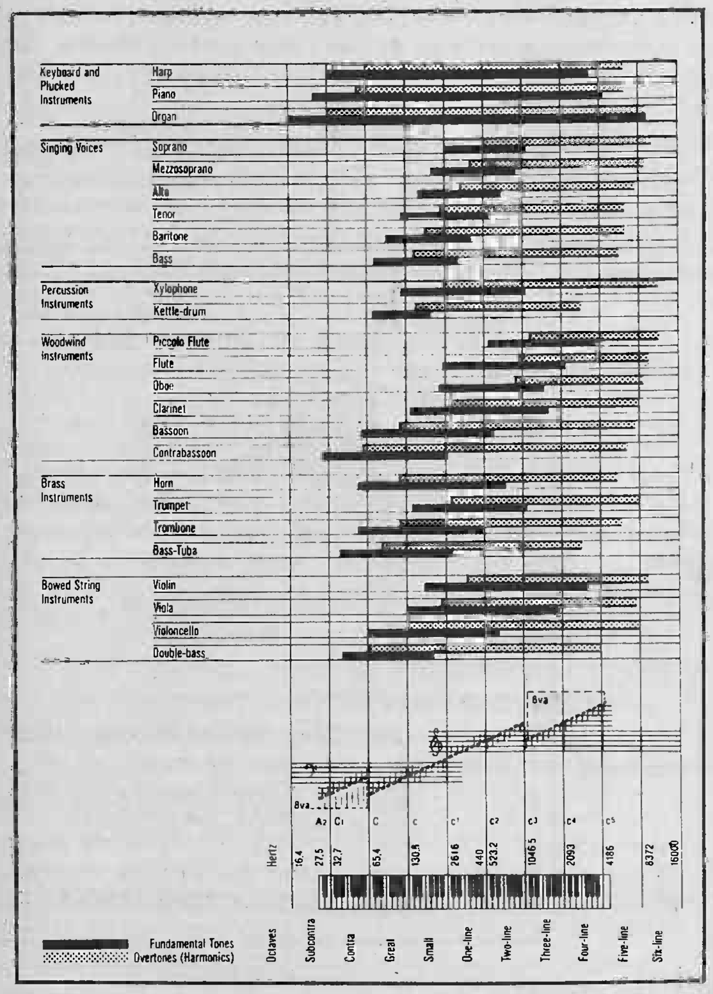
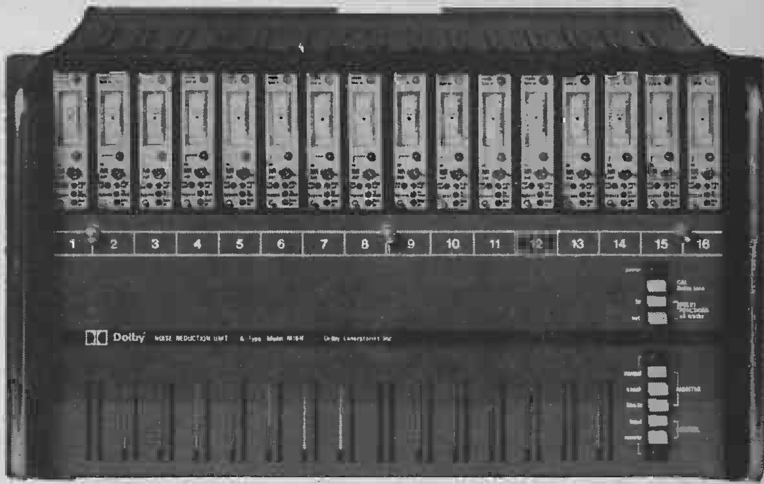


Fig.4 The perceived pitch of a note falls within the range of fundamental frequencies which can be sounded by the instrument, but the overtone frequencies extend beyond the highest note that can be played.



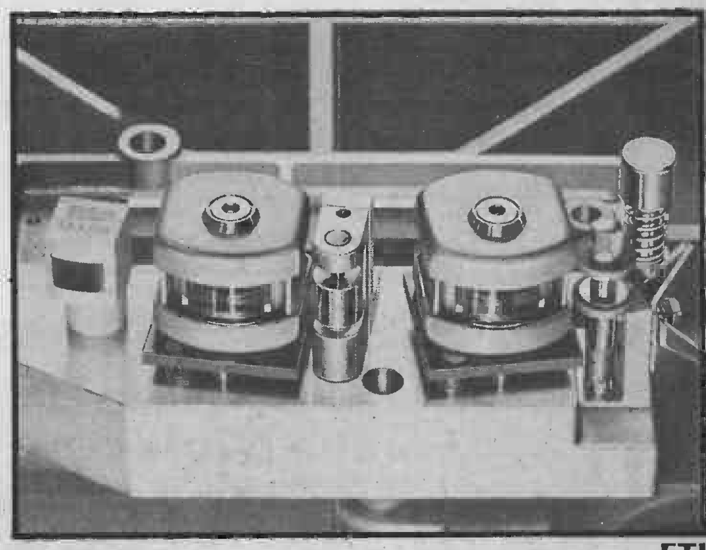
A professional noise reduction unit from Dolby Laboratories Inc.

master from which all other discs are made. Until recently, disc cutting was strictly a manufacturing process, carried out by skilled tradesmen wearing white dustcoats, in a room that was simply part of the pressing plant. Now, however, as much care and attention is paid to the cutting room facilities as is given to the recording studio and the mastering engineer, whose name often appears on the album cover together with those of the engineer and producer, is as much a part of the creative process as they are.

Technological advances in the cutting lathe and its associated equipment — the magnetic/electrical/mechanical interface for this process — now enable the mastering engineer to actively contribute in creating a record which is 'better' than it otherwise might have been. Using monitoring facilities which are equal to those in the best studios, and therefore better than most, he is able to detect and correct any imbalance in the tonality of the master tape; or he may use frequency-selective compression to limit high power/high frequency signals — which can destroy the cutting head — to produce a 'hot' cut. This gives a higher level on disc, for the same playing time, but does not affect the overall programme content.

Once the master disc is cut, however, the creative process has reached the end of the line. Pressing operations are strictly mechanical and, although there is much that can be done, or not done, to ruin a pressing, there are no creative options left. Except, of course, for those clever, hard-working advertising and marketing men who 'interface' between the records and our wallets!

Our thanks to Shure for providing the lead photograph of this article.



A standard half track mono or stereo headblock (Studer A80/RC).

ETI

DIGEST

Tiny TVs

A new low-price version of the Hanimex 531-2 is the HTV 531-1 unit which incorporates a 5" black and white TV, 3 band AM/FM/LW radio and UHF tuner. The unit has a built-in condenser microphone and auto-stop cassette deck, a four-way power facility and built-in charger circuit for nickel cadmium batteries. Hanimex are currently working on a colour version which should be available for Christmas this year.

Hanimex think that with the advent of breakfast TV the portable television will really come into its own.

It's A Gas

British Gas moved 40 million tons of earth last year by digging half a million holes in Great Britain at a cost of about £250 per hole. To do this job more cheaply and efficiently they need to be able to detect pipes accurately and quickly, and existing plans don't always tell the excavating teams what other pipes there might be lurking under the turf and tarmac. So, for some years, British Gas have been developing a device which can quickly and simply locate pipes, even in built-up areas. The result is an instrument called Gascopact (GAS COrporation Pipe And Cable Tracer). The instrument has two parts — a transmitter induces currents in the ground which then concentrate into the pipe. A receiver then detects the magnetic field associated with these currents, an array of sensors measures the shape of the field, calculates the position of the pipe and gives an 'X' indication when it is directly above it. Moving away from the line of the pipe gives a 'O' reading. The Gascopact can locate plant at depths of up to 2½ m with positional accuracy of about the width of a spade. At the moment extensive field trials are being carried out and possible manufacturers are being invited to compete for the contract for the final version.

Cheap Computing

Sinclair Research Ltd are making an offer to all UK secondary schools for a low-cost subsidised computer package. For a mere £65 schools can have the Sinclair ZX81 personal computer, the 16K add-on memory pack and the ZX81 BASIC manual. Normal price is £119.50. Each school is entitled to one package from the scheme which will be administered by Griffin & George Ltd. Unfortunately the ZX81 and the RAM pack will not be available separately at subsidised prices. This scheme is an extension to the Government scheme currently open to schools which do not have a microcomputer. Clive Sinclair, Chairman of SRL, commented, "We felt, however, that the scheme failed properly to assess school needs and to review the available equipment which could have provided schools with a wide and economic choice. At £65, about half the cost to schools of the reported price of the cheapest Government-recommended computer, we are offering a full facility computer with a proven track record — some 15,000 ZX81s are already in use". This scheme is open to orders received by Griffin & George between 1st June and 31st July. Their head office is at Ealing Road, Wembley, Middx HA0 1HJ.

Space Age Charity

Video games have become a very popular way of spending money, and profits from the machines can be considerable. As 1981 is the International Year of Disabled People, it would be nice if some of these profits went to help those people who will never play these games themselves. The Muscular Dystrophy Group of Great Britain is launching a space age fundraising campaign, in which they will hire video games for installation in public areas and share the profits with the owner of the site. Suggested sites are factory canteens, social or sports clubs, working men's clubs, hospital waiting rooms, launderettes and take-away shops. If anyone is interested in taking part, further information can be obtained from Fran Willison, Head of Fundraising and Public Relations, The Muscular Dystrophy Group of Great Britain, 35 Macauley Road, Clapham, London SW4 0QP; telephone 01-720 8055.

SPOT DESIGNS

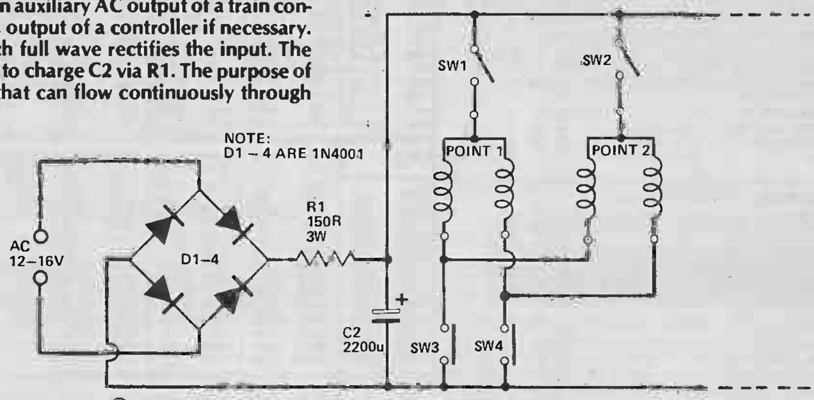
Points Controller

The electric points used in model railways are operated by a simple mechanism which utilises two solenoids, providing movement in opposite directions. Thus, briefly connecting power to one solenoid changes the state of the point and a short burst of power to the other solenoid changes it back to its original state. It is essential that the power is only applied in short bursts, since the current consumption of each solenoid is typically in the region of 1-2 A, and continuous power would almost certainly cause the solenoid to burn out.

It is possible to eliminate any chance of accidentally destroying a solenoid by using a capacitor discharge controller such as this simple design. This takes its power from an auxiliary AC output of a train controller. It will also work from a DC output of a controller if necessary. D1-4 form a bridge rectifier, which full wave rectifies the input. The resultant pulsing DC signal is used to charge C2 via R1. The purpose of R1 is merely to limit the current that can flow continuously through the points solenoids to a safe level (about 100 mA or so).

The high current required to activate the point is available from C2, but this can only provide short bursts of current as it quickly discharges through the low impedance provided by the solenoids. The value of C2 is chosen to give pulses of power that are sufficiently long to give reliable operation of the point, but offer no possibility of burning out the solenoids. SW1 or SW2 is closed to select the appropriate point and then either SW3 or SW4 is operated to switch the point over.

Although only two points are shown in the circuit diagram, obviously any number of points can be connected to the unit, an additional switch being required for each one. SW3 and SW4 must be heavy duty types having a current rating of at least 2 A.



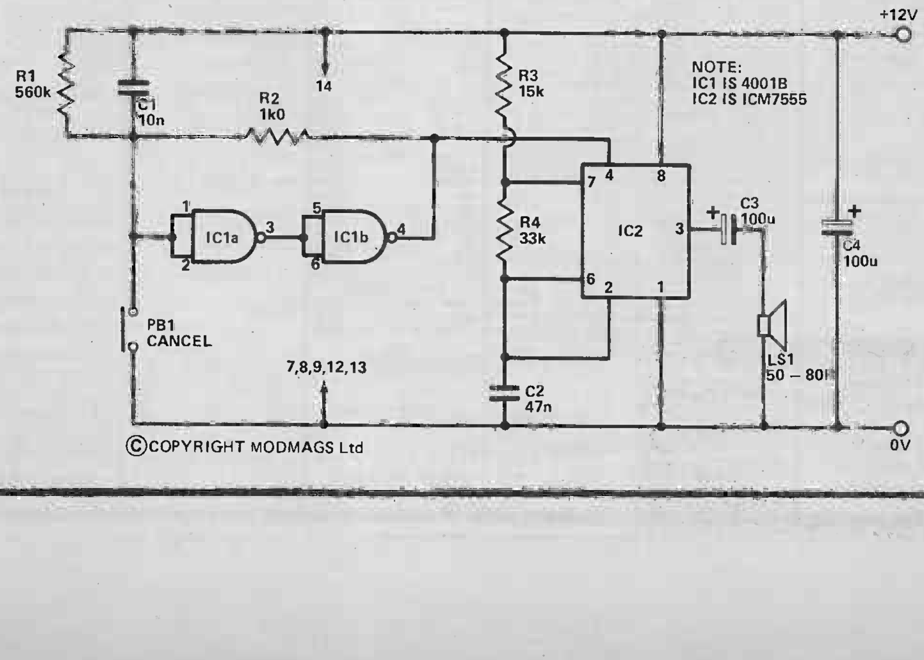
Seatbelt Reminder

This simple seatbelt reminder circuit connects to the battery via the ignition switch, and sounds an audible reminder signal when the ignition switch is turned on. The audio signal is cancelled by operating a push button switch.

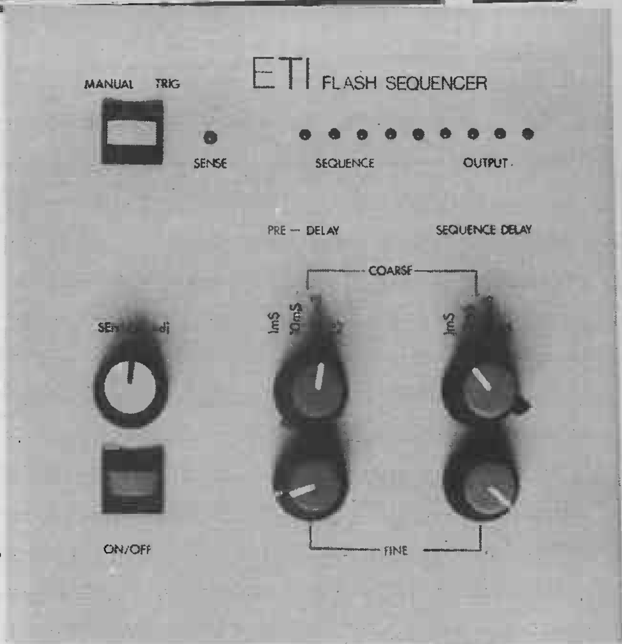
The audio signal is generated by IC2 which is an ICM7555 (the CMOS version of the standard 555) used in the astable mode. The CMOS version is preferable in this application since it has a higher maximum supply voltage rating and is less likely to be damaged by an excessive supply voltage. C3 couples the output of IC1 to a high impedance loudspeaker which should not have an impedance of less than 50R.

The oscillator is controlled by the voltage fed to pin 4 of IC2; with a voltage of less than about 0V5 the oscillator is disabled. IC1 is a CMOS quad two-input NOR gate, but only two of the gates are used, and these have their inputs wired in parallel so that they operate as simple inverters. They are connected in series and have DC positive feedback via R2 so that a simple latch circuit is produced. C1 provides a positive input pulse to the latch so that initially the input (and therefore the output as well) assumes the high state. Thus the oscillator operates when power is first applied to the circuit, and the reminder signal is produced. Briefly operating PB1 takes the input (and output) to the low state, so that the unit functions properly when the ignition switch is operated again.

As the quiescent current consumption of the unit is less than 1 mA the unit does not have a detrimental effect on the car battery (which has a very high capacity), especially as the unit is only connected to the battery when the ignition switch is turned on.



FLASH SEQUENCER



Here's a project that lets you do something new with a flashgun, or nine. Make mobile matter into marvellous multiple images. Design by Plamen Pazov. Development by Steve Ramsahadeo.

The human eye is an extremely complex sensor, capable of discriminating shapes and colours in an extremely high optical noise environment. Researchers are only just realising the computational complexity of image analysis, shape recognition, perspective compensation, telemetry and all the other tasks that our eyes and brain perform continuously.

Yet, probably because of this complexity, there is one area in which the eye's performance is rather poor: speed. Any discrete movement shorter than an eighth of a second becomes blurred and at a repetition rate of 18 to 25 events per second, everything blends into a continuous movement. Unfortunately, in real life, events happen a lot faster than that and to try to analyse and understand them, we need some device that can effectively reduce the flow rate of information.

Strobe Shots

When the number of images of the subject is small, we can simply superimpose them. The result is strobography. The pioneers of this method (Mach, Foucault, Helmholtz) used the high intensity discharges of Leyden jars (the first capacitors) as the light sources. Nowadays, you can obtain much more accurate results with a battery of electronic flashes, a camera and our flash sequencer!

The basic idea is that when an event occurs, it triggers a series of flashes at a constant and known rate. Being of very short duration (of the order of a microsecond), the flashes effectively immobilise the subject in its consecutive positions, which can be recorded with a camera. As an example of possible analysis, the change in position during a known time can give the velocity. The change of velocity to the next frame gives the acceleration, and so on. The position, intensity and colour of each flash can give further meaning to the recorded positions.

Multi-option Multiflash

One of the basic practical problems in strobography is the triggering of the sequence. The most useful sensors are contact, audio and optical, and all three are provided for in our design. Furthermore, an adjustable pre-delay allows the flash sequence to commence a short time after the triggering pulse, should this be necessary for the right effect. Both the pre-delay and sequence delay can be continuously adjusted from 100 us to 1 s per flash and thus practically all situations are catered for. A manual trigger and LED readout are provided to make the time settings for a particular picture easier to adjust. A camera X flash socket allows for a sequence to be triggered from the camera, in exactly the same way as a single flash. In this case, make sure that the exposure time is longer than the total sequence time.

Using The Unit

There are two possible modes of operation for this device. It is either:

- Triggered by the event to be recorded, using a suitable sensor (adjusted just below the triggering point) and the pre- and sequence delays being set for the desired effect. In this case, the camera shutter has to be open during the whole of the sequence. Therefore a manual operation of the shutter (B setting) and very little or no ambient light will be necessary. The diaphragm should be set to one stop less than the calculated aperture for the flash and subject-camera distance used.
- Set off by the camera in exactly the same way as a single flash. This method is somewhat more flexible for event durations of the same order as the human reaction time (about a tenth of a second). In that case, the synchronisation is made through an ordinary extension lead from the X socket of the camera to the 'contact' input of the sequencer. Care must be taken to set an exposure time, longer than the total sequence time. The diaphragm setting is determined in the same way as in the previous method.

