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Space Shuttle

first step to the stars?

Wheel Of Fortune

MK 14 Review

STAC Timer

Amplifiers

FM Tuner

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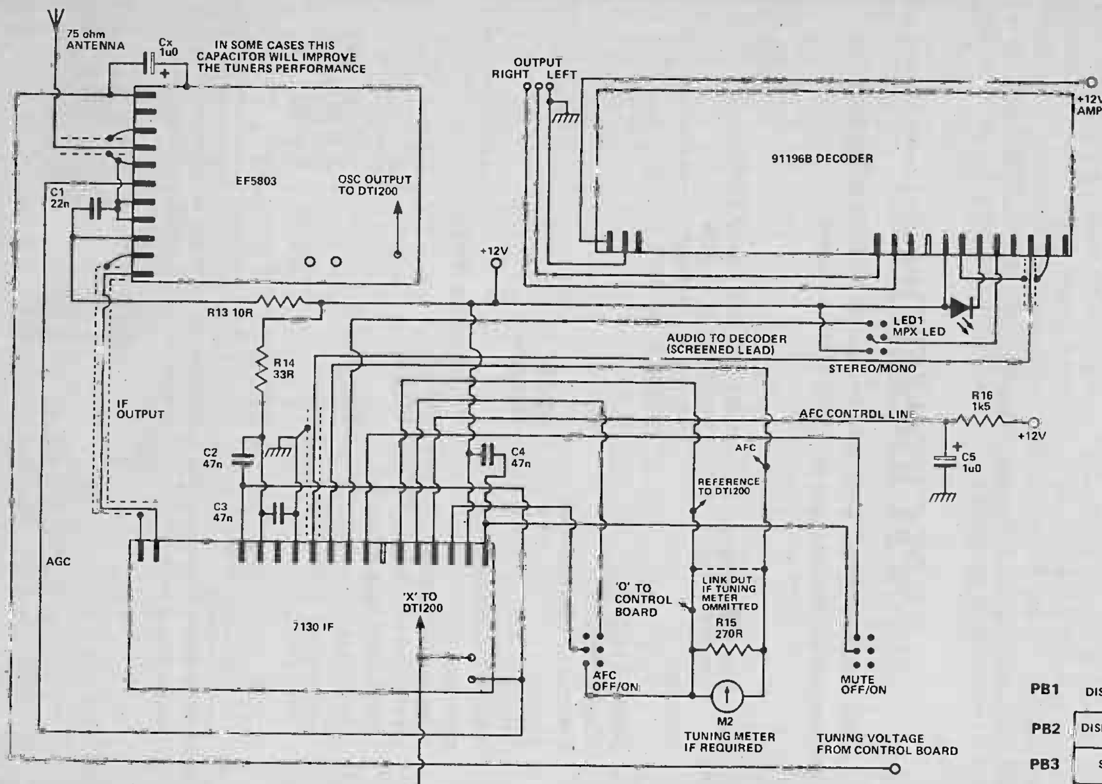


Fig. 1. Interconnection details of the three major modules that form the International Mk 3. LED 1 is included in the DTI-200 but should be a TIL 209 if the analogue frequency meter is used.

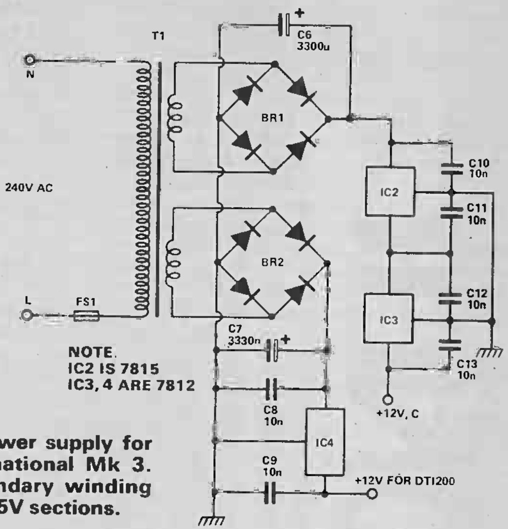


Fig. 3. Power supply for the International Mk 3. T1's secondary winding are both 15V sections.

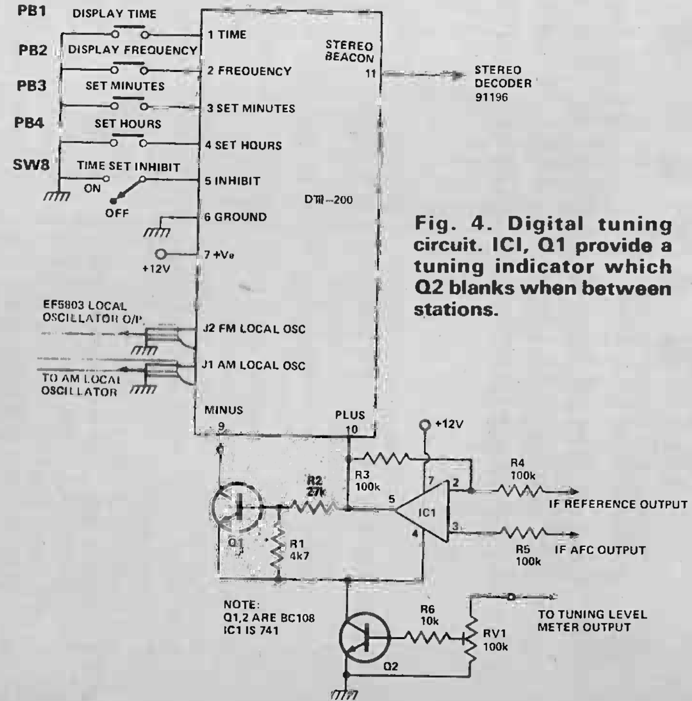


Fig. 4. Digital tuning circuit. ICI, Q1 provide a tuning indicator which Q2 blanks when between stations.

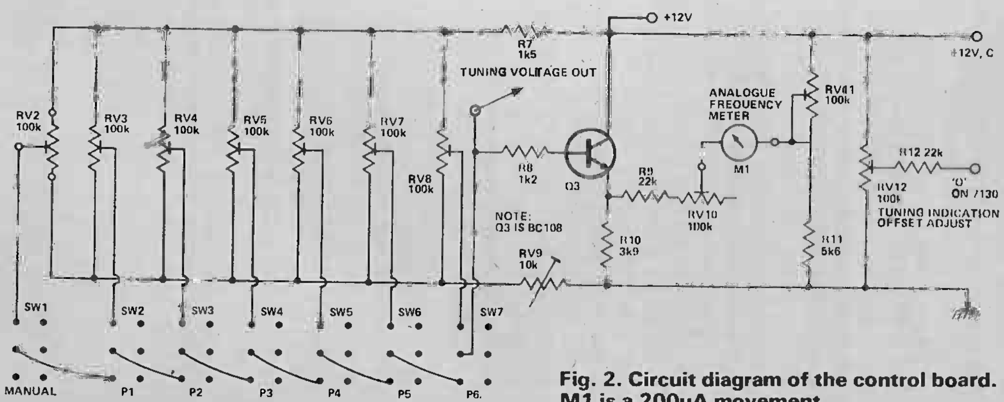


Fig. 2. Circuit diagram of the control board. M1 is a 200uA movement.

HOW IT WORKS

To start at the beingging, all radio receivers have an antenna. This should ideally deliver about 1mV of the desired stereo FM station in most tuner applications, although this system is designed to operate with rather less. The tunerhead system comprises two similar dual gate MOSFET stages, using low noise types of VHF devices from either the BF900 or MEM680 series. Each stage provides 22dB of gain, which can be readily controlled along the gate 2 line with AGC from the main IF amplifier system. The interstage coupling is very loose — imparting a narrow peak to the coupling passband for best rejection of the spurious signals encountered in FM band two tunerheads.

By the time the amplified RF signal reaches the mixer, it is processed through five tuned circuits at the RF frequency — and these must be made to match each other in a process known as tracking. It is not much use having 3 circuits at 89.4 and the other two at 89.1 MHz since signal would only be lost in the detuning effects, but the susceptibility to spurious signals will increase as the overall bandpass response begins to acquire some humps in odd places:

To assure good tracking of the RF — and also the oscillator, at this frequency, the EF5803 employs totally symmetrical layout of all frequency determining components so that all circuit strays will be balanced in each individually screened compartment.

At the input to the mixer stage, the signal is fed into the signal gate of the MOSFET — and the local oscillator is fed into the control gate, producing a multiplicative mixing effect for good dynamic range and isolation of the oscillator frequency from the effects of strong signals that tend to 'pull' the oscillator in some bipolar mixer designs. The products of mixing are signal frequency plus oscillator, and signal frequency minus oscillator. The latter is the desired IF signal, and this is selected out of the drain circuit at 10.7MHz in a bandpass pair. The drain also provides a wideband derived AGC signal for the second RF stage to prevent exceptionally powerful RF signals exceeding a level where the mixer tuned circuit has volts of RF signal — which may then be rectified in the varicaps and superimposed on the tuning voltage, creating some very undesirable cross modulation effects in the whole front end. This AGC circuit only operates at inputs of more than about 5mV — when the AGC that is derived after the IF selectivity has usually already taken the line of zero volts. It is therefore aimed at signals just outside the IF bandpass, but still sufficiently close to the RF bandpass to cause problems.

The IF sections comprise a MOSFET preamp, with AGC from the IF AGC line, followed by the first of two linear phase filters. Correct termination of the filter leads to a very smooth bandpass characteristic that permits low distortion stereo to pass through unhindered or deformed in any way. The full multiplex composite spectrum is an AF signal bandwidth of 55kHz — and in the FM system, which is too complex to explain here — a transmission bandwidth of around 200kHz is considered necessary. Ceramic IF filters are a lot better than they used to be — but the coil/capacitor arrangements of linear phase filters have superior stability, and much better skirt and spurious responses in strong signal environments such as the EF5803 will provide. A second MOSFET/Filter stage precedes the main IF element, a rather comprehensive device that performs IF amplification functions, including limiting, detection, signal level meter drive, AGC drive, centre tuning drive, noise muting and deviation, muting systems. The IC which performs all these functions is the CA 3189E.

The IF of an FM tuner is probably one of the key areas of the whole tuner specification. It determines just about every 'audible' parameter and so must be given close attention for its effect on sound quality. Of the key subject areas of sound quality as applied to FM tuners, the signal to noise ratio is one of the most important — and so a wide dynamic range is necessary. This is ultimately determined by the choice of IF IC, and to a lesser extent the stereo decoder — at present, the specification of the CA3189E is capable of coping with the broadcaster's specifications. Distortion of the device is largely up to the external circuitry that is used in the detector circuit. Here the transfer characteristic of the discriminator is up to the board layout, and the quadrature components. Ad double tuned circuit, with critical coupling, is used to provide the detector with a THD of less than 0.05% when everything is correctly adjusted. The detector cannot be set up using the transfer curve method very satisfactorily, an audio spectrum analyser is best, with distortion factor meter next best — although a lot slower.

The IF system also provides an accurate muting method, that cuts out interstation noises when tuning the band. It operates in two ways – firstly by noting the signal to noise ratio of the incoming signal at the detector stage, and cutting in when the S/N is sufficiently degraded. However, although this method has been considered satisfactory for a long time past – there are certain shortcomings when tuning through a strong signal, where the edge of the discriminator curve can provide two additional detection transfer slopes at either edge of the desired passband. This leads to some loud rasping as only half the signal is being processed in this way.

So the secondary muting technique is employed, whereby the signal is muted after it passes sufficiently off-tune to begin to become distorted. This method is known as deviation muting – and the control voltage is readily obtainable from the AFC voltage – which is in fact the DC level present at the detector, though decoupled from audio. If this voltage exceeds a predetermined level — the mute operates. This feature also assists greatly in fine tuning the unit — since it is not possible to listen to a detuned and thereby distorted signal, when the muting circuit is switched on.

The muting voltage may also be taken to the stereo decoder to prevent chattering of the stereo switching circuits as the unit is tuned through the band.

There are also two signal level voltages available – one for driving the tuning meter, and one for driving the AGC. The two are related, so that the AGC threshold may be preset to operate at any signal level – thus avoiding the tendency of the AGC to operate too suddenly in conjunction with high gain, high signal level handling tunerheads such as the EF5803. In this circuit, AGC begins to operate at about 1mV of antenna signal.

In the stereo decoder, the signal first passes through to the 'birdy filter', which restricts the audio bandwidth to below 55kHz – preventing an adjacent channel signal from beating with any of the decoder pilot tone frequencies and products creating the faint warbling that can appear on stations in crowded conditions. This filter is rather crucial, and an LC arrangement, in the form of the common delay line, is used for the most readily adjustable combinations of HF signal attenuation and AF signal attenuation. Many IF systems pour forth many millivolts of 10.7MHz and 21.4 MHz in the audio line – and the 'active' filter arrangement is not as effective in attentuating these frequencies and maintaining good phase response.

The decoder IC itself is the HA1196. Most people will know about the MC1310 – the original PLL stereo decoder IC – and much of the HA1196 is similar, except that the distortion is rather better, it possesses an adjustable separation facility, and best of all, it provides low distortion AF gain specifically derived to drive the pilot tone filter. Attenuation of 19 and 38kHz components of the AF voltage is essential to prevent HF intermodulation in the amplifier – and the BLR3017N unit also provides a steep cutoff after the audio bandwidth of 15kHz. The HA1196 drives the conventional LED beacon – and as mentioned already, has a stereo muting facility via an external control voltage – which may be supplied at high impedance.

The Control Sections

Apart from the signal processing, the control aspects of this tuner require explanation and comment. First and foremost the digital frequency readout unit.

The DFM unit is a ready made 'black box,' incorporating FM and AM frequency and 12 hour quartz clock functions. It is unique at present — but it should be pointed out that the DT1200 is primarily based on USA markets, and so the count resolution is alternate 100kHz channels in 88-108, and 10KHz channels in the medium and long wavebands. Purists will no doubt realise that there are stations in the UK broadcasting in between these frequencies — although the BBC sticks fairly well to the alternate 100kHz pattern. This design can be run with the tunerhead powered continuously — since the clock/frequency counter needs continuous power — and so achieve a stability otherwise unheard of. The varicap tunerhead also permits a selection of preset stations, through switched multiturn potentiometers.

For those of you not sufficiently enthusiastic about digital tuning, an option is described for an analogue frequency meter indicator — driven from the main tuning voltage line. The accuracy is not overwhelming — but the narrow spread of the UK FM band means that most listeners quickly appreciate the relative locations of their local transmissions. The meter is driven from an emitter follower circuit to isolate the actual tuning voltage from the dangers of picking up stray hash and noise along the meter lines. If driven directly from the tuning voltage, tapping the meter can reproduce a hollow mechanical clunk in the audio.

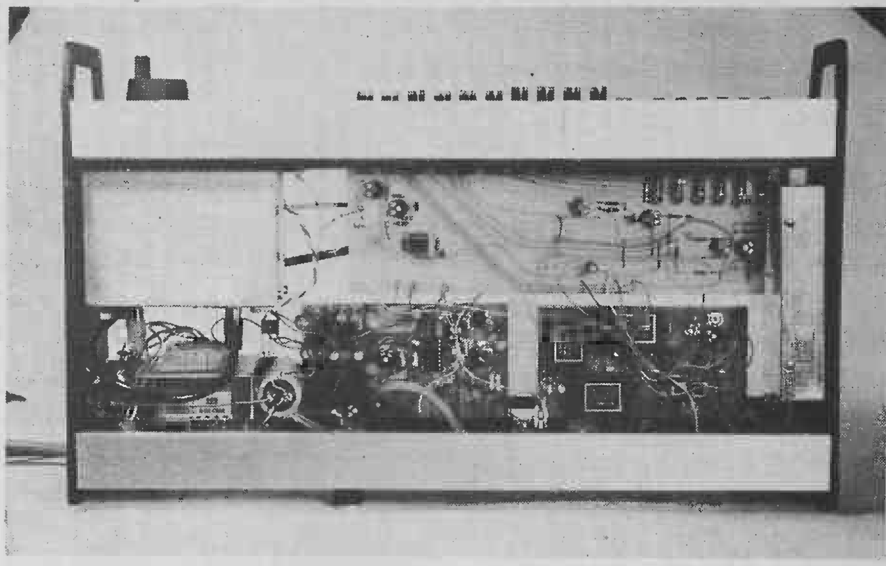
Finally, the PSU looks straightforward enough, but it must be carefully decoupled to prevent RF noise getting any further around the tuner than essential. Most voltage regulation sources are producers of wide band RF noise — and so careful filtering and decoupling is used as close to the source as possible. The supply for the audio monitor stages of the decoder board (2xLM380) is separately derived — and need not be stablised — to prevent modulation peaks detuning the whole thing.

The supply for the DT1200 requires careful filtering at the exit of the box — since the display of this unit is strobed at about 500Hz. The main tuning voltage to the EF5803 is decoupled at the entry of the shielded can, since this relatively high impedance line can be prone to picking up any radiated hash that is floating around.

considerations generally limit the levels used. This approach trades off a little ultimate distortion for a few dB signal to noise ratio. Subjectively, this is more than justified.

In fact, the decoder used here incorporates a 2W per channel monitor amplifier feature for persons requiring the unit to be self supporting as a very classy bedside radio — or as a means of monitoring programmes without upsetting the whole Hi-Fi operation. This is mentioned briefly here, and will not be covered in great detail in the text, but if it is to be incorporated, please follow the directions supplied with the module carefully.

Metering facilities are provided both in the DT1200 module — where FM detune is indicated by illumination of the + and — on the display — and separately with moving coil meters if desired. A signal level meter is considered to be a desirable feature in a unit of this sophistication (to make certain you are getting the most local transmission from the multiple relays of the BBC), and the centre zero tuning indication is essential for the very best fidelity in narrow IF systems. The pedantic may also like to run a pair of PPM/VU audio level meters driven from the decoder output — but that is something considered unnecessary here.



Picture showing the internal layout of the International Mk 3 FM Tuner. The modules can clearly be seen mounted in their edge connectors with the PSU bottom left. The RF shielded DTI is top left.

Construction

The modules are fitted (Fig. 1) in the order shown, and it is desirable to follow the earth path layout shown on the wiring connection diagram (Fig. 2) if the problems of HF and VHF earth loop instability are to be avoided. Such instability is the curse of RF, and the reason why otherwise competent engineers have been known to lock themselves away in the loo when asked to "just debug the VHF frontend" AF instability has the delightful quality that it can be heard, and so progressive fault tracing can be a relatively simple and speedy matter. With RF, the engineer's 'ear' is the spectrum analyser (just as the ear is a reasonably good audio spectrum analyser). Whilst the home constructor is usually blessed with an ear or two, the latter instrument is not as commonplace as it ought to be.

In other words, the unit may sound quite healthy on reasonable signals, but on weaker signals, the whole thing oscillates around an unforgiving earth loop and the signal sensitivity appears to be unreasonably impaired. A quick check for stability is to listen to interstation noise with the mute 'off.' The noise should be smooth and white, clean and bright etc. (Sung to the tune of Eidelveis) It should not be crackling and broken up, or buzzing with a low level hiss.

When the system is really well set up, generator EMFs of 0.63 uV can provide full limiting on mono. This is very close to the theoretical limit of the system, and whilst some of it may be due to leakage effects, it still illustrates that not only is the nature of the signal VHF —but you are dealing with amplification levels vastly in excess of anything likely in an audio environment.

Edge Connectors

The modules fit into 0.2in edge connectors for ease of assembly, and it is recommended that the edge connectors should be very carefully wired with the modules in situ outside the case, the whole lot being transferred to the inside when it has been ascertained that the system is 'go.' The Swiftcase lends itself very nicely to this approach, since it comes virtually completely apart into a stack of plates and screws. In fact, it is rather better to solder the units together to avoid the dangers of interconnection degradation, but many people still feel happier if the units can be dismantled easily, although this is really not necessary (hopefully). The PSU is simple enough, but remember that RF environments call for extra attention to potential RF noise sources such as the regulator device itself. The curse of tuners is frequently next door's fridge thermostat or the slightly noisy fluorescent tube fittings. An IEC type of mains filter is very useful here, and it also doubles for the mains input socket. One of the bolt-on extras envisaged for this unit is a noise blanker system to take out any residual click type interference that inevitably starts up during the quiet passages of Beethoven's 6th.

Interconnection of RF and IF signal paths should be made with RF coax. The antenna input should certainly use good 75/50 ohm coax — though the use of lesser types of screened cable is permissible for the IF connection — and of course the audio connections. Always use stranded cables for the rest of the wiring, since single solid cables will send you completely up the nearest wall if you ever have to manipulate the circuitry in the case. Units wired in this way will also not be eligible for the alignment service that is being offered to the constructor.

The connection of the frequency counter should also be made via coax of an RF nature, but since this is well buffered from the actual tunerhead oscillator, it may not be essential. The take off for the external connection of the AM local oscillator (when your MW/LW tuner is ready) should be made with the same coax.

PARTS LIST

REFERENCE SERIES MODULES

7130	IF Strip
91196(91196B)	Decoder
EF5803	Tuner Head
DTI200	Digital Tuning Indicator

RESISTORS

R1	4k7
R2	27k
R3, 4, 5	100k
R6	10k
R7	1k5
R8	1k2
R9, 12	22k
R10	3k9
R11	5k6
R13	10R
R14	33R
R15	270R

CAPACITORS

C1	22n polyester
C2, 3, 4	47n polyester
C5	1u0 electrolytic
C6, 7	3 300u 35 V electrolytic
C8-13	10n polyester

SEMI CONDUCTORS

IC1	741
IC2	7815
IC3, 4	7812
Q1, 2, 3	BC108

POTENTIOMETERS

RV1, 10, 11, 12	100k preset
RV2-8	100k diode law-type AB47

SWITCHES

SW1-7	Double Pole Charge Over
SW8	Single Pole on-off
PBI-4	Push to make, release to break

MISCELLANEOUS
West Hyde Swift Case, Meters (200uA), edge connectors, transformer (15-0, 0-15), screened lead etc.

BUYLINES

Ambit International of 2 Gresham Road, Brentwood, Essex will be supplying a complete kit of parts for this project.

The cost of the tuner with DTI-200 will be £139.00. Without the Digital Tuning option the kit will be £99.00.

Switch On And Test

Never complete a project of this complexity and simply press the mains switch. Always build up gradually, starting with the PSU on its own — ie disconnect the supply feeds to the rest of the works — and check the voltage. Leave the PSU running for an hour in this fashion, since experience has shown that many PSU reservoir capacitors are at their most fragile during this period. The slim chance of 500-1 is sufficient odds to let the PSU have a good soak before endangering the rest of the works. Next, hook up the power to the frequency counter, and check out the time function, following the setting details in Fig. 3. It is quartz referred, so it should be very accurate indeed. If nothing happens, check your switching wiring very carefully and try again.

Now monitor the supply current to the rest of the circuit and connect. Over 150mA means you have a problem, so then you must methodically disconnect each module's supply in turn, until offending connection is located. The usual trouble is 'frilly' wire terminations, so do not immediately despair that all is blown up and disasterously defunct. Also check that any decoupling electrolytics on the connectors and harnesses are correctly rated and polarized.

It is hoped that the process of test will quickly get you to a state where noises are apparent — and don't forget to set the audio output level pots on the decoder so that they are about halfway. And remember to leave the mute and AFC buttons in the "off" state until you have started to get recognisable sounds through the system.

The function of the tunerhead can be verified to a certain extent by switching the display to FM frequency readout. You will be able to see the frequency — to the nearest 200kHz — as the tuning is varied. Unless you live in a really bad location, a degree of sound from a BBC transmission will be readily obtained with a simple piece of wire poked into the antenna socket. If you get the right sort of no-signal 'hiss' — but no stations, and the DEM is indicating tuning is going on, check the RF and IF signal leads for shorts and problems at the connectors.

The muting used is a combination of deviation and noise muting — which means that unless a station is reasonably accurately tuned to start with, the mute cannot open to pass the signal when switched on. Furthermore, the mute is tied in with the operation of the stereo switching of the decoder, so that stereo is automatically inhibited as the signal goes off tune, preventing the jittering crashes that are sometimes found in such systems. The mute will not always completely kill all background noise, since it is set to lift on the slightest vestige of a signal. Usually 1uV or so.

ACC Circuitry

The AGC threshold point and operating level are factory set in the IF module, but those of you who know what you are doing may wish to tweak these controls to optimize for a particluar location condition. The unit cannot be seriously detuned with these controls — though most of the others should be left well alone. If you feel it is essential to have a tweak of the coils to get the thing going, then do not under any circumstances do so. The problem will only be worsened by a quick tweak of a trimmer, and must be sought elsewhere.

The AFC function of the tuner is readily confirmed. Slightly detune the transmission, and switch in the AFC. The signal will be pulled closer to the centre of the passband. Some listeners believe that operation of the AFC is detrimental to listening quality. In this tuner it is not so, since AFC controls all the tuned circuits of the VHF tunerhead, and not merely the oscillator. So the tracking of the tuned circuits is not in any way impaired when the AFC is operational.

As mentioned in the 'How it Works' section, the AFC is also programmable in its effect, so you may increase it up to the point at which it becomes overpowerful with respect to ease of tuning.