In both cases, the flashes are connected to the sequencer either directly or through ordinary extension leads, and can be distributed along the path of the subject or grouped as a battery. It is generally easier to work with flashes of the same type or at least of the same intensity. In the case of flashes with auto-exposure circuitry ('computer' flashes), the setting should be to manual (or the sensors masked with opaque tape), in order to avoid interactions from the previous flash.

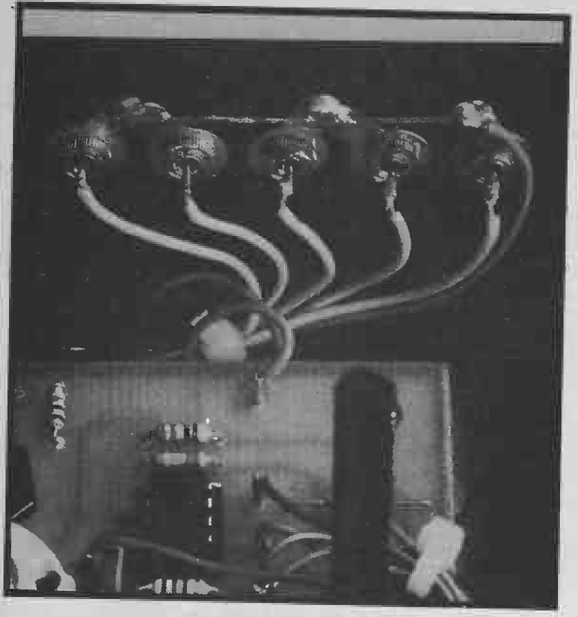
Construction

Construction should be fairly straightforward, as all the components (except PB1, SW1, SW4 and the LEDs) are mounted on a single PCB to minimise interwiring. Note that the number of output stages is entirely up to you; if you want less than nine flashguns to be triggered, leave out the unwanted triacs together with their associated LEDs, resistors and sockets.

When fitting the rotary switches, the tags will have to be trimmed off to fit the PCB holes. Some of the unused tags have been cut off completely to allow PCB tracks to pass through — the overlay shows which ones remain.

The board can now be mounted on ¾" pillars and holes marked out and drilled to accommodate the LEDs and switch and potentiometer spindles. A cut-out measuring 14 mm x 14 mm will have to be made to accept the two push-buttons.

The next stage is to wire the LEDs, sockets and off-board switches. When this is completed a visual check of the whole project should be made.



Close-up of the flash output sockets. These are held in place on the front panel by the knurled nuts, which are rather hard to solder to; we drilled out solder tags to the right size to make the earth connections.

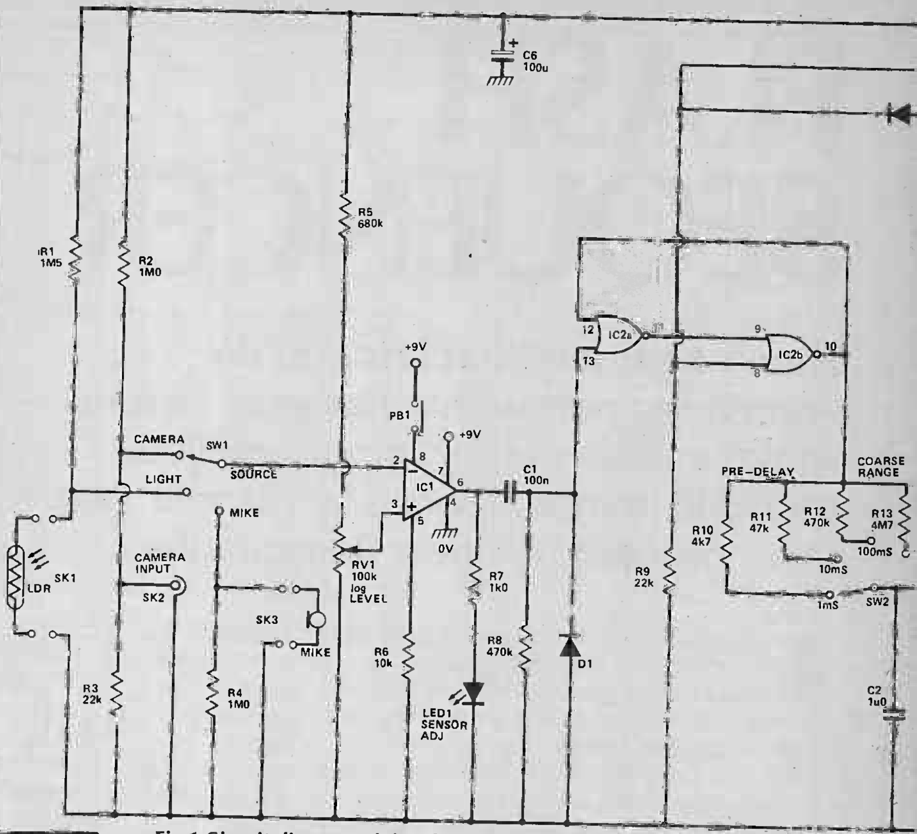


Fig.1 Circuit diagram of the Flash Sequencer.

HOW IT WORKS

The sensor is a simple level crossing comparator, the signal source being selected by SW1. Resistor R6 at the offset compensation input of IC1 provides an imbalance of the input differential stage, so that RV1 can set the threshold level both over and under 0 V. LED1 is an aid to adjustment and should be just flickering when RV1 is properly set.

A manual trigger is provided for test purposes. Irrespective of the input conditions of IC1, when PB1 is activated the strobe pin (pin 8) has direct control of the output. Thus pin 6 of IC1 is sent high and the flash sequence is triggered.

The next two stages are the pre- and sequence timers. They work in the same way, except for the enable conditions. The difficulty here is a reliable and repeatable delay. In all monostables, energy is charged and discharged between two levels. Our problem is that the stable level must be the same as the power off state; it must be reached in a much shorter time than the shortest delay (100 uS) without a negative supply to draw on; and the time control must be linear.

The stage works as follows: IC2a and IC2b lock to the charging position (low and high outputs respectively) as soon as an impulse is received through C1. C2 begins to charge through the range resistor. The output of IC3a is high, so IC4's inverting input is higher than its non-inverting input and its output is low. This output is fed back to IC3a's input so we have a steady state. As soon as the voltage on C2 reaches the limit set by RV2, IC4 switches high; IC3a switches low; the voltage at the inverting input of IC4 is zero, giving a Schmitt trigger action. A change of state can only occur now when C2 is discharged to 0 V (power off state). This is done by IC3b and C3, which was initially positively charged. As IC4 goes high, IC3b goes low. The voltage on the negative side of C3 is driven below zero potential. D4 is now forward biased and passes a current from C2 into C3. D3 and IC2b suppress the charging current. This continues until the voltage on C2 is just under 0 V, at which point IC4 switches over to its low position again.

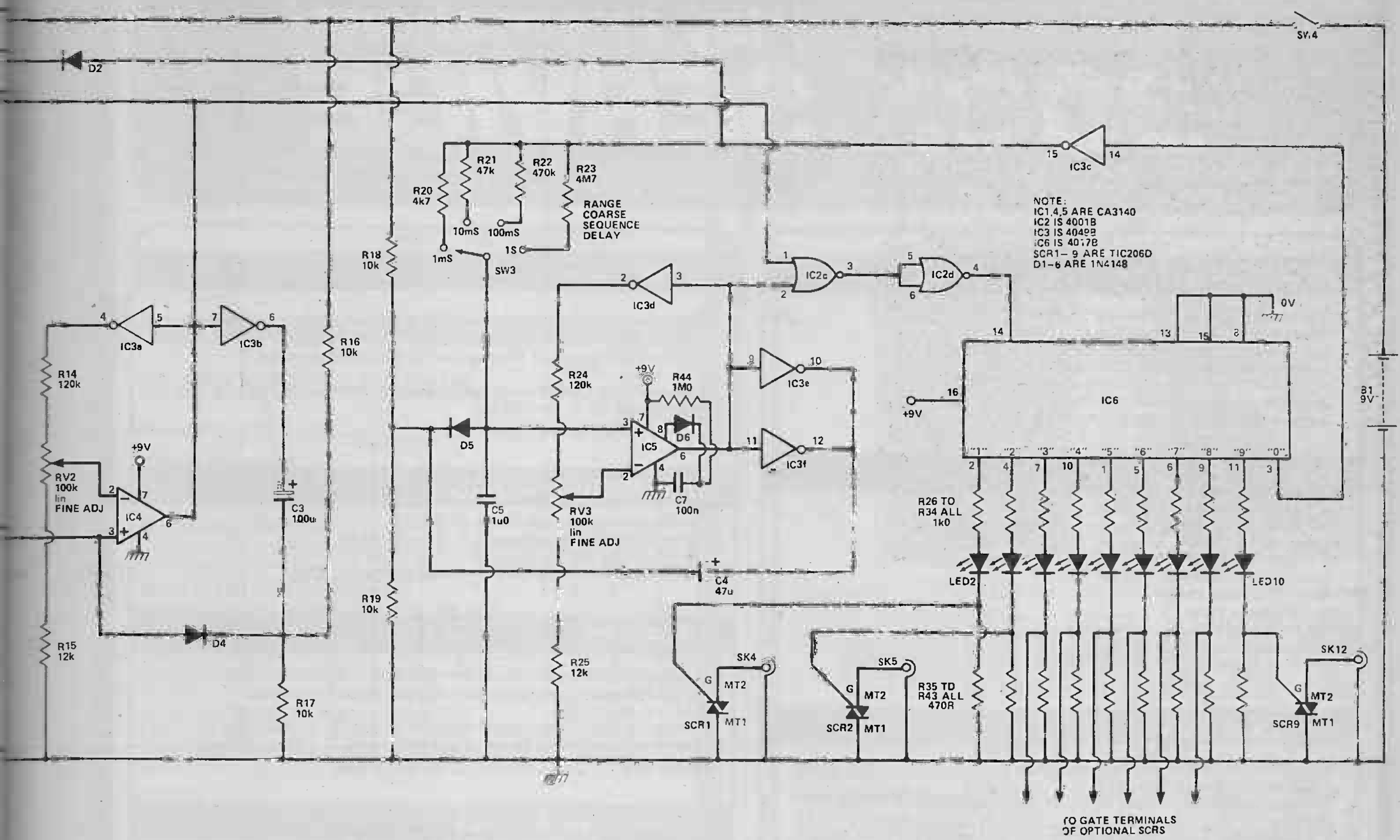
The low to high transition of IC4 has been transmitted through IC2c and IC2d to the IC6 clock input. This results in the '0' output, which was at logic 1, going low, thus disabling the pre-delay stage and enabling the sequence timer through IC3c. At the same time output '1' goes high, triggering triac SCR1 and firing the first flash. The sequence timer (still enabled) operates in the same way as the previous stage, except that the charging current is continuously enabled and astable operation results. Each one of the impulses shifts the high output of IC6, successively triggering the following flashes until the '0' output is reached again. This disables the sequence timer and allows the pre-timer to receive a new impulse from the sensor, ready for a new cycle.

The output control elements are triacs, rather than SCRs, because not all electronic flashes conform to the 'positive centre/earth shield' standard connection.

PARTS LIST

Resistors (all ¼ W, 5%)	
R1	1M5
R2,4,44	1M0
R3,9	22k
R5	680k
R6,16,17,18,19	10k
R7,26-34	1k0
R8,12,22	470k
R10,20	4k7
R11,21	47k
R13,23	4M7
R14,24	120k
R15,25	12k
R35-43	470R
Potentiometers	
RV1	100k logarithmic
RV2,3	100k linear
Capacitors	
C1	100n polycarbonate
C2,5	1u0 polycarbonate
C3,6	100u 10 V tantalum
C4	47u 16 V tantalum
C7	100n ceramic
Semiconductors	
IC1,4,5	CA3140
IC2	4001B
IC3	4049B
IC6	4017B
SCR1-9	TIC206D
D1-6	1N4148
LED1	0.125" yellow LED
LED2-10	0.125" red LED
Miscellaneous	
PB1	push-button type SRM (see Buylines)
PB2	push-button type SRL (see Buylines)
SW1,2,3	one-pole rotary switch
SK1,3	phono socket
SK2,4-12	3 mm coaxial flashgun sockets (see Buylines)

Snap-in PP3 battery holder (Vero order code 202-21392J), case (see Buylines).



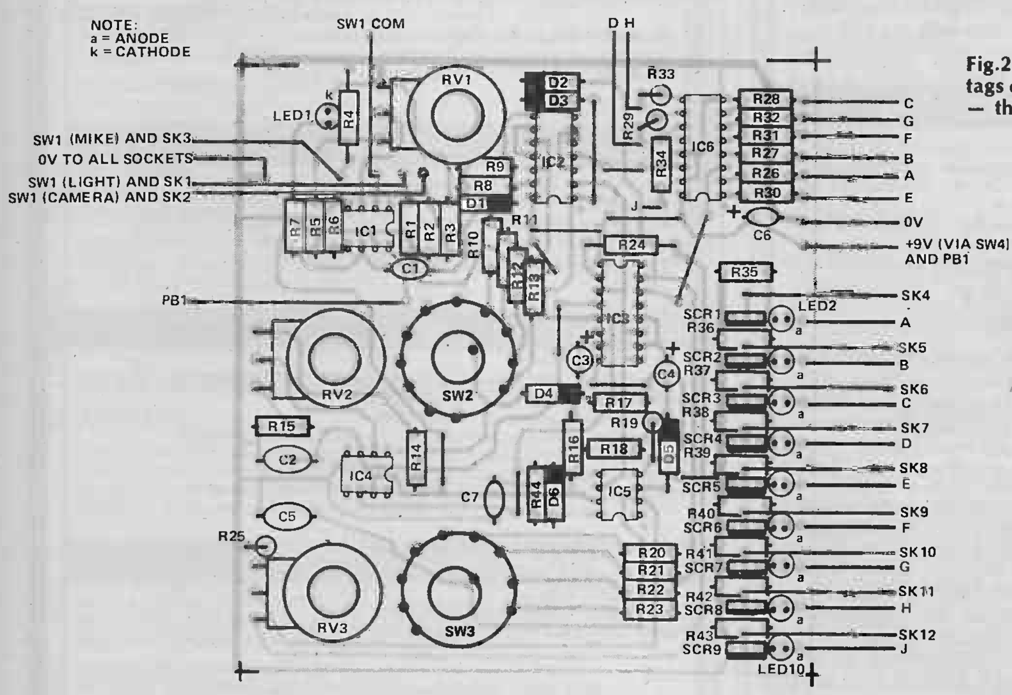
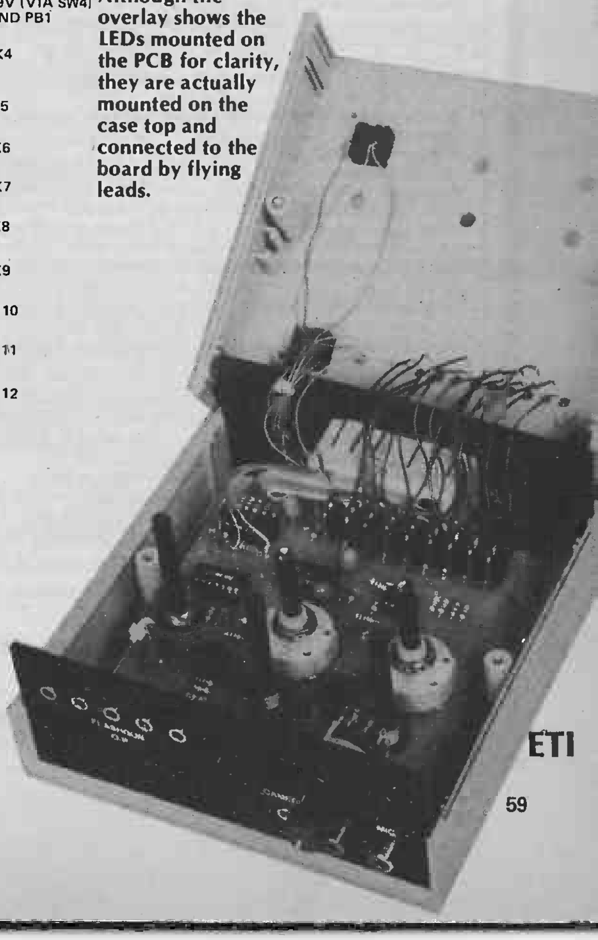


Fig.2 (left) Overlay for the sequencer. Some of the tags of SW2 and SW3 are cut off before mounting — the dots indicate which tags remain.



Below: inside the sequencer. Although the overlay shows the LEDs mounted on the PCB for clarity, they are actually mounted on the case top and connected to the board by flying leads.

BUYLINES

The 3 mm coaxial sockets (and a camera extension lead, if required), can be obtained from Vanguard Instruments, 233 High Street, Brentford, Middlesex. Telephone: 01-560 7667.

The case was chosen from the CM range offered by OK Machine and Tool Ltd, Dutton Lane, Eastleigh, SO5 4AA, telephone 0703 610944/5. The case order code is CM6-225.

PB1 and PB2 are available from Watford Electronics, and all the other components are common types and should cause no problems.

DESIGNER'S NOTEBOOK

In this month's edition of Notebook, Ray Marston first looks at high impedance 'bootstrapping' techniques, and concludes by showing some unusual 4001B/4011B CMOS monostable and bistable circuits.

Bootstrapping is an in-phase (positive) feedback technique that can be used to greatly increase the apparent (AC) value of a resistor or reduce the apparent value of a capacitor. The technique is of particular value in the design of ultra-high input impedance AC amplifiers. We'll take a brief look at some practical examples of the technique in the first part of this edition of Notebook.

The easiest way to understand why the bootstrapping technique is needed is to look at the simple AC emitter follower circuit of Fig. 1. A major attraction of the emitter follower circuit is that it is capable of presenting a high input impedance to external signals, the actual impedance (looking into the base of the transistor) being equal to the product of the transistor h_{fe} (current gain) and the emitter load impedance (R_e): thus, the base impedance of the Fig. 1 circuit is equal to 220k. In practice, however, the emitter follower circuit cannot work unless it is DC-biased in some way, and in Fig. 1 potential divider R1-R2 is used as the biasing network. Unfortunately, this network is effectively in parallel with the input (base) of Q1 and thus reduces the true input impedance of the circuit to a mere 10k or so. Not very good.