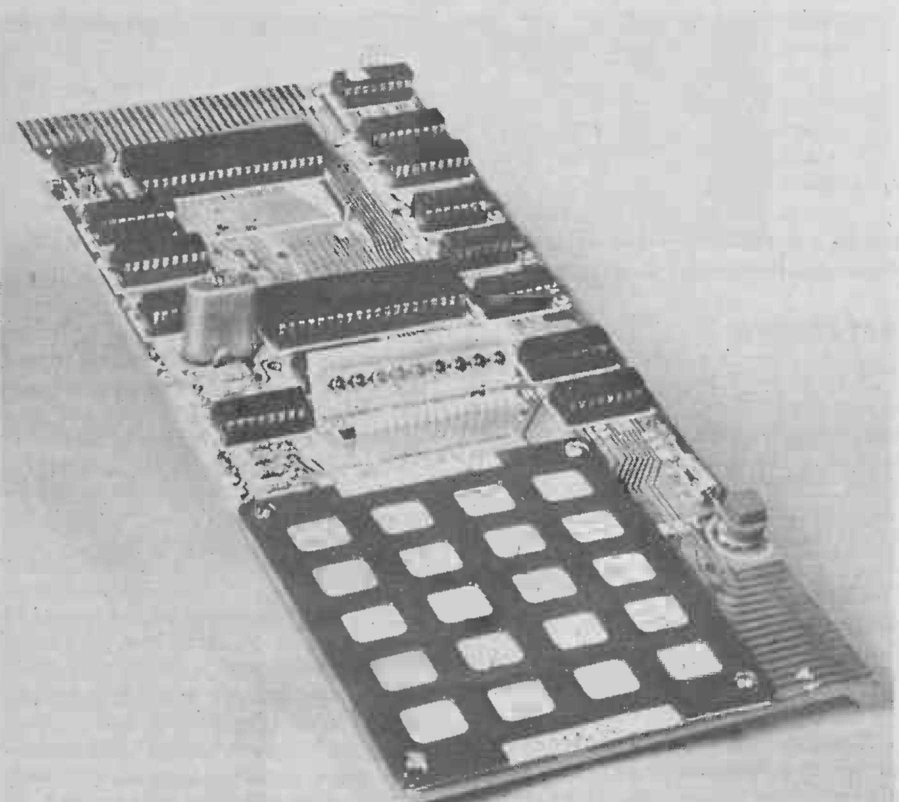
In Conclusion

An alignment service for tuners constructed according to the contents of this article will be available for approx £10.00. But the units must be working to a degree where alignment will consist of final trimming and tweaking to optimize the final unit. It cannot encompass trouble shooting of smoking regulators and vapourized ICs at the basic charge.

ETI

SCIENCE OF CAMBRIDGE'S MK 14 REVIEWED

Gary Evans has built and used Science of Cambridge's MK14, a kit that seems to offer a true low cost development system for National's SC/MP microprocessor. Here is his report.



The MK14 development kit from Science of Cambridge. The kit show has the optional add-on RAM I/O chip, top of board, and RAMs fitted. The edge connector at the top of brings out the I/O connections while the connector at bottom right (below reset switch) allows a remote keypad to be added to the system.

THE MK14 WAS LAUNCHED by Science of Cambridge earlier this year. The product, described as a microcomputer kit, sells for around £40, and features a SC/MP II microprocessor together with keyboard, display, 256 bytes of RAM (two 256×4), 512 byte monitor program (two 512×4) and various other items of hardware, that together provide the means by which machine language programs for the SC/MP may readily be written and debugged. The MK14 will also prove valuable to those who wish to learn more bout the ins and outs of using a typical 8 bit MPU, without having to spend the rather large sums of money associated with the purchase of some other development systems.

At this low price however just what does the MK14 have to offer in terms of performance and what corners, if any, have been cut to meet this low price tag.

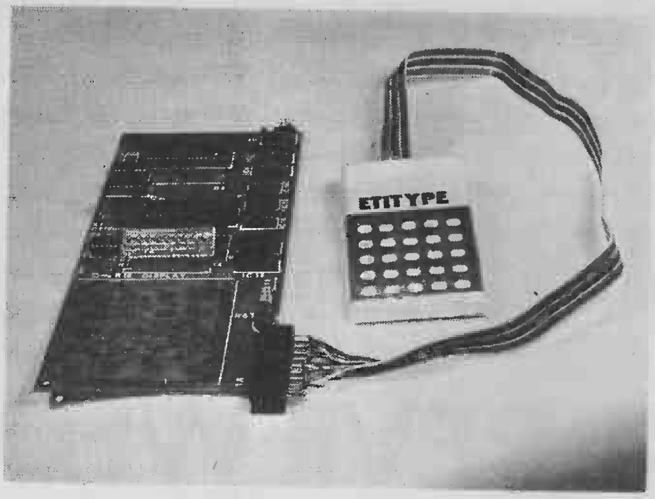
Demanding Supply

Before going on to describe the kit in detail though, it's as well to mention the supply problems that Science of Cambridge have had in meeting the demand for MK14 kits over the past few months. Initial problems with supplies of semiconductor devices and later, more acute troubles with production of PCBs, have lead to a large backlog of orders building up.

This situation is slowly being rectified as alternative suppliers are sought where the original has failed to keep to delivery dates and the 21 day delivery time quoted by Science of Cambridge should be met on all new orders and the backlog soon cleared.

Now to the kit itself and first let's ▶

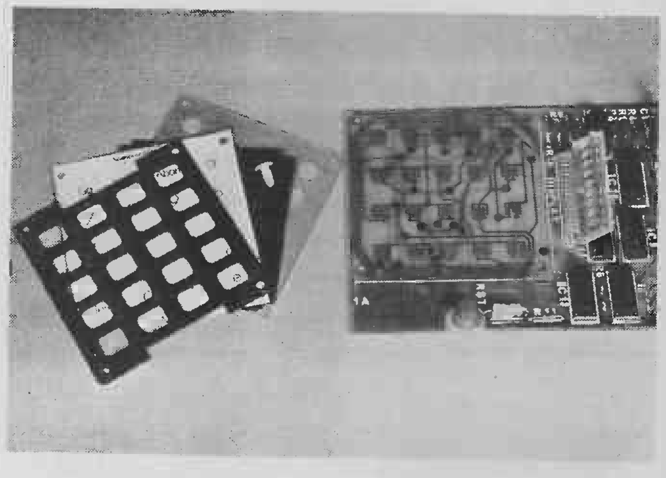
say that we found the MK14 to be a very good product and the comments that follow should be read with this in mind. We remark upon a number of features which in our view detract from the overall performance of the kit, but with these rectified, the Science of Cambridge are looking at some of them at present, we would have no hesitation in recommending the MK14. Suffice it to say that even



One way around the bad keyboard of the MK14, a cheap calculator is modified to provide the system's input.

The MK 14 is a kit and is supplied as a plated through PCB together with some 14 ICs, a four part keyboard, display, reset switch, crystal and various resistors and capacitors, which must be carefully assembled according to the detailed instructions in the MK14's manual. The only equipment required is a soldering iron, solder and a pair of side cutters.

The manual assumes very little knowledge of electronics providing a guide to the identification and orientation of the various components supplied. The manual does however assume a knowledge of the resistor colour code and a section describing this might be a valuable addition to help those who have little experience of electronics .



The pieces that go to make up the MK14's keyboard.

The kit is not supplied with sockets though the manual "most strongly recommends" that sockets are used, a view we share, the extra cost of sockets proving its worth if any fault finding/system expansion proves necessary.

Assembly Point

Assembly of the kit is straightforward. The only area we thought likely to confuse was around ICs 12 and 13. At first sight it seems that there are 18 DIL holes on the PCB whereas the ICs to be fitted are 16 pin devices. A closer look however reveals that the pair of holes nearest the edge of the PCB are unused, a remnant of some previous layout? A very minor criticism however and if the manual's instructions are carefully followed and a reasonable standard of soldering maintained (notes on soldering technique mean that even those who have not soldered before should be able to tackle this kit) the assembly of the electronic components should pose no problems. With all the electronic work complete the keyboard and display can be fitted. The display is an eight digit calculator type and is connected to the PCB via a short length of ribbon cable. Again a couple of spare holes on the PCB but it is fairly obvious where everything goes.

Key Feature

The keyboard is one of the areas where cost cutting is apparent. It is a sandwich type construction consisting of a metal plate with some 20 holes, corresponding to the 16 hex character and the four command words (more of these later), under which a ledgend sheet is positioned. The next layer consists of a sheet of conductive rubber. The last layer of the construction is a sheet of card with a matrix of holes similar to that of the top plate punched out of it.

The assembly is held together by a set of four plastic pegs. These will prove almost impossible to fit unless they are first "squeezed" with a pair of pliers. Even when fitted they are not really up to the job and, as you can perhaps see in our photograph, on our kit we used four 4BA bolts in place of these pegs with far more satisfactory results.

This arrangement is mounted above an area of the PCB that has a pattern of interlocking "fingers" etched onto it. The idea is that the conductive rubber, usually seperated from the PCB by the layer of card, will bridge the gaps between the "fingers" when sufficient pressure is exerted on the foam to force it down onto the board through the holes in the separator. That's the theory, in practice the operation is to, say the least, clumsy.

Science of Cambridge are aware of the difficulties of using the keyboard and are working on a number of solutions. These include providing plastic buttons to enable a more even pressure to be applied to the conductive foam or, a more expensive proposition, the provision of individual switches for each switch function.

A further solution is to connect an external keyboard to the MK14 via the keyboard edge connector. No details of the connection pattern for this are included in the manual, although

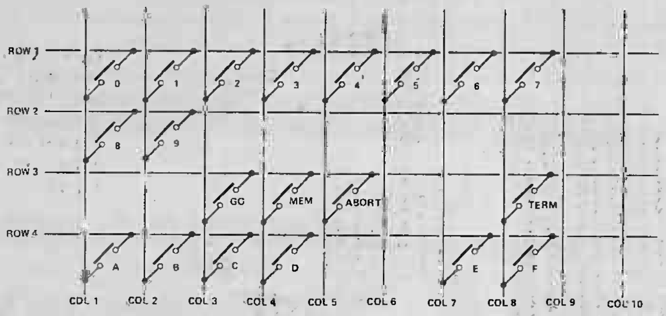
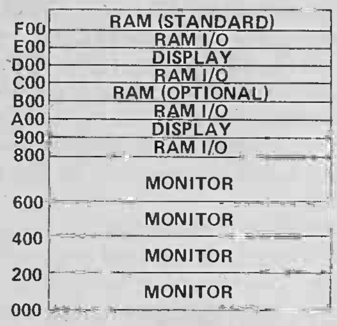
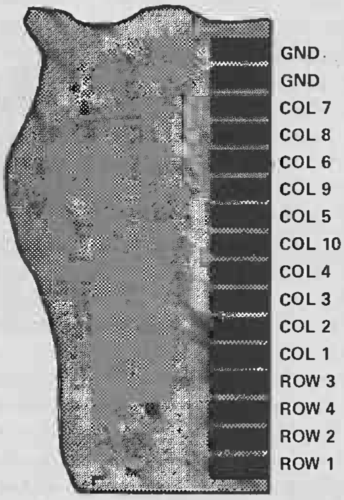


Fig. 1. The manual does not show it, but here is the connection information for that add on keyboard.

Fig. 2. Below, the memory map of the MK14 shows how the partial memory decoding of the kit results in the monitor and RAM I/O appearing all through the lower 4K of memory





it is fairly easy to trace the PCB tracks and work out how the extra keyboard should be wired up.

With assembly of the MK14 complete the manual suggests that you put the kit to one side for 24 hours (to rest the eyes) before inspecting the PCB for signs of solder splashes or of IC pins that have not been soldered. When satisfied that all is well it's time to power up and begin to get to know the machine.

Working Model

As the MK14 features an onboard 5 V regulator, a power supply with a DC output in the range 7-35 V can be used, although the regulator will require a heat sink if supplies near the upper limit are used. In addition if the supply has a lot of ripple on it an additional capacitor of about 2 000u should be fitted in the space provided on the PCB.

Upon switch on, if all is well, the display should show a series of dashes in the four leftmost positions followed by two blank displays and a further two dashes in the righthand positions. The group of four characters will form the address field, while the group of two digits will become the data field.

If instead of a nice row of dashes you get some other display try pressing reset. If things are still not right, turn off the power and check PCB again. Science of Cambridge tell us that they have had very few kits returned and the faults have been due, in the main to hairline solder splashes or, in some cases to PCBs that have been incompletely etched in some areas. Another reason for return is the apparent faults thrown up by an inadequate power supply. The supply must not drop below 7 V when on load and must not present too much ripple to the MK14 (this latter problem manifesting itself as apparent keyboard bounce).

If your MK14 will not go after all reasonable attempts to get it up and running Science of Cambridge offer a get you going service at little more than the cost of postage and replacement parts — expect to pay more if you haven't used sockets though.

Routine Example

The monitor program used by the MK14 is the same as that of the National Introkit plus Keyboard kit combination (KITBUG) and as such features four command words: GO, MEM, ABORT, TERM.

The dashes referred to above indicate that the MK14 is awaiting a GO or MEM command. The first of these to be introduced by the manual is the MEM key. This allows the user to display the contents of the MK14's memory. After pressing the MEM key a four digit hex number may be entered via the keyboard, the MK14 echoing this number in the address field as it is entered and displaying the contents of the memory location pointed to by the entry in the data field.

To examine the next memory location all that it is necessary to do is to operate the MEM key again whereupon the number in the address field will be incremented by one and the contents of the corresponding memory location displayed in the data field.

End of Term

The MEM key is also used in conjunction with the TERM key to modify the contents of the MK14's RAM. The location which is to be modified is first pointed to using the MEM instruction as above. The TERM key is now pressed and the two digit hex character we wish to enter can now be input via the keyboard, it being echoed as input in the data field's display. Further operation of the MEM key will increment the address pointer as before, the TERM key preceeding any data input. In this way a program can be built up in the system's RAM.

To execute a program entered in the above manner the GO key is used to set the address pointer to the memory location at which we wish to enter our routine.

The ABORT key will return the system to a condition in which it ex- ▶

pects either a MEM or GO command. A reset will have much the same effect except that ABORT will not destroy the contents of the SC/MPs registers.

The manual takes the user through the operation of the command keys by describing the entry and execution of a sample program. The manual however fails to give an exact definition of their use or function and a section expanding on this aspect would be valuable.

The manual also makes no mention of how to input and output data from a user program. This together with the fact that sections on the basic principles of the MK 14 and a section on SC/MP architecture and instruction set would still leave the person with no knowledge of microprocessors a trifle lost is a little disappointing.

Science of Cambrige tell me however that the reason for this state of affairs is that a section covering programing, which will cover some of the above points was inadvertantly omitted from the manual. These details will however be included in all kits sold from now on.

In addition to the addendum covering programming, a section making the use of the various programs listed in the manual a little clearer will also be included with all MK14s. There are some 22 program listings under the headings of Mathematical (multiply, divide etc.), Electronic (pulse delay, random noise etc), system (single step, relocator etc), games (moon landing, mastermind, etc), music (organ, etc) and miscelleaneous (message reaction timer, etc). Together these provide a good way of becoming familiar both with the MK14 and with the SC/MP MPU.

For those who start with a little more idea of MPU operation the complete monitor listing included in the manual will prove a valuable aid in trying to get the most out of the system. There is also a full circuit diagram for those who wish to extend the basic system.

I/O, I/O, It's Off To Work ......

That then is the basic MK14, but what of expansion? The PCB includes space for the addition of a further 256 bytes of RAM and of the National INS8154 RAM I/O device. This latter IC will greatly extend the scope of the MK14 kit providing as it does a set of 16 lines (configured as two seperate eight bit ports) each of which may be seperately defined as either input of output under program control. The IC also provides a further 128 bytes of RAM. The manual describes the use of the RAM I/O chip's various features including a section of the IC's use in handshaking mode. Connections to this IC are brought out to an edge connector at the top of the board. Again, although the manual describes the device, the explanation is brief, and for those unfamiliar with the IC, will leave questions unanswered.

As well as the extra memory and I/O chip referred to above, Science of Cambridge plan to introduce a number of other MK14 expansion aids. First on the cards is a cassette I/O using a simple tone burst system together with a new monitor to include the software for this interface and to provide for easier data entry (getting rid of the MEM-TERM-MEM approach) and providing an offset calculation function. The space for these extra routines has been found by tidying up National's original monitor. Note that these new ROMS will be compatible with existing hardware and the cassette I/O can be used with Mk1 monitors by storing the necessary software in user RAM.

Plans also include a PROM programmer for a fusible link PROM and a low cost VDU.

Last Night Of The PROMS

The basic kit, as a cost saving measure, adopted a system of partial memory decoding and the basic board's maximum RAM complement of 640 bytes cannot be extended without alterations to the board — hardly worth bothering with a VDU. However the alterations to the PCB, involving the use of gates, at present used, enable up to 4K of memory to be addressed are not that major details of such modifications will be made available.

A second volume of programs is also in preparation. This will highlight the MK14's rôle as a control system with programs that should find a wide range of applications.

To sum up, at £40 the MK14 while not perhaps a "microcomputer" is an easy to build development kit that provides an excellent way of getting to know about MPUs. The system is let down at present by its poor keyboard and by some ommissions in the kit's manual.

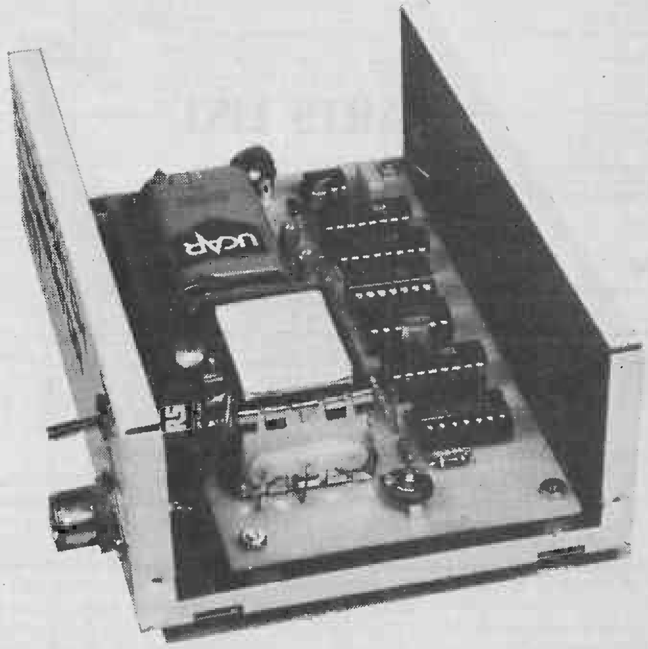
Science of Cambridge are however aware of these faults and are working on them. As for value for money, the MK14 is certainly the cheapest development kit that we know of, and with the cost of components bought individually coming to more than the kit price, its got to be a good buy.

The MK14 is not a toy and with the low cost addons planned by Science of Cambridge, should prove a powerful tool to those wanting a versatile MPU development system at under £80.

ETI

CROSS HATCH GENERATOR

The ETI Project team have hatched up this neat piece of test gear.



THE COLOUR television picture is created in the receiver picture tube by three separate electron guns — one each for red, green and blue. As these guns cannot be in the same physical position they need to be converged into one spot on the screen.

The process of converging at the centre of the screen is called static convergence and is performed by magnets on the yoke assembly.

Green Cross Code

However, the green ot the picture tube is not everywhere coincident with the deflection plane and this causes errors when the beam is deflected away from centre. These deflection errors are corrected electronically by 12 or more controls and the process is known as dynamic convergence.

An important part of the process is the use of a crosshatch generator to provide horizontal and vertical lines on the screen. Using the generator, the convergence errors are immediately apparent and the controls on the set are usually labelled with the effect each has on a crosshatch pattern.

In addition to setting up convergence the generator pattern may also be used to set up horizontal and vertical linearity and to orientate the deflection yoke coils on both black and white and colour sets.

Carry Her Power

Most of the inexpensive pattern generators, which are currently available, produce a video waveform, which must be injected into the correct place in the TV, and require a synchronizing signal from the TV set. Such generators are thus fiddly things to use.

The ETI crosshatch generator produces a combined horizontal and vertical sync waveform and this, together with the crosshatch video, is modulated on to a carrier frequency operating on UHF channel 36. Thus to use the generator one simply attaches it to the antenna terminals and selects channel 36.

Construction

Assemble the PCB according to the overlay starting with the links, resistors and diodes. The 555 IC and capacitors next with the CMOS ICs last. Solder the power supply pins of the CMOS (7 and 14 or 8 and 16) first. This allows the internal protection diodes to protect the inputs of these ICs.

We mounted the unit into a small metal case, this prevents drift due to the presence of hands, etc, as it provides good shielding.

Alignment

This is easiest if a frequency counter or oscilloscope is available. Monitor the output on pin 1 of IC4 and adjust RV1 to give 50 Hz.

Connect the unit to the TV set and select channel 36. RV2 should be adjusted to give vertical lines of about the same width as the horizontal.

The cross hatch generator is now ready for use. May we suggest however, that before making any adjustments to your TV you consult the manufacturers' service guide for the set. ETI

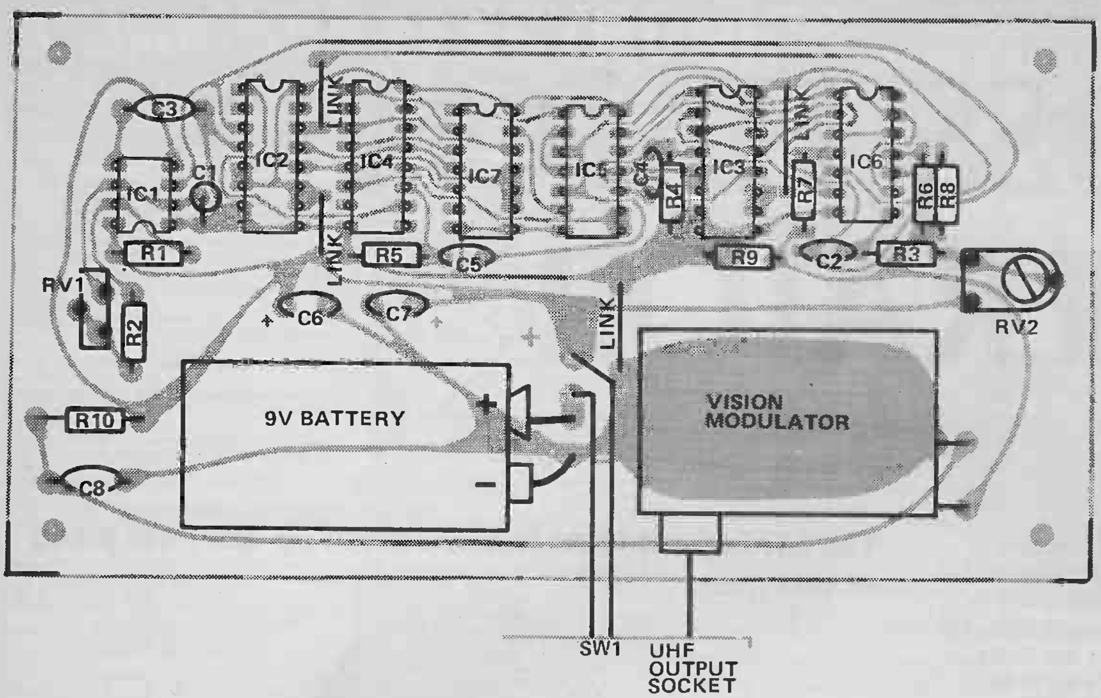


Fig. 1. The cross hatch generator's overlay is shown left.

PARTS LIST

RESISTORS

R1	1k0
R2,7,8,	4k7
R3,4,5,6	10k
R9	330R
R10	110R

POTENTIOMETERS

RV1	5k miniature preset
RV2	25k miniature preset

CAPACITORS

C1	180p ceramic
C2	22p ceramic
C3,8	10n polyester
C4,5	100p ceramic
C6,7	33u 16 V tantalum

SEMICONDUCTORS

IC1	555
IC2,3	4027B
IC4	4040B
IC5	4011B
IC6	4001B
IC7	4012B

MISCELLANEOUS

PCB as pattern, case to suit, output socket, single pole toggle switch, 9 V battery, Astec UMIIIIE 36

BUYLINES

The only component liable to be difficult to obtain is the Astec UHF modulator. These are available from most suppliers of TV game kits, Watford Electronics and Teleplay are examples. Make sure you get a vision modulator, sound modulators look the same but will not work in this application! All the CMOS and other components is widely available. The PCB will be available from usual suppliers who advertise regularly in the magazine.

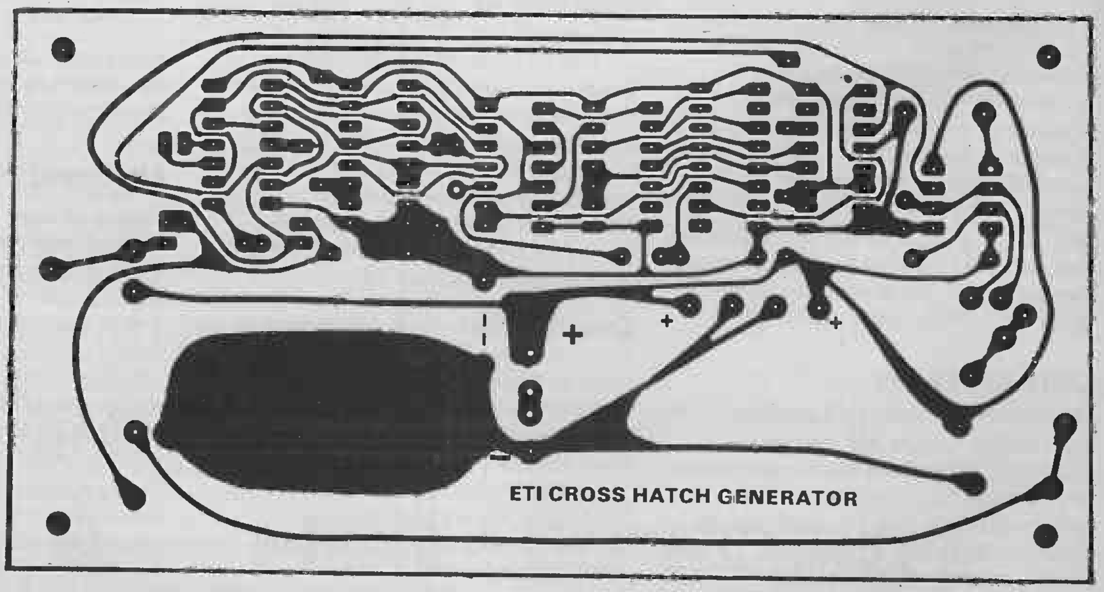


Fig. 2. The foil pattern of the cross hatch generator is shown full size on the right.