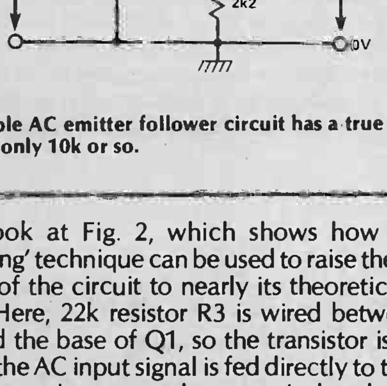


Fig.1 This simple AC emitter follower circuit has a true input impedance of only 10k or so.

Now look at Fig. 2, which shows how the so-called 'bootstrapping' technique can be used to raise the true AC input impedance of the circuit to nearly its theoretically-attainable maximum. Here, 22k resistor R3 is wired between the R1-R2 junction and the base of Q1, so the transistor is still correctly biased, and the AC input signal is fed directly to the base of Q1. The important point to note, however, is that the output of the emitter follower is AC-coupled back to the R1-R2 side of R3, so that in-phase AC signals appear on both sides of R3. What's the effect of this action?

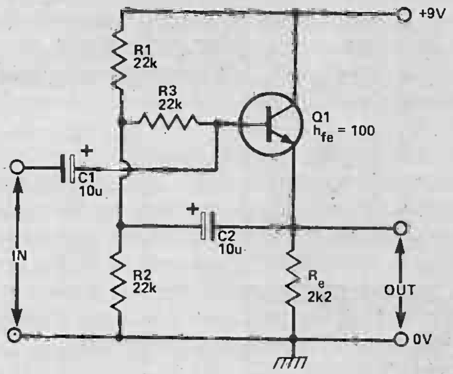


Fig.2 This 'bootstrapped' version of the AC emitter follower has a true input impedance of about 180k.

Suppose that our Fig. 2 emitter follower has an AC voltage gain of 0.98 (a reasonable figure). In this case, when an input signal is applied, all (100%) of the input signal appears on the 'base' side of R3, and an isolated but in-phase copy of this signal, with 98% magnitude, appears on the R1-R2 side. Consequently, the signal current flowing in R3 equals only 2% of that which would be expected from the original input signal alone. In other words, the AC input signal sees R3 as having a value of 100/2 x 22k, or 1M1: this impedance is in parallel with the base impedance of Q1, so the final input impedance of this bootstrapped emitter follower circuit works out at about 180k. Pretty good.

You can see, then, that the bootstrapping principle is very simple. By feeding an input signal to one side of passive component and a less-than-unity in-phase copy of the signal to the other side, the apparent impedance of the component can be increased. If 50% feedback is used, the impedance is doubled (100/50), if 90% feedback is used, the impedance increases by a decade (100/10). 99% feedback raises the impedance by a factor of one hundred (100/1), 100% feedback raises the impedance to infinity. If bootstrapping is applied to a resistor, the apparent resistance is increased: if the technique is applied to a capacitor, the apparent capacitance value is reduced. Clever stuff.

Bootstrapped Op-Amps

The basic bootstrapping technique can easily be applied to op-amp circuits, to produce non-inverting AC amplifiers with ultra-high input impedances, as shown by the examples of Figs. 3 to 7. In our first example (Fig. 3), the op-amp is biased by R1-R2 so that it can operate from a single-ended supply, and the input

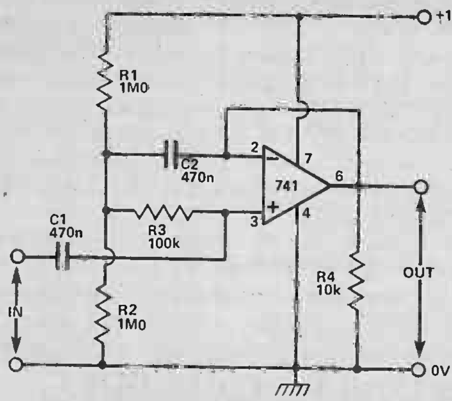


Fig.3 This single-supply version of the unity-gain non-inverting AC amplifier has an input impedance greater than 100M.

signal is AC-coupled to the op-amp side of R3 while the other side of this resistor is bootstrapped (via C2) from the output of the op-amp. The gain of the op-amp is so close to unity that the apparent (AC) impedance of R3 is increased to near-infinity, giving the circuit a true AC input impedance in excess of 100M. Without bootstrapping, the input impedance would be a mere 600k.

Note at this point that the attainable input impedance of the bootstrapped op-amp circuit is so high that in practice the true impedance is actually determined by the surface leakage impedance of the PCB and IC socket, etc. An easy way around this problem is to provide the area of the PCB surrounding the op-amp input pin with a 'guard ring', as shown in Fig. 4. This guard ring effectively bootstraps the leakage impedances of the PCB and raises them to near-infinite levels.

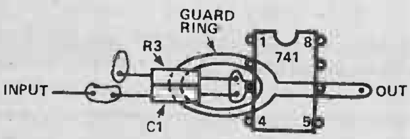


Fig.4 Method of providing a guard ring on the PCB, around the op-amp input terminal of the Fig.3 circuit, so that the PCB leakage impedances are effectively bootstrapped.

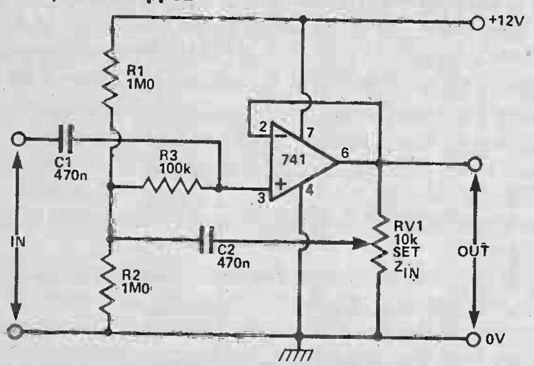


Fig.5 A variable input impedance unity-gain non-inverting AC amplifier. The impedance can be varied from 100k to 100M using RV1.

Figure 5 shows how the Fig. 3 circuit can be modified so that it acts as a variable-input-impedance circuit in which the impedance can be varied from roughly 100k to 100M using RV1. With RV1 slider set to the top of the pot, 100% bootstrapping is applied and the input impedance is 100M. With RV1 slider set to the bottom of the pot, zero bootstrapping is applied and R2 is bypassed to ground via C2, so the input impedance is about 100k.

Figure 6 shows how to bootstrap the unity-gain non-inverting op-amp circuit when operating it from split supplies. R1-R2 are the DC bias resistors, with R1 bootstrapped from the op-amp output via C2. The circuit has an input impedance capability of about 500M.

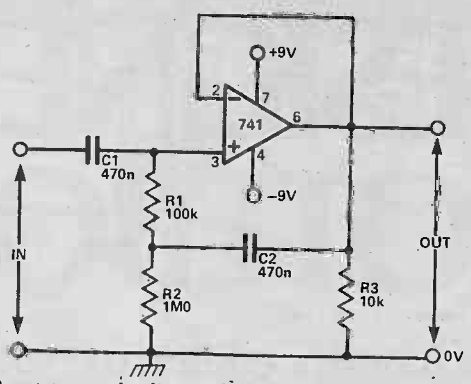


Fig.6 This bootstrapped split-supply unity-gain amplifier has an input impedance of about 500M.

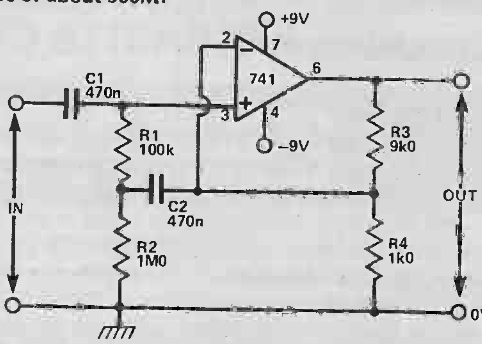


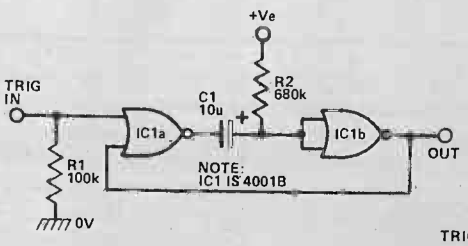
Fig.7 A high input impedance, x 10 non-inverting AC amplifier. Note that the guard ring of this circuit (if used) should be taken from across R4 (not from the op-amp output).

Finally, Fig. 7 shows how to apply the bootstrapping technique to a non-inverting amplifier with a gain greater than unity. Here, the gain is determined by the R3-R4 values and equals 10 with the values shown. Note that the bootstrapping signal is taken from the output of R4, rather than directly from the op-amp output. Also note that if a guard ring is used on the PCB, it must be bootstrapped from the same source.

Modified Monos

Now for a complete change of topic. If you've ever used the 4001B or 4011B CMOS ICs in the standard monostable configuration you'll know just how useful these circuits are in non-precision applications. They are easy to trigger, give clean outputs, and can cover a very wide timing range. The only trouble is, they're non-resettable; once they've been triggered they simply latch on until their timing periods end naturally. Figures 8 and 9 show a couple of easy ways of modifying these circuits to give easy reset operation.

Figure 8a shows the circuit of the conventional 4001B version of the standard monostable: with the R2 value shown, the circuit gives a timing period of about 0.5 s per microfarad of C1 value. The circuit is triggered by a positive-going input signal and generates a positive output waveform which is direct-coupled back to one input of IC1a to effectively maintain a 'trigger' input once the true trigger signal is removed.



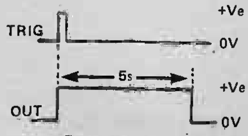


Fig.8a Standard 4001B monostable with a 5 s timing period.

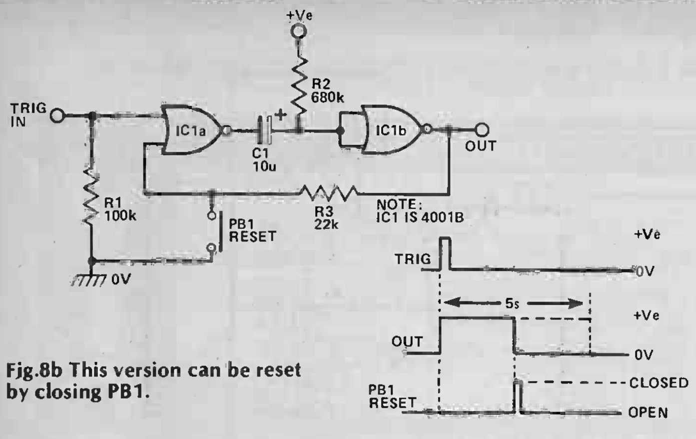


Fig.8b This version can be reset by closing PB1.

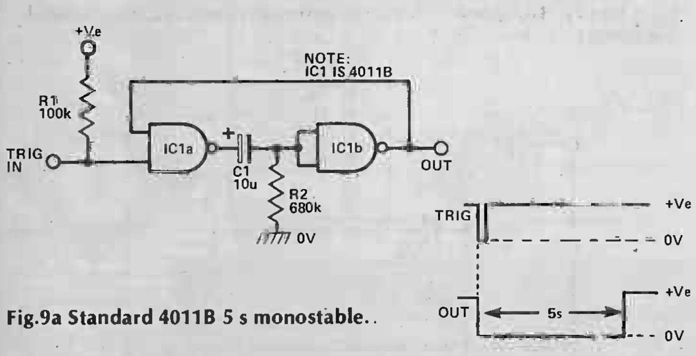


Fig.9a Standard 4011B 5 s monostable..

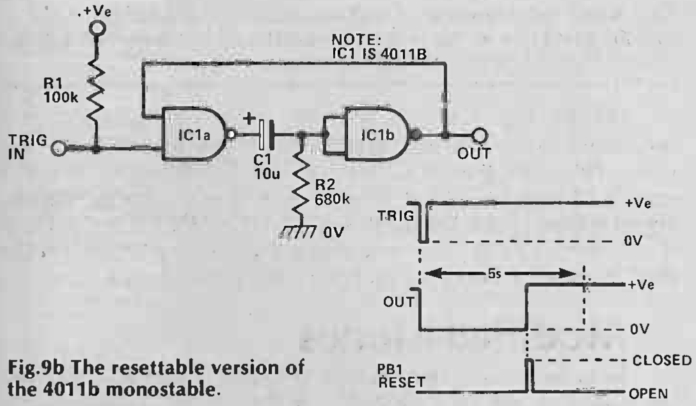


Fig.9b The resettable version of the 4011b monostable.

Figure 8b shows how the above circuit can be modified to give resettable operation. Here, the feedback connection from the IC1b output to the IC1a input is made via R3. Consequently, once the circuit has been triggered *and the original trigger signal has been removed*, the circuit can be reset at any time by simply pulling the feedback input of IC1a to ground. In the diagram we've shown this reset function accomplished using a simple push-button switch, but in practice it can be done using a gated transistor or CMOS switch, for example.

Figure 9 shows the 4011B version of the monostable unit, with the standard design in (a) and the resettable version in (b). Note here that the circuit is triggered by a negative-going input pulse and generates a negative or low output waveform.

New-fangled Flip-flops

Finally, to complete this edition of Notebook, Figs. 10 and 11 show a couple of unusual CMOS bistable circuits, each capable of being built using simple CMOS inverters or inverter-connected 4001B or 4011B gates. You'll sometimes find in project design that at some stage you'll have a couple of 'spare' 4001B or 4011B gates in a circuit, and at the same time need to use a simple bistable in the design, only to find that the spare gates are not compatible with the kind of bistable operation that is needed. The conventional 4001B bistable, for example, needs positive set and reset pulses, while the 4011B bistable needs negative set and reset pulses. The Fig. 10 or 11 circuits may solve your problems in such cases.

The operation of Fig. 10 circuit is pretty simple. Normally, the input of IC1a is held low by R1, the output of IC1a and the input of IC1b are high, the output of IC1b is low, and the circuit is in a stable state. If a positive 'set' pulse is momentarily applied across R1, the output of IC1b flips high and D1 then pulls the direct input of IC1a high and latches the circuit into this state, irrespective of subsequent actions of the set signal. The circuit can be reset by momentarily applying a positive pulse to the input of D2, thereby driving the output of IC1b low and latching the circuit back into its original state. Note that this circuit is triggered (set and reset) by positive-going input signals.

Figure 11 shows an alternative version of the bistable circuit, which in this case is triggered by negative-going inputs. The circuit is similar to that of Fig. 10, except that the polarities of the two diodes are reversed and the input of IC1a is normally biased high by R1. Note that both of these circuits have two outputs, thus providing either type of output polarity.

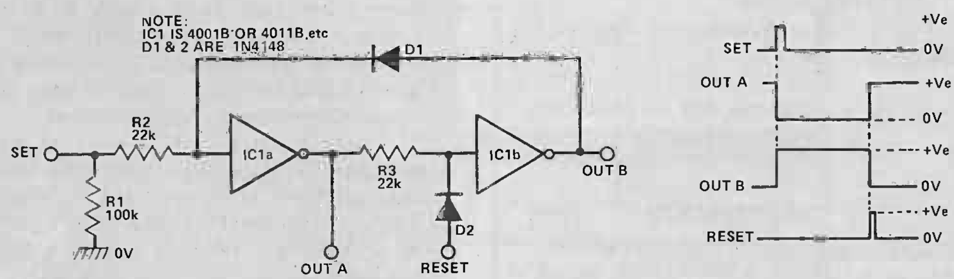


Fig.10 Positively-triggered bistable circuit.

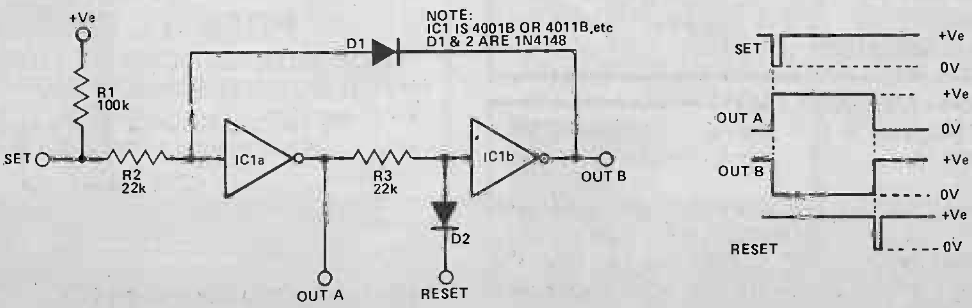


Fig.11 Negatively-triggered bistable circuit.

HAND CLAP SYNTHESISER

Does your snare drum suffer from nervous skin tension, lack of timbre? Then revive it with the ETI Hand-clap Synthesiser. Designed to simulate the staccato effect of multiple hand-claps, the unit can be triggered by a microphone or footswitch. Design by Roger Shore. Development by Steve Ramsahadeo.

It would seem that no disco record is complete without the familiar hand-claps that faithfully accent the snare drum's down beat. One can imagine a group of people centred around a studio microphone, palms reddening, acting like human metronomes. We are happy to report that such a form of torture is now unnecessary in this electronic age!

It's generally accepted that the advent of the synthesiser in the late 60s was the commercial starting point of electronic music, not so much in the way of percussive synthesis but with such effects as tremolo, fuzz, flanging, reverberation and phasing, all of which are added to give expression to a piece of music.

Synthetic Control

The reproduction of synthetic voices, be it in digital or analogue form, requires precise control of all levels contributing to the original make-up of the sound. Musicians, producers and arrangers are continually striving for new creative sounds, and the pressure eventually falls on the engineer who is called upon to wave his magic wand and come up with the latest synthetic sound which will send the fans wild.

As in any new venture, whatever approach is used will be an expensive one. At present there are some systems commercially available. The Fairlight CMI (Computer Musical Instrument) is one, retailing at around £15,000 with its sophisticated electronics and hardware. It can create any musical sound you care to name. With such technology available creativity is limited only by the operator. However, if you prefer a dedicated instrument the LM1 Drum Computer can offer some interesting prospects. The unit is programmable, capable of accepting 100 drumbeats in real time, and there are real drum sounds — digital recordings stored in computer memory. Twelve percussive voices are provided (all tunable in pitch), there are facilities for versatile editing, a 'human' rhythm feel is made possible by special timing circuitry, and so the list goes on.

It is therefore not surprising that when designing the ETI Hand-clap Synthesiser some ground work was required.