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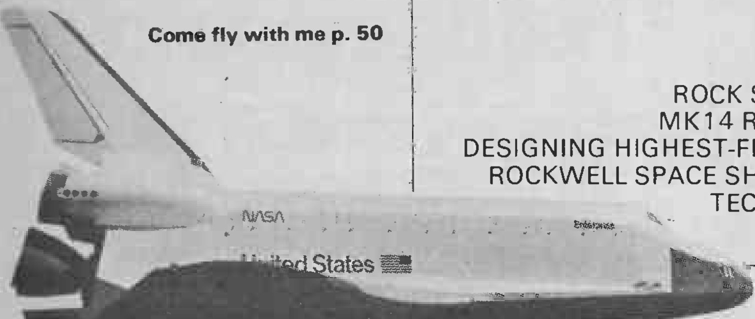
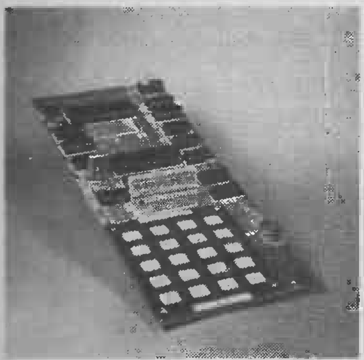


Come fly with me p. 50

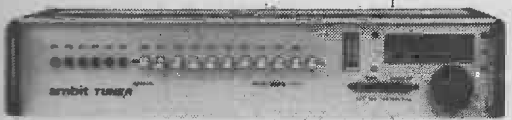
FEATURES

PROJECTS



Key kit ? p. 27

NEWS



Tune in and count on p.21

INFORMATION

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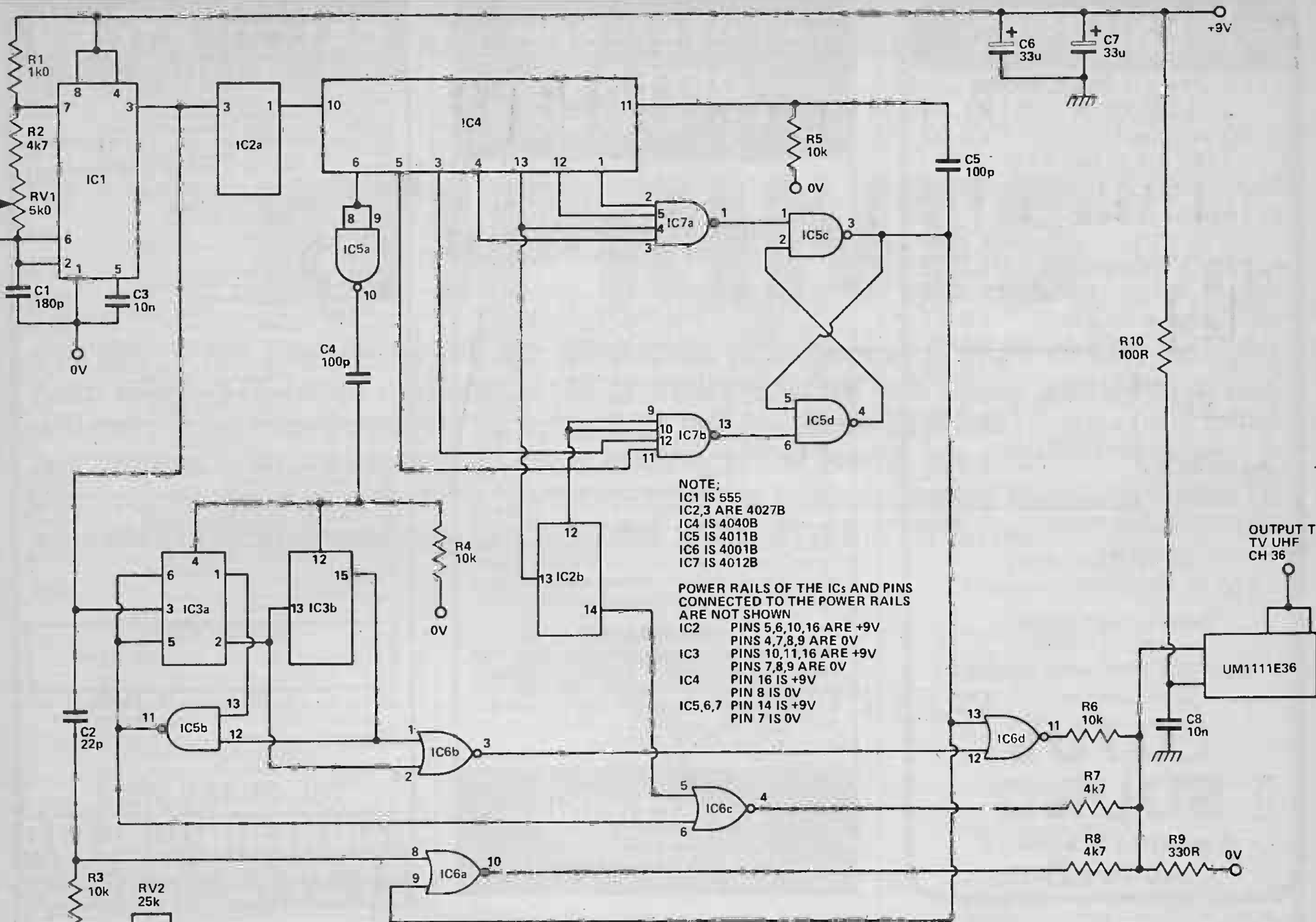


Fig. 3. The full circuit diagram is shown above.

HOW IT WORKS

A TV picture is made up of a series of horizontal lines equally spaced down the screen with the information transmitted in a serial form along with the necessary synchronization pulses. There are 625 lines in each complete picture but these are transmitted as two "frames" each of 312½ lines with the second frame interlaced between the first giving a total of 625 lines. This is to reduce flicker of the picture which would otherwise occur.

To simplify our circuit and prevent a double horizontal line we have used 624 lines which eliminates the interlacing. The TV set automatically accepts this change.

To synchronize the TV set we need a 192μs wide pulse every frame (20ms) and a 4μs wide pulse every line (64μs). All pulses, including the information, are derived from a single 249.6 kHz oscillator IC1. This is divided by 2 in IC2a and then by 2496 by IC4 giving an output of 50 Hz. This IC is a 12 stage ripple counter which, while normally dividing by 4096, can be forced to divide by 2496 by decoding (IC7) the outputs from the 7th, 8th, 9th and 12th stages and reseting IC4 back to zero. The output of IC7 toggles the RS flip flop IC5/c, IC5/d which resets IC4 via C5. This flip flop is reset by the decoded output from the 4th and 5th stages of IC4. This occurs 192μs later; thus the output from IC5/c is the frame sync. pulse.

To generate the line sync. pulse the output from the 3rd stage of IC4 (15,600 Hz) is used to reset both halves of the dual JK flip flop IC3. This IC is then toggled by the 249.6 kHz clock until, after three pulses, both "Q" outputs are '1' when IC5/b detects this and disables IC3/a, IC6/b decodes the second of these clock periods and this becomes the line sync. pulse. These pulses are combined in IC6/4 to give a combined sync. pulse.

The 249.6 kHz is differentiated by C2/R3 and after being quared up by IC6/a is used to generate 16 white spots on each line which results in vertical lines. These pulses are deleted during the frame sync. period to prevent interference to synchronization. Due to variations in the CMOS a trim potentiometer is provided to give equal width to the vertical and horizontal lines.

The horozontal line is generated by IC2/b (JK flip flop) and this IC is toggled by the 8th output (487.5 Hz) of IC4 and is reset by the output of the 4th stage (64μs later). This gives a single white line every 16 lines. To prevent this line interfering with the line sync. pulse the output of IC2/b is combined with that of IC5/b which is high for a period 4μs before the line sync. pulse to 4μs after the pulse. This gives a short black region on both ends of the line (normally off the screen). The outputs of IC6/b, IC6/b and IC/c are combined by R6-R8 to give a composite video signal. Note that the video information gives positive pulses while the synchronization pulses are negative.

The video signal is fed to the UHF modulator. This is a ready built unit that is adjusted at the factory to operate on channel 36. R10 and C15 decouple the supply to the modulator.

DESIGNING HI(EST)~FI AMPS PART 2

Stan Curtis considers which parameters matter most in super-fi audio, and how they can best be optimised.

FOR MANY YEARS it has been the standard practise to specify and compare amplifiers through their ability to handle a continuous (steady state) sine-wave signal. Thus such a signal is used to measure power-output frequency response, harmonic distortion, crosstalk, input overload capability, intermodulation distortion, damping factor, and gain! Unfortunately many engineers and Hi Fi pundits still believe that such information is ALL that is necessary to quantify an amplifiers performance and to compare it with others. Not so.

Steady-state sine-wave testing can however, tell only part of the story and can often be misleading. Music contains complex wave forms with a spectral content of greater than eight octaves and dynamic ranges of up to 100 dB. Yet such complexity is readily understood by the human brain which, in mastering the subtleties of spoken language, has evolved the ability of extraordinary auditory sensory perception. The music signal, as with all audio signals, can be considered in terms of two variable qualities: — the frequency domain, and the time domain.

Frequently Timely

The frequency domain is the area that has monopolised engineers thought for so long. Even the most complex music signal can be represented by a Fourier Analysis.

This develops mathematical equation which lists separately each frequency. making up the signal, together with its phase and amplitude. However, a Fourier Analysis is only complete in the case of simple waveforms, with more complex waveforms it becomes only a convenient approximation.

Of course, in order to make a Fourier analysis of a signal the components of that signal have to be analysed over a period of time such that complete cycles of the lowest frequency can occur.

Thus we take consideration of the Time Domain.

Where steady-state signals are concerned the Time Domain is not normally considered as the signal is of a continuous unchanging nature between any two periods. If the "time window", during which the signal is Fourier analysed is reduced progressively it becomes apparent that an accurate spectral analysis becomes less possible. It can then be seen that the important characteristics of the signal are amplitude and rate of change. In other words it's envelope.

What is required is the amplification of an audio waveform in such a way that the ear can detect no degradation.

What Do We Want?

Let us consider ways in which such degradation can occur. The waveform envelope can be distrorted by amplitude changes of any component or by changes in the phase relationship of the component harmonics.

Experimental work has established that changes in the relative amplitudes of the harmonic structure of the waveform are readily detectable.

Other work has shown that the qualitative characteristics of a complex sound depend upon the phase relationships of the component harmonics. It would seem that as a phase difference must be interpreted as a time delay between the component parts of the signal then a sufficient phase shift in a system must eventually become audible as these component parts are moved in respect to each other in time. In practise large phase shifts are very audible and indeed telephone lines are often subjected to phase and delay correction to render speech intelligible. However, establishing an acceptable degree of phase shift is extremely difficult.

Following the arrival of the "linear phase" loudspeakers great controversy has raged over whether phase shifts effect sound quality. A study of the experimental work performed to date shows that

i). It seems to be very difficult to repeat someone else's experiment (and get the *same* results!)

ii). It seems, on balance, that where recurrent waveforms (steady state) such as sine-waves (and instruments producing a "continuous" although decaying tone) are concerned; then quite large phase shifts, between the extremes of the frequency band, have no identifiable effect on sound quality.

However, a phase non-linerarity on the leading edge of a true transient appears to be audibly more perceptible. Particularly on speech and percussive sounds. ▶

Bandwidth and TID

Transient signals cause many problems for amplifiers of which phase lineratity is but one. Other problems are; instability and ringing, clipping, slew-rate limiting, and transient intermodulation distortion. Transient intermodulation distortion (TID or TIM) is an effect that has been much in vogue in the past 3 or 4 years but which is often misunderstood. TID can be predicted mathematically but such a description is out of place here. TID most commonly occurs when an amplifier, with overall negative feedback over several stages, is driven by a large enough signal whose frequency (or equivalent rise time) is above the open loop bandwidth of that amplifier.

Because the feedback loop is fed from the output of the amplifier, it cannot be operating until signal current flows at the output. i.e. during the open-loop rise time of the amplifier.

The outcome is very large signals occuring in the intermediate stages of the amplifier causing those stages to distort or even to clip. With some amplifiers this clipping (which cannot occur with any steady-state signal) can cause the stage to latch-up for a time until the operating conditions restabilise.

Thus not only is the leading edge of the signal severly distorted — in some cases it is removed completely.

TID is therefore a form of overloading that is dependent upon both amplitude and time. This is audibly (but at a higher signal level) similar to cross-over distortion, as both effects cause phase and amplitude modulation of the signal due to momentary change in gain. (Remember that at the corss-over point zero, there is a no current flow in the output stage and hence no feedback current and so the amplifier is momentarily open-loop).

Making Big Bands

TID can be avoided by careful design an amplifier whose open-loop bandwidth is greater than the highest freqiency of the input signal. The maximum bandwidth can then be defined at the input by a passive RC Filter. Thus if we decide upon a maximum signal bandwidth of 20 KHz than our filter will limit the signal waveform rise-time to T = 0.35

$$T = \frac{0.35}{20kHz}$$

i.e. 17.5 uS

Our amplifier's open-loop bandwidth should be designed to be, say 23 kHz, giving it an open-loop rise-time of 15 uS. and freedom from TID. If however, in the interests of a good specification, and possibly better reproduction, we decide upon a closed-loop bandwidth of 100 KHz (i.e. a rise time of 3.5 uS.) then our amplifier will need an open-loop bandwidth of greater than 100 kHz to maintain freedom from TID effects. In a power amplifier such performance is not easy to obtain.

Fast power transistors are notoriously easy to blow-up and are expensive. The common form of lag compensation (used where the open-loop bandwidth is perhaps 2 kHz) has to be replaced by lead compensation:—

Another technique is an extension of the first in that the preceeding stage of the power-amplifier is designed to have a lower open-loop band width than the next.

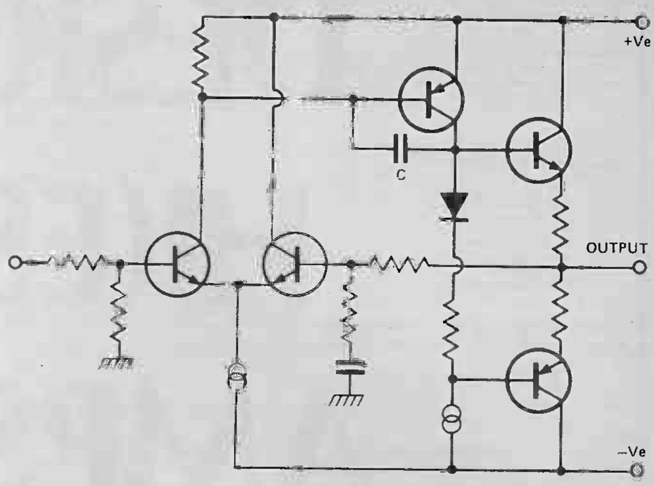


Fig 1. Circuit diagram of a typical amplifier circuit which employs lag compensation techniques — provided by C.

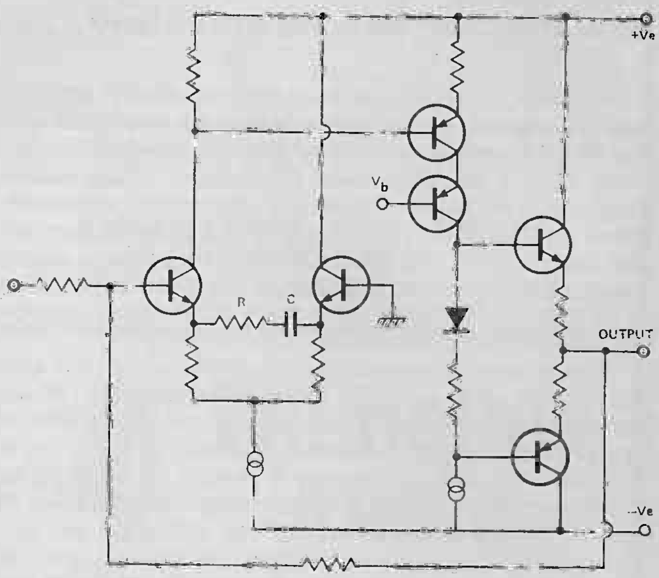


Fig 2. The other method. Lead compensation illustrated. Components R and C provide the time constant.

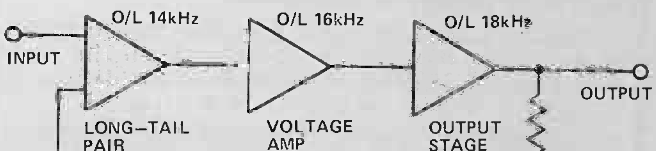


Fig 3. Third method of avoiding TID. Making each stage in the design have a wider B/W than the preceding one.

Important or not?

Many people now consider that TID is unimportant or that it doesn't exist. This is partly because it is very difficult to measure and only readily visible in the laboratory in the "clipping" stage. To reach this stage with most amplifiers (but not TID — free designs) requires either fast rise-time or high signal levels or both.

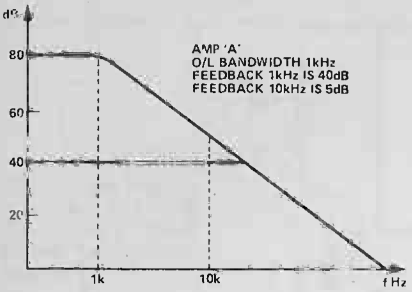


Fig 4. This amplifier design has a limited open loop bandwidth and the THD will rise with frequency.

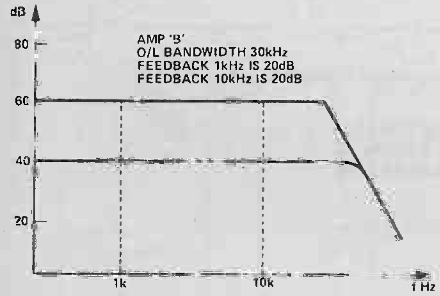


Fig 5. Contrast this with figure four above. The bandwidth here is much wider, resulting in a more linear THD response.

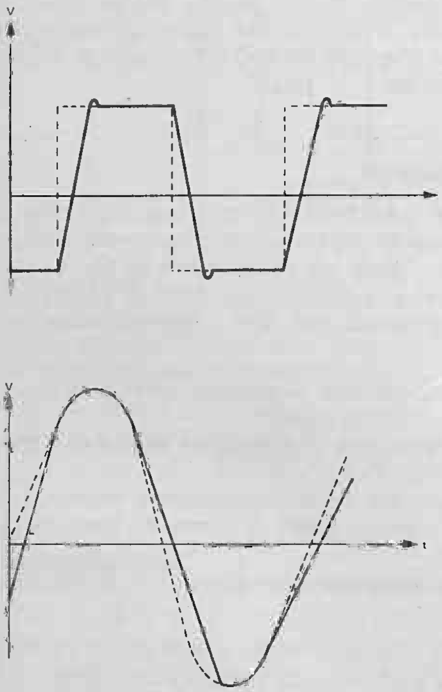


Fig 6. The effects of slew-rate limiting on a signal passing through an amplifier prone to this fault. Top: a squarewave, note the slight overshoot. Below that a sinewave. In both cases the dotted line represents the input.

Conditions that are unlikely to occur in practise.

However, a large degree of non-linearity and hence bad intermodulation will still occur with more realisable input signals. Although this cannot be measured yet (how do you measure say, 5% IM over a period of 5 milliseconds??) it can be predicted mathematically and, just as important, heard. Amplifiers free of TID have a very "open" quality with accuracy of depth.

Benefits Conferred

An amplifier designed with a wide open-loop bandwidth, for low TID often has other, more tangible, benefits. The high frequency THD is usually no higher that at the mid-point; in stark contrast to more traditional designs. This is because gain is still available at high frequencies for negative feedback.

Slew Who?

Such amplifiers also usually have much higher slew-rate. Slewing-rate defines the speed with which the amplifier can deliver output voltage to the load. For example, if an amplifier has a maximum output of 100 volts p/p and a rise-time of 10 uS. then the amplifier, if it were perfect, should have an output of about 80 volts after 10 u secs in response to a suitable square wave input. In other words the output voltage would have risen at the rate 8V/uS.

However, amplifiers do not generally respond to large changes as fast as their small signal characteristics predict, for circuit and transistor capacitances can be charged only as fast as their driving circuits allow. In its simplest form the slew-rate of an amplifier defines how fast the output voltage can change for large signal conditions, and it is normally quoted in Volts per micro second. The maximum slew-rate of an amplifier is usually limited by the slowest stage in its circuit.

That stage will have an operating current I (as set in the design) and a capacitance C (usually a frequency compensation capacitor)

$$\text{Slew-Rate} = \frac{T}{C}$$

Thus it a transistor stage has a standing current of 100 u A and is compensated by a 43 pF capacitor then its Slew-Rate will be

$$\frac{100}{33}$$

i.e 3 V/u S.

Depending upon the design some circuits have a different Slew-Rate depending upon whether their output is negative-going or positive-going. Slew limiting also defines the full-power bandwidth; a figure more commonly quoted by manufacturers.

$$fp = \frac{SR\,(10^6)}{2\pi E\,op}$$

E op = peak output swing in volts
fp = Full power bandwidth in Hertz.

Thus in a 100 Watt (into 8 Ohms) amplifier having full-power bandwidth of 20 kHz the required minimum slew-Rate would be about 5V/uS. This is, however, the absolute minimum figure and experience suggests that such an amplifier would have a hard, gritty high-frequency sound. Such an amplifier should have a Slew-Rate of greater than 20V/us to be certainof avoiding the increase in distortion caused by the gradual onset of slew-limiting. ▶

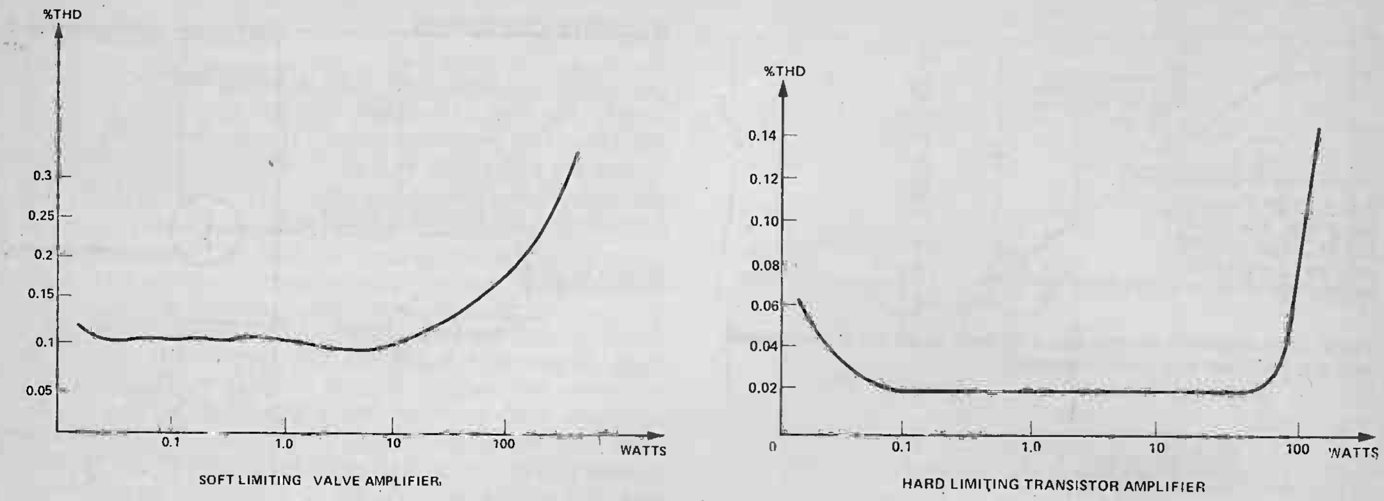


Fig 7. A comparison of the limiting characteristics — in general — of both transistor and valve amplifier types. There is a body of opinion which holds these curves to be the whole truth as to why valve amplifiers are preferred by many musicians.

Unfortunately the higher the power output of the amplifier the greater the required slew-rate as more volts are swing at the output in the same period of time and so as our 100 W amp needs a 20V/uS an otherwise identical 50 W amp needs 14V/uS and a 20W amp needs only 9V/uS.

Clip Around The Ear

But these forms of distortion tend to give subtle audible effects compared to the most common amplifier problem — that of clipping. Clipping occurs when an amplifier is overloaded by high level signal peaks. Such peaks occur frequently in much music material and so the manner in which the amplifier clips determines its audibility. A soft, clipping effect where the distortion rises gradually (typical of valve amplifier circuits) is audibly preferable to the hard clipping typical of transistor circuits.

Worse still, some amplifiers tend to suffer saturation effects on clipping, and take a time to recover; thus artificially extending the length of time the signal is clipped. The use of overall negative feedback to reduce distortion unfortunately makes things worse. Overall feedback effectively linearises the clipping making it hard the distortion changes from 0.01% (say) to 10%, and quite suddenly too.

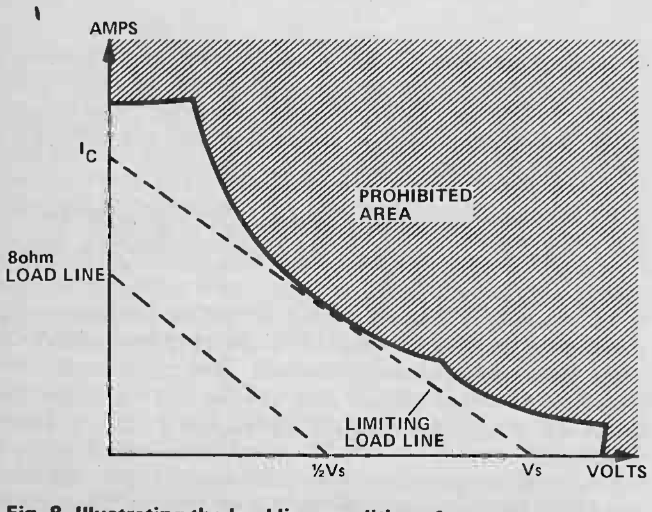


Fig. 8. Illustrating the load line conditions for output stages.

Designing A Designer

We have covered just a few of the requirements a designer must consider when working upon the design of power-amplifier. There are many more to be considered to even rough out a design specification before the circuit hardwave is considered. The following sequence is mandatory:

1. What parameters are important to prevent any audible degradation of the signal?
2. Detail a performance specification that meets the requirements of (1).
3. Decide upon the circuit technology necessary; Bipolar; MOSFET; Valve; Class A; Class B; Switching; fact; slow; etc; etc.
4. Perform a development programme to produce a prototype.

At this point the designer has to admit that it's a real world and that his performance specification cannot be achieved in a way that is acceptable to the accountants, salesman, customer, customer's wife or whoever else is around. Trade-offs are necessary and much to the "art" in amplifier design is in the deciding which defects and degradations are more acceptable than others.