No Applause Please

Multiple or 'ensemble' hand-clapping may be analysed subjectively in two distinct sections:

1. A general 'crash' — which may be simulated with a short burst of tuned noise.
2. Individual claps — this can be simulated by generating pulses which cause a multiple feedback band-pass filter to ring. Several different combinations of individual claps were

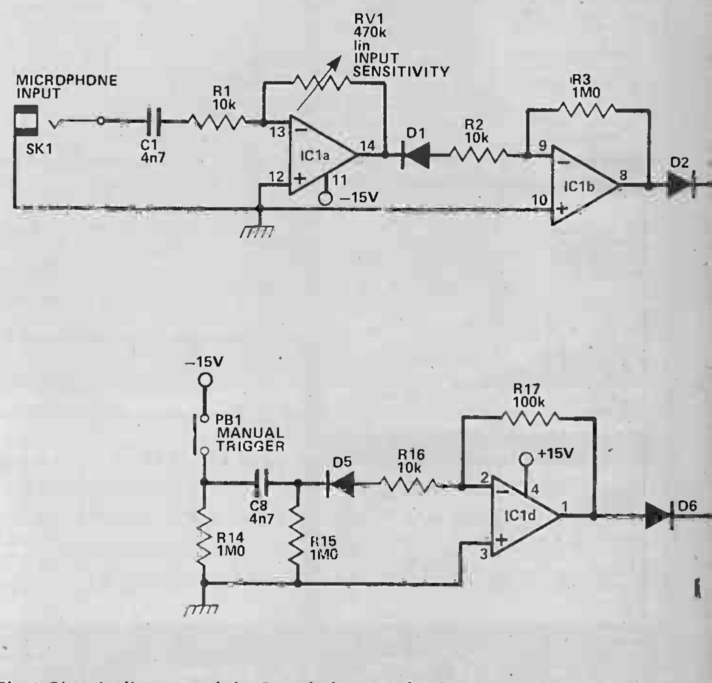


Fig.1 Circuit diagram of the Hand-clap Synthesiser.

tried from one to seven, at both regular and irregular intervals, but two provided the best subjective results.

Setting up a unit such as this will depend on personal preferences and also on the type of amplifying system used. It is preferable to use a unit with reverberation where possible as this will greatly enhance the effect.

The problem of which variables should be external and which should remain preset is also one of personal taste. As circumstances dictate different settings we decide to make all seven controls external.

Construction

No problems should be encountered in constructing the Hand-clap Synthesiser. The power supply section should be built first; care should be taken to sleeve the mains terminals on the PCB and the on/off switch.

When this is completed, connect a voltmeter across the output pins of the supply. A reading of +15 V and −15 V should be available at the output. If all is well the rest of the control circuit can be constructed observing the usual CMOS handling procedure and the orientation of polarised components.

Back panel of the synthesiser. Sockets are provided for the manual trigger (an external footswitch) or a microphone, triggered by the snare drum for example.

HOW IT WORKS

The unit can be triggered from either a momentary push-button (PB1) or from a suitable transducer, eg a microphone placed near a snare drum.

In the first case, pressing PB1 causes a negative-going pulse to be developed across C8. This is steered via D5 to the inverting input of IC1d, causing a positive pulse to appear at the cathode of D6.

Alternatively, an input signal from a microphone is differentiated by C1 and R1. This prevents false triggering from other nearby sources. The signal is amplified and inverted by IC1a with RV1 acting as a sensitivity control. Further inversion by IC2b is required to provide a positive pulse at the cathode of D2. These trigger pulses appearing at the cathodes of D2 or D6 are fed to both the anode of D3 and pin 1 of IC2a.

When D3 is forward biased by the trigger pulse it allows C3 to charge positively. The rate of discharge is determined by R5 and the setting of RV2; this ramp is buffered by IC1c, the output of which is connected to D4 and C4 via R8.

The base-emitter junction of Q1 is reversed biased to produce the required noise. A low noise transistor is chosen to give a cleaner noise source. This noise is amplified by IC3a and fed to the cathode of D4. When a trigger pulse causes a positive ramp to appear at the output of IC1c, D4 conducts allowing noise to pass via D4 and C4 to the band-pass filter formed by IC3b and associated components. The length of this noise pulse is determined by the setting of RV2, the ramp discharge time.

R9 normally holds the anode of D4 at approximately −1V5 to prevent noise peaks from turning D4 on intermittently.

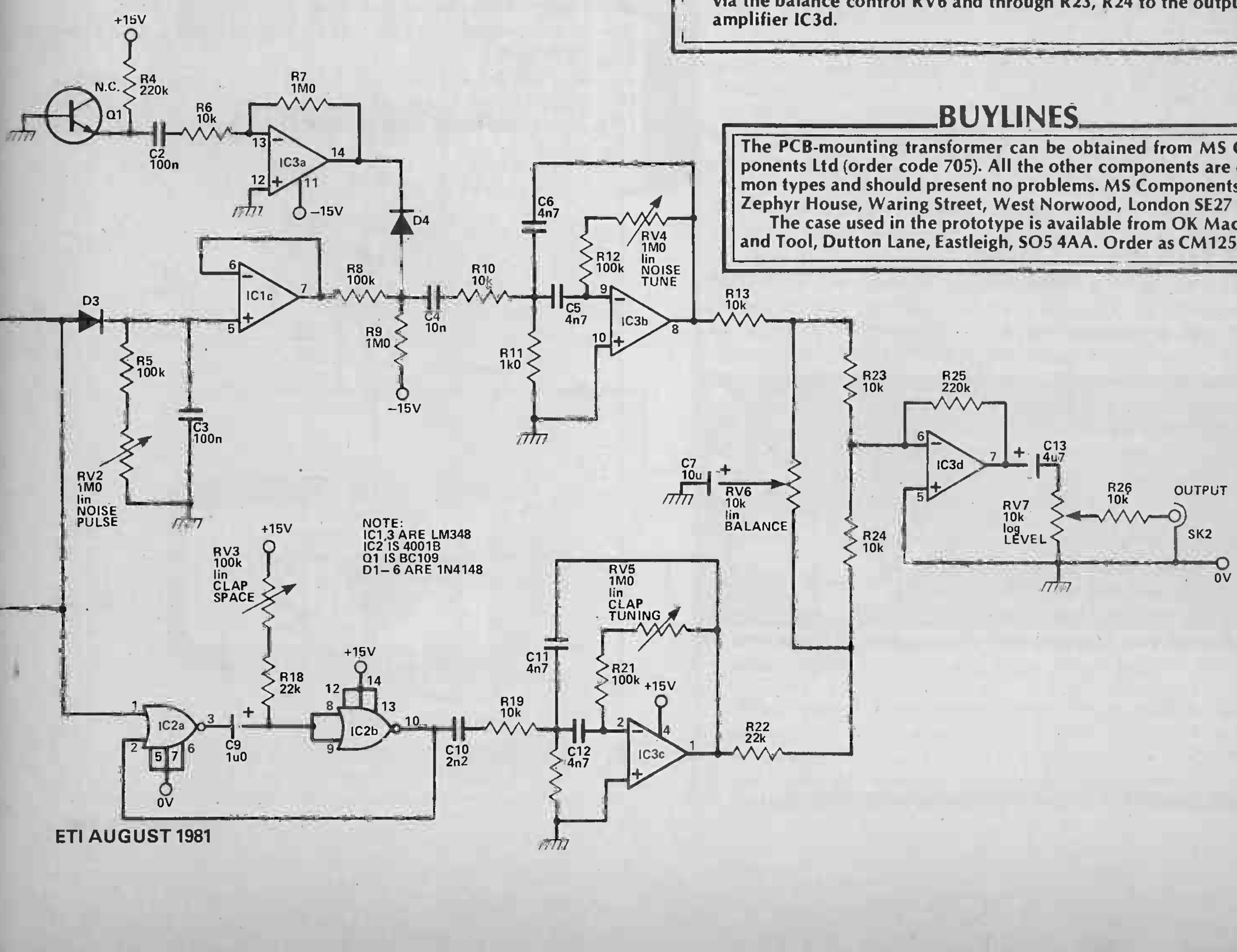
The band-pass filter is tuned over the 'useful' part of the noise spectrum for this application. Although the Q of the filter network will vary (because RV4 is not 'ganged' with R10), this does not pose any problem in this non-critical situation.

At the same time as the noise pulse is generated, the trigger pulse is applied to pin 1 of IC2a, turning on the monostable formed by IC2a and IC2b and allowing pin 10 to assume a high state. This positive voltage is developed across C10, causing the band-pass filter formed around IC3c to ring at a frequency determined by the position of RV5. (The two band-pass filters are of identical design.) At a time determined by RV3 and C9 the monostable will reset and the negative-going edge at pin 10 of IC2b allows a second ringing pulse to be generated by the band-pass filter. These two ringing pulses are the individual claps and are mixed with the noise pulse via the balance control RV6 and through R23, R24 to the output amplifier IC3d.

BUYLINES

The PCB-mounting transformer can be obtained from MS Components Ltd (order code 705). All the other components are common types and should present no problems. MS Components Ltd, Zephyr House, Waring Street, West Norwood, London SE27 9LH.

The case used in the prototype is available from OK Machine and Tool, Dutton Lane, Eastleigh, SO5 4AA. Order as CM125-225.



PARTS LIST

Resistors (all ¼ W, 5%)	
R1,2,6,10,13,16,19,23, 24,26	10k
R3,7,9,14,15	1M0
R4,25	220k
R5,8,12,17,21	100k
R11,20	1k0
R18,22	22k
R27	1k2
Potentiometers	
RV1	470k linear
RV2,4,5	1M0 linear
RV3	100k linear
RV6	10k linear
RV7	10k logarithmic
Capacitors	
C1,5,6,8,11,12	4n7 ceramic
C2,3	100n polycarbonate
C4	10n polycarbonate
C7,16,19	10u 35 V tantalum
C9	1u0 35 V tantalum
C10	2n2 ceramic
C13	4u7 35 V tantalum
C14,17	1000u 25 V axial electrolytic
C15,18	220n polycarbonate
Semiconductors	
IC1,3	LM348
IC2	4001B
IC4	78L15
IC5	79L15
Q1	BC109
BR1	50 V, 1 A bridge rectifier
D1-6	1N4148
LED1	0.125" red LED
Miscellaneous	
T1	15-0-15, 3 VA PCB-mounting transformer (see Buylines)
SW1	DPDT miniature toggle
PB1	momentary push-button
SK1	¼' jack socket
SK2	phono socket
FS1	50 mA fuse and holder
Case (see Buylines), seven collet knobs.	

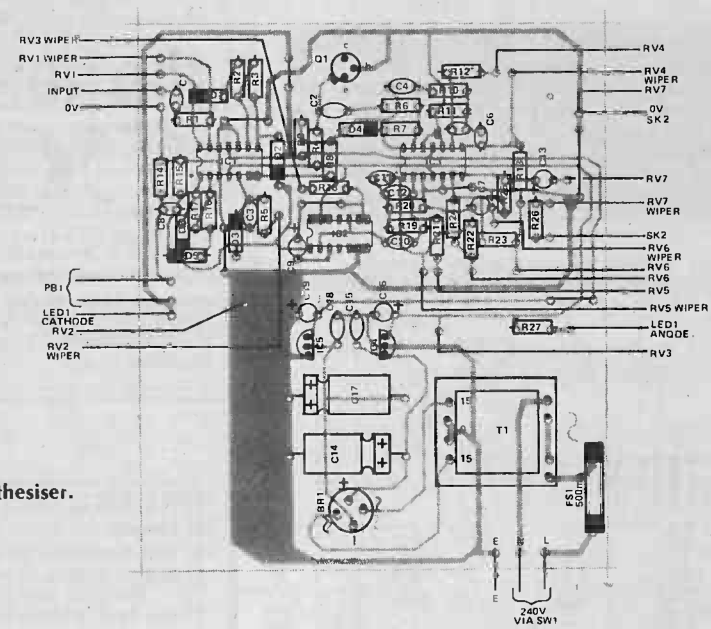


Fig.2 Component overlay for the Hand-clap Synthesiser.

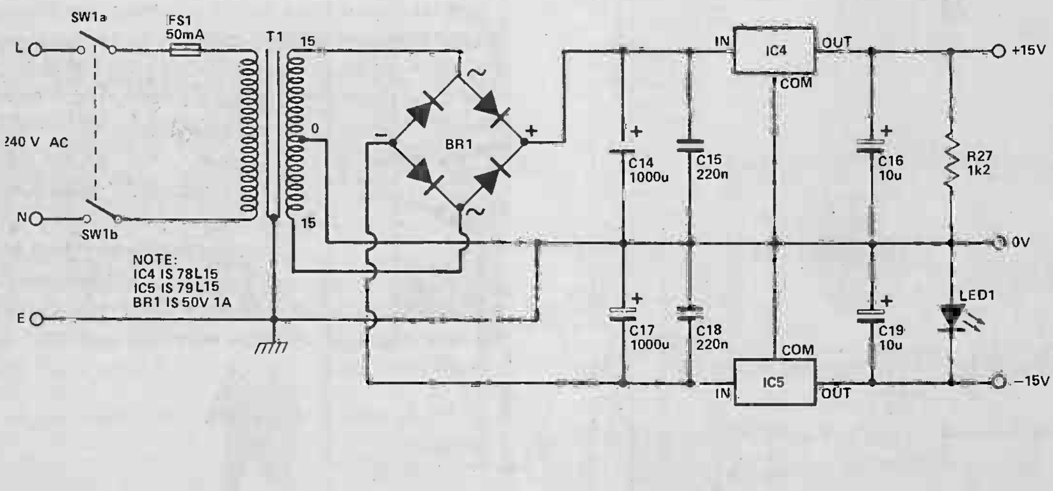
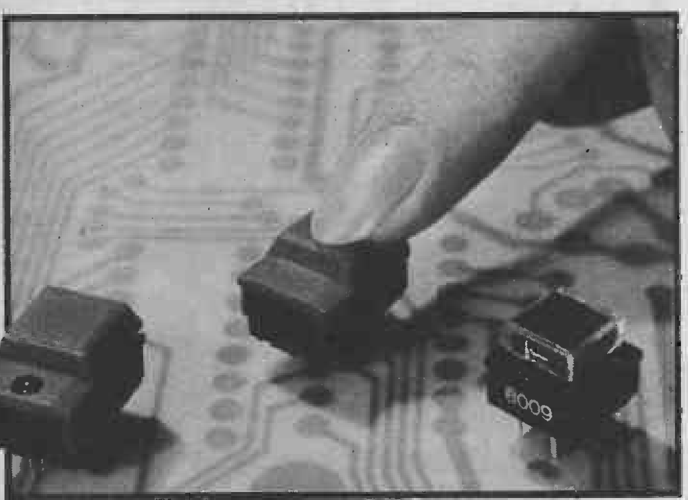


Fig.3 Circuit diagram of the power supply for the synthesiser.

Components

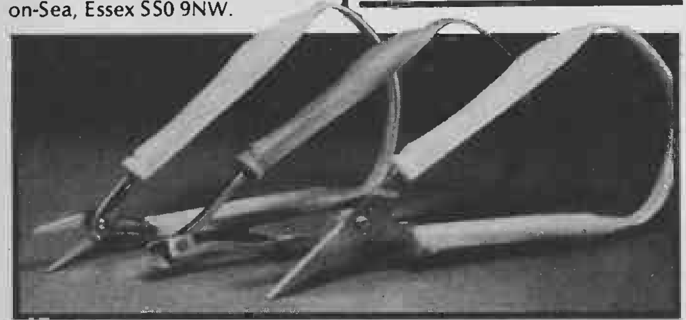


Handy Size!

Tele-Production Tools have announced the introduction of a set of three 'Easi-Grip' miniature handtools designed for use in electronics and fine modelling. All of them have self-opening handles and are operated with finger tips for fine control and ease of operation. The set consists of miniature carbon steel side cutters, fine nosed stainless steel tweezer pliers and a serrated stainless steel scissor/shear for fine wires, boards, foil, etc. The cost is £3.75 each including p&p and VAT, or £10 for the set of three from Tele-Production Tools Ltd, Stiron House, Electric Avenue, Westcliff-on-Sea, Essex SS0 9NW.

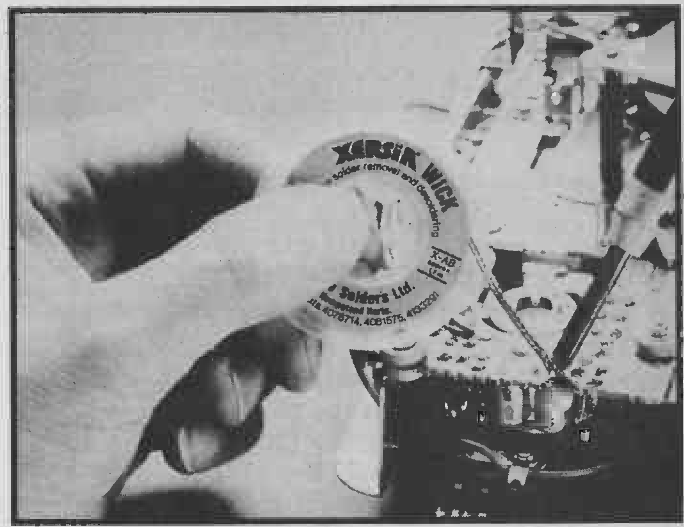
Tap Switches

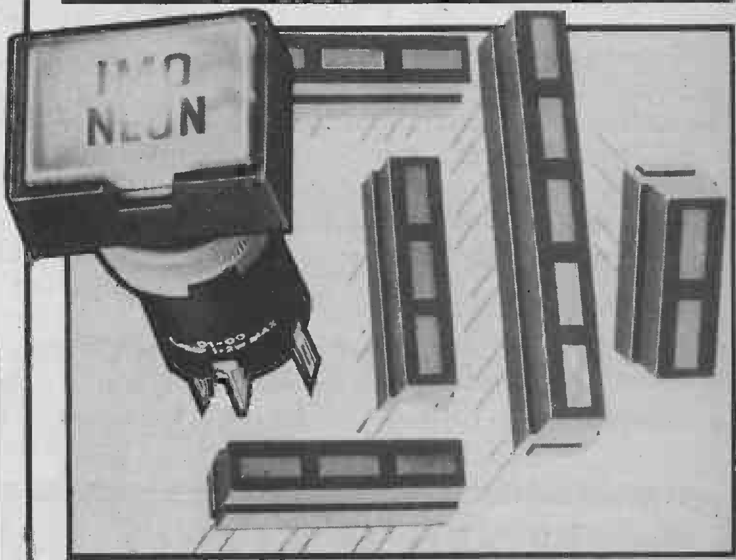
A new range of miniature 'tap' switches for direct mounting on PCBs is now available from Sifam Ltd. There are three basic models of the switches: an ordinary square push-button design, a hinged cap type and a hinged cap type with LED to indicate switch status. Each type is available with colour coded caps in orange, blue, black, grey or red, with a choice of a transparent cover for the square cap type. LED indicators can be red, green or yellow. For further information contact Sifam Ltd, Woodland Road, Torquay, Devon TQ2 7AY.



Desolder Wick

Xersin Wick is a new kind of desoldering wick, using a unique flux/preservative developed by Multicore which doesn't have the oxidation problems usually associated with natural rosin fluxes. It comprises of high quality copper braid specially deoxidised prior to its xersin coating and, unlike conventional wicks, it doesn't get brittle and flaky, but remains pliable even after years on the shelf. Supplied in four different width sizes, ranging from 0.8 mm to 2.7 mm, Xersin Wick is dispensed from a plastic spool in 1.5 m lengths. The Wick is manufactured by Multicore Solder Ltd, Maylands Avenue, Hemel Hempstead, Herts HP2 7EP.





Available in 110 V AC and 240 V AC 1 amp contacts these mains-operated (neon) illuminated push-button switches are from IMO Electronics Ltd, 349 Edgware Road, London W2 1BS.

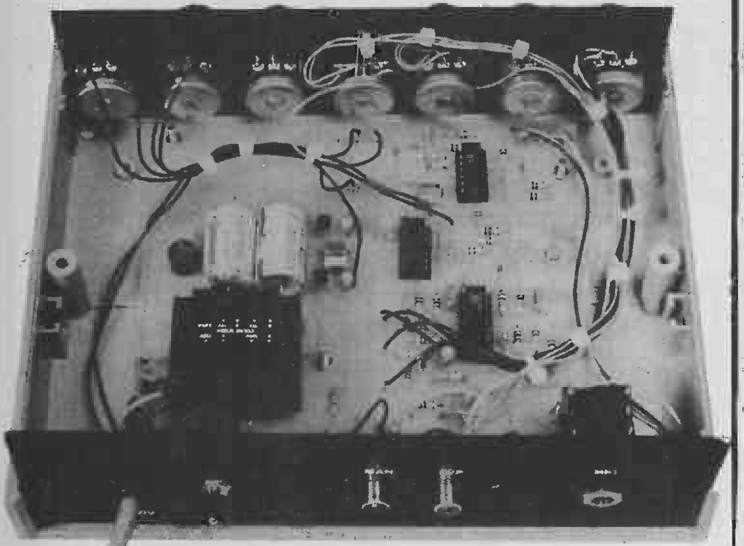
Multi-LED arrays are available from Zaerix Electronics Ltd in 2,3,4 or 5 segment lamp units in red, yellow or green from 46 Westbourne Grove, London W2 5SF.

Low Cost Frequencies

A complete frequency meter built into a module less than ½" deep is now available and costs £19.95. The FM77T will directly measure and display frequencies up to 3999.9 kHz. With external prescaling, this can be extended to 39.999 MHz or 399.99 MHz. Stability is better than ±1 digit over a 10°C to 30°C range and is defined by a built-in crystal timebase. The display is high contrast reflective LCD with 9 mm high characters and user selectable decimal points and kHz/MHz legends. For radio receiver applications the user can select any one of 23 pre-programmed standard IF offset frequencies, enabling a reception frequency to be displayed by measurement of the Local Oscillator. The FM77T operates from a single power rail of between 4½ and 7 V and consumes only 1 mA. Overall size if 2¾" x 1½" x 7/16" and the display is mounted behind a textured bezel. It is available from Thurlby Electronics Ltd, Office Suite 1, Coach Mews, The Broadway, St Ives, Huntingdon, Cambs PE17 4BN.

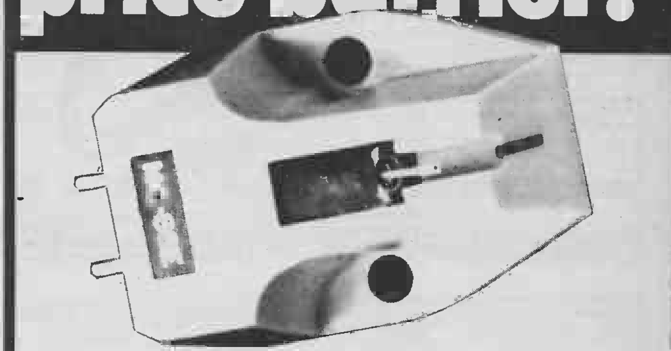
Bin It!

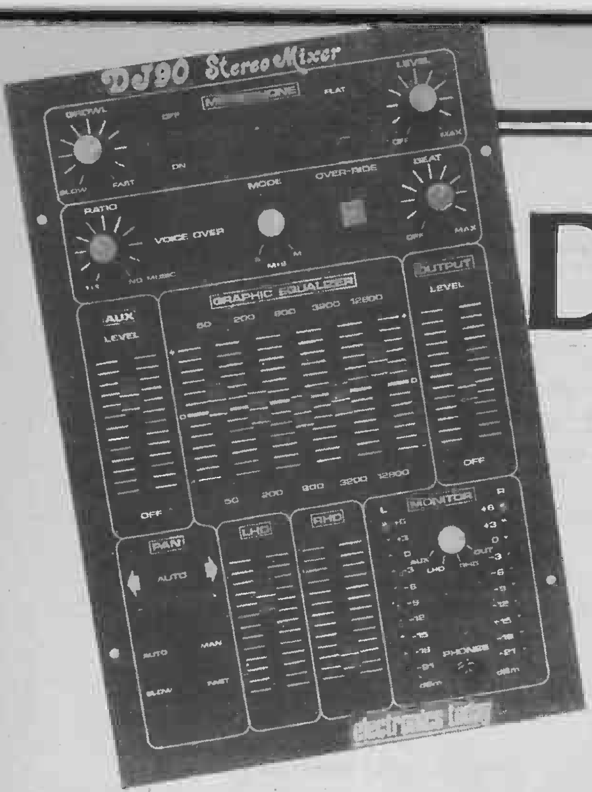
Improved bins for small parts and components cost around £12 (plus VAT) for 50 from Bankers Box Storage Systems. The parts bins are made from corrugated fibre-board and are a development of an existing product range from the company. Contact Bankers Box Storage Systems, Doncaster Road, Kirk Sandell, Doncaster DN3 1HT for further information.



The Hand-clap Synthesiser with the lid removed. The sockets and switch mount on the rear panel, and the potentiometers and LED on the front; all the other components, including the mains transformer, fit onto the single PCB, making construction extremely easy. Follow the overlay and this photograph when interwiring and position the wiring looms exactly as shown, in order to minimize hum and noise. Plastic cable ties will keep everything neat and tidy.

ETI

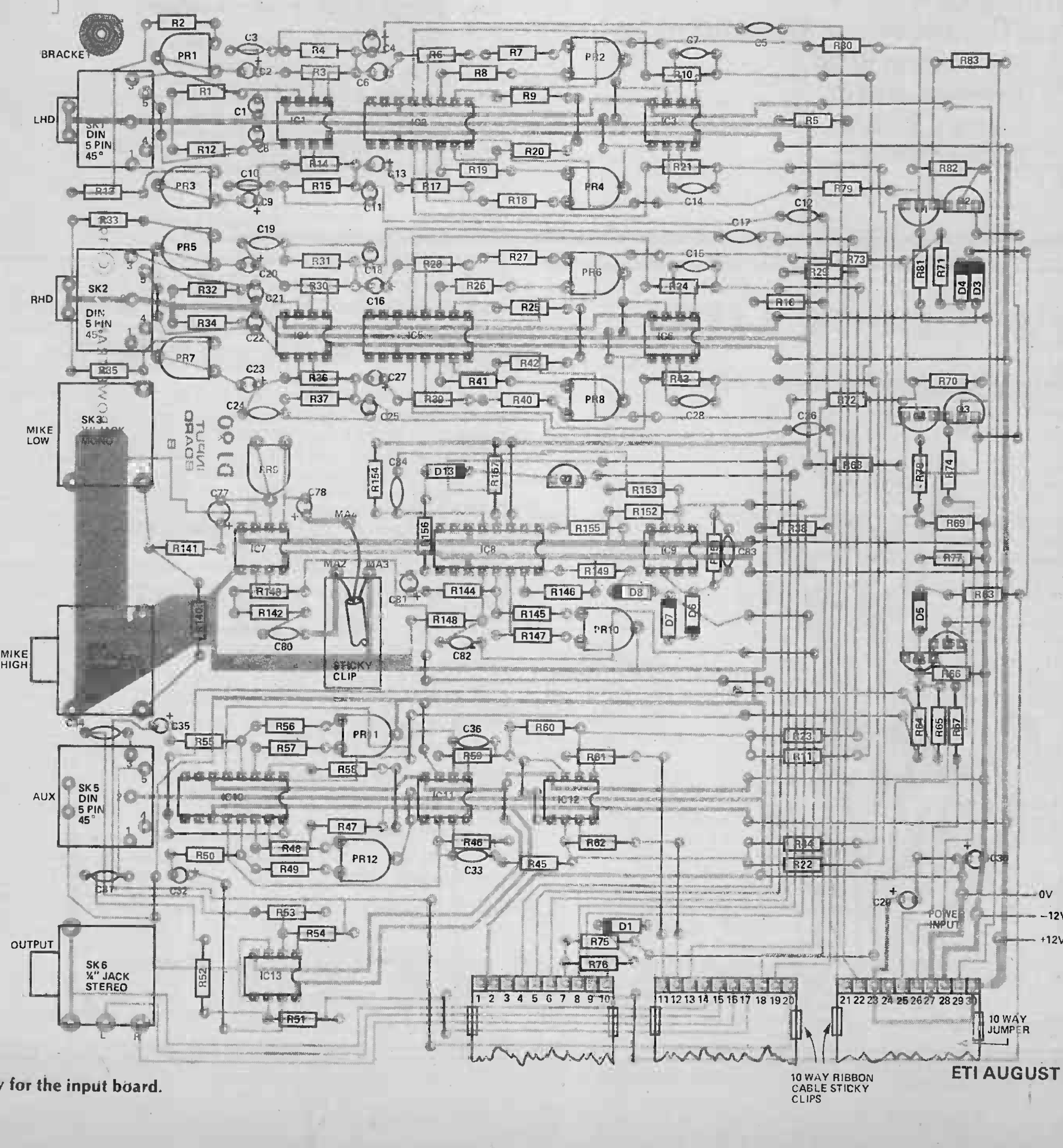




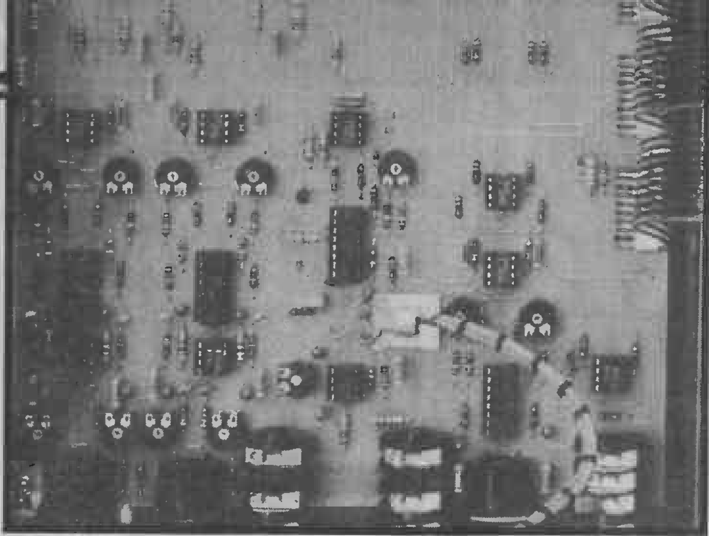
DISCO MIXER

Part 2 of the DJ90 Stereo Mixer describes the input and equaliser circuitry, and construction of the input board. 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



Overlay for the input board.



The input board. The input and output sockets are all PCB-mounting and connections to the panel board are made via multiway cable and a Molex connector.

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.

PARTS LIST

Resistors (all ¼ W, 5%)

R1,12,32,34,52,53,99,117,142, 160,162,164,170,172,176	47k
R2,13,33,35,69	1k0
R3,5,14,16,29,30,36,38,87,152	1M0
R4,15,31,37,51,54,89,97,143, 179,180,182,188,189,191,199, 206	100k
R6,17,28,39,50,55,144	5k6
R7,8,18,19,26,27,40,41,48, 49,56,57,145,147	390R
R9,11,20,22,23,25,42,44,45,47, 58,60,61,62,67,98,101,106,107, 108,109,110,111,112,113,114, 115,119,124,125,126,127,128, 129,130,131,132,134,146,148, 158,165,166,171,202,203	10k
R10,21,24,43,46,59	6k8
R63,71,81	1k3
R64,65,72,73,79,80	1k8
R66,70,82,95	68k
R68	2k0
R74,83	8k2
R75	130k
R76	39k
R77	36k
R78	1k2
R84,175	2M2
R85,86	10M
R88,92,93,94,100,102,105,116, 118,120,123,135,163	150k
R90	820R
R91,181,183,190,192	200k
R96	560k
R103,104,121,122,154,156, 169,174	18k
R136,139,149,177	22k
R137,138,140	27k
R141,173	1k5
R150,184,193	2k2
R151,153	330k
R155,197,204	220k
R157	180R
R159,178,185,194	4k7
R161	470k
R167	12k
R168	33k
R186,195,198,205	3k3
R187,196	47R
R200,207	2R7
R201,208	15R

Note that R133 is not used

Potentiometers

RV1	100k dual antilogarithmic rotary
RV2	10k logarithmic rotary
RV3	10k linear rotary
RV4	10k dual linear rotary
RV5	10k logarithmic slider
RV6,7,8,9,10	100k dual linear slider
RV11	47k dual logarithmic slider
RV12,13	47k logarithmic slider
PR1,3,5,7	1k0 miniature horizontal preset
PR2,4,6,8,10,11,12	220R miniature horizontal preset
PR9	470k miniature horizontal preset
PR13	4k7 miniature horizontal preset
PR14,15	22k miniature horizontal preset

Capacitors

C1,8,21,22	47p ceramic
C2,9,20,23,29,30,40,111,112	10u 16 V tantalum
C3,10,19,24	3n3 polycarbonate
C4,11,18,25	680p ceramic
C5,12,17,26,31,34,37,38,39,41, 55,80,82,92,95,100,103,105	100n polycarbonate
C6,13,16,27,32,35,75,76,78,81, 89,90	1u0 35 V tantalum
C7,14,15,28,33,36,50,51,64,65	1n0 polycarbonate
C42,56,87,88	15n polycarbonate
C43,45,57,59	680n polycarbonate
C44,47,58,61,84,85,86	150n polycarbonate
C46,60	33n polycarbonate
C48,52,62,66	8n2 polycarbonate
C49,63	39n polycarbonate
C53,67	470p ceramic
C54,68	2n2 polycarbonate
C69	2u2 16 V tantalum
C70,91,93,96	470n 35 V tantalum
C71,73	220n polycarbonate
C72,74	100p ceramic
C77,99,106	4u7 16 V tantalum
C79,83	4n7 polycarbonate
C94,97,101,102,104,107	220u 16 V electrolytic, PCB type
C98,108	22n polycarbonate
C109,110	2200u 25 V axial electrolytic
C113,114	47n 50 V disc ceramic

Semiconductors

IC1,3,4,6,7,11,12,13,18,19,20, 21,22,23,24,25,26	RC4558
IC2,5,8,10	LM13600
IC9,15,27	741
IC14,16,17,28,31,33	1458
IC29	TL081 or equivalent
IC30	LM1877N-9
IC32,34	LM3915
IC35	7812
IC36	7912
Q1-7	BC212L
Q8	BC182L
D1-24	1N4148
D25-28	1N4002
ZD1	6V2 400 mW
LED1-3,5-13,15-23	0.2" red LEDs
LED4,14	0.2" green LEDs

Miscellaneous

SW1,2,5,6	DPDT slide switch
SW3	four-pole three-way rotary switch
SW4	SPDT toggle switch
SW7	three-pole four-way rotary switch
SW8	push-to-make non-locking switch
SK1,2,5	five-pin DIN sockets (PCB type)
SK3,4	¼" mono jack socket (PCB type)
SK6	¼" stereo jack socket (PCB type)
SK7	¼" stereo jack socket

Transformer (15-0-15 @ 300 mA), fuse (500 mA), fuseholder, mains lead, cabinet, knobs to suit, eight-pin DIL sockets (27 off), 16-pin DIL sockets, (four off), 14-pin DIL socket (one off), 18-pin DIL sockets (two off), flexible multiway cable and Molex PCB plugs, mounting hardware to suit.

PROJECT : Disco Mixer

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.

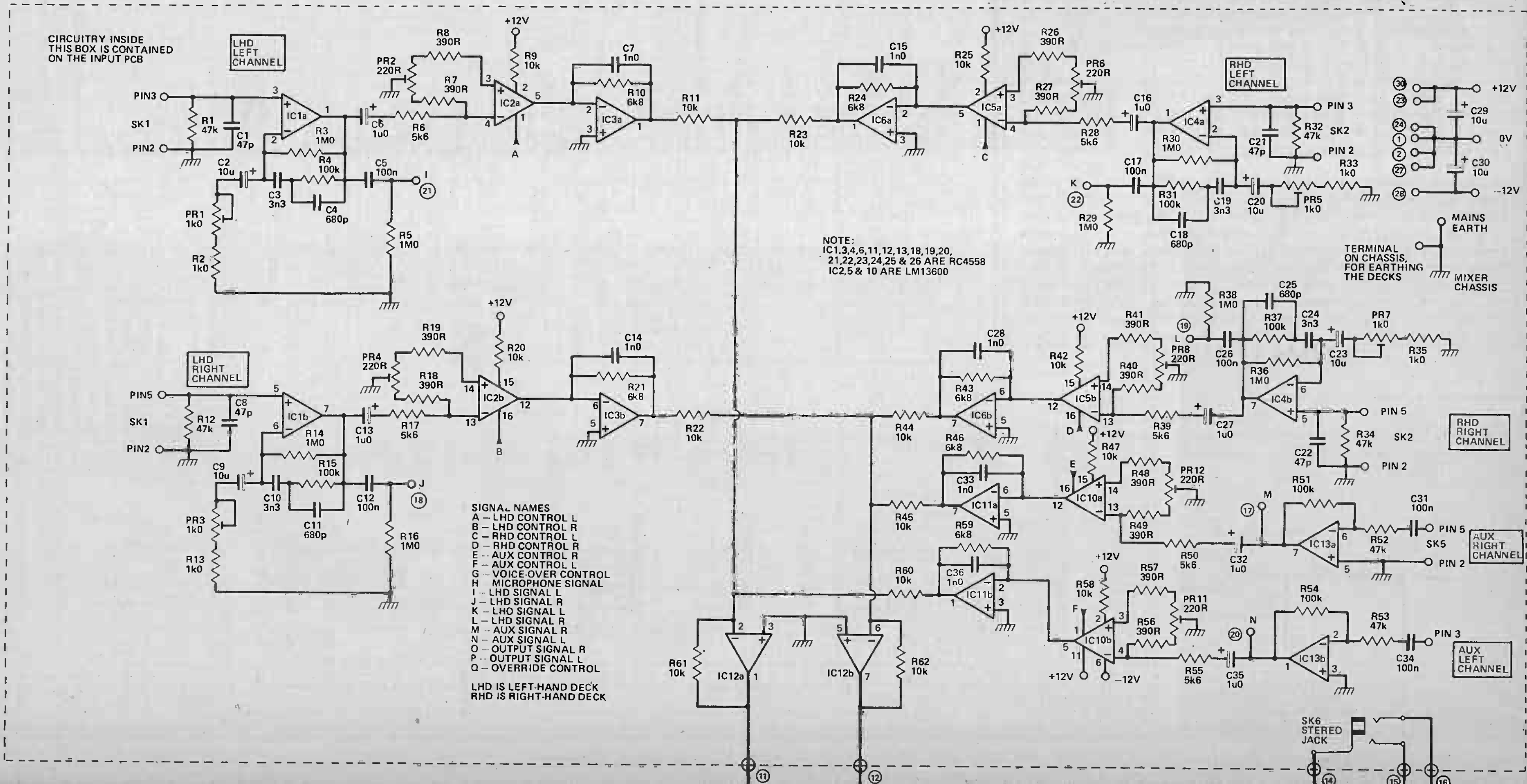
Voltage controlled amplifiers have been used to control the signal levels in all seven audio paths. They have the ability to produce automatic cross-fades and ducking, as well as reducing crosstalk. The signal is not transmitted anywhere until the control voltage is correct. Therefore it will not crosstalk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

TABLE 1.