As an illustration of the changes in design approach over the years we will briefly illustrate three designs for which the author has been responsible:

1. Cambridge Audio P60 (P80) (designed 1974)
2. Lecson AP3 Mk II (designed 1976)
3. Mission Electronics Voltage Amplifier (designed 1977)

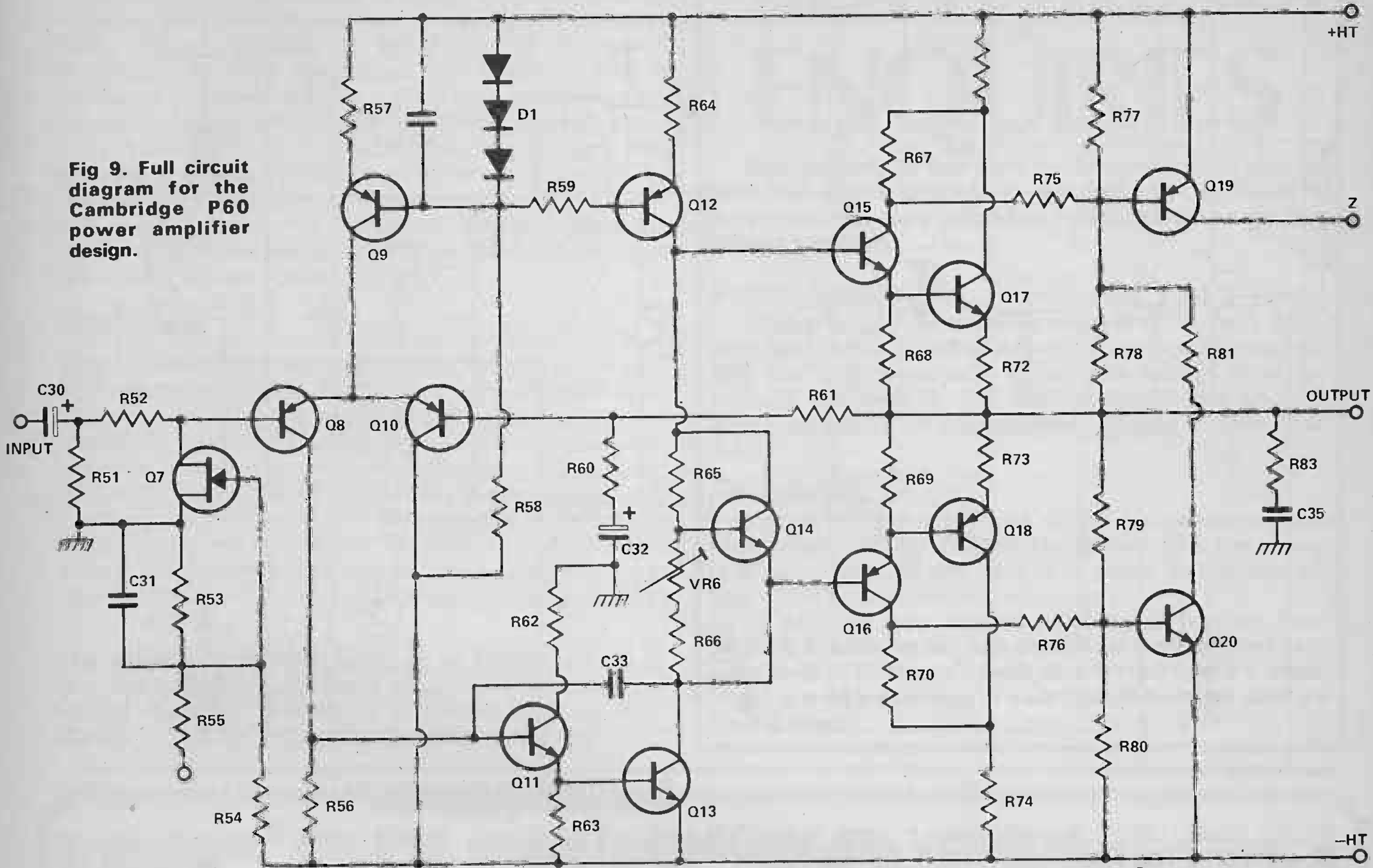


Fig 9. Full circuit diagram for the Cambridge P60 power amplifier design.

HOW IT WORKS—Cambridge P60

The P60 power amplifier is of a conventional design but with care being taken to optimise each stage. Q8 and Q10 form a long-tailed pair with Q9 as their emitter current source. Q8 and Q10 must be very closely matched for minimum DC offset and for maximum common-mode rejection to avoid H. T. ripple appearing at the output. The next stage is the Q13 voltage amplifier which is loaded by a current source (Q12) instead of the more common "bootstrapped" resistors. Note that Q13 is buffered from the long-tail pair by an emitter follower (Q11) to prevent any loading of that stage worsening the distortion characteristics.

Capacitor C33 gives lag compensation which defines the dominant pole of the amplifiers. The open-loop bandwidth is quite high (for this type of circuit) at 12 kHz but none the less this amplifier is prone to TID effects. The protection circuit is very unusual in that the output is limited by an FET (Q7), Q19 and Q20 each form conventional V-I summing circuits which monitor the loading on the output stage.

If either Q19 or Q20 turns-on, the gate of the FET Q7 (normally biased-off by R54 to the negative HT) is biased positive and it starts to turn-on. It then acts as a potential divider with R52 and thus attenuates the audio signal. This protection only turns on at the equivalent of 50 W into 2 Ohms load and when it turns on it only adds moderate distortion (0.2% typically) as distinct from clipping.

Improvements

The P60 is capable of good mid-band performance (THD 0.01% at 1 kHz 30 W) but its high frequency distortion is poor because of the limited open-loop bandwidth. Generally this amplifier performs well at low and moderate levels but at high levels its sound quality becomes hard and aggressive. Some improvements to this circuit can be quite simply made as follows:

1. A resistor is fitted between Q10 collector and the negative rail to give better balance between Q8 and Q10.
2. A cascode transistor is fitted to Q13 collector to reduce "Early effect" distortion due to the collector-base capacitance of Q13.
3. An emitter resistor is fitted to Q13 to provide local negative feedback.

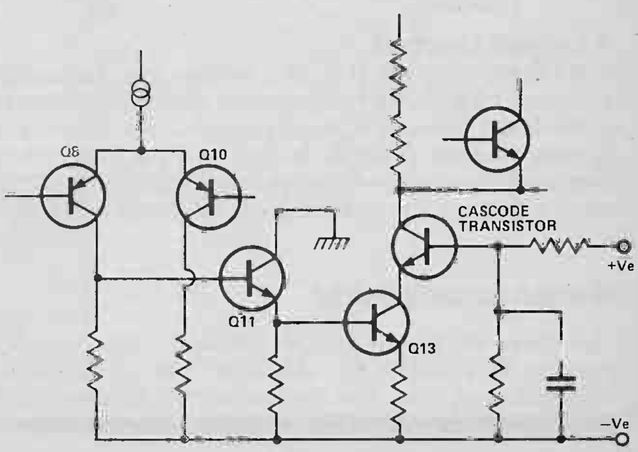


Fig 10 (Right). Showing how some of the improvements mentioned can be added to the P60 basic design.

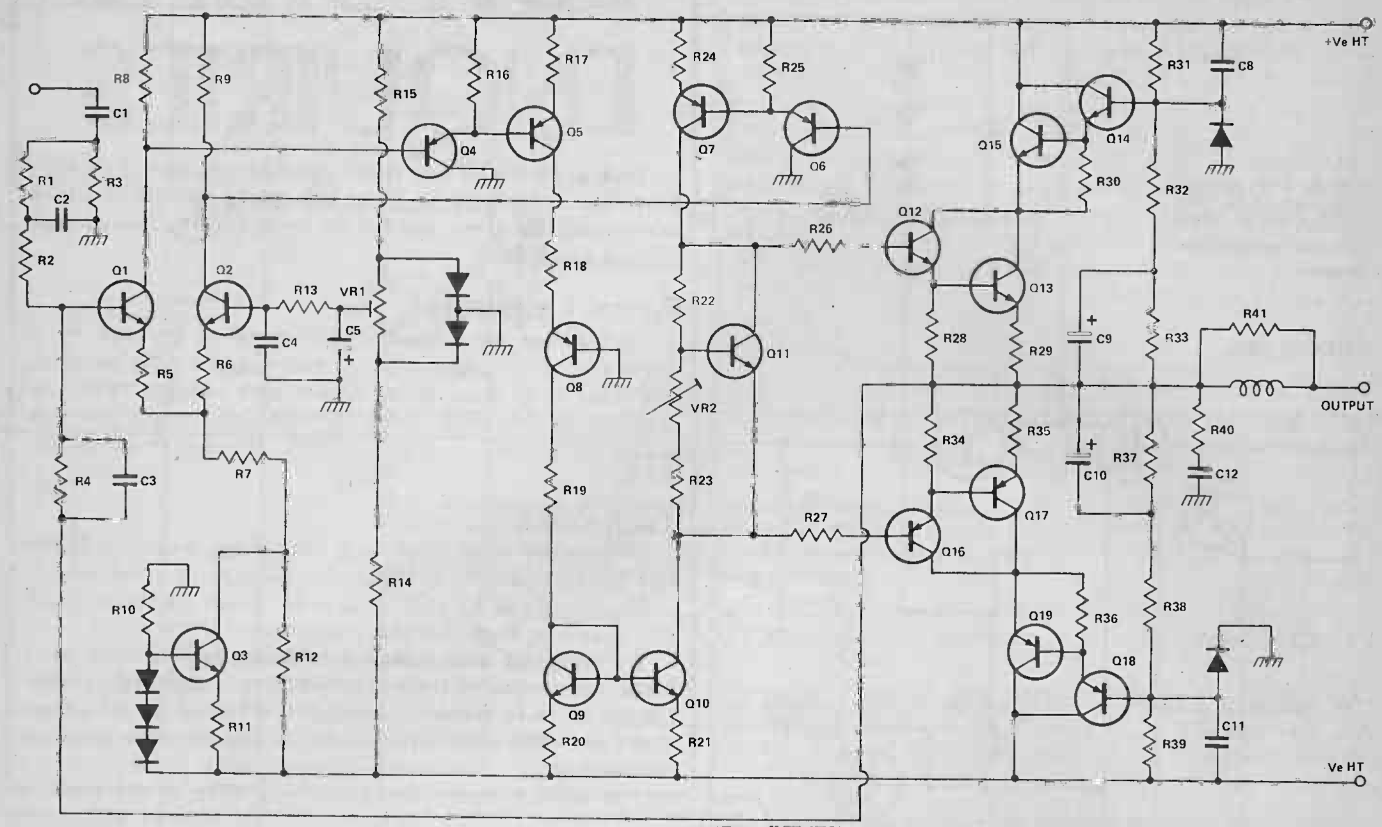


Fig. 11. Full circuit diagram for the Lecson AP3 power amplifier design, producing around 150W.

HOW IT WORKS–Lecson AP3

Transistors Q1 and Q2 form a long-tailed pair differential amplifier with Q3 as the emitter current source. Local feedback is applied in the form of emitter resistors R5 and R6. The base of Q2, instead of being grounded, is connected to a potential divider RV1 which permits the DC offset at the output to be set to zero. The input signal to Q1 is passed through a low-pass filter (R1, C2) which sets the bandwidth to 22 kHz (i.e. below the open loop bandwidth for no TID effects). The bi-phase outputs of the long-tail pair feed a second differential amplifier Q5 and Q7. Transistor Q5 has a constant current load (Q8) whilst is terminated by a current mirror (Q9 and Q10). Transistor Q10 will always deliver the same current as transistor Q9 hence the term "Current Mirror" and the excellent symmetry and balance this stage achieves. Functionally, however, Q10 can be considered as an active load whilst Q7 is a voltage amplifier from whose collector the drive to the output stage is taken. Note that Q5 and Q7 both have local emitter feedback (R17, R24) and that both are buffered from the long-tail pair (Q4 and Q6 emitter followers).

Transistors Q12, Q13, Q16 and Q17 each form conventional Darlington emitter follower stages. Each stage is series connected to a further power transistor (Q14, Q15 and Q18, Q19 respectively) which is permanently biased ON. Their emitter potentials are determined by the ratio of the base potential dividers. This ratio was chosen such that Q13 and Q15 each has half the supply rail across them.

The whole amplifier is in the inverting mode with overall shunt feedback through R4 and C3.

This amplifier is quite fast having an open-loop bandwidth of about 27 kHz. The circuit is stable without the usual compensation capacitors within the loop. THD is low being typically (at 100 W into 8 Ohms) 0.004% at 1 kHz and 0.02% at 10 kHz. The HF distortion can be further improved by selection of transistor Q7 for a device with a low collector-base capacitance.

No conventional protection circuits are used as extremely high power transistors are fitted and these can survive a short-circuit condition in the time taken for the power supply to shut down.

A Lesson Learned

The Lecson AP3 Mk II is an amplifier that incorporates much of the thinking in this article and is representative of the latest types of high performance amplifiers. It is a directly-coupled Class B design using a fully complementary output stage of series connected transistors and gives a power output of around 150 Watts per channel.

Mission Accomplished

The New Mission Voltage Amplifier represents an attempt to produce an amplifier that performs well irrespective of load. The circuits cannot be described at this stage as they are the subject of patent applications. However, a brief description will illustrate the philosophy behind the design.

The casing contains two completely separate Mono Amplifiers each with its own power supply. A separate module carries the DC-voltage offset protection circuits; the delayed switched-on circuits; and the thermal protection circuits. Particular attention has been paid in the design to achieving:

1. Low distortion with a very low order of overall feedback
2. Wide open-loop bandwidth with an excellent slewing rate
3. Minimum time and phase distortion
4. A high transient power capability with virtual freedom from clipping effects.

The output stages have a very high current capability but have no protection circuits. The output transistors being designed to sink the full energy of the power-supply into the load. A patented form of voltage feed to this stage gives the smplifier a short term power delivery capability of about 600 Watts. (Compared to the rated 150 Watts 8 Ohms). This, of course, represents a 6dB increase in power availability over the rated figure. The voltage amplifying stages are designed to clip softly and this combined with the low-overall feedback gives the amplifier overload characteristics similar to those of an equivalent valve amplifier.

Conclusion

This article had discussed just some aspects of modern audio amplifier design. Much attention is focused upon whether an amplifier is designed around bipolar transistors, FETs, valves, or switching transistors. Designers are now finding that the major stumbling block is not designing a circuit using any of these technologies but in deciding upon what is the performance specification required *that will give faithful reproduction of the sound source.* Until this problem is solved there will continue to be an element of uncertainty in amplifier design. ETI

The Mission Amplifier referred to in this article is due for release very soon now, and we will be taking a closer and more detailed look at this design — results as soon as possible in ETI.

EDITORIAL ENQUIRIES

Due entirely to our own vacillation, some confusion has arisen as to how we reply to our readers enquiries, be they postal or telephonic! So to put things straight:

Postal Enquiries

These are, of necessity, answered by use of a standard letter. Replies can take up to two weeks. We can only deal with questions which refer to articles within ETI, and cannot undertake design work on behalf of our readers. Please enclose an SAE.

Telephone

Technical enquiries can ONLY be answered on the telephone on a Monday afternoon between 2.30 pm and 5.30 pm. This is to allow us the rest of the week to turn out the magazine!

In addition any queries relating to current projects will usually be answered by listening to our recorded 'ETI News'. Just dial 01-434 1781 after 6.00 pm any evening — any points to note will be found there.

THE S

THE FIRST FREE-FLIGHT test of the Space Shuttle, watched by 60,000 people marked the commencement of the final phase of months of testing at Edwards Air Force Base in the Mojave Desert of Southern California. At the end of January 1977 the Shuttle was moved from its assembly facility at Palmdale, along 58 km of specially widened roads to Edwards AFB, for the first Approach and Landing Test (ALT)

The Space Shuttle is the first of a new breed of spacecraft which is designed to be reused. Previously, the technology available meant that each spacecraft could be used only once, but for any long-term program of space research this is extremely wasteful. Everything was built to the highest standards and then used only once. The Space Shuttle changes this. The Space Shuttle Orbiter vehicle is designed to land intact in the same manner as an aircraft, and the solid rocket boosters used to provide the enormous thrust at takeoff are also reusable. In fact, a Space Shuttle can be launched as quickly as 160 hours after landing from the previous mission, although a two-week ground turnaround is the goal in actual use.

Up Up And . . .

The Shuttle is launched vertically, attached to an external tank which contains the ascent fuel burnt by the Orbiter's main engines, and two solid rocket boosters. At lift-off all the engines fire in parallel, the SRB's each generating 11,800,000 Newtons of thrust and the three Orbiter engines each generating 2 100,000 N. The two SRB's are jettisoned once they burn out and are recovered after a parachute descent. The external tank is jettisoned before the Orbiter attains orbit

The orbital manoeuvring system is used to make any adjustments to the orbit or any manoeuvres that may be

ROCKWELL SPACE SHUTTLE

Hidden in the wake of the Apollo's the Shuttle Orbiter heralds the beginnimg of a new age — the age of the true spacecraft. With its initial testing completed, we take a detailed look at the first (proven) spaceship to land on Earth!

required during the mission. The jets for this system are mounted near the nose and in pods on the upper rear of the fuselage. These jets can pitch, roll or yaw the Orbiter.

The Orbiter is designed to carry a crew of seven (early missions call for four), including scientific and technical personnel and a payload up to 18 m long and 5 m in diameter. Because of the low g forces at launch, only 3g and less than 1.5g on re-entry, space flight is no longer limited to intensively physically trained astronauts — now experienced scientists and technicians can have access to zero g, vacuum conditions.

Payloads up to 29,500 kg can be placed into orbit. These can range from small satellites to fully equipped scientific laboratories, and not only can the Space Shuttle launch payloads into orbit, it can also retrieve and return them, and service or refurbish satellites in space. The versatility of the Shuttle's cargo opens up whole new areas, i.e. space manufacturing.

Down

Upon completion of the various mission duties, the crew will prepare the Orbiter for re-entry — this is when the Space Shuttle really flies. The Orbiter, since it moves in the two media of air and vacuum, has two separate manoeuvring systems. One is the orbital manoeuvring system referred to above, and the other is a set of aerodynamic control surfaces that act in much the same way as conventional aircraft.

There are seven aerodynamic control surfaces on the ▶

news digest.......

viewdata... prestel...

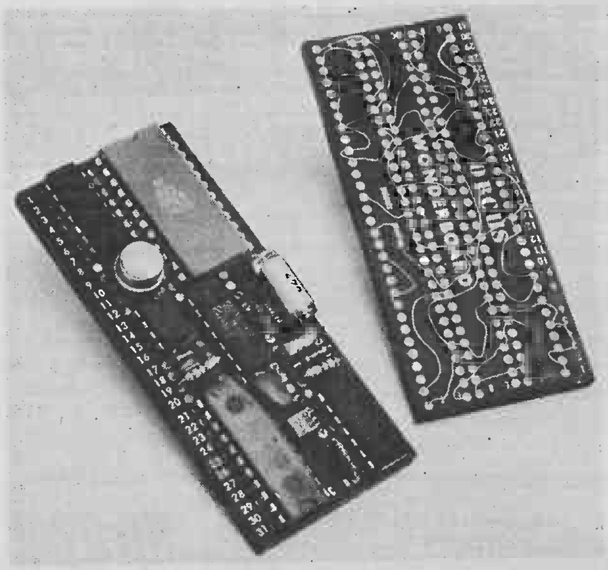
THE Post Office seems to be having a lot more success outside the U.K. with Viewdata/Prestel than it is having at home. As well as the negotiations with A.T. & T in the States the P.O. has sold the Hong Kong Telephone Company the know how to enable it to set up a system. Part of the sales pitch involved making a portable system (it weighed 56 kilos in a rather large case) and taking it to Hong Kong — a successful 'round the world' link was set up via satellite and undersea cable to the P.O. research station at Martlesham.

Meanwhile back at the ranch, the ambitious plans for a U.K. network have suffered some rather embarassing setbacks — the expected 1,500 sets by the end of 78 has been revised to 10,000, and at present only about 100 are installed. Of the presently installed sets the vast majority are with information providers, not customers. Also the department that specified the electronic design parameters forgot to check with the department that certifies all equipment fit to be connected to the P.O. system. The result was that all the sets have had to be modified in case they tried to send nasty kilovolts down the line.

As well as the mechanical hitch the computer data banks are still not quite ready, all this means that instead of marketing trials the P.O. will have a basic 'test service' until the real public service starts — no definate date has been set for this yet though.

Finally, the reason for a sudden change from Viewdata to Prestel as a name has been discovered. The P.O. application to register Viewdata as a trade mark has been rejected by the trade mark office, the word Prestel has been submitted as an alternative name — but even this has not been accepted, yet. Informed opinion has it that Prestel will also be rejected, as an Italian company has used it since 1968 in the U.K. Any suggestions for a third alternative should be sent to . . . ?

french connection



Wonderboards are a new bread boarding aid manufactured by Orcus International. Unlike normal solderless bread boards, which use metal sockets, the Wonderboards use conductive elastomeric contacts to provide the means for inter-connecting all the components. A benefit of this technique is that connections can be made to both sides of the board, giving far denser layouts than possible with conventional bread boards. Contact resistance is 10 milliohms and insulation resistance 10,000 megohms between contacts.

Two sizes are available — Small Wonder (81x35x4 mm) and Big Wonder (81x140x4 mm) and naturally enough the contacts are on a 0.1 inch matrix to accommodate DIL packages. They are made in France and are available in the U.K. from Charcroft Electronics Ltd., Charcroft House, Sturmer, Haverhill, Suffolk, CB9 7XR. Price of Small Wonder is £2.80, and Big Wonder is £11.20 inclusive.

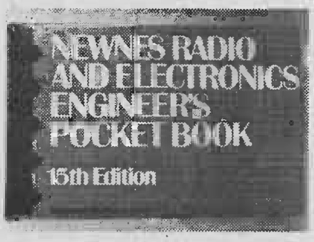
close encounters

Is it a bird? Is it a plane? No, it's a smartie? Everyone seems to like thinking up new acronyms, SMARTIE stands for Submarine Automatic Remote Television Inspection Equipment — probably thought up by a Mr S. Alik! Smartie is a microcomputer controlled submersible for use in the North Sea, to investigate the murky depths around oil platforms and conduct general surveys.

Equipped with multiple TV cameras, the device uses a submersible pump instead of a propellor to move around. Benefits brought by MPUing include a simple hold command, which tells Smartie to stay where it is — with automatic compensation for water currents. Unlike conventional submersibles Smartie has a very thin (5mm diameter) umbilical cord — previous units have used bulky multicore cables.

Smartie has been developed by Marine Unit Technology Ltd, with the support of the Department of Energy via the offshore Energy Technology Board.

pocket size

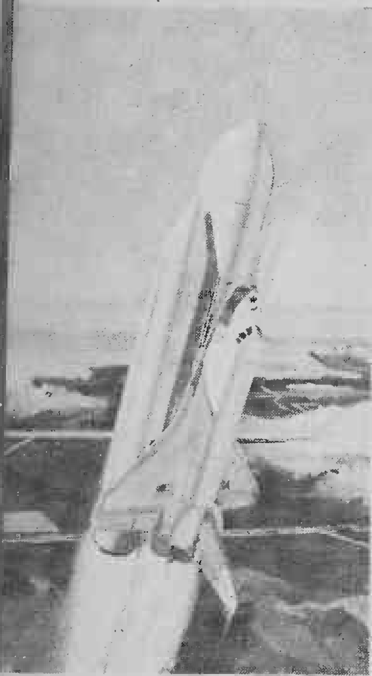


Ever needed to know how to convert furlongs per fortnight into chains per nano second? If you have then you must be a loony! However for the rest of our devoted readers, we would like to recommend the new Radio and Electronic Engineers Pocket Book. Full of useful information from CMOS data to frequency allocations, this the 15th edition has been updated by the editorial team that put the fun into electronics (you guessed, the ETI staff). We don't get commission and we still think you should buy a copy, so it must be good! Most decent (and some indecent) book shops should stock it, so keep your eyes out and have a look when you get a chance.

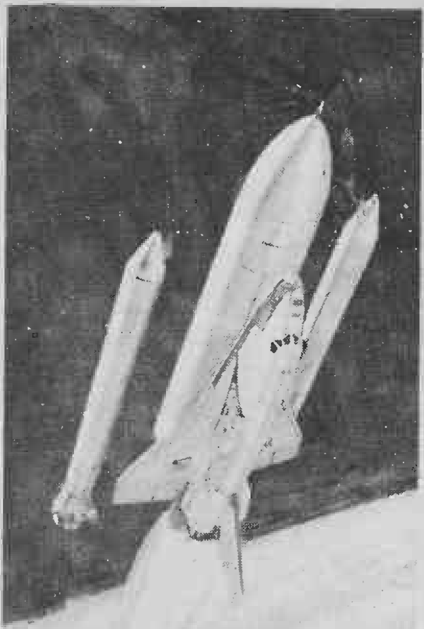
buzz buzz

A new range of solid state buzzers are available from FieldTech Limited. A minimum output of 65dB (at 3 feet) is buzzed by the 1V5 and 3VO versions while the 6, 9, 12 and 24V versions give a beefier buzz of 70dB. Each device incorporates a silicon transistor oscillator, with no mechanical bits to arc or fall apart. Further details from FieldTech Ltd, Components Division, London (Heathrow) Airport, Hounslow, Middlesex.