Noise chart for converting dBm into voltage.

0 dBm	775 mV_{RMS}
−10 dBm	245 mV_{RMS}
−20 dBm	77.5 mV_{RMS}
−30 dBm	24.5 mV_{RMS}
−40 dBm	7.75 mV_{RMS}
−50 dBm	2.45 mV_{RMS}
−60 dBm	77.5 uV_{RMS}
−70 dBm	245 uV_{RMS}
−80 dBm	77.5 uV_{RMS}
−90 dBm	24.5 uV_{RMS}
−100 dBm	7.75 uV_{RMS}
−110 dBm	2.45 uV_{RMS}

Close-up showing the wiring to Molex connector 'A' (on the input board).



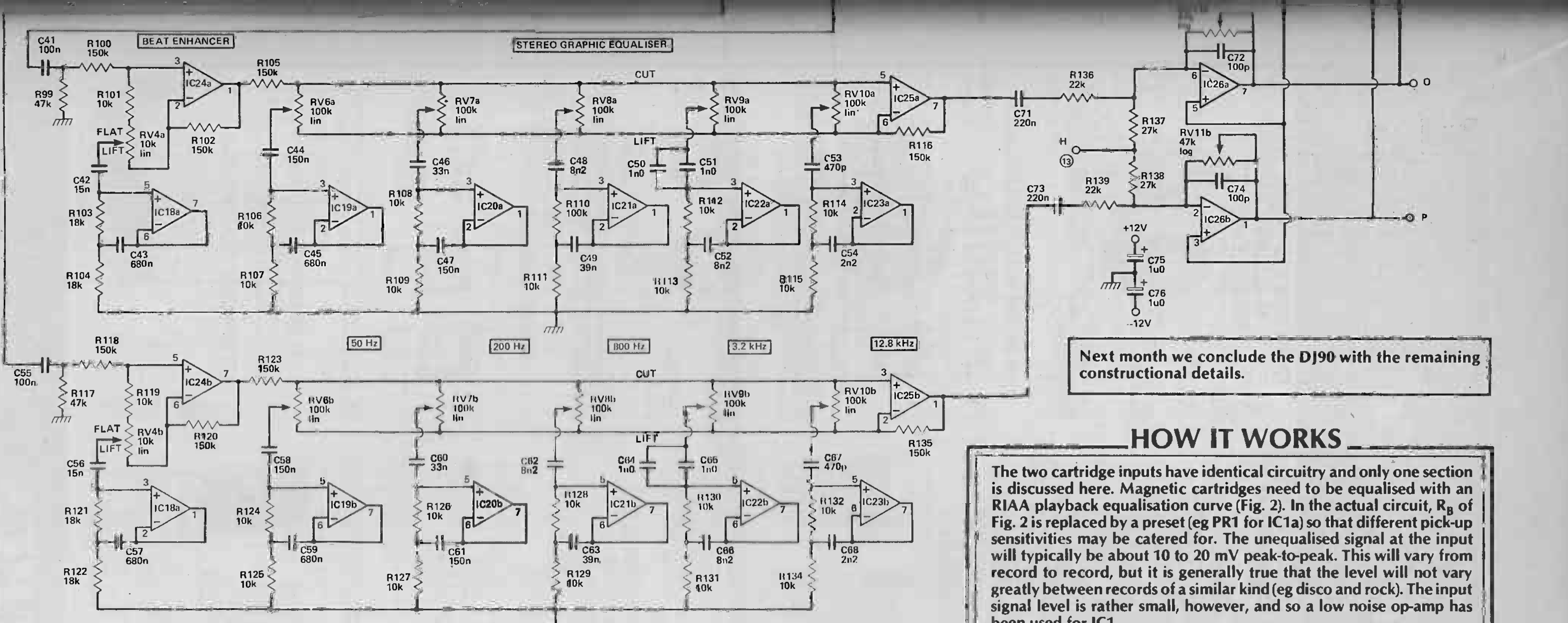


Fig.1 Circuit diagram for the music input and graphic equaliser sections.

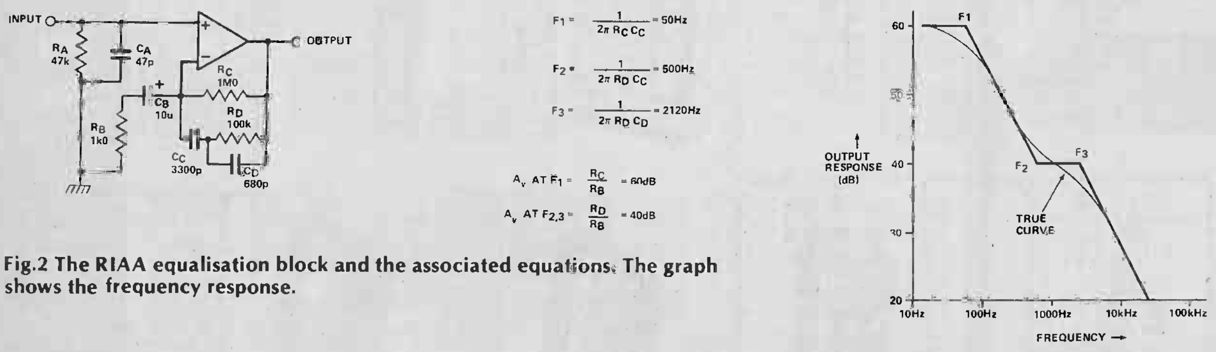
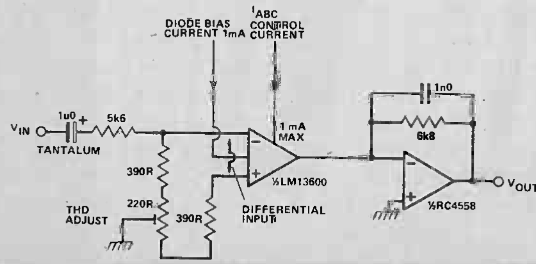


Fig.2 The RIAA equalisation block and the associated equations. The graph shows the frequency response.



TYPICAL MEASUREMENTS

MEASUREMENTS TAKEN OVER A 20 kHz BANDWIDTH AT 100 Hz, 1 kHz AND 10 kHz. I_{ABC} SET AT 1 mA. THE PRESET IS SET TO THE BEST THD POSITION.

V_{IN} dBm	V_{IN} Vpp	DIFFERENTIAL INPUT mVpp	OUTPUT THD%	OUTPUT S/N RATIO
+12dBm	8V8	80mV	0.8%	92.5dB
+6dBm	4V4	40mV	0.2%	86.5dB
0dBm	2V2	20mV	0.055%	80.5dB
−6dBm	1V1	10mV	0.04%	74.5dB
−12dBm	0V55	5mV	0.02%	68.5dB

Fig.3 The basic VCA unit and a table showing the performance figures (see text).

Next month we conclude the DJ90 with the remaining constructional details.

HOW IT WORKS

The two cartridge inputs have identical circuitry and only one section is discussed here. Magnetic cartridges need to be equalised with an RIAA playback equalisation curve (Fig. 2). In the actual circuit, R_B of Fig. 2 is replaced by a preset (eg PR1 for IC1a) so that different pick-up sensitivities may be catered for. The unequalised signal at the input will typically be about 10 to 20 mV peak-to-peak. This will vary from record to record, but it is generally true that the level will not vary greatly between records of a similar kind (eg disco and rock). The input signal level is rather small, however, and so a low noise op-amp has been used for IC1.

The VCA unit uses the LM13600 Operational Transconductance Amplifier; in the actual circuit this section comprises IC2 and IC3. The gain of the unit depends on the control current I_{ABC}, so the unit should actually be called a current controlled amplifier (ICA); however, we shall stick to the misnomer of VCA. The LM13600 has a diode predistortion network included in the IC which actually reduces the THD of the unit by as much as 10 dB. It was decided to operate the VCA at a V_{IN} level of 0 dBm (2V2 peak-to-peak). This gives a THD of 0.055% (typical best) and a signal-to-noise ratio of 80.5 dB. It also gives an overhead of about 18 dB. The output noise at 0 dBm is well below that of the RIAA stage and so it can be ignored.

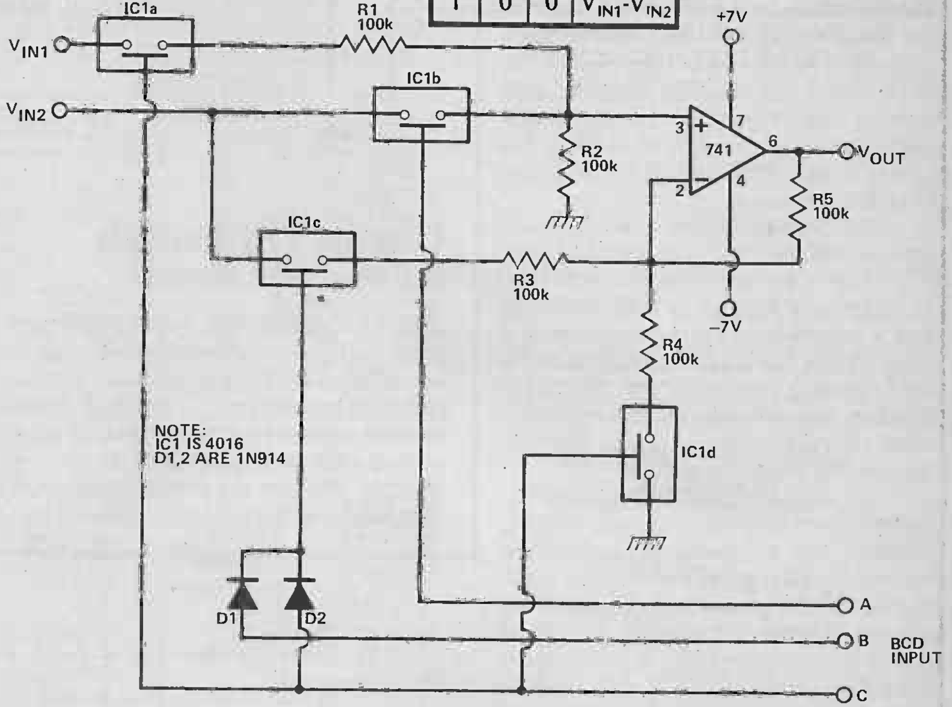
The AUX input uses the same VCA (IC10, 11), but it does not have an RIAA circuit. Instead it has a +6 dB gain, flat frequency response preamplifier, built around IC13.

All three music inputs are mixed together by IC12 and fed into the beat lift and graphic equaliser section. The beat lift circuit is a peaky 90 Hz resonator, designed around IC18 and IC24, that can provide a variable lift. This tends to emphasise the beat in the music. The graphic equaliser (ICs 19 to 23 and IC25) uses the same circuit as the beat lift except that it can provide both lift and cut.

The circuit operates as follows. Each section (IC19a, C44, R106, R107, C45, for example) is a fixed frequency resonator. At resonance the resonator looks like a short to the ground. By connecting this resonator to the wiper of a pot which is connected across the input and the feedback of an amplifier, the amplifier can be made to provide cut and lift respectively at the resonant frequency. The resonator selectively shorts out the input signal or the feedback signal. The output of the equaliser is fed into a variable gain mixer, IC26, together with the output from the microphone section. Throughout the system, the operating signal level is 0 dBm, although gain can be provided in the equaliser. The output is also capable of generating 6 dB of gain.

TECH TIPS

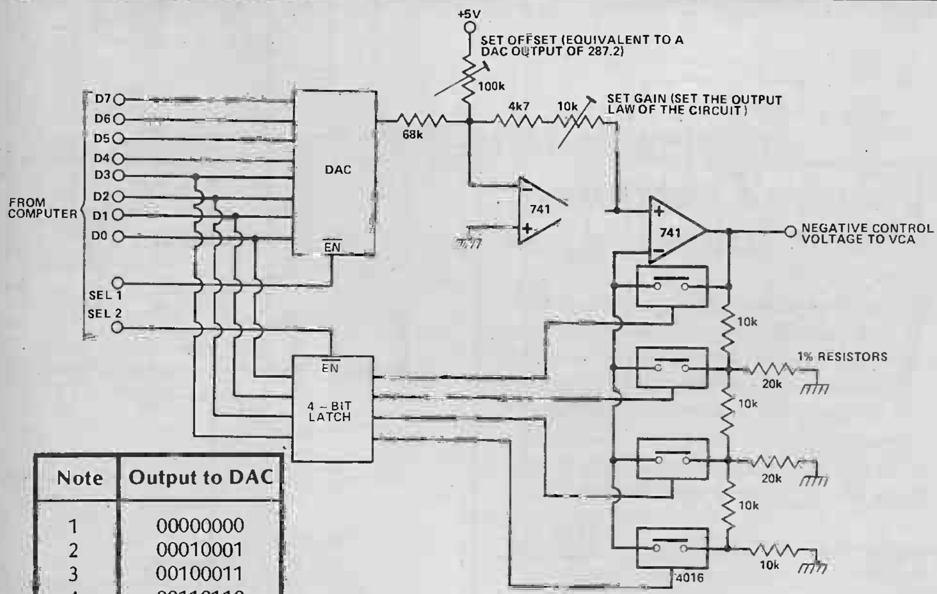
C	B	A	V_{OUT}
0	0	1	V_{IN2}
0	1	0	$-V_{IN2}$
1	0	0	$V_{IN1}-V_{IN2}$



Programmable Op-amp

J. P. Macauley, Crawley

This circuit was developed for experimental purposes and enables an op-amp to be operated in either the inverting, non-inverting or differential mode. This is accomplished by switching inputs into the circuit by means of IC1, a quad bilateral CMOS transmission gate. A diode matrix connects the control voltage for the gates together so that the operating mode can be controlled by a simple BCD-encoded word. As presented here the gain of the op-amp is unity, although this can be altered by the simple expedient of altering the respective resistor values. Note, however, that the output may distort if resistor values of less than 10k are used. This is due to the transmission gates, which like to 'see' at least this resistance as a load.



Note	Output to DAC
1	00000000
2	00010001
3	00100011
4	00110110
5	01001011
6	01100000
7	01110111
8	10001111
9	10101001
10	11000010
11	11011110
12	11111111

Octave	Output to Latch
1	00000001
2	00000010
3	00000100
4	00001000

Computer-controlled Synthesiser Keyboard

K. Wood, Ipswich

Referring to P. McChesney's idea published in Tech Tips, May '81 issue, it is possible to obtain an output voltage of an accuracy similar to that of the 12 bit code given without resorting to a second digital-to-analogue converter.

The logarithmic property of music allows this to be done. For each octave higher a note is, its frequency (and hence the control voltage of a linear oscillator), must double. Thus it is possible to generate voltages for the top octave of a keyboard, and obtain control voltages for lower octaves by dividing it by two, four, eight and so on, according to the number of octaves required.

An alternative arrangement is outlined to cope with these ideas. The note is output to the DAC, and its octave number is sent separately to the divider network by the computer system. A bias is added to the voltage to eliminate another bit from the 12 bit code, reducing it to the eight bit capability of the DAC.

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.

LED Peak Meter

G. Durant, Selby

This circuit gives a bar display when connected to an audio source, such as the speaker outputs of an amplifier. Each channel is fed to one of the drivers IC1a, IC1b, which are op-amps wired as peak voltage detectors; they also rectify the signal and give the output voltage the characteristic PPM fast attack and slow (1 s) decay times.

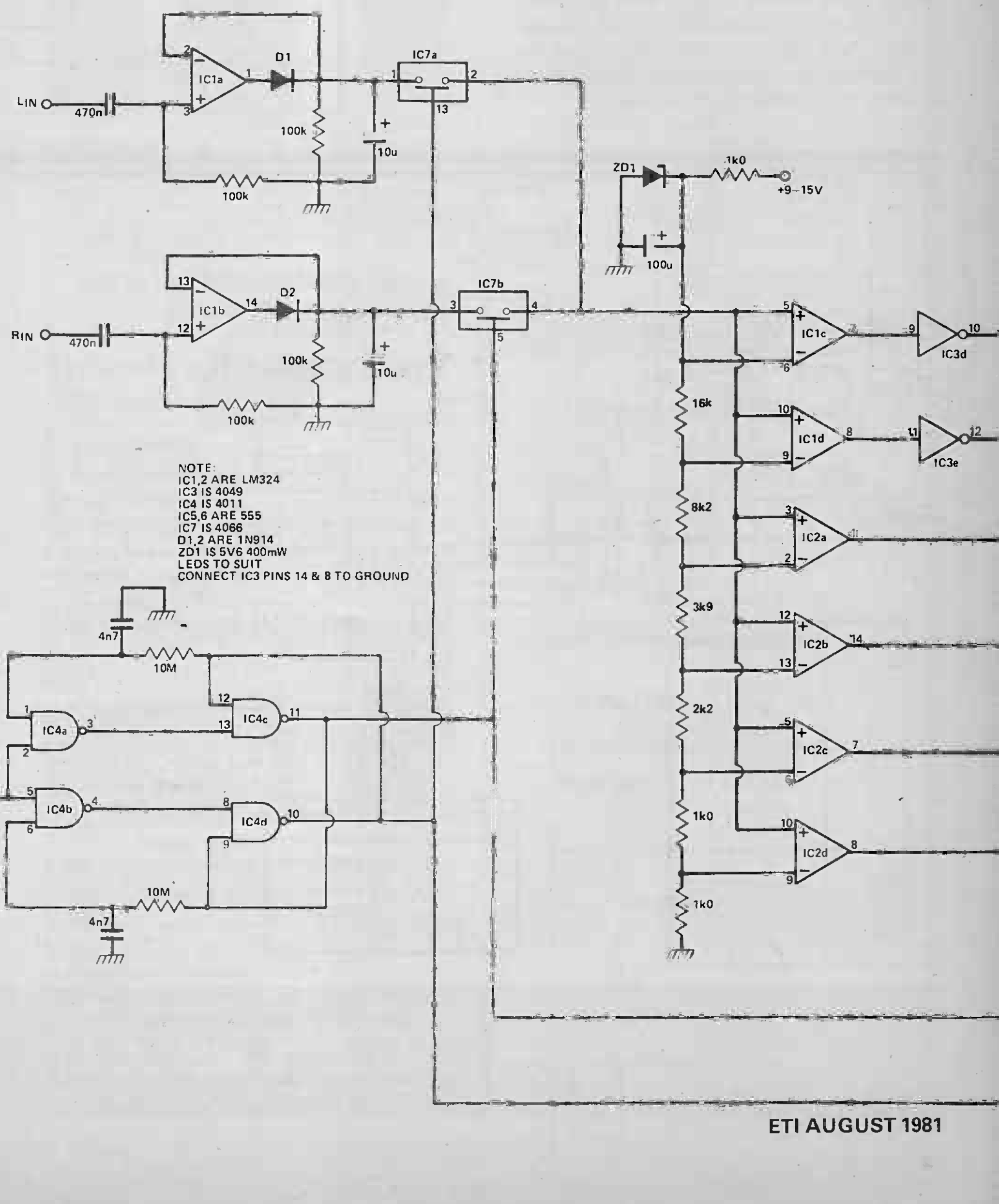
The display section is multiplexed for low cost. A row of comparators (IC1c-IC2d) have their inverting terminals held at a particular voltage by a resistor chain and a reference voltage derived from a zener diode. The audio signal is fed to the non-inverting terminals of the comparators and as it reaches the reference level on each inverting input, the corresponding output goes high.

The output of each comparator is taken to two LEDs via a current-limiting resistor. Two independent displays are formed by taking the cathodes of each pair of LEDs to one of two common lines; each line can be connected to ground by one of the switches IC7c, IC7d. The two remaining switches in IC7 select which channel is to be sampled by the comparator section.

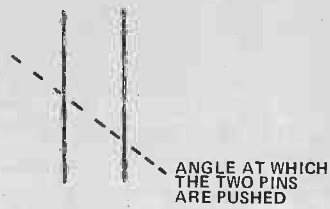
A clock formed by IC3a,b,c and running at about 100 Hz feeds a retriggerable flip-flop built around IC4. The two outputs control the IC7 switches so that each channel is measured and displayed on the appropriate LEDs in alternation.

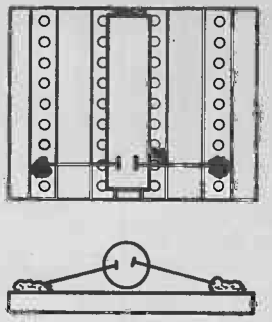
A brief peak which reaches the top two LEDs on each channel may be too short to notice. When the top two comparators are triggered, they activate the peak hold circuitry built around IC5 and IC6. These are 555 timers wired in the astable mode and prolong the time for which the LEDs are illuminated to about 600 ms and 1.2 s respectively. This gives the same impression as a commercial peak hold meter.

The circuit requires a single 15 V supply, and use can be made of the existing supply when the circuit is built into the equipment. The LEDs, which can be rectangular if preferred, are mounted in two rows of six and labelled with the level in dB (which is only approximate). The number of display LEDs can be extended by adding more resistors and comparators to the existing chain.

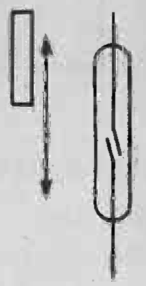


Below and right: Constructing a cheap tilt switch.





Below: 'Magnetic' tilt switch.

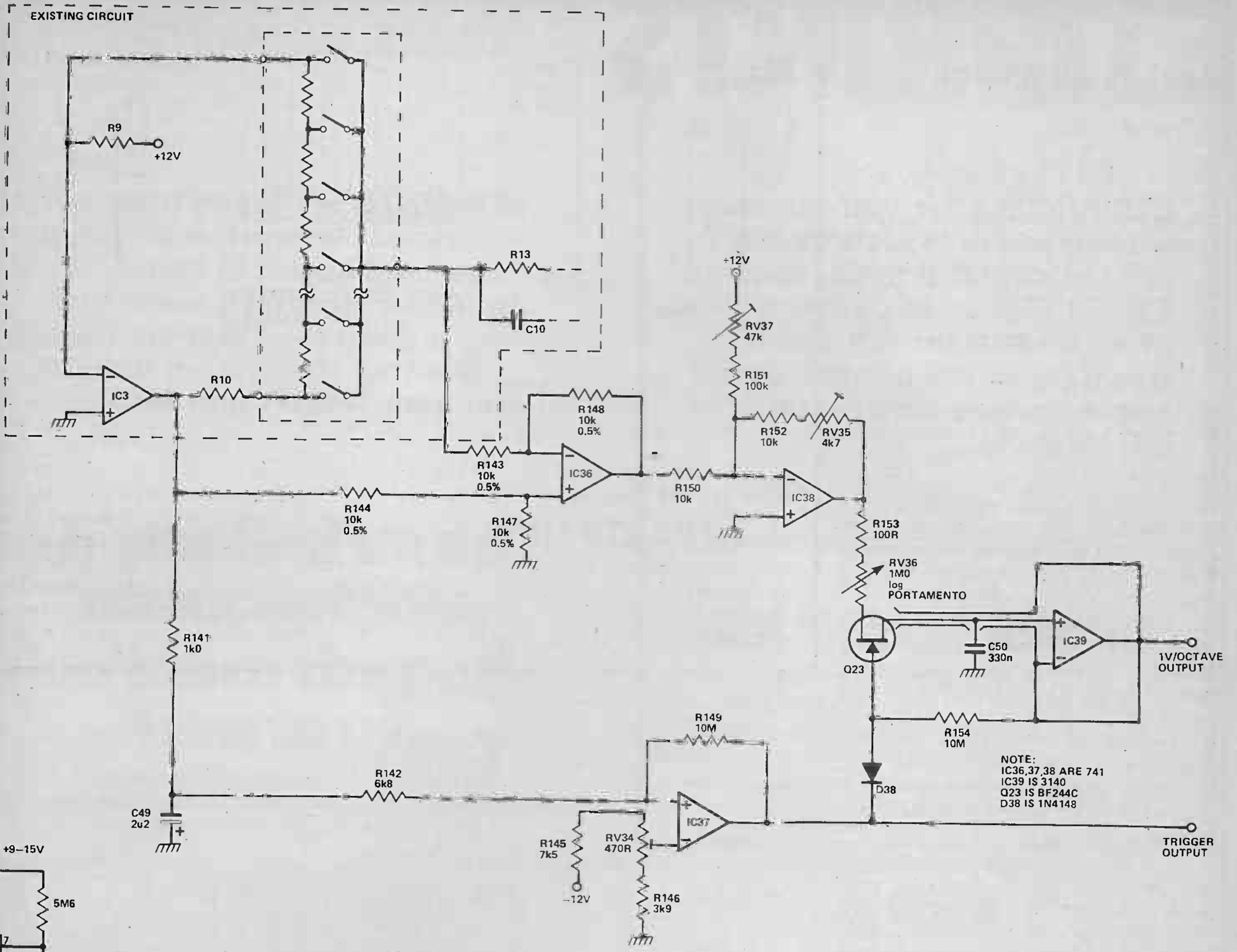


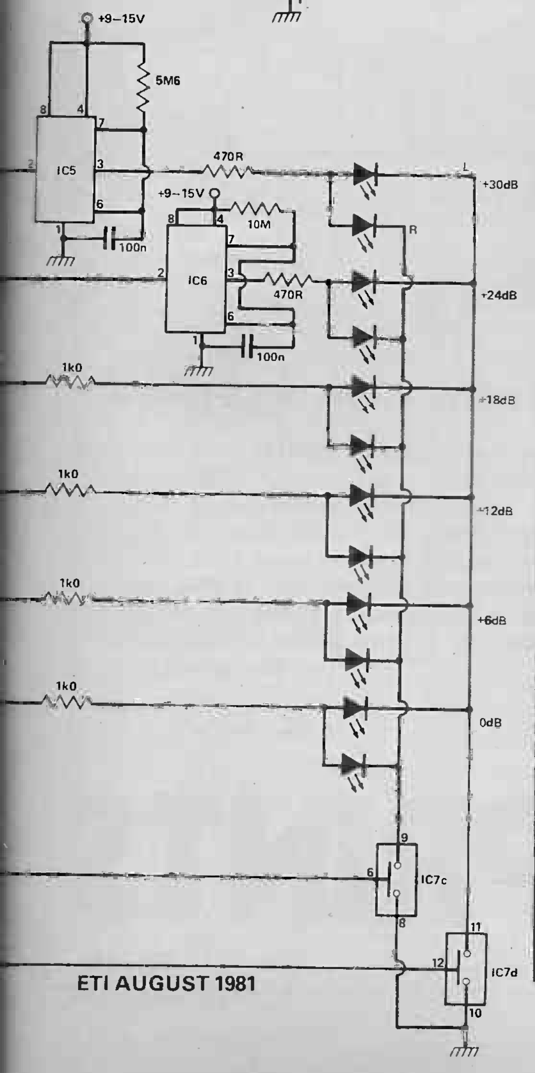
Cheap Tilt Switch

M. J. Woodbridge, St. Albans

The ETI Musical Box needs a mercury tilt switch, but there are cheaper ways of providing one. One way is to use a plastic biro refill end about 1.5-2 cm long. Thick pins are pushed into the soft plastic tube at one end as shown, at 180° to one another. The pins are then bent down at right angles to the tube and soldered to a small piece of Veroboard. The end of the tube nearest the pins is blocked up using glue, or by melting the plastic with a hot object. A small ball bearing is inserted, which will short the pins when it touches them. Blocking up the other end of the tube leaves you with a passable tilt switch.

A second way of making a tilt switch is to have a magnet moving up and down in a plastic channel near a reed switch.





Duophonic Synthesiser Keyboard

P.R. Williams, Stevenage

Most synthesisers, including the otherwise excellent ETI Transcendent 2000, are strictly monophonic; only one note can be keyed at a time. True polyphonic synthesisers are, however, complex and expensive. The circuit described here is a very simple modification which can be made to the Transcendent 2000 to make it a duophonic instrument; that is, any two keys can be pressed simultaneously to produce two notes.

The circuit relies on the fact that the keyboard resistor chain is fed by a constant current source, IC3. When more than one key is pressed, one or more resistors in the chain are short-circuited, resulting in the output of IC3 becoming more positive to keep the current constant. The output voltage of the normal keyboard circuit is then equal to that corresponding to the highest note pressed. The change in voltage at IC3's output is thus proportional to the number of keys between the highest and lowest pressed.

Thus, to obtain the voltage corresponding to the lowest key pressed, the change in voltage at IC3's output must be subtracted from the normal keyboard voltage. This is done at IC36. IC38 provides a scaling factor to achieve the common 1 V/octave output, which is adjustable by RV35. An offset is also introduced at this point to put the voltage in a useful range. RV37 controls this.

A trigger signal derived from the change in IC3's output is produced by the level detector, IC37. R141 and C49 de-bounce the contacts, while R142 and R149 provide some hysteresis for additional triggering reliability. RV34 is set so that IC37 will reliably detect when two or more keys are simultaneously pressed. Q23 and IC39 form a sample and hold circuit, which has been duplicated from the Transcendent 2000 design. RV36 could be ganged with the existing portamento control.

The output can then be used to control either an external VCO or another synthesiser.

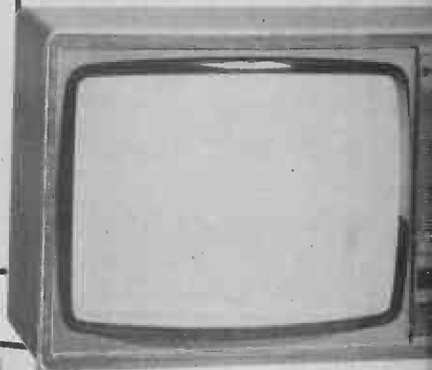
Although this circuit was primarily designed as a modification to the Transcendent 2000, it could be easily adapted for use with any synthesiser that uses a constant current keyboard resistor chain.

Cheaper Video

Sony are now aiming to corner the market in Video machines in every way, by launching their new C5 recorder. It follows in the footsteps of their revolutionary C7 machine which was launched last year (News Digest, May 80). The C5 is based on the C7 chassis but does not claim to be revolutionary. As Sony say: if the C7 is everything you could *want* in a recorder, the C5 is everything you *need*. It has all the functions of the normal second generation video recorder, including picture search, which is in black and white. Sony have reiterated their intention not to enter the consumer video disc market, partly because they believe the players themselves will be comparable in price to a cheap tape machine. This is where the C5 will fit into the video picture because its expected retail price will be around £450, and because it uses tape it has the facility to record — something which the video disc does not have. Sony also prefer to back the forthcoming Compact Disc (see news item entitled 'Mini Discs'), because they believe the consumer will prefer to buy this to update his audio equipment when it becomes available.

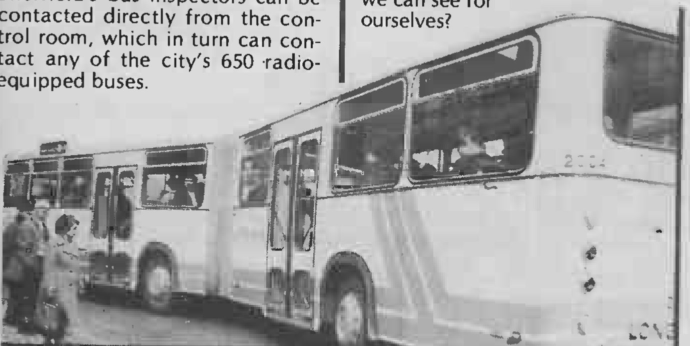
Text In Colour

Mitsubishi Electric have commenced delivery of their new 26" teletext colour television. The CT2606TX is based on the standard Mitsubishi 26" set and features soft touch controls, full function remote control, touch-button channel selection and 2 W audio output with tone control. There are also sockets for audio, earphone and video. It has a recommended retail price of £699 including VAT.



Buses On Call

Pye Telecommunications have been awarded a £34,000 contract to supply the South Yorkshire Passenger Transport Executive (SYPTE) with hand-held P5001 radio telephones and associated equipment. These pocketfones will complement the Executive's existing radio communications network. Sheffield has been called 'Bus City' as it is the County Council's policy to provide the cheapest and most efficient bus service possible and SYPTE's bus fares are, in fact, a third of those in other areas. The latest additions to their fleet are the 60 ft long articulated 'Cityliners' which hold 120 passengers and each have a running cost of £30,000 per year. The pocketfones will help to maximise efficiency by putting vehicles into service more swiftly. Now any of Sheffield's bus inspectors can be contacted directly from the control room, which in turn can contact any of the city's 650 radio-equipped buses.



Laser Loot

If you've been lucky enough to get your mits on a new £50 note, you might have noticed that the security thread embedded in the note is wider than in any other denomination of Bank of England notes and that it has one edge contoured in a regular pattern. This thread is produced on novel equipment developed and built for the Bank at the Culham Laboratory of the UK Atomic Energy Authority. The precise details of the equipment are obviously confidential, but the pattern on the tape is cut accurately at high speed by a focused laser beam steered through a microprocessor-controlled optical system programmed with a special code. The laser is manufactured in the UK by Electrox Power Beam Products Ltd in Bedfordshire. Would anyone out there like to donate a £50 note to our editorial department so we can see for ourselves?

Nickless Strips!

OK Machine & Tool Company have a new manual wire stripper suitable for 14-22 AWG (0.6 — 1.4 mm) solid or stranded wire. Virtually all current insulators including kynar, vinyl, rubber and neoprene can be cut. The tool is heavy duty and costs £5.95 (excluding VAT and postage) from Dutton Lane, Eastleigh, Hants SO5 4AA.

Street Music

Following the huge success of their Stowaway portable stereo cassette player, Sony have introduced the Walkman 2 which is said to have a better sound from a smaller, lighter and less expensive unit. The Walkman 2 is only slightly larger than the cassette it plays and has all the features of the Stowaway, but with the added bonus of playing metal cassettes and having a muting button on the headphones. The Walkman's controls are more conveniently placed than those of its predecessor, being at the front of the unit. It is supplied with MDR 4L1 headphones, a demo cassette, a battery case, which, using 2 HP2 batteries will give 69 hours of playback time, and a carrying holster. The retail price of the Walkman 2 is expected to be around £90.

HOME SECURITY SYSTEM

Have you seen the price of meat these days? It's much too expensive to keep a watchdog. Our ever-vigilant electronic version will protect your valuables without creating ruinous butcher's bills. Design by Ray Marston. Development by Steve Ramsahadeo.

The ETI Watchdog acts as the heart of a top-quality home security system. When it is coupled to suitable input sensors (window and door-mounted security switches, pressure mats, thermostats, panic buttons, and so on), and to an output sound generator (alarm bell or siren), it gives excellent protection against burglars and front-door thugs, as well as acting as an automatic fire alarm unit.

The Watchdog is powered by a PP9-style Ni-Cd battery, which is intended to be permanently trickle-charged by a simple mains-powered charger unit. The Watchdog will operate for about two weeks with mains power removed. Particular care has been taken in the design of the entire home-security project to ensure that it has excellent reliability, with a high degree of immunity against false alarms induced by lightning strikes and radio interference. The system incorporates features such as timer-controlled auto-turn-off of the main (external) alarm, a built-in audio-visual alarm recorder, and automatic exit and entry delays on the main burglar alarm system.

Operating Modes

The Watchdog has three basic operating modes selected by a Yale-type keyswitch, these being 'standby', 'reset', and 'on'. When the unit is in the standby mode the basic burglar alarm circuitry is disabled but the fire alarm and panic facilities are fully armed: if a fire is detected by one or more of the thermostats, or if one or more of the panic buttons are momentarily closed, the external alarm bell or siren will be immediately activated by the Watchdog's built-in timer-controlled relay. Simultaneously, the internal audio-visual alarm recorder will latch on, giving a permanent indication that an alarm activation has occurred: after five minutes or so the external alarm will automatically turn off, but the internal alarm recorder will remain latched on until the key-switch is moved to the reset position.

When the Watchdog is set to the on mode the fire and panic alarm systems are immediately fully armed, as described above, but the burglar alarm system is only semi-armed for the first 50 s. This delay gives the owner time to check (from a built-in LED) that none of the security sensor switches are active and to then pass through armed doors and so on, without sounding the main alarm. At the end of this 50 s delay the burglar alarm system becomes fully armed, and the alarm then sounds and self-latches (for a timer-controlled period) if any sensors are subsequently activated. The burglar alarm system can be temporarily disabled at any time for another 50 s period by briefly operating a remote re-entry switch, which can take the form of either a key-switch or a concealed push-button switch, thus enabling authorised persons to re-enter the building without sounding the main alarm.

Sensors And Alarms

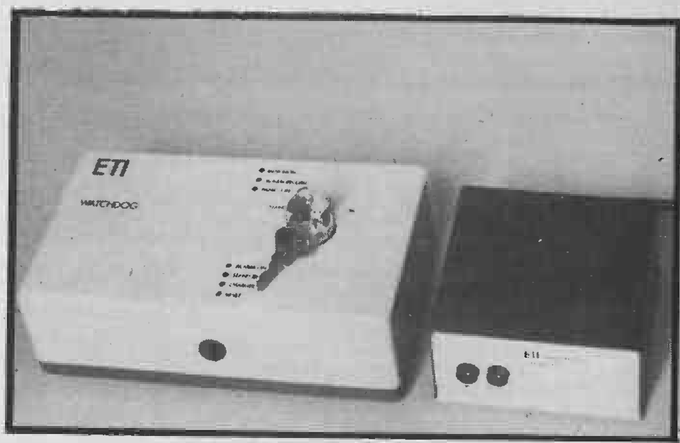
The Watchdog can be used with a variety of types of anti-burglary and fire-detecting input sensors, which can be coupled into the system via terminal strips that are built into the unit. The anti-burglary sensors can take the form of normally-open parallel-connected devices such as pressure mats, and normally-closed series-connected devices such as microswitches and magnetically-activated reed relays mounted on doors and windows. Fire protection can be obtained by wiring normally-open thermostats in parallel, and thug protection can be obtained by wiring normally-open panic (push-button) switches in parallel.