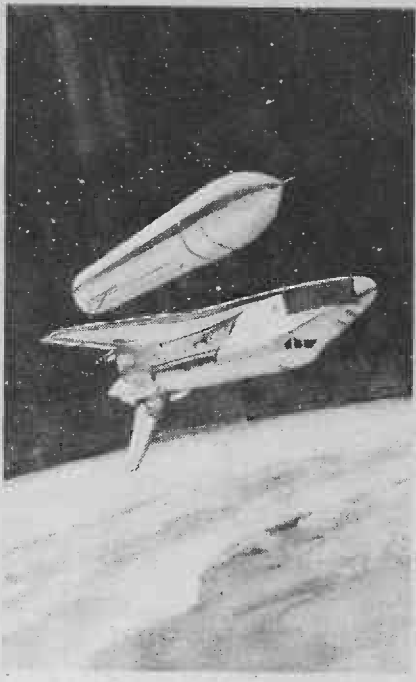
HOW IT WORKS



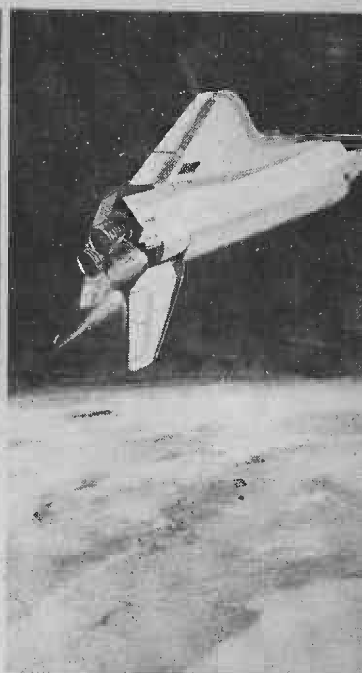
LAUNCH



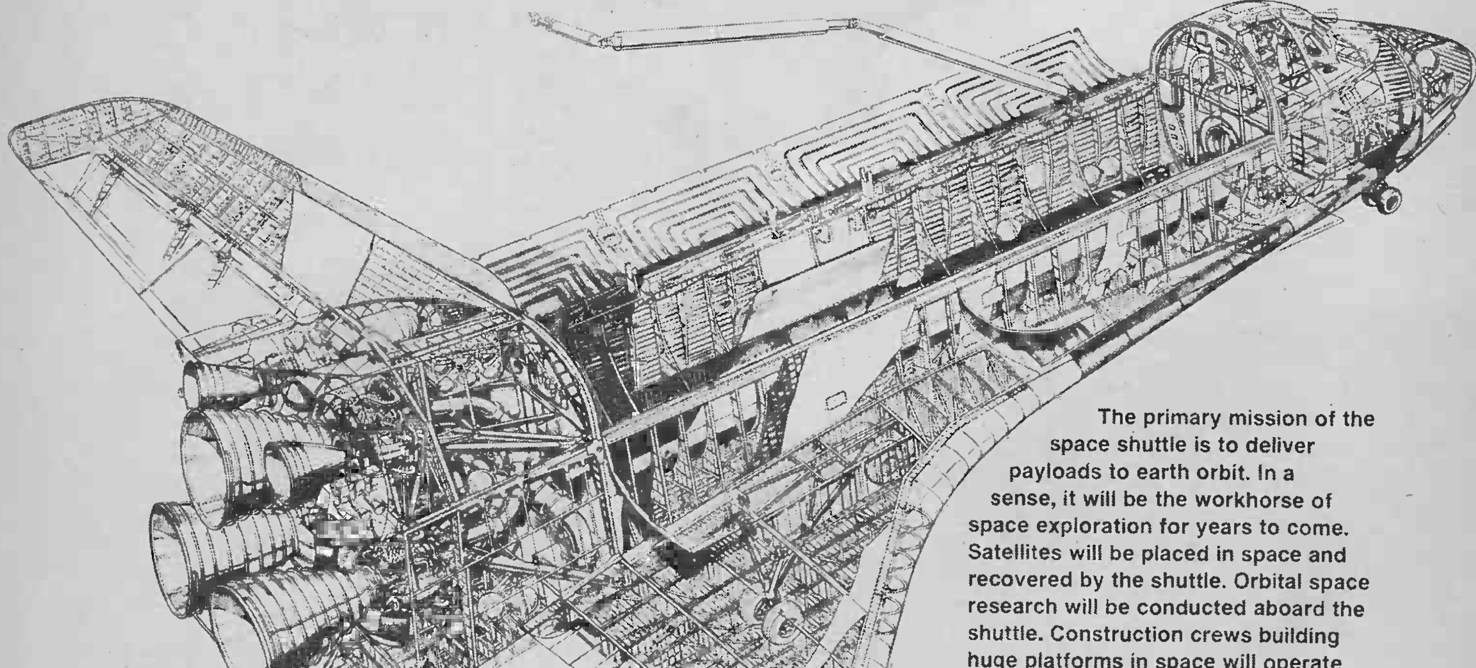
SEPARATION OF SOLID-ROCKET BOOSTERS
HEIGHT: 46 kilometers (28 miles)
VELOCITY: 5008 km/hr (3112 mph)



SEPARATION OF EXTERNAL TANK

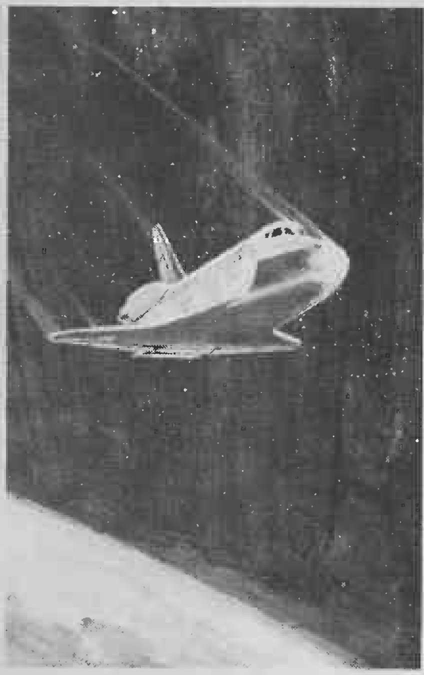


ORBIT INSERTION AND CIRCULARIZATION
HEIGHT: 185 kilometers (115 miles, typical)
VELOCITY: 28,300 km/hr (17,600 mph)



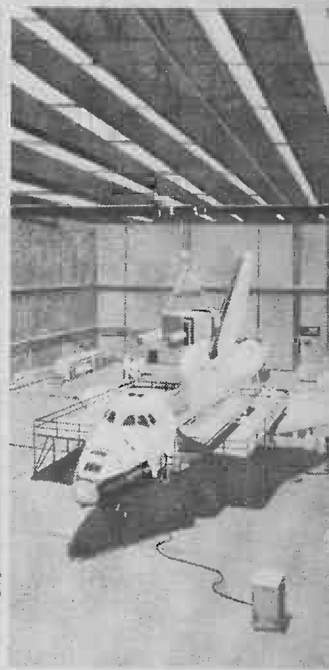
The primary mission of the space shuttle is to deliver payloads to earth orbit. In a sense, it will be the workhorse of space exploration for years to come. Satellites will be placed in space and recovered by the shuttle. Orbital space research will be conducted aboard the shuttle. Construction crews building huge platforms in space will operate from this unusual orbital craft. Astronomers will greatly expand their knowledge of the heavens with telescopes borne aloft by the shuttle. In time, the shuttle may be flying earthmen to space factories and even space cities.

ORBITAL OPERATIONS
HEIGHT: 161-966 kilometers (100-600 miles)
DURATION: 7-30 days

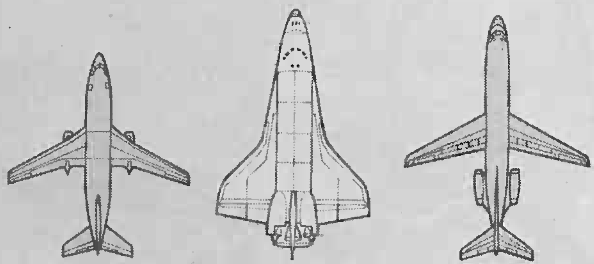


ATMOSPHERIC ENTRY
HEIGHT: 122 kilometers (76 miles)
VELOCITY: 26,765 km/hr (16,633 mph)

LANDING
CROSSRANGE: ±2011 kilometers (±1250 miles) (from entry path)
VELOCITY: 335 km/hr (208 mph)



SERVICING FOR RELAUNCH



Orbiter. Four of these are on the rear of the wings and are called 'elevons' — they combine the effects of elevators and ailerons. The fifth surface is at the bottom rear of the fuselage between the wings, and assists the elevons in controlling the pitch of the craft. It also protects the rocket engine nozzles from buffeting in the airstream during re-entry. The two remaining panels are on the rear of the vertical tail and can be used as a rudder or spread apart to form a 'speedbrake' by increasing the drag. This is used to limit the airspeed during landing.

At low speeds these surfaces act in a conventional manner. However, at supersonic speeds above Mach 1.5, the effect of some of the control surfaces is reversed, or not the expected one, which makes flying in a conventional manner impossible! To get round this problem, the Space Shuttle, unlike most aircraft, which use mechanical or hydraulic links between pilot and controls, uses a digital 'fly-by-wire' Flight Control System. This is based on three on-board IBM System/4 Pi AP-101 computers which monitor their own operation to provide a measure of fail-safe redundancy.

SPECIFICATION

LENGTH
SYSTEM: 56.1 meters (184 feet)
ORBITER: 37.1 meters (122 feet)

HEIGHT
SYSTEM: 23.1 meters (76 feet)
ORBITER: 17.4 meters (57 feet)

WINGSPAN
ORBITER: 23.8 meters (78 feet)

WEIGHT
GROSS LIFT-OFF:
1.99 million kilograms (4.4 million pounds)
ORBITER LANDING:
84.8 thousand kilograms (187 thousand pounds)

THRUST
SOLID-ROCKET BOOSTERS (2):
11.6 million newtons (2.6 million pounds) of thrust each
ORBITER MAIN ENGINES (3):
2.1 million newtons (470 thousand pounds) of thrust each

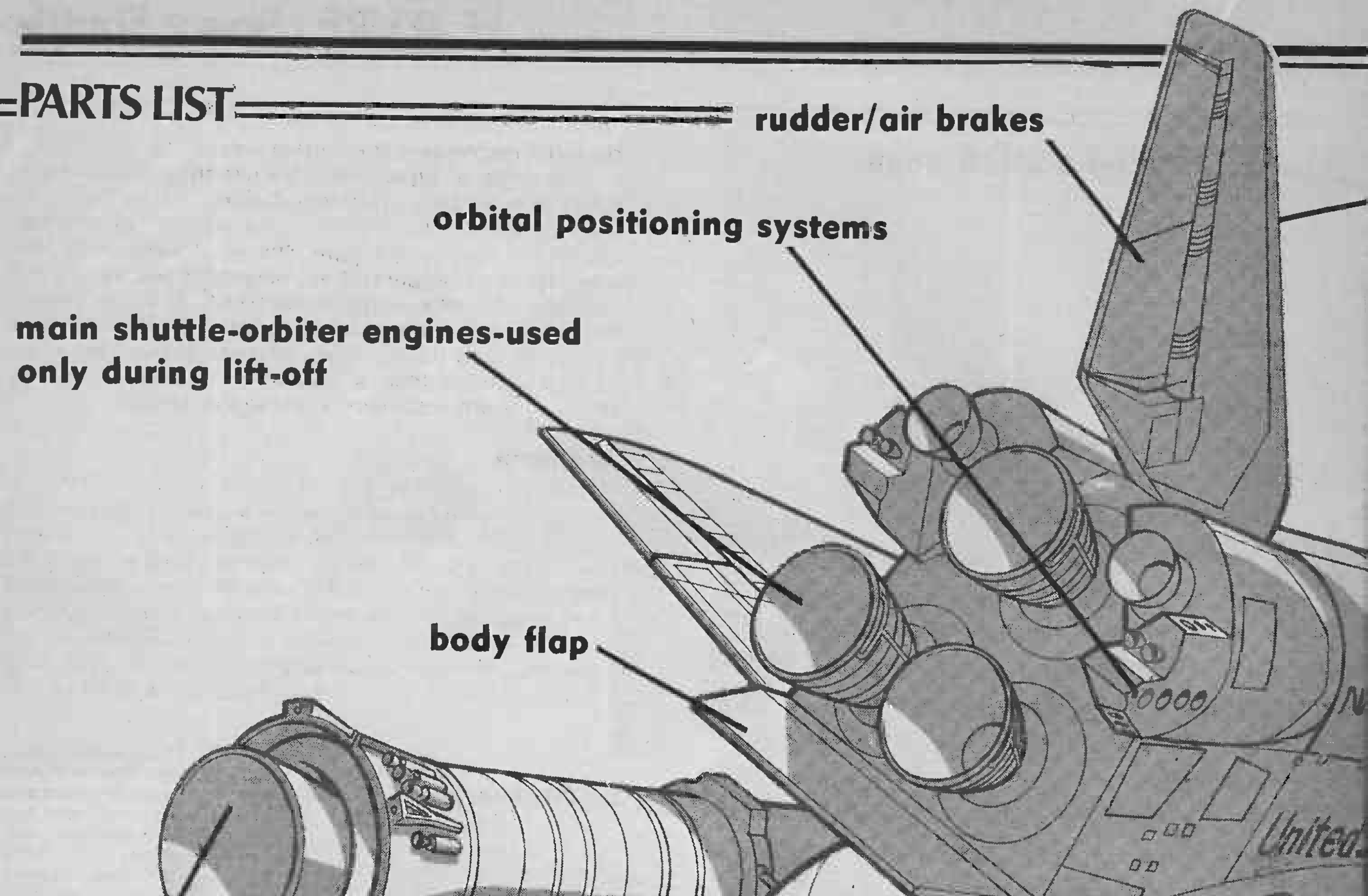
CARGO BAY
DIMENSIONS:
18.3 meters (60 feet) long, 4.6 meters (15 feet) in diameter
ACCOMMODATIONS:
Unmanned spacecraft to fully equipped scientific laboratories

Flight Modes

The Flight Control System (FCS) can be operated in three modes: Direct (DIR), Control Stick Steering (CSS) and AUTO. The mode can be selected separately for pitch, roll/yaw, speedbrake and body flap controls.

In DIR mode, the pilot grips a small stick called the Rotational Hand Controller and ordinary pedals. Movements of these inputs to the FCS produce movements of the control surfaces in the same way as a conventional ▶

PARTS LIST



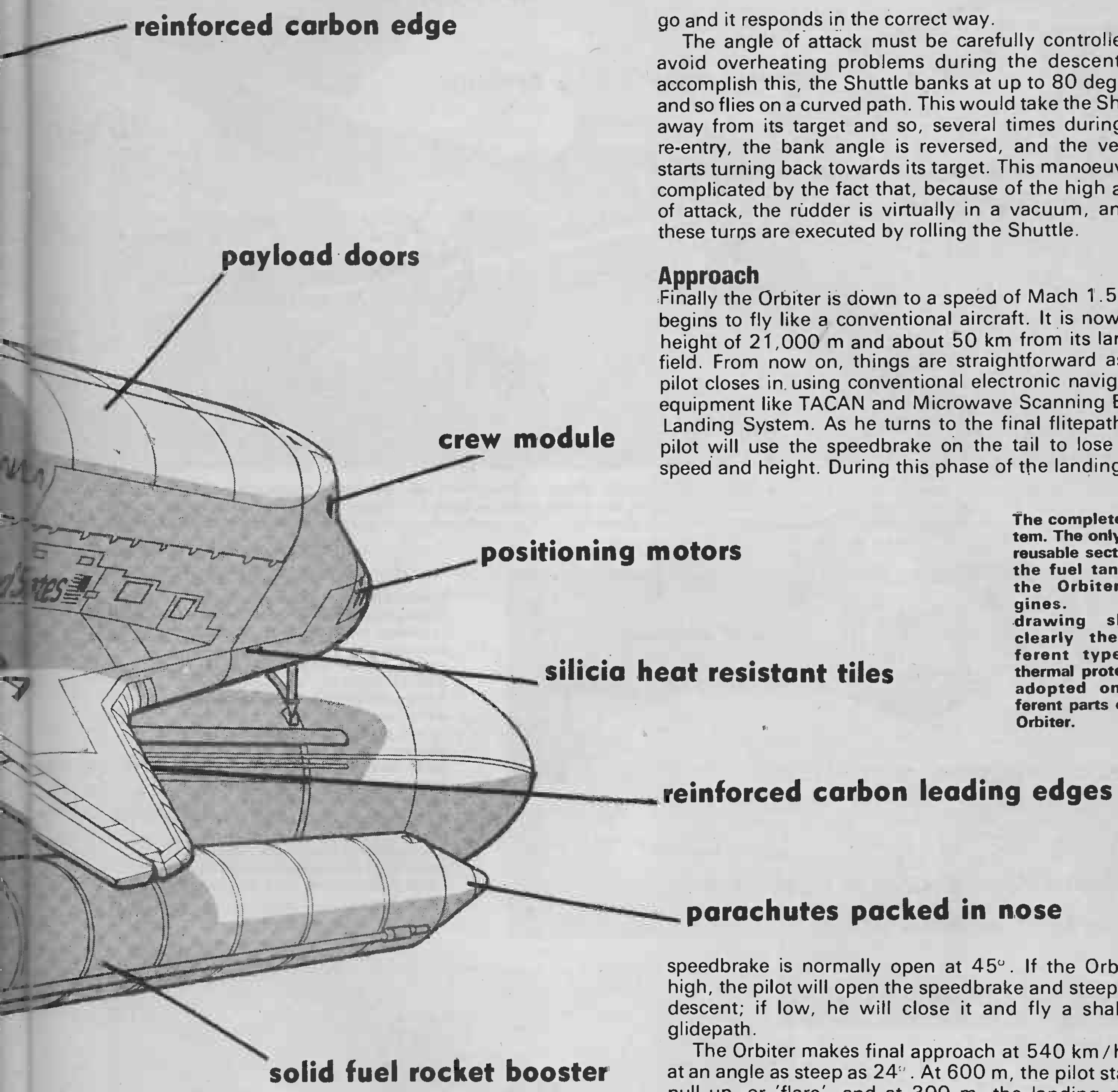
aircraft would respond — at subsonic speeds, at least. Above Mach 1.5 things go haywire — the result is like trying to ride a bicycle with your hands crossed, and only with considerable training can the pilot avoid making involuntary, incorrect movements of the RHC stick.

In the AUTO mode, the FCS takes inputs from star sensors, inertial measurements units, rate gyros, accelerometers, and air data sensors, and compares this with the desired trajectory, automatically making corrections to keep on the path. In fact, the Orbiter can land itself from orbit completely automatically, with the only pilot intervention required being landing gear extension and operating the brakes on the runway!

In the CSS mode, the Flight Control System interprets between the pilot and the control surfaces. The pilot uses the Rotational Hand Controller and pedals, but the FCS accepts these inputs as rate commands in pitch, roll or yaw — in other words, the way the pilot wants the Orbiter to move. These commands are compared with inputs from the rate gyros and accelerometers, and generates control signals to implement the pilot's commands. In this mode the FCS automatically takes account of the reverse effects produced by the aerodynamic surfaces at high airspeeds.

Re-entry

The Orbiter starts re-entry at a high angle of attack, around 30 to 40 degrees, so that the bottom of the wing and fuselage are exposed to the airstream. The under surface is covered with a high-temperature structure of reinforced carbon-carbon on the leading edges and



The complete system. The only non-reusable section is the fuel tank for the Orbiter engines. This drawing shows clearly the different types of thermal protection adopted on different parts of the Orbiter.

special silica tiles over most of the other surfaces to maintain the airframe within acceptable temperature limits.

Unfortunately, because of the high angle of attack, moving the RHC to the left in the DIR mode causes the Orbiter to roll to the right. This is because the right elevon is deflected downward, but this causes drag, and turns the vehicle to the right. This increases the lift on the left wing, so it lifts, causing the right roll. In the Control Stick Steering mode, though, this problem is taken care of by the Flight Control System, and the pilot simply moves the stick the way he wants the vehicle to go and it responds in the correct way.

The angle of attack must be carefully controlled to avoid overheating problems during the descent. To accomplish this, the Shuttle banks at up to 80 degrees, and so flies on a curved path. This would take the Shuttle away from its target and so, several times during the re-entry, the bank angle is reversed, and the vehicle starts turning back towards its target. This manoeuvre is complicated by the fact that, because of the high angle of attack, the rudder is virtually in a vacuum, and so these turns are executed by rolling the Shuttle.

Approach

Finally the Orbiter is down to a speed of Mach 1.5, and begins to fly like a conventional aircraft. It is now at a height of 21,000 m and about 50 km from its landing field. From now on, things are straightforward as the pilot closes in using conventional electronic navigation equipment like TACAN and Microwave Scanning Beam Landing System. As he turns to the final flitepath, the pilot will use the speedbrake on the tail to lose both speed and height. During this phase of the landing, the speedbrake is normally open at 45°. If the Orbiter is high, the pilot will open the speedbrake and steepen his descent; if low, he will close it and fly a shallower glidepath.

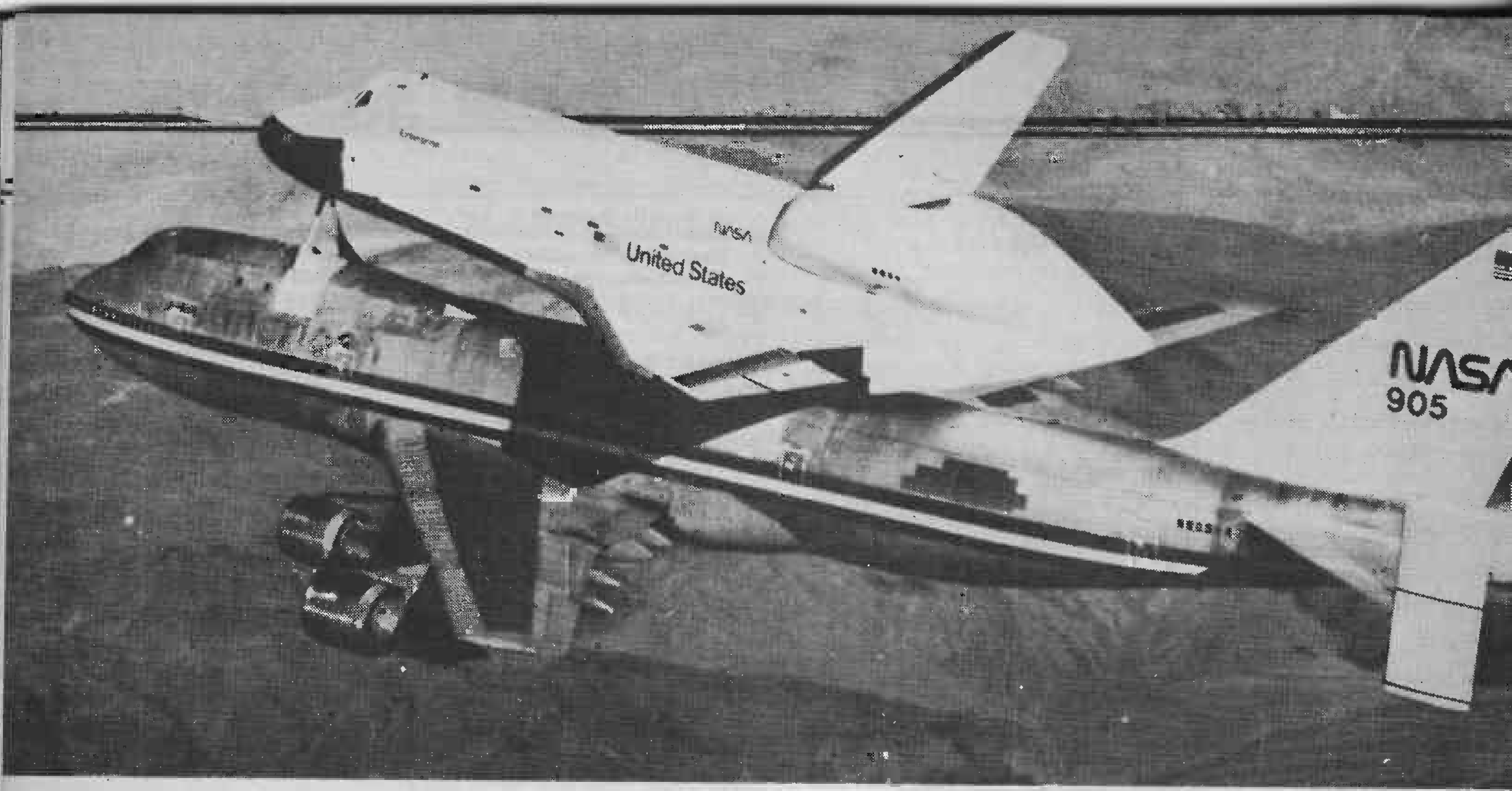
The Orbiter makes final approach at 540 km/hr and at an angle as steep as 24°. At 600 m, the pilot starts to pull up, or 'flare', and at 300 m, the landing gear is dropped. The vehicle touches down at 350 km/hr; at this point it is losing 9 km/hr of speed every second and stalls at 280 km/hr, which is why the land is at such high speed.

The Approach and Landing Test were designed to check out the performance of the Shuttle during this phase of the mission. They were also designed to check the performance of that now-famous 747/Space Shuttle combination which will continue to fly, delivering Orbiters to the launch site from the production line and landing sites.

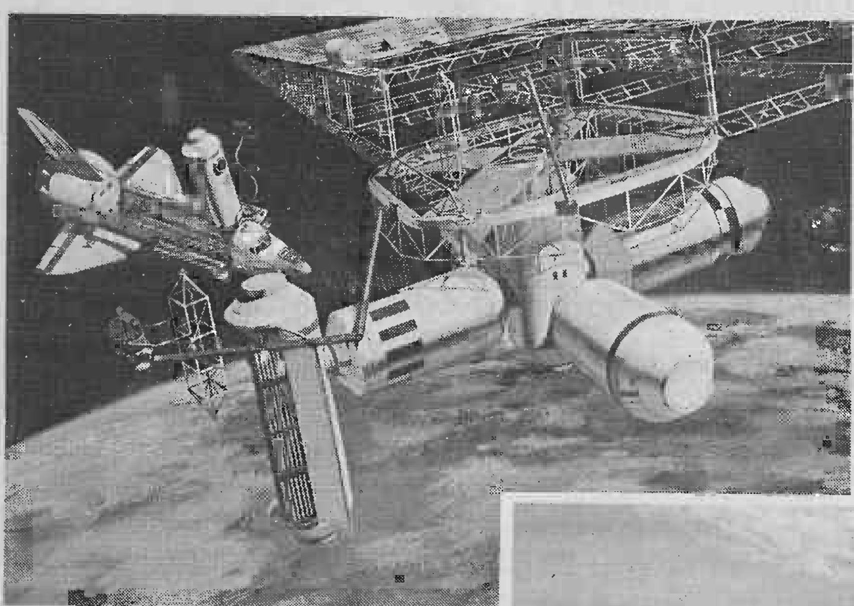
First Flights

The first flight of the Space Shuttle took place on 12th August last year. At 8 AM, the 747 Shuttle Carrier

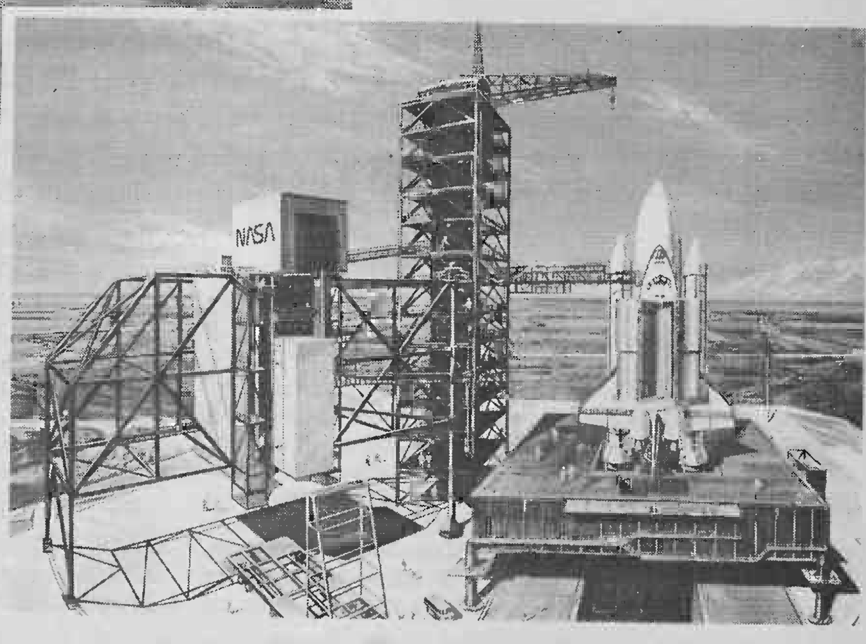
▶



Space Shuttle Orbiter 101 rides "piggyback" atop NASA's 747 Carrier Aircraft in the first series of captive approach and landing tests conducted at NASA's Dryden Flight Research Center at Edwards Air Force Base in California. With the Orbiter unmanned and its systems inactive, the highly successful first tests verified the safe operation of the combined vehicle configuration, Photo was taken at about 16,000 ft. above the California desert.



Space Shuttle can deliver both the materials and the machinery required to build large space structures, such as this demonstration satellite solar power station. After being fabricated and assembled in low earth orbit, a power station would be transferred to its permanent place in geosynchronous orbit (about 22,000 miles out in space). There it would beam a continuous stream of microwave energy to earth receivers, which would convert the energy to electricity. When completed the station would be 1000 feet square and 25 feet thick.



The Shuttle orbiter cargo bay — which is larger (60 by 15 feet) than most freight cars — will accommodate a great variety of payload combinations. Payloads can be installed or removed while the orbiter is either horizontal or in the vertical position on the launch pad, as shown here, which greatly enhances operational flexibility. The payload "changeout" room is located in the white structure on the left.

Aircraft with its piggyback Orbiter took off on time — the only problem had been a fault in one of the AP101 computers, but that unit was quickly replaced.

At 8.47 the pair were at 8,539 m, and the Boeing started a 7° dive. At a speed of 280 kts, and a height of 7.346 m, the Boeing pilot informed the Shuttle crew that they were ready for separation. The crew, Haise and Fullerton, fired the separation bolts and lifted away, rolling to the left while the 747 dropped to the right. Following a pair of right and left rolls to put some distance between the two craft, Haise tried a practice flare and some banking manoeuvres. This gave the computers at Johnson Space Centre the opportunity to calculate any deviation from the predicted lift/drag ratios, information which would allow a more accurate landing. In fact, the JSC ground controllers muffed it by assuming that the Orbiter was in level flight, whereas it was actually climbing, so they concluded that the lift/drag ratio was lower than predicted.

Haise could not open the speedbrake beyond 45°; this was a mission constraint to avoid steep glideslope angles. Performing a flare at 270m, Haise touched down 600m beyond the expected touch down point at a speed just over 360 km/hr. The overshoot was no problem, as runway 17 at Edwards AFB is 11 km long, but with the wheels on the ground, Haise opened the speedbrake to 90° and the nose wheel came down. The flight had lasted just 5 min 23 sec.

The first three flights were made with a streamlined tail fairing covering the dummy rocket engines at the tail. The fourth flight, on 12th October, was made with the fairing removed, giving a slightly reduced lift/drag ratio. Otherwise, the vechicle did not behave significantly differently.

Next Comes Nothing

With all the approach and landing tests completed, the Shuttle programme moves into its next phase which takes it into space. In the middle of 1979 the Orbiter will be lifted from Cape Kennedy for its first real flight. At present the projected date is sometime in June, but this may well change.

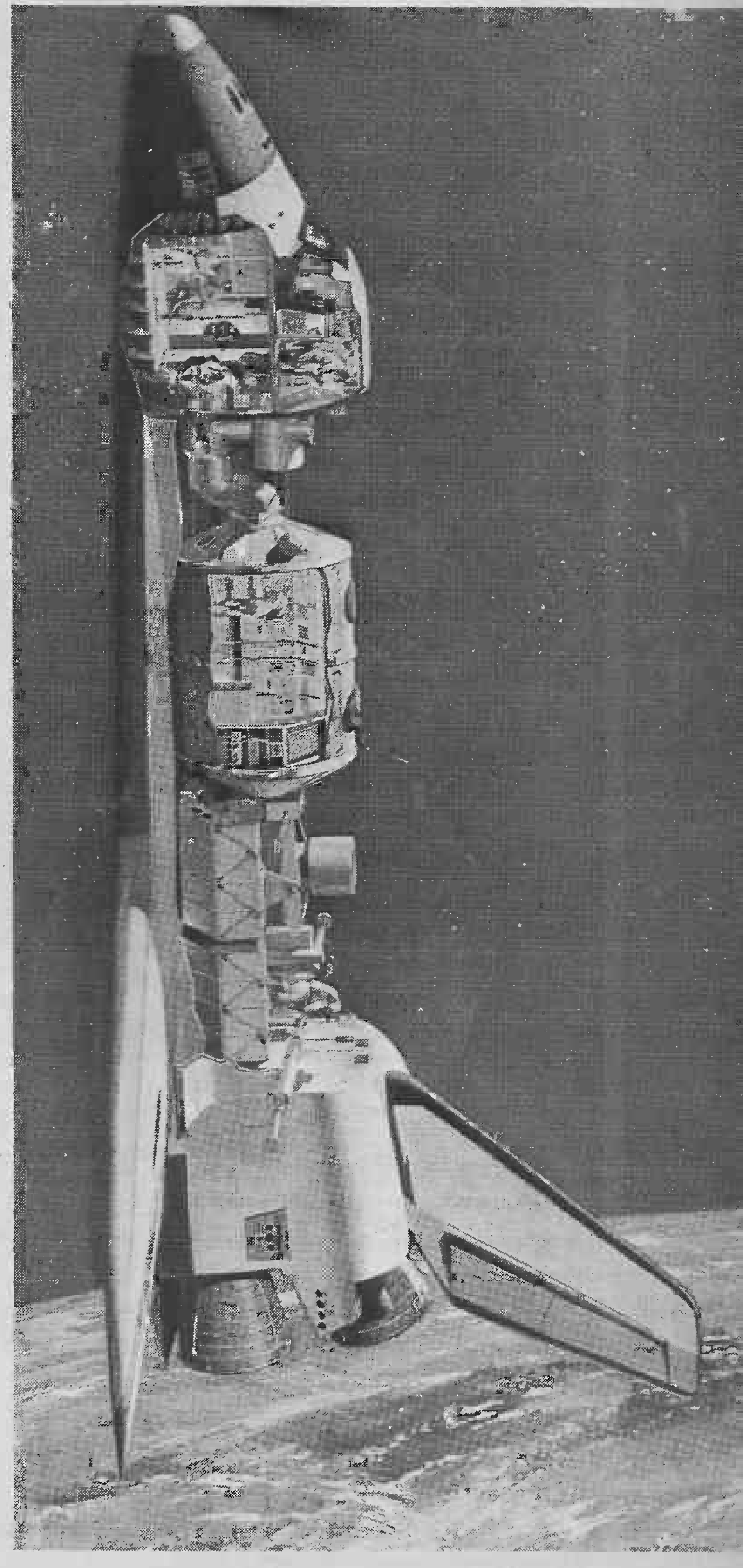
Rockwell are already selling space in the cargo bays — and doing very well too. One of the first payloads will be the Euro Space Lab, which will use the Orbiter's ability to stay put in space for up to a month or more. Cargos are being accepted from commercial firms too — so if you fancy sending a package into space this is your chance. Move quickly though because space in space(!) is harder to get than Star Wars tickets and bookings stretch out a few years into the future.

Hopeful Sign

Of course the Shuttle gives us the capability to build space stations at last, with all that implies — solar power, weather control, observatories and starships. It may be a long time before Man does reach for the stars, but at least we've taken the first step. ETI

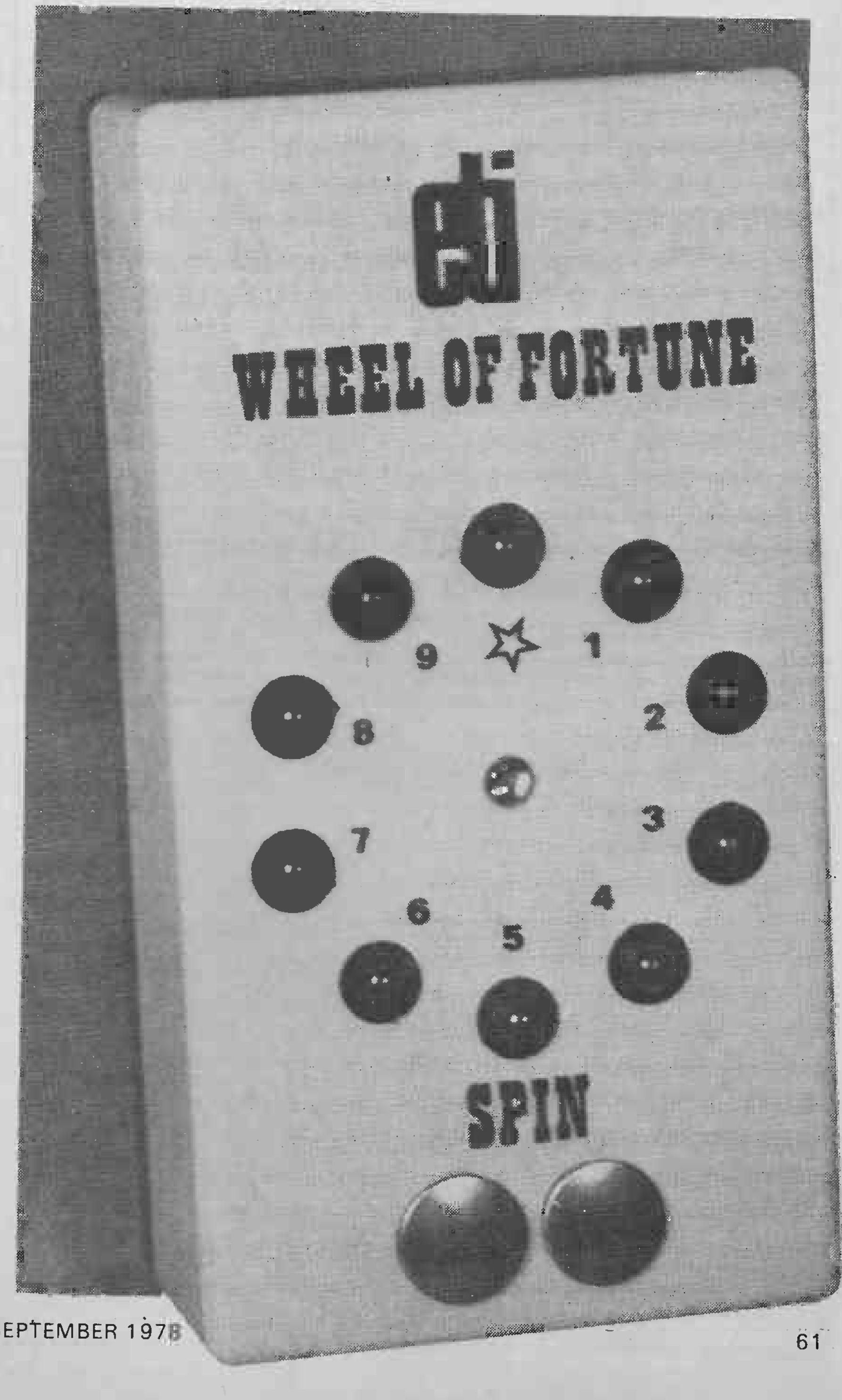
Our thanks to Rockwell International — Space Division — for their assistance in the preparation of this article.

A key Shuttle payload is Spacelab, center, a multipurpose laboratory that will enable scientists to conduct experiments in the gravity-free environment of space. The lab is being produced by the European Space Agency (ESA), a consortium of European nations, in cooperation with the National Aeronautics and Space Administration.



WHEEL OF FORTUNE

ETI's project team is in a real spin this month with their Wheel of Fortune game.



ONE ARMED BANDITS with no arms, Pinball tables with an MPU at their centre — the world of electronics has a lot to answer for. Is nothing sacred?

The answer to that last question as far as we at ETI are concerned is not a lot. We've taken the liberty of implementing that traditional fairground attraction, the Wheel Of Fortune in our own electronic fashion. The game usually features a large wooden wheel and ratchet arrangement, the stall either accepting bets on which of the ten numbers will be under the pointer when the wheel stops, or, perhaps, suggesting that a message under the pointer will give an indication of what the future holds in store for you — you will meet a tall dark stranger, you will marry young and have 2.4 mortgages, etc.

Will O Fortune

Our game accurately apes the real thing, the circle of LEDs simulating the spin of the Wheel getting under way as a pair of touch contacts are crossed with you palm (or more likely finger). The movement of the LEDs will then slow down to, it seems, an excrutiatingly slow speed until it finally stops. All this visual activity is at the same time accompanied by a clicking sound that simulates the ratchet sound of the real game.

Wheel Meet Again

It's easy to become a trifle blase about electrical games, particularly in the face of the never ending stream of things that we see in the shops at present, but even the most hardened people, and we've got some fairly hardened people here at ETI, found ▶

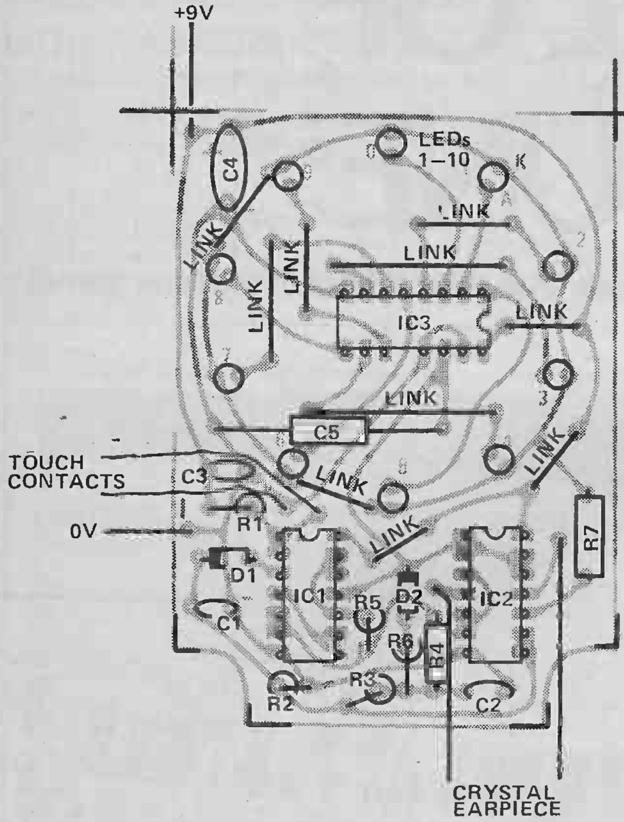
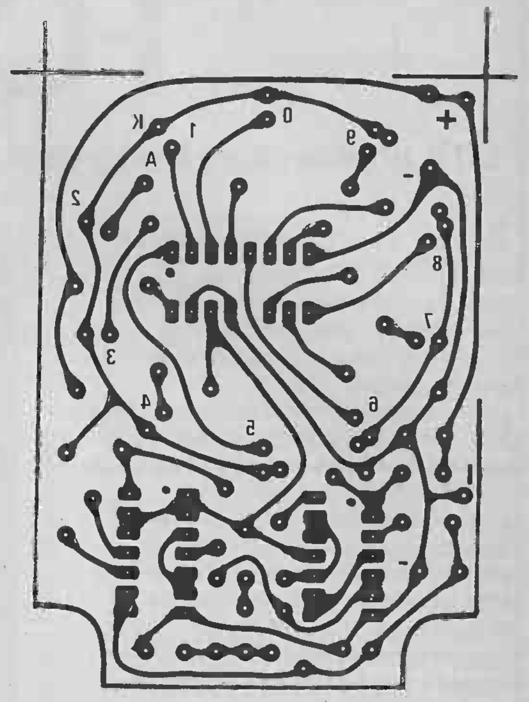


Fig.1. (left) shows the overlay for the Wheel of Fortune game while Fig.2. (right) is the full size foil pattern of the game's PCB.



PARTS LIST

RESISTORS (all ¼W 10%)	
R1	2M2
R2	1M0
R3	100k
R4	470k
R5	4M7
R6	10k
R7	330R

CAPACITORS	
C1	100u 10 V tantalum
C2	1u0 10 V tantalum
C3	22u 10 V tantalum
C4	100n polyester
C5	1n0 polyester

SEMICONDUCTORS	
IC1,2	4011B
IC3	4017B
D1,2	1N914
LED1-10	TIL209

MISCELLANEOUS
Battery, crystal earpiece, drawing pins, vero box, PCB as pattern

BUYLINES

None of the components used in the Wheel of Fortune game should prove hard to find as most will be stock items in many component shops. Make sure that the tantalum capacitors specified for C1, 2 and 3 are used as the circuit makes use of the low leakage characteristics of these components.

the Wheel of Fortune to be fun. If you start thinking about building it now it might just get finished for Christmas.

Construction

Start by mounting all the components on the PCB with the exception of the LEDs. Pay attention to the orientation of the polarity sensitive devices and, for choice, mount the ICs in holders. In order to squeeze everything into the small box we used, the PCB tracks have been made quite fine so be careful when soldering that no excessive amounts of heat are applied to any sections of the board.

As can be seen from the internal photograph of the game, the back of the crystal earpiece is removed before mounting the device in the case. This is to ensure adequate room between the IC and earpiece. The touch contacts formed by two drawing pins are glued to the front panel. When the case has been prepared place, but do not solder the LEDs, into the PCB and offer them up to the case. Solder one lead of each LED. At this stage make sure that all the devices are properly seated, then solder the second lead.

That just about completes the construction, just connect up to a battery and place your bets. ETI

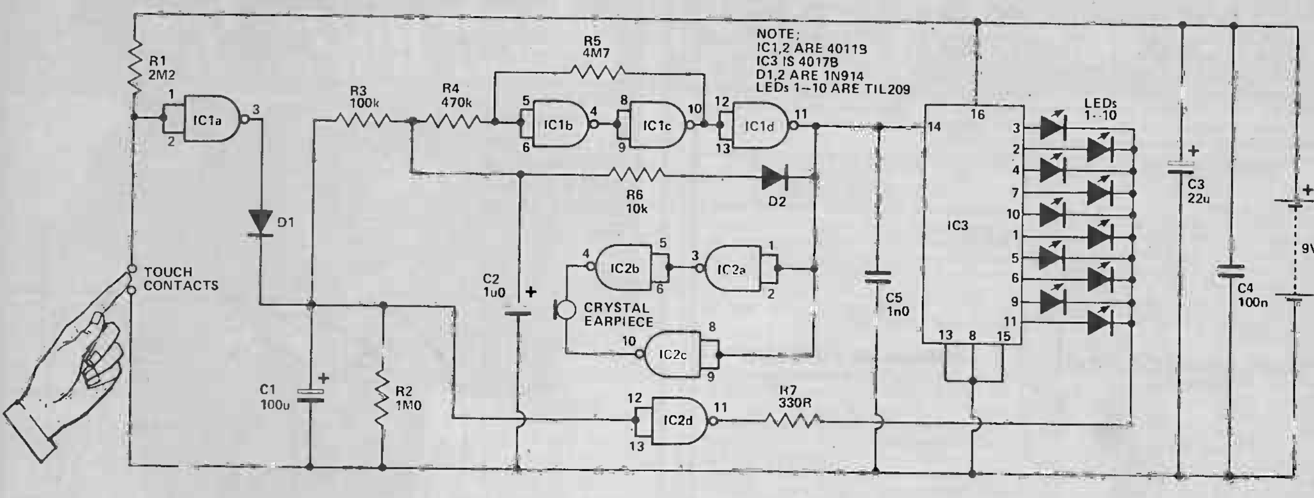


Fig.3. Full circuit diagram of the Wheel of Fortune game.

HOW IT WORKS

THE Wheel of Fortune circuit can be broken down into a number of distinct sections; the display circuitry, an audio stage, a VCO, and a touch sensitive/monostable configuration.

In the "off" state R1 holds the input of IC1a high and hence the output of this gate, wired as an inverter, is low and C1 is discharged. Bridging the touch contacts causes the gate's output to go high and C1 to be charged up via D1. When the finger is removed from the touch contacts and the output of IC1a returns low, C1 is prevented from discharging into this gate as D1 is now reverse biased, instead C1 discharges slowly via R2.

The VCO is formed by the components associated with IC1b, c and d. The circuit in fact generates a series of constant duration negative going pulses separated by "spaces" whose duration can be varied by the control voltage.

When the control voltage (the voltage on C1) is below a threshold level that is equal to half supply voltage the circuit will not oscillate. If we now assume that the voltage on C1 rises to supply, as would be the case when the touch contacts are bridged, C2 will start to charge up. The voltage avross C2 is applied, via R4, to the schmitt trigger formed by IC1a and b. As the voltage applied to the schmitt crosses its upper switching threshold the output of IC1d, which inverts and buffers the schmitt's output, will go low. This will cause C2 to be rapidly discharged via the relatively low impedance path offered by R6 and D2. As the voltage on C2 crosses the lower threshold of the schmitt the output of IC1d returns high and C2 once more begins to charge. The time taken for the voltage on C2 to reach the schmitt's trigger point is dependent on the voltage across C1. Thus when the voltage on C1 is large, C2 quickly reaches the trigger point and the VCO produces a high frequency, this requency reducing as the voltage of C1 falls.

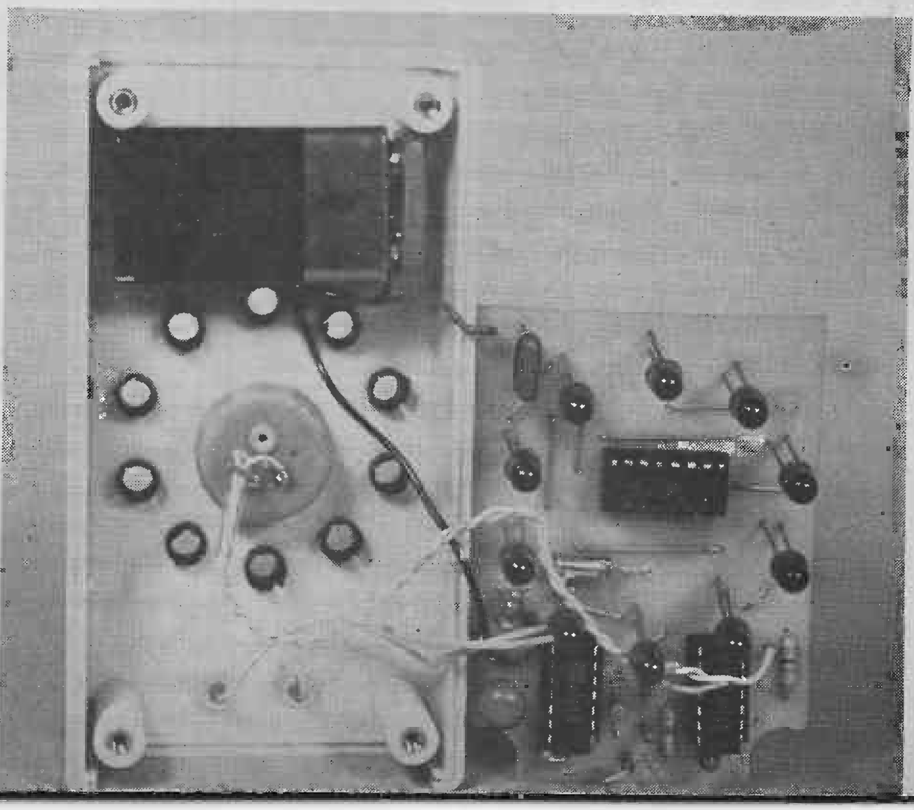
The output from the VCO is fed both to IC3 to drive the ring of LEDs and to IC2a, b and c to produce the audio output.

The crystal earpiece that provides the "clicking" is driven from a bridge circuit. This effectively doubles the voltage applied to the transducer and hence, from $P=V^2/R$, doubles the audio output.

The LEDs driven by IC3 have their cathodes connected via R7, to the output of IC2d. The output of this gate will normally be high, going low when the voltage on C1 is above half supply. As IC3 outputs are active high the display is thus enabled for a period of time that is slightly longer than the duration of the VCO's oscillation.

C3 and C4 are included to decouple the supply while C5 is needed to prevent any RF interference affecting the circuit's operation.

Photograph of the game's inards. Note that the back of the crystal earpiece has been removed to ensure sufficient clearance between it the IC directly below when the box is assembled. The drawing pins that form the game's touch contacts are glued to the front panel with an epoxy adeshive, the tips of the pins can be seen at the bottom of the picture.



data sheet

MM57160 STANDARD TIMER AND CONTROLLER (STAC) NATIONAL

The standard timer and controller chip is a preprogrammed member of National's Controller Oriented Processor (COP) family. The device is designed for use in repetitive timing application where 1 to 4 outputs are to operate at 4 user-programmed times. Minimal external hardware is needed for complete system implementation due to direct display drive capability and a key-switch interrogation feature. Strap selection for 50/60 Hz input and 7-day/8-day mode has been included for added versatility.