The external alarm can be any electro-mechanical or electronic siren or bell that is provided with its own power supply and draws an operating current of less than 5 A; the external alarm is activated by the contacts of relay RLA, which is built into the Watchdog unit.

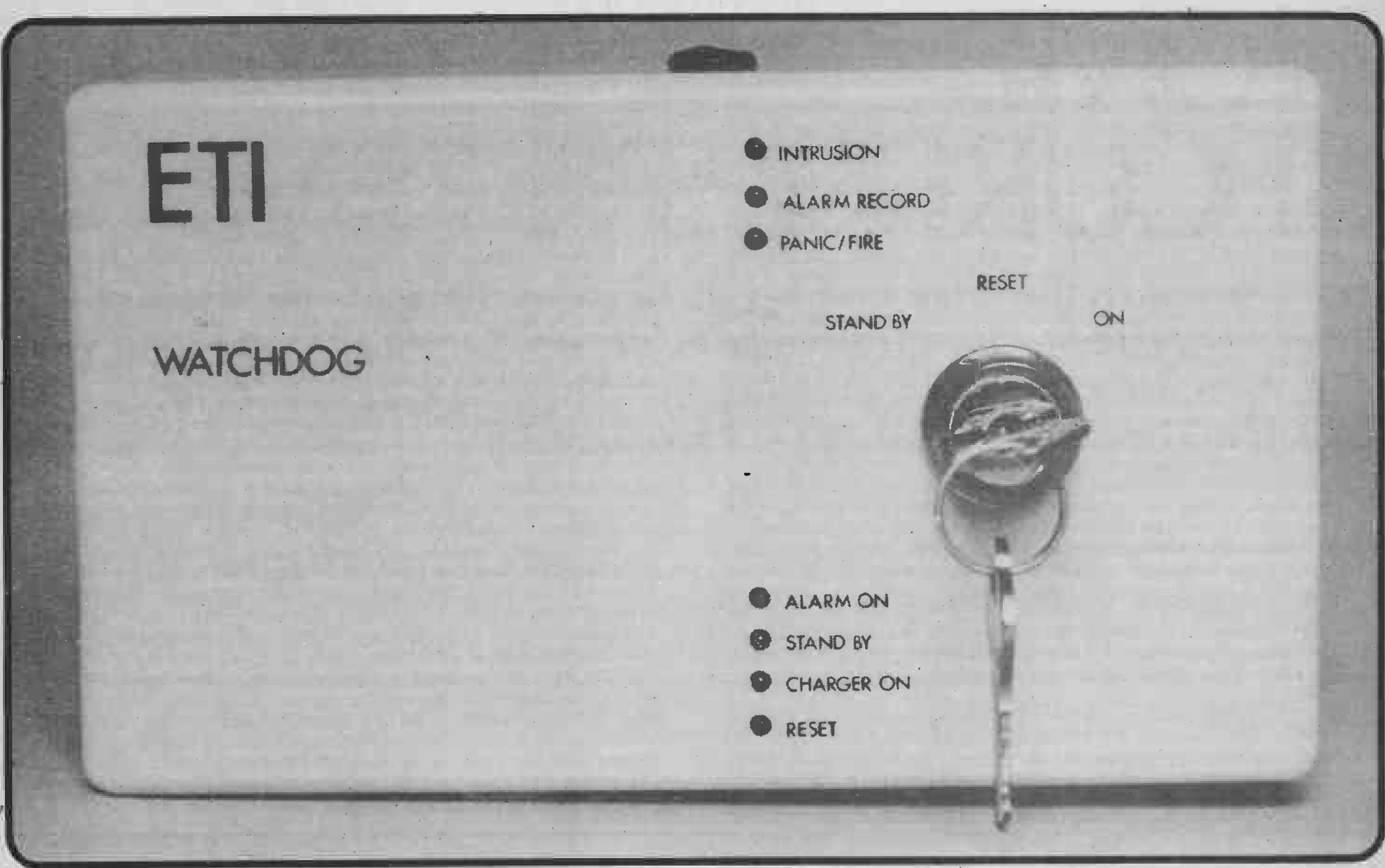
Construction

The Watchdog system consists of two independent units. The main unit is intended to be wall-mounted and contains the main electronics of the system, together with the key-switch, LEDs, alarm-driving relay and the Ni-Cd battery. The second unit is a simple mains-powered Ni-Cd trickle-charger and can be mounted on a skirting board, close to a power socket, and coupled to the main unit by a twin lead.

Construction of the main unit should present very few problems, provided that the specified components are used and that the usual care is taken to observe semiconductor polarities and so on.



The Watchdog main unit and the trickle-charge unit.



The ETI Watchdog Security System is keyswitch operated and has plenty of LEDs to let you know exactly what's going on.

Start the construction by assembling all indicated components on the PCB, as shown by the overlay, and then test-fit the board into the specified case, together with the PP9-style Ni-Cd, taking care to ensure that clearance is available to give access to the wall-mounting knock-out holes moulded into each end of the case bottom.

Next, modify the top half of the case to accept the key-switch and the seven LEDs, etc, and the lower half of the case to accept the charger input socket and the PB-2720 transducer, then complete the circuit interwiring, noting that current-limiting resistors R11, R12 and R25 are wired directly in series with LEDs 3, 4 and 7 respectively.

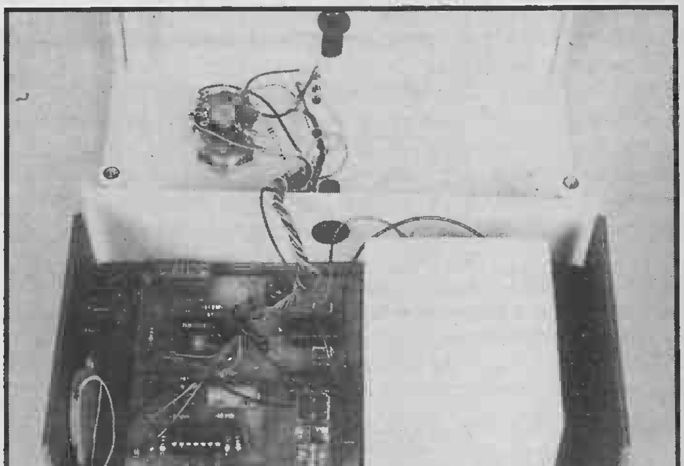
At this stage you can give the unit a brief functional check. Switch to the standby mode and check that the alarm can be activated by briefly shorting one of the N.O. thermostat or panic-button inputs. Check that the main (external) alarm turns off after a few minutes, but that the alarm recorder remains permanently latched on and produces a pulsed-tone audio signal. Check that the recorder can be turned off by switching to reset. Similarly, check that the burglar alarm circuitry functions as already described by switching to the on mode.

The charger circuit is so simple that its construction should present absolutely no problems at all. Note, however, that Q2 must be mounted on a small heatsink. When construction is complete switch the unit on, connect a DC current meter directly across the unit's output terminals, and check that a current indication of roughly 70 mA is given. Finally, connect the output of the charger to the charger socket on the main Watchdog unit, switch on, and check that LED7 (on the main unit) illuminates.

Installation And Use

The installation of a home security system is a fairly major undertaking, with many fine points to consider and individual decisions to be made regarding the degree of protection that is required and the types of sensors that are to be used. In the next few paragraphs we've outlined some very basic principles of installation; however, it's up to the individual reader to work out the details of his own sensor and alarm generator networks, and then couple these networks to the main Watchdog unit.

A complete home protection system should contain three distinct types of sensor networks, these being designed to give (1) fire protection, (2) anti-thug or panic protection, and (3) anti-burglary protection. Fire protection can be obtained by mounting normally-open thermostats close to the ceiling in each room, then wiring all the thermostats in parallel and



The main unit contains the alarm sensing circuitry and the PP9 Ni-Cd battery. Connections to the sensors depend on the application and none are shown on this prototype — but all the necessary wiring can enter the box through the two grommets in the lid.

HOW IT WORKS

The circuitry of the main unit can be broken down into two distinct sections, with the basic alarm circuitry plus alarm recorder to the right of key switch SW1a, and the burglary-detection circuitry to the left. The circuitry to the right of SW1a is permanently enabled and can be activated at any time by the panic and fire inputs: the burglary-detection circuitry is active only when the system is turned fully on by SW1.

The operation of the basic alarm circuitry and the alarm recorder is fairly simple. IC2a-IC2b is a long-period monostable (several minutes) and can be triggered by applying a high (logic 1) signal to R13 via the D2-D3 OR gate; the mono can thus be triggered by closing any of the fire-detecting thermostats or the panic buttons, or by a high output from IC1d. When the mono is triggered, the output of IC2b goes high for the duration of the monostable period and thus drives RLA (and the external alarm) on via Q1 for a preset period. Simultaneously, the output of IC2a goes low and causes special-purpose bistable IC2c-IC2d to self-latch into a state in which the output of IC2d goes low, driving LED6 (the visual alarm recorder) on and activating the IC3-IC4 audible alarm recorder circuitry. When the monostable turns off at the end of its timed period the external alarm also turns off, but the audio-visual alarm recorder remains active, giving a permanent indication that an alarm action has occurred. The monostable and the recorder can both be reset by briefly moving SW1 to the reset position.

The audible alarm recorder circuitry is quite simple. IC3a-IC3b form a low-frequency astable, which is gated on by a low input signal. IC3c-IC3d form a high-frequency astable (a couple of kilohertz) which is gated by the output of IC3b. The output of the high-frequency astable is fed to acoustic transducer TX1 via bridge-configured driver IC4. Thus, when the circuitry is activated by a low input signal, an audible pulsed-tone signal is generated by TX1.

The burglary-detection circuitry is designed around IC1 and is active only when SW1 is switched to the on position. Here, the N.O. and N.C. burglar-detecting security switches are wired in such a way that a high voltage is normally applied to R3, but this voltage goes low if any of the N.C. switches are opened or the N.O. switches are closed. The R3 voltage is inverted by both IC1a and IC1b, so that LED2 turns on and a high voltage is fed to one input of IC1c if an intrusion occurs; C1 and R5-C2 filter the signals from R3, to eliminate the effects of lightning-induced signals and transients. The other input of IC1c is controlled by the C3-R7 time-controlled network, which causes IC1c to be effectively disabled for a minute or so after SW1 is first moved to the on position or after the optional re-entry switch is momentarily closed. The output of IC1c is inverted by IC1d and fed to the D2 input of the D2-D3 OR gate, where it can control the action of the main alarm circuitry.

Thus, when the burglar alarm circuit is first switched on by SW1 the IC1a-IC1b section is fully enabled, so that LED2 will turn on if any of the security switches are incorrectly set, but IC1c-IC1d are disabled by the C3-R7 network, so that the main alarm will not activate under this condition. After a minute or so, however, the IC1c-IC1d section becomes fully enabled, so the main alarm will sound instantly if any subsequent intrusion is detected.

Note that the main Watchdog unit is powered by a single PP9-style Ni-Cd battery, which is intended to be permanently trickle-charged by an external mains-powered charger circuit. Also note that the circuit is provided with a total of seven LEDs, which give visual indications of the existing operating mode, the presence of sensor faults/actions, the presence of the charger current, and the record of an alarm action.

The charger circuit is very simple and is intended to apply a permanent trickle-charge current of roughly 70 mA to the Ni-Cd battery of the main Watchdog unit. Here, the mains voltage is stepped down, full wave rectified and smoothed by T1-D9-D10-C8, to provide roughly 13 V DC across C8. This voltage is used to power the constant-current generator that is designed around ZD1-R26-Q2-R27. ZD1 sets a standing voltage across emitter resistor R27, which thus determines the emitter current of Q2: since the emitter and collector currents of an active transistor are virtually identical, the collector of Q2 effectively acts as a constant-current source and is used to feed a trickle-charge current to the Watchdog Ni-Cd via D8. LED7 (in the main unit) illuminates when the charger is active.

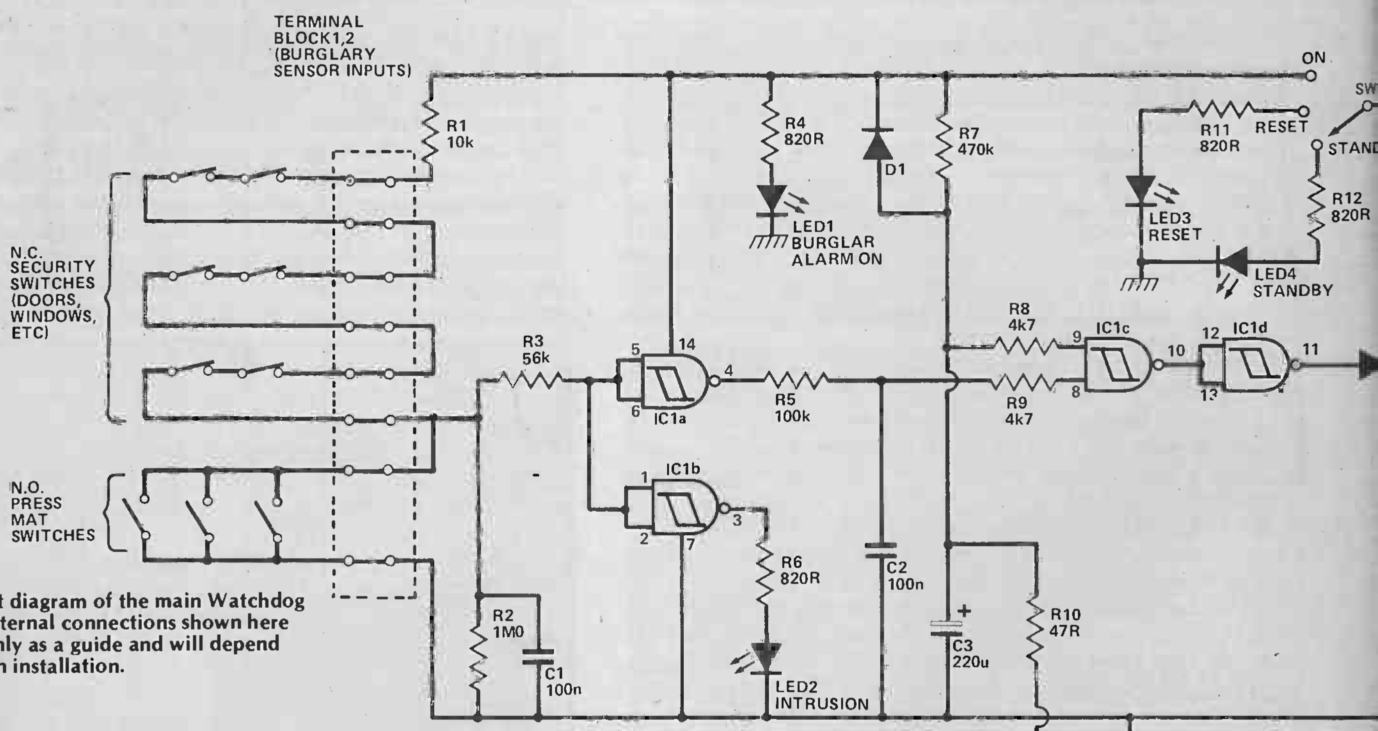


Fig.1 Circuit diagram of the main Watchdog unit. The external connections shown here are given only as a guide and will depend on your own installation.

PB1 CONCEALED RE–ENTRY SWITCH

coupling them to the appropriate input of the Watchdog unit.

Anti-thug or panic protection systems enable the owner to activate the alarm system manually if he is attacked by thugs inside the house or suspects the presence of intruders in the house. Protection can be obtained by mounting normally-open push-button switches in readily accessible points near the front and rear doors and in the lounge and bedrooms and then wiring all of the switches in parallel and coupling them to the Watchdog unit.

BUYLINES

The piezo-electric transducer can be obtained from Ambit International. Electrovalue are stockists for the PCB terminal blocks. The 6 V relay, DC socket and key-switch are available from Watford Electronics.

All the other semiconductor devices used in this project should be available from advertisers in this issue eg Technomatic, Greenweld etc. The Samos 002 case can be obtained from West Hyde Developments.

Special-purpose security sensors, panic buttons and alarms are available from specialist security companies such as Strathand Security, 44 St Andrew's Square, Glasgow G1 5PL.

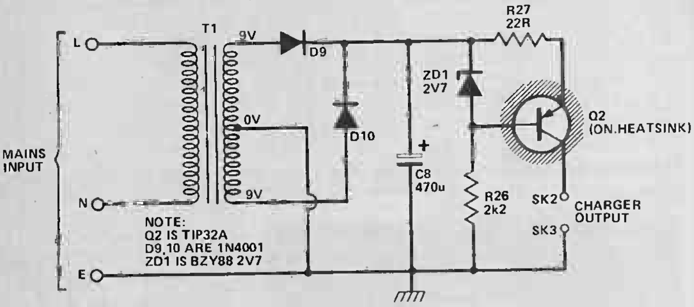


Fig.2 (Above) Circuit diagram of the trickle-charger.

Burglary protection can be obtained either by using a perimeter defence' system that detects the intruder as soon as he enters the building through a protected door or window, or by using a 'spot defence' system that detects him only after entry has been made (as he opens internal doors or treads on concealed pressure mats), or by using a combination of the two systems.

Selected doors and windows are easily protected with a reed-relay/magnet combination: the magnet is installed in the door or the opening window, opposite a reed-relay installed in the frame, so that the relay is normally closed but opens when the door/window is opened. All the relays are then wired in series and connected to the appropriate input of the Watchdog unit, so that the alarm activates when any of the protected doors/windows are opened.

Pressure mats come in a variety of sizes and are easily hidden under rugs and carpets. They are usually normally-open devices, so any number can be wired in parallel and fed to the appropriate input of Watchdog.

Modus Operandi

When planning the installation, the house-owner must try to think like a burglar. Normally, the burglar enters a house from an easy access point that is obscured from the view of the neighbours, ie a back door or window. Often, he breaks in using tools 'borrowed' from the owner's shed or garage, so these two places should be included in the owner's defence system. Invariably, the burglar's first action on entering the property is to secure a rapid escape route, ie to open the back door. He then starts hunting for stealable goodies.

Once you've designed and installed your own sensor systems you can couple them up to the main Watchdog unit, which should be mounted in a central part of the house, such as on a landing or at the foot of the stairs, so that it can be operated with maximum convenience. The external alarm generator,