Initialization

Power for the device is a single power supply of 7V9 to 9V5. Proper initialization will occur internally if the supply rise time is between 11 µs and 1 ms. If the supply rise time to final value exceeds 1 ms, an external RC network with a time constant in excess of the supply turn-on time should be placed on the Power On Reset (POR) pin. This delays initialization until the power supply voltage is within specifications. Initialised conditions are: (a) time (real-time clock) at 00:00, (b) all set point times to 00:00 and all outputs off, (c) all days valid, (d) present day counter to day 1, and, (e) real-time clock mode.

Setting the time is performed in the normal real-time clock mode by depressing the SET HOURS (10) or SET MINUTES (9) keys. Each depression will cause an increment of the hours from 0-23 or minutes from 0-59, respectively, holding the appropriate key depressed will cause the numbers to roll (slew) at a 4/second rate. Normal operation is to slew the value close to the desired setting and then "bump" it to the final value.

OPTION SELECTION

Strap switches can be used to implement key functions. Figure 1 illustrates "strapping" of keyswitch functions 1-5.

Features

- 24-hour real-time clock with 4-digit display
- 60 Hz (50 Hz option) timing derived from the power line
- 4 Control outputs at each set point time
- 4 set point times may be programmed with repeat every 24 hours
- Valid day programming to "skip" certain days
- Manual mode to verify programming
- Transducer input to force to a preset condition
- Time of day reset to ease time setting or to allow use as a sequence timer
- High speed "demonstration" mode for verification of capability
- Single 9V power supply

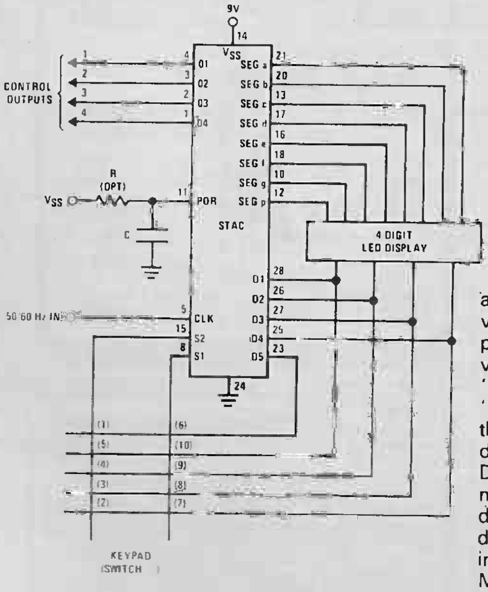


FIGURE 1. Typical STAC Connection

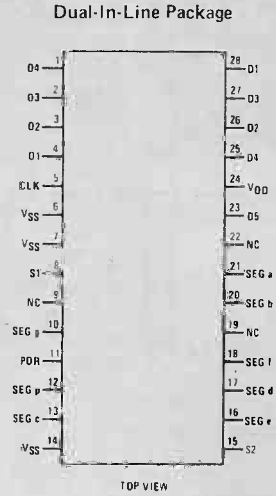


FIGURE 2: Pinouts

Programming

For proper operation, the system must have 1 or more of its set point times loaded. To load (or program) set points, the DATA ENTRY key (5) must be depressed momentarily to take the system from the normal real-time clock mode to the data entry mode. Upon activation, 1 of the set point times will be displayed and its output status will be shown on the decimal points of the display. After power-up, this will be 00:00 and the decimal points will be off. To examine or go to another set point, the ADVANCE SET POINT key (6) is depressed in the data entry mode for each new time. The 4 values are held in a revovling stack (similar to a calculating stack) and each advance causes it to roll 1 position. Four advances returns to the original position.

To activate a set point, the hours and minutes will be loaded with the same SET HOURS (10) and SET MINUTES (9) keys used in setting the real-time clock. In addition the SET STATUS (8) key is activated and is used to load the output(s) to be activated at the programmed time. Depresssion of the SET STATUS key causes the 1st decimal point to turn on (which will correspond to output 1 turning on at run time). If this output is the only one to be used at this programmed time, one can go to the next set point by using the ADVANCE SET POINT key. If, however, the desired output is to be either output 2, 3 or 4, the set status key should be pressed again to advance to number 2, 3 or 4. Each advance turns off the previous decimal point.

If a *combination* of outputs is designed (such as numbers 2 and 4), the HOLD STATUS key (2) is used to hold the number 2 decimal point on *before* the SET STATUS key advances through 3 to number 4. With the use of the HOLD STATUS key and the SET STATUS key, any combination of the 4 outputs can be programmed at each set point. If an error in programming occurs, using the SET STATUS key from position 4 will clear all data (including that set by the HOLD STATUS) and the proper information may be re-entered by following the proper sequence.

If conditions permit the programming can be verified on the actual outputs by using the MANUAL key (1). This key, when depressed in the data entry mode, transfers the decimal point set-status data to the output latches; thus, the motor, solenoid, valve, or whatever is being controlled will be activated. When all 4 times and their respective output conditions have been programmed, the system is returned to the real-time clock mode by another depression of the DATA ENTRY key. If the valid day information is not used, the system is ready to operate.

Depression of the DAY MODE key (7) enables setting and display of the current and valid day information. The current day is displayed in the left-ost digit of the display and the validity of the day in the right-most digit with a "1" for a valid day, and "0" for an invalid "off" day. As the clock steps through the week, the programmed conditions occur on all valid days and do not occur on invalid days. The SET DAY key (10), when depressed in the day mode, advances to the next day upon each depression. The SET STATUS key (8), in the day mode, is used to change the validity information. Another depression of the DAY MODE key will return the system to the real-time clock mode.

Closure of the HOLD STATUS/DEMO key (2) will provide a means to rapidly cycle through the programmed sequence or set up an "in store" display. With this key closed in the real-time clock mode, time is advanced at the rate of 1 hour per second; thus, a 24-hour day requires 24 seconds to verify and a 7-day week requires less than 3 minutes.

Closing key 6 during the real-time clock mode (either normal or demo operation) will reset the clock time to zero without changing the set point timing but will reset the valid day information.

External Inputs

The MANUAL / REMOTE TRANSDUCER key (1), when depressed in the real-time clock mode, will override any time-related programming and immediately force output 1 on and 2 through 4 off. This condition will remain until the next valid set point occurs.

Using It

A table of key functions and an example program are given on the next page, the permutations are endless! ▶

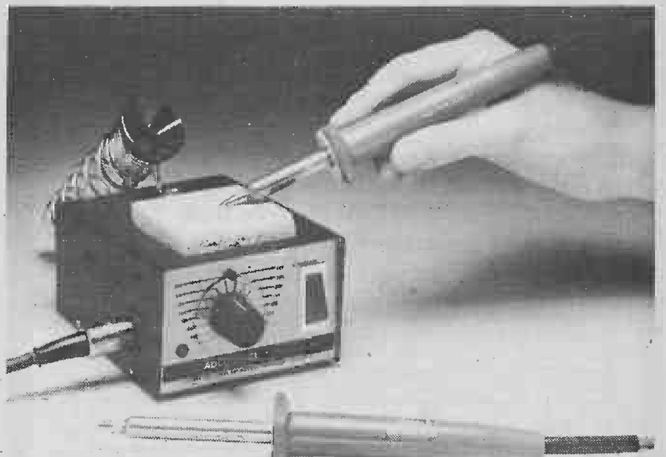
Further details available from **National Semiconductor Ltd., 19 Goldington Road, Bedford, MK4 03LF.**

news... ...digest

distorted truth

In the July Oscillators article we mentioned the Intersil 8038 function generator IC — in fact we said that distortion changes with frequency, and frequency is not a linear function of control voltage. Both statements are only true under certain conditions. Jayen Developments have pointed out that within the audio range both distortion and deviation from linearity are negligible (<0.1%) the device only goes haywire above approximately 100kHz and below about 20Hz. As we said in our July 1977 Data Sheet on the 8038, it is an inherently versatile device with some drawbacks – but overall it has a lot going for it!

sawn off



Adcola have gone and cut 22mm off the length of their 101 temperature controlled soldering iron, leaving it with a barrel only 45mm short. The new model (101TS) is also lighter than its brother (sister?) by 16 per cent at 42gms. The idea behind the amputation is to give more precise control of the hot end — needed with modern components, which can be easily damaged by excess heat.

The temperature control is provided by a thermocouple feeding an op amp and special power control c, which uses the zero crossing technique to eliminate RF interference. Control is within 2% of the set temperature as shown on the control unit/stand dial. Full details and spec sheets from Adcola Products Ltd, Adcola House, Gauden Road, London SW4 6LH.

boris slain

Regular readers (aren't you all!) will have seen the item in last months News Digest about Boris the chess machine. Fidelity Electronics who make the Challenger felt that Boris's challenge should be taken up, and arranged a seven game tournament at the recent Chicago Electronics Show.

Boris was set on 3 minute response time and the Challenger set at a similar level. The result was Boris 0 Challenger 7, a veritable wipeout! The average response time of the Challenger was only 2 minutes 15 seconds. The game of the century would be to pit Boris and Challenger 10 against each other on their largest response times (99 hours and 24 hours respectively) – but a game like that could well take so long it would be the game of next century!

MM57160 STANDARD TIMER AND CONTROLLER (STAC) — NATIONAL

KEY NO.	KEY SWITCH NAME	FUNCTION		
		REAL-TIME CLOCK MODE	DATA ENTRY MODE	DAY MODE
1	MANUAL/ REMOTE TRANSDUCER	Remote transducer input; forces output 1 ON, outputs 2–4 OFF until next valid set point after switch is off	Manual verification mode; allows data to be transferred to outputs 1–4	(None)
2	HOLD STATUS/ DEMO	Allows rapid demonstration of sequence by advancing clock at rate of 1 hr/sec	Holds output N ON while programming advances to output N+1, N = 1–4	(None)
3	8 DAY	Specifies 8-day cycle in lieu of 7-day	Specifies 8-day cycle in lieu of 7-day	Specifies 8-day cycle in lieu of 7-day
4	50 Hz	Specifies 50 Hz line frequency input	Specifies 50 Hz line frequency input	Specifies 50 Hz line frequency input
5	DATA ENTRY	Places unit in the data entry mode	Returns unit to the real-time clock mode	(None)
6	ADVANCE SET-POINT/ RESET TIME	Resets time of day to 00:00 without changing set points but resets all days to valid	Advances display to the next set point so that it may be verified or altered	(None)
7	DAY MODE	Places unit in the day mode	(None)	Returns unit to the real-time clock
8	SET STATUS	(None)	Controls programming of outputs; resets output N to "0" (unless preceded by HOLD key) and advances to output N+1	Alternate action key; changes day from valid ("1") to invalid ("0") and vice-versa
9	SET MINUTES	Advances minutes display of real-time clock	Advances minutes display of selected set point	(None)
10	SET HOURS/ SET DAY	Advances hours display of real-time clock	Advances hours display of selected set point	Advances display to next day—must be set to current day before returning to real-time clock mode

Programming Example

1. Output 1 should turn on at 2:00 a.m., and turn off at 4:00 a.m. each valid day.
2. Output 2 should turn off at 2:05 a.m. and turn back on at 4:00 a.m. each valid day.
3. Output 3 should turn on at 2:00 a.m. and turn off at 2:05 a.m. each valid day.
4. Output 4 should turn off at 3:01 a.m. and turn on at 4:00 a.m. each valid day.
5. Monday through Friday are valid days — Saturday and Sunday are invalid.
6. It is now Monday, the time is 1:00 a.m.

Given these conditions, it is now advisable to construct an "output truth table":

TIME/OUTPUT	O1	O2	O3	O4
2:00 AM	ON	ON	ON	ON
2:05 AM	ON	OFF	OFF	ON
3:01 AM	ON	OFF	OFF	OFF
4:00 AM	OFF	ON	OFF	ON

The following key sequence may be used to load the preceding program into the STAC memory.

KEY DEPRESSED	DISPLAY	NOTES
	0000	Initial display
Data Entry	0000	
Set Hours	0100	
Set Hours	0200	
Set Status	0.200	Set point 1 at 2:00 a.m.; output 1 ON
Hold Status	0.200	Hold output 1 ON
Set Status	0 2.00	Output 2 ON
Hold Status	0.2.00	Hold output 2 ON
Set Status	0.2.0.0	Output 2 ON output 3 ON
Hold Status	0.2.0.0	Hold output 3 ON
Set Status	0.2.0.0.	Output 4 ON
Advance Set Point	0000	
Set Hours	0100	
Set Hours	0200	
Set Minutes	0201	
Set Minutes	0202	
Set Minutes	0203	
Set Minutes	0204	
Set Minutes	0205	
Set Status	0.205	Set point 2 at 2:05 a.m.; output 1 ON
Hold Status	0.205	Hold output 1 ON
Set Status	0.2.05	Output 2 ON
Set Status	0.20.5	Output 2 OFF, output 3 ON
Set Status	0.205.	Output 3 OFF, output 4 ON
Advance Set Point	0000	
Set Hours	0100	
Set Hours	0200	
Set Hours	0300	
Set Minutes	0301	
Set Status	0.301	Set point 3 at 3:01 a.m., output 1 ON
Advance Set Point	0000	
Set Hours	0100	
Set Hours	0200	
Set Hours	0300	
Set Hours	0400	
Set Status	0.400	Set point 4 at 4:00 a.m.; output 1 ON
Set Status	04.00	Output 1 OFF, output 2 ON
Hold Status	04.00	Hold output 2 ON
Set Status	04.0.0	Output 2 ON, output 3 OFF
Set Status	04.00.	Output 3 OFF, output 4 ON
Data Entry	0000	Present time
Day Mode	1 1	Day 1, valid
Set Day	2 1	Day 2, valid
Set Day	3 1	Day 3, valid
Set Day	4 1	Day 4, valid
Set Day	5 1	Day 5, valid
Set Day	6 1	Day 6, valid
Set Status	6 0	Day 6, invalid
Set Day	7 1	Day 7, valid
Set Status	7 0	Day 7, invalid
Set Day	1 1	Return to current day
Demo	(Running)	Run thru at least one 24 hour cycle intermittently (use Hour & Minute keys to "nudge" display to set points) to verify output settings. After passing set point just prior to present time, release Demo key
Set Hours	0100	Present time

Programming of the STAC is now complete. The program will continue in 24-hour, 7-day cycle until manually altered.

STAC TIMER

The odds were STACed against ETI's projects team this month, but once again they've come through with the goods

THE NAME OF this project is derived from that given by the manufacturer to the IC around which it is built. STAC stands for Standard Timer And Controller and the device is part of National's COPS (Calculator Orientated Processor System) group, a series of, what are in effect, dedicated microprocessors.

The STAC provides a 24-hour clock with four digit display, much as any clock IC, but has four control outputs which may be programmed to turn on, turn off, or to retain their current status at any one of four preset times during the day. STAC also has the facility to ''skip'' certain selected days within its seven or eight day (selectable) cycle.

The IC is thus a perfect basis for many control applications from central heating installations to fish tanks and hi-fi systems. We will not give details of the interfaces between STAC and the outside world, as with so many potential uses, the circuitry will have to be selected with the particular environment in which you wish to use your STAC in mind.

A STAC In Time

Setting up the STAC is quite straightforward and is rather like using one of the programmable calculators with which many of us are familiar.

At switch on the STAC is initialised to a state where the clock is at 00.00, all set points are zero and outputs off, all days are valid with the present day set to one. The display will show the clock output.

Setting up the clock follows the usual procedure adopted with any digital clock. Pressing the SET HOURS or SET MINUTES will advance the appropriate digits at a rate of four per second.

The next task is to enter the four set points, the times at which the outputs will change and the exact manner in which they will change. To program the STAC it must be taken out of the clock mode and put into the data entry mode by pressing the DATA ENTRY key.

At this stage one of the set point times will be displayed. These values are held in a revolving stack and to examine the next the ADVANCE SET POINT key is pressed, after four ''advances'' the original value is displayed.

Any one of the set point time is set up with the SET HOURS and SET MINUTES keys as with the clock. The conditions that the outputs adopt at the set point are set up with the SET STATUS and HOLD STATUS keys.

Indication of the condition of the four outputs is provided by the decimal points of the display, if the decimal point is on the corresponding output is on the left-hand point represents output one. At power up all decimal points, thus outputs, are off.

Operation of the SET STATUS key will cause the first decimal point to turn on (output one on at run time). Each subsequent operation will cause ▶

the current output to turn off and the next to turn on.

If a combination of outputs is required the HOLD STATUS key may be used to hold the current decimal point on when moving to the next. To do this the key must be operated before SET STATUS is used to advance to the next.

Status Symbol

Operating the manual key while in the data entry mode will cause the decimal point status information to be transferred to the outputs for verification.

When all programing is complete, STAC may be returned to the clock mode by a second operation of the DATA ENTRY key.

While in the data entry mode, the valid days may be set up. The DAY MODE key will cause the current day to be displayed (as a number from one to seven) in the left-most digit of the display. The current status ''1'' for valid, ''0'' for invalid, will be displayed in the right-hand digit.

SET DAY will advance to the next day while SET STATUS will change the validity.

DAY MODE will return the system to the clock display.

If the HOLD STATUS is operated in the clock mode, time is advanced at a rate of one hour per second, this enables program information to be checked.

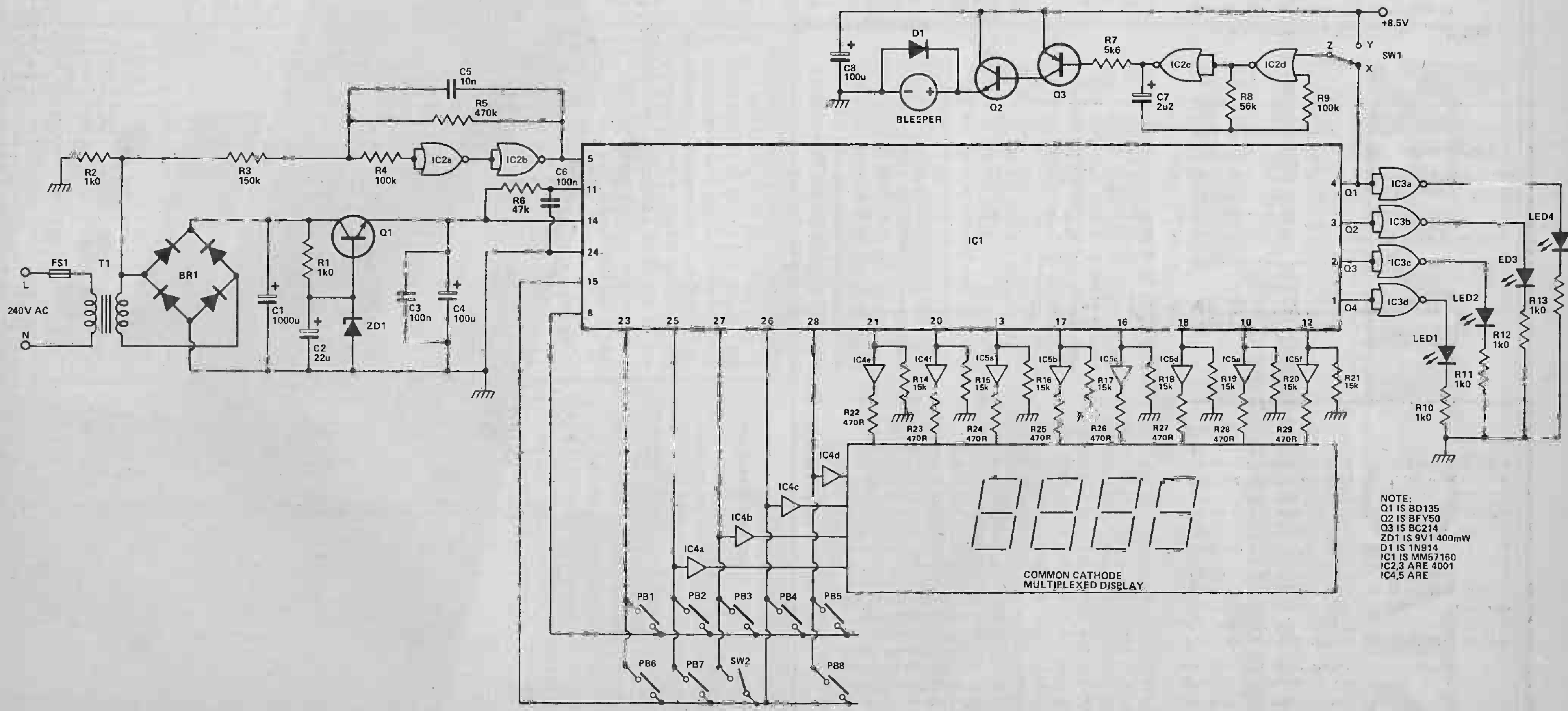
The ADVANCE SET POINT KEY if used in the clock mode will reset the clock but leave set point times unaltered although the day information will be reset.

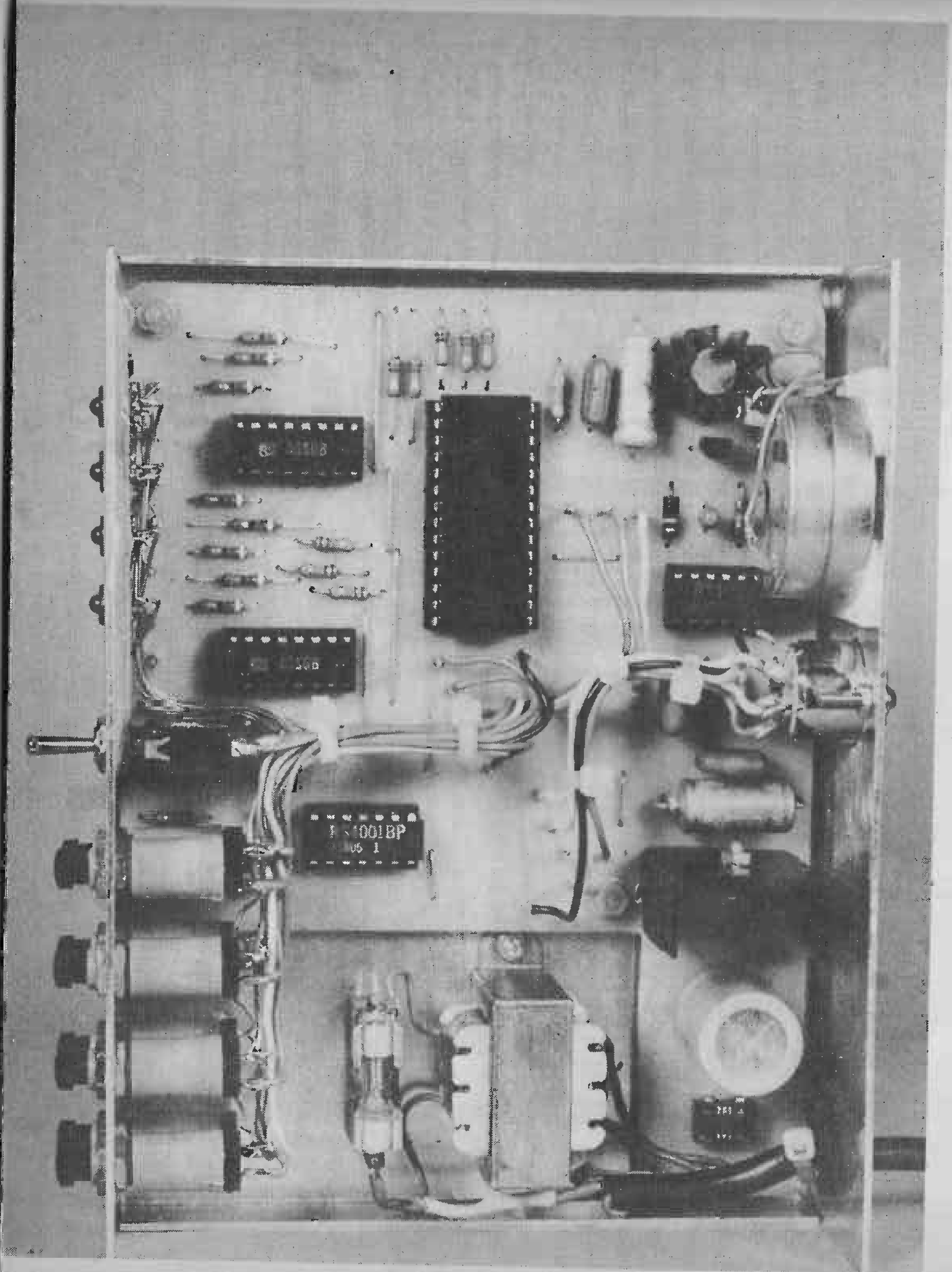
Needle In A STAC

When programming the STAC it is best to draw up a table of set point times and the state of outputs at each of these as an aid to entering the data in a logical fashion.

An example of programming STAC is shown in the ETI data sheet elsewhere in this issue.

Fig. 1. Full circuit diagram of the STAC timer. Resistors R 14-21 are necessary because the segment outputs will not provide logic level swings without pull down resistors. On our prototype, SW2 was replaced by a wire link.





HOW IT WORKS

The power supply for the STAC timer is regulated by A1 after having been smoothed and rectified by C1 and BR1 respectively.

C6 and R6 ensure that the rise time of the voltage on pin 11 is such that proper initiation of the timer takes place.

The unsmoothed output of the transformer is taken to the shaping circuit provided by IC2a and IC2b together with associated components. This acts as both a schmitt, to clean up the wave form, and a monostablee, to ensure that any transients on the mains are not counted by the timer's input circuits.

The operation of the STAC IC is described in the main text, the programming switches referred to being PB1-8 and SW2. The display is driven via the buffers in ICs 4 and 5. Note that R14-21 are required to pull down the segment outputs of the STAC in order to provide a suitable display drive signal.

The outputs of the STAC are active low and drive LEDs 1-4 via the buffer invertors in IC3 to provide an indication that a particular output is 'on'. The invertors ensure that a LED is lit when output is active.

Output 1 can be applied, via SW1, to the astable formed by IC2 c and d. When the output goes low it enables the oscillator which drives the buzzer via Q2 and Q3. The buzzer produces an audible tone when a DC voltage is applied to it. D1 prevents anh back EMF generated by the buzzer causing damage to Q2 or Q3.

BUYLINES

The STAC timer will be available from National Semiconductor suppliers and the rest of the components should be generally available.

In case of difficulty a suitable display can be obtained from Audio Electronics in Edgeware Road for £1.25. They can also supply a suitable buzzer at 25p.

The case can be obtained from Marshall's and Watford, although there are a lot of similar cases around in most loca shops.

Construction

Construction of the STAC timer should not pose any special problems if the overlay shown is followed carefully. Bear in mind, though, that the power supply, due to the size limitation placed upon the transformer by the case used, is run near its maximum rating. This means that the buzzer, which increases the current drawn by the unit from the 45mA with the buzzer inactive but display and LEDs on to 90mA with buzzer active, should only be run for a maximum of about half an hour.

It also means that although the power supply connections are brought out they should only be used, at most, to power an interface circuit that does not draw excessive current from the main unit.

The STAC's outputs are capable of sinking 20mA and if they are to be used to control any devices that require more drive than this, these limitations should be borne in mind and suitable drive circuitry provided.

By the way, if you happen to come up with some ingenious application for your completed STAC timer, perhaps you would let us here at ETI know about them. ▶

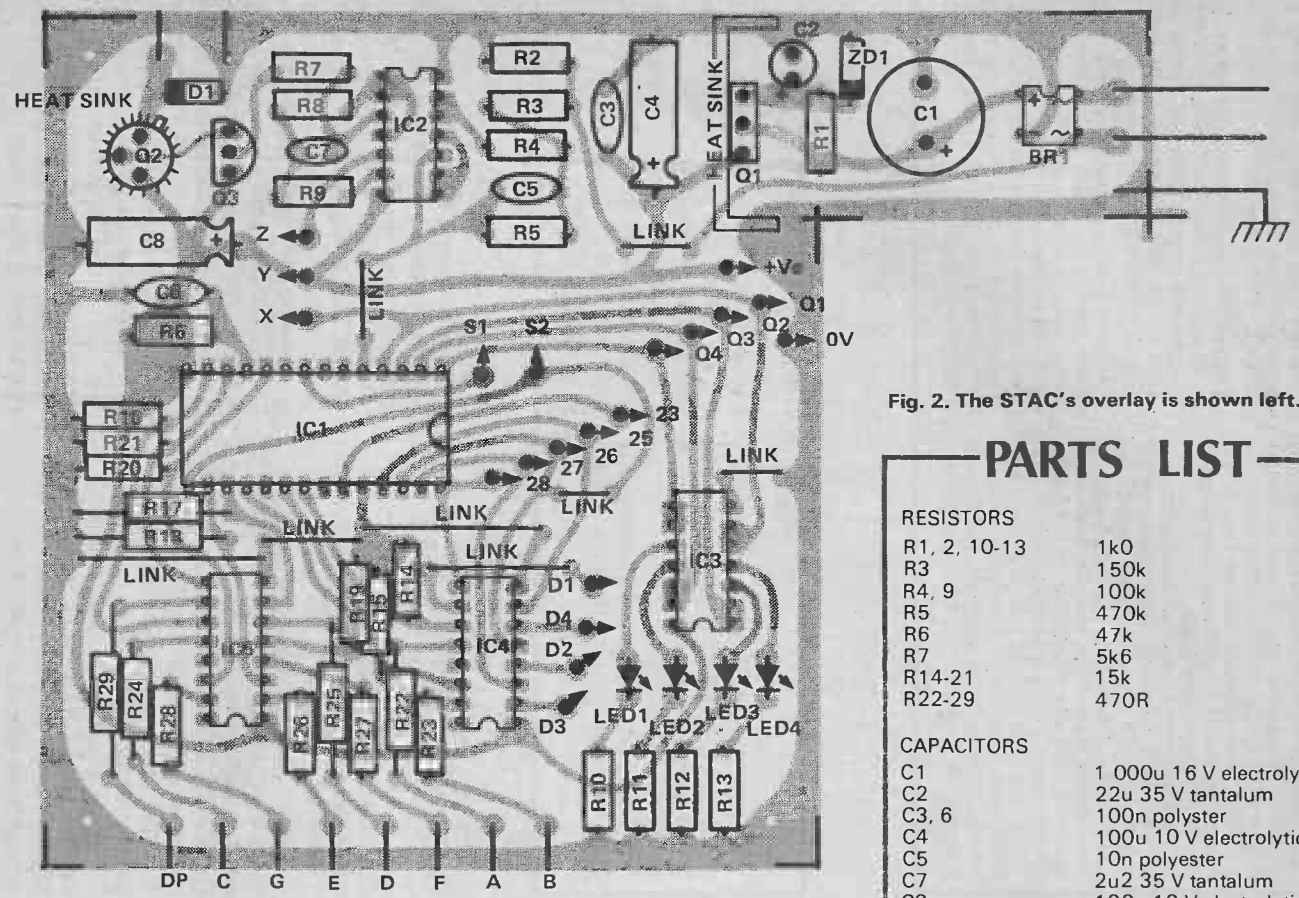


Fig. 2. The STAC's overlay is shown left.

PARTS LIST

RESISTORS

R1, 2, 10-13	1k0
R3	150k
R4, 9	100k
R5	470k
R6	47k
R7	5k6
R14-21	15k
R22-29	470R

CAPACITORS

C1	1 000u 16 V electrolytic
C2	22u 35 V tantalum
C3, 6	100n polyster
C4	100u 10 V electrolytic
C5	10n polyester
C7	2u2 35 V tantalum
C8	100u 10 V electrolytic

SEMICONDUCTORS

IC1	MM57160
IC2, 3	4001B
IC4, 5	4050B
Q1	BD135
Q2	BFY50
Q3	BC214
D1	1N914
ZD1	9V1 400 mW
BR1	0.9A 400 V
LED1-4	TIL 209

SWITCHES

PB1-8	mush to make release to break
SW1	SPDT
SW2	SPST

MISCELLANEOUS

PCB as pattern, four digit common cathode display, 240 V/12 V 50 mA transformer, bleeper, case to suit, display filter, connecting wire, etc.

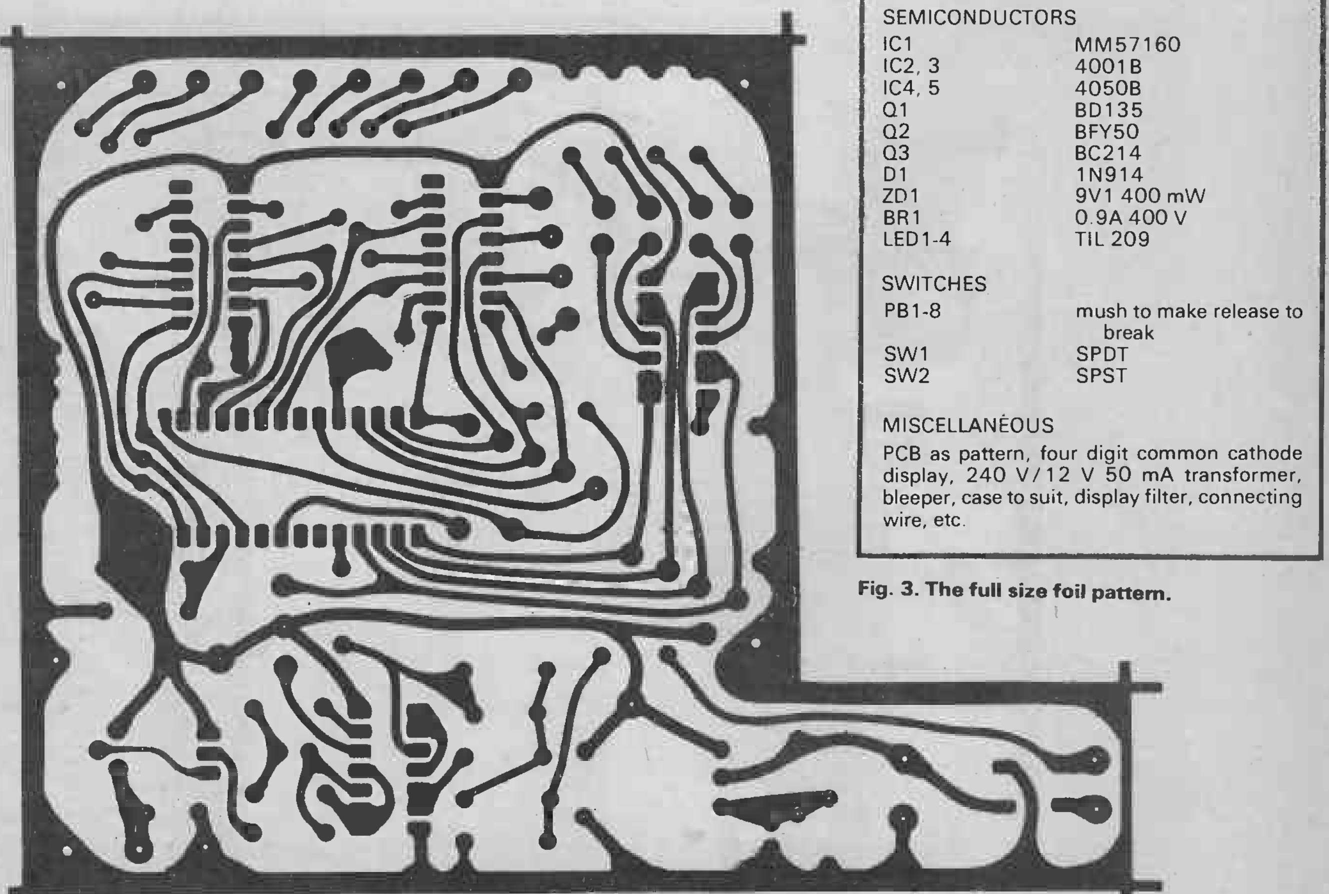


Fig. 3. The full size foil pattern.

ETI

audiophile......

With the vast numbers of amplifiers available today, choosing the one most suitable for your own particular requirements can be a daunting task. Ron Harris explains the steps you can take to make it as easy on the tranquilisers as possible.

AMPLIFIERS are perhaps the most extensively specified hi-fi unit, and whereas this could be a good thing if all the manufacturers agreed which specifications to quote, (and how to quote them) there seems an ever increasing divergence of opinion and technique.

This of course provides the hardened enthusiast with hours of harmless amusement meandering along the twisted webs spun across the ad pages. Great fun to figure out whether the Xplam 500 with its 2.0 MW (UHF) really is more powerful than the Tinne Special at a mere 800.17 W (RMS at 100.3 Hz). Isn't it?

When attempting to select yourself an amplifier, either as a first system or an upward move, there are a few things you can remember to make life easier for yourself.

Watt to do first

Before anything else you need to decide how much power you're going to need. This really depends on how big your listening room is, and how efficient the chosen loudspeakers are at turning electrical power into sound energy.

So, strange as it may seem, the first step in amplifier selection is made with a tape measure — find out the size of the room in which the amp has to work. Ignore protesting females and displaced cats during this operation.

Once you know the volume of the room, a good estimation of how many watts are wanted can be gained by allowing 25 W, good old fashioned RMS watts — but we'll return to that later, for the first 1000 cu. ft. and then 10 W per thousand cubic feet thereafter. For example if your living room is 10ft x 20ft x 8ft = 1600 cu. ft. you need 35 W a channel MINIMUM.

This assumes loudspeakers of average efficiency, always a dangerous thing to do I know, but unless you're using horn-loaded units — in which case you'll have far too *much* power, or transmission lines — for which add 15 W to the estimate, this will generally be O.K. Efficiency varies from manufacturer to manufacturer, with the extremes being represented by the Wharfedale 'E' series at the high end, down to the KEF 104 and IMFs at the other.

A power of good

Let's go back for a minute to the question of how that power rating should be quoted. Perhaps the most meaningful figure is the half-power bandwidth. This tells you the frequency range over which the amp will deliver at least HALF its rated power into a given load.

This is of more use than even an RMS figure, as these are usually quoted at 1 kHz only. For example consider these two units.

AMP A 50 W RMS 1kHz Power Bandwidth 40Hz-10kHz into 8 ohms.
AMP B 40 W RMS, from 20Hz to 20kHz into 8 ohms.

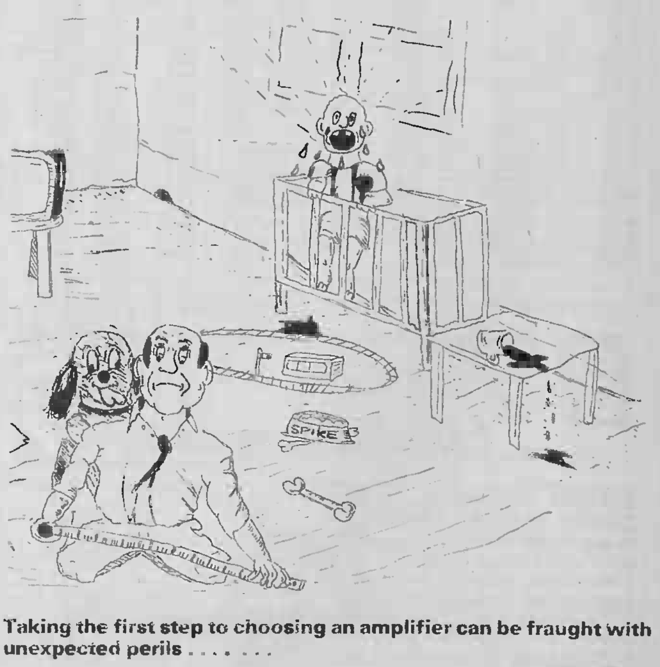
Differently specified, and at first glance Amp A is more powerful. But this is not useful extra power at all. At 40 Hz the unit is only capable of delivering 25 W into the load, and above 10kHz the power is similarly restricted. Amp B, however, can produce 40 W at both these frequencies, and would thus handle extreme bass and treble much more confidently.

Amp B is thus more honestly specified. Look for the range of frequencies over which power is available, and remember the audio spectrum is approx 20Hz to 20kHz.

Ample funds?

Having worked out how much power you need, you can scan the ads within your price range to find out which units are suitable. If you're at all serious about hi-fi *don't* scrimp on the output to save pennies. Nothing sounds worse than a 10 W amplifier trying to pretend it's a 50 W and fooling no-one. Reserve power is a necessity, not a luxury.

Most systems incorporate at least two sound sources; tape, records, radio, etc, and so the next stage is to decide what peripherals the unit has to control. Do you need filters? Tone controls? Three tape deck inputs? Two



Taking the first step to choosing an amplifier can be fraught with unexpected perils . . . . . .

record decks? There is a great variety of available combinations to choose from, and only you know which facilities you really need.

However, remember that the more stages you force the signal to travel through on its path to the speaker cones, the more it gets modified. For best reproduction keep things as simple as possible, filters and tone controls in particular should be avoided if possible, as a high quality source will need only minimal equalisation. If your room is particularly bad acoustically, buy a graphic equaliser and do the job properly.

By now the number of models to choose from will have fallen quite considerably, and it's possible that a shortlist can be compiled. (It's probably wise to let the feminine member of staff have a look over the prospective additions to the family — just to make sure that the Pioneer you've set your heart on doesn't clash with the frame around Aunt Nellie's picture, and gets heaved out the first week.)

Specifake shunned?

Every company produces masses of literature on their produce, all loaded with loaded figures. I haven't forgotten these specs at all, I just don't regard most of them as particularly useful in selecting hi-fi. Once you've got your shortlist make the final choice on grounds of compatibility with other equipment — and how the amplifier SOUNDS.

Try to hear your choices in a direct comparison against each other if possible, and even more importantly through the type of speakers you will be playing it with at home. All amplifiers are load sensitive to some degree, and the resulting sound can be changed dramatically by this fact.

Different speakers will present a different electrical load to an amplifier's output stages, which will mean that when you take it home the result may be totally different to that produced in the showroom. Most dealers carry a good range of speakers, and some are willing to arrange home demonstrations.

The main trap to avoid though is to start comparing numbers studiously, and conclude that an amp with 0.04% THD will sound better than another with 0.1%. It *might* do — but it's just as likely to sound worse. Leave the numbers alone, and give your ears a chance.

In summary then.

1. Decide how much power is necessary in your room.
2. Set your price limit.
3. Decide what facilities you need.
4. Draw up a shortlist.
5. Make the final selection by listening and comparing models through the same speakers — which should also be the type you use at home.

ETI

microfile.......

Gary Evans has been trying to do-it-himself this month but only landed in trouble with a COP

THE TITLE Do-it-Yourself Computer Show would have been more apt if, when applied to the event held at the West Centre Hotel between the 22 and 24 of June, it had been preceded by a negative. Most of the exhibitors required little more of the user than to plug their machines in, insert a disk and hit return, and as such reflected the current US trend towards the slick, glossy, expensive hardware/software package.

In the States the Personal Computing Industry is seen, by the people with the cash, as one of the major growth areas of the next few years. As such it is attracting a lot of the venture capital that is looking for a profitable outlet. At the same time it is realised that the gains that these injections hope to promote are unlikely to come from the low-cost home DIY products, but from the education/small business sector.

HI Finance, Bye DIY

This latter market demands the ready built, cosmetic package of hardware, together with readily available software packages and support and considers the £2,000-£5,000 price bracket of such systems to be far cheaper than any viable alternative.

We shall then soon see a polarisation of the micro computer market. Thus at the DIY show we still saw the likes of Bywood, Micros, NASCO, Newbear, and Science of Cambridge with products that require that people do it themselves, but the pleasant "club like" atmosphere of last year's show was missing. The event, instead of being a meeting of keen, often naive. (in terms of computing) hobbyists, was instead of gathering of calculating, if adventurous, businessmen.

It seems that a large section of the DIY computing field has passed through the first few tentative steps of youth and has already reached a, to me, saddening maturity.

Osbourne On Finance

As well as the exhibition, the DIY show also featured a number of lectures throughout the first and second days. One of the first speakers was Adam Osbourne, a well known figure in the States, who has been involved in the development of the Personal Computing Industry right from the early days.

During his talk he put forward the following, if unusual, nevertheless sound advice. He said that when choosing between systems of similar performance, one should not look at the detailed specs. of each product, and make the choice on these grounds, but at the financial stability of the companies marketing the products. In this connection Osbourne highlighted the questionable tactics employed by some of the concerns trading in the US, one of these being the adoption of a scheme that, for want of a better phrase, can be termed forward financing.

The idea is, briefly, this. You have a product for which you feel there would be a ready market but no money to take the idea to the production stage. Approaches to banks and other financial institutions meet with rejection as the venture is seen (by them) to be too risky. Fairy god mothers not being too thick on the ground, even in the States, the solution adopted by some is to advertise the product heavily, this takes supprisingly little cash, gather in the money sent in response to the ad. and use this as your development fund. If all goes well, and it rarely does, you might be able to ship the first units before the customers start screaming for their goods (which may not even be designed) or their money (which you no longer have). Even if you manage this tight-rope act the first few batches of the product are likely to be riddled with faults because of the hurried nature of the development.

Forward financing can, and has, worked but it is at best sharp practice.

I'm not suggesting that these tactics have been employed to any great extent in this country, but at least one company I know of in the personal computer area (marketing low-cost terminals) is in some financial trouble.

Scech ...... Do You Know Who?

The advertising department of ETI drop the odd clanger or two (spot the error on page 48 of the July issue) but then a certain Project Editor is covered with something other than glory at present. It's nice to know however that we are not the only human beings (less than perfect that is) about, some work at the offices of Commodore's PR agency. ▶

Below we see Kit Spencer and Derek Rowe of Commodore but who's that on the right?

Said agency sent out a photo showing Kit Spencer and Derek Rowe of Commodore extolling the virtues of PET to some, unnamed, customer. It would have been better to name the person however as some might have recognised Chuck Peddle as the man who conceived and designed the PET and who probably knows more about the machine than anyone. He was pictured on a brief visit to London but nobody told the PR people that.

I get a lot of letters detailing the activities of various Computer Clubs around the country and it seems like a good idea to collect all these together and publish the list in ETI. So please if you run such a club, and would welcome new members please drop me a line. If you have already written to me please write again as my filing system is, shall we say, in a mess and your letter is as likely to be filed under "threatening memos from the editor" as anywhere else.

Just to be corny, could you please mark your letters club call.

By the way if your club is in the habit of inviting guest speakers along, I'll be only to happy to come along and say a few words but will probably spend more time listening to what you, the reader, have to say and I mean that most sincerely.

Blue Chip News

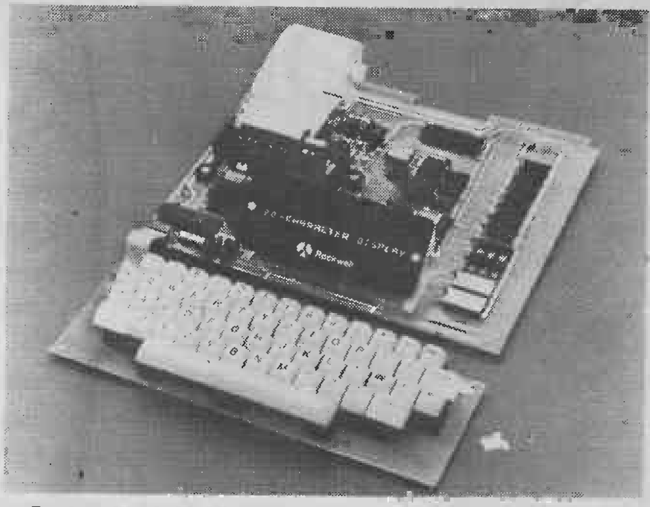
The series of single chip MPUs from National that go under the generic title of Calculator Orientated Processor Systems, or COP Systems, include devices that have been programmed by the manufacturers to provide various dedicated control functions, including timers and a number cruncher for general purpose use.

National however hope that design engineers will develop their own programs for the COP series that will suit their own particular needs. Because the memory of a COP is not normally available to the outside world, and is at any rate not alterable it having been masked at the manufacturing stage, some form of development system that can imitate the performance of a COP must be provided to the software engineer.

Had National taken a leaf out of the SC/MP book and called this dedicated series of processors Simple Microprocessors an obvious name for this machine would have been SIMulator. With a name like COP the name is equally obvious but far less repeatable — long live National.

Before leaving this subject area when you next hear someone working with an Intel 8085 exclaim SOD it, technically speaking he's requesting that data be output via the devices Serial Output Device port. Of course if he's not speaking technically then yet another of murphy's laws has probably come to light.

By the way, while not promising to publish all suggestions, if you can come up with any likely ideas for pin description of IC's of the future, send them to me at ETI.



Rockwell's AIM-65 interface module featuring the 6500 MPU, keyboard and thermal printer.

Well Rock On

Rockwell is a company that, as far as the amateur is concerned, seems to have kept a low-profile in the micro/computing fields. Their calculators are well known however and their products are well known to industry. One of their MPUs is at the heart of the Monitel telephone charge monitor mentioned in News Digest recently for example. Their latest release is also likely to bring the name of Rockwell to the attention of the aforementioned computer hobbyist.

Described as an Advanced Interface Module and designated AIM-65, the machine (pictured) features a full alphanumeric keyboard, 6500 processor with ROM monitor, dual cassette plus TTY interfaces and, the main attraction, a 20 column printer.

The 20 character wide display is formed from 16 segment characters providing the usual 64 character ASCII subset. The printer features built in memory, decoding and drive circuitry.

Not much to say about the keyboard and the 6500, by now familiar as the device around which the PET and KIM-1 are built.

The cassette interfaces can be switched between two standards, an ASCII KIM-1 standard and a binary blocked file assembler compatible.

The Monitor/Debug commands are too numerous to detail here, suffice it to say they are far more comprehensive than the minimal functions provided by many systems.

The AIM-65 provides on board sockets for a 4K Assembler or for an 8K BASIC Interpreter thus making the system easily expandable.

For full details of the AIM-65 contact Pelco (Electronics) Ltd. at:

Enterprise House
83-85 Western Road
Hove
Sussex
BN3 1JB.

Point Of Scale

One of the many ways in which micros have improved the quality of life is in the area of Point Of Sale (POS) terminals, cash registers to you and I. A few years ago parting with your money would be accompanied by a series of whirrs, groans, clicks, with a final, puny, ting. Nowadays things are almost musical, the entry verification tones providing the melody while the chatter of the thermal printer takes care of the rhythm.

In fact I'd swear that I heard one of the things rendering the song "money money" the other day — well at least the machine was honest.

ETI

electronics tomorrow......

by John Miller-Kirkpatrick

A COUPLE OF MONTHS ago I suggested some refinements which would be rather useful in connecting a cassette recorder to a microprocessor.

The intention of the original article was to suggest a standard form for recording files on a cassette tape, regardless of the actual bit recording form (CUTS, Tarbell, etc). One requirement is some form of identification Of Start Of Record, this cannot be a 'special' character as any of the 256 possible characters coudl appear on the tape as data, similarly a sequence of characters may exist on some tape somewhere as data. What is required is a marker which is not data and could not appear by accident — a reflective strip is one possible answer, other solutions are a hole, blank tape, a tone or a highspeed signal. Most of these have limitations when applied to the various types of tape machines available including Audio Cassette, digital cassette or even paper tape. The idea of recording and playing back an indication on the tape at high speed is an interesting idea (for which I am indebted to Mr Fielden of Suffolk). On a digital cassette drive, those of the high price and low dropout, the motor speeds are usually the same in both directions which makes this idea perhaps only applicable to these machines. My extension of this idea conforms with the requirements to have a signal which cannot be recognised as data because the data is written at high speed so that it will be ignored by a normal speed read. The tape can be initialised in either of two ways —

Initial Reaction

1. A series of tape marks can be written on the tape during a high speed forward write operation, these are intersperced with blank tape of enough duration to allow for a tape stop, transfer to slow speed, read data record and then revert to high speed. Using this method the format of the records must be fixed length, the formatting program will calculate how much blank tape is required from the given record length.

2. The first record written onto the tape is an End Of File indicator written at fast forward speed and containing information about the tape, number of records, units of tape used, etc. An alternative is to record this information twice, once as a tape header record and once as a trailer. In this case the header and trailer could form an index to the contents of the tape and the theoretical space available on the tape.

Each header record would be required to be read in either direction and thus the bit pattern of the fist byte in the record should be the inverse of the last, usually this bit pattern is used to act as a START byte and thus act as a double check that a record starts here. If a UART or similar device is used within the system it will cuase 11 bits instead of 8 to be recorded on the tape, this system assumes that 8 bits are recorded but coudl be used with inginuity wit an 11 bit system. If we choose the bit pattern x'5A' or 0101 1010 it can be recognised by reading in either direction, even by a UART. We should also include a direction byte to indicate which direction the tape is reading, I would suggest '01' for forward and x'FF' for reverse (ie +1 and —1), this indicates both the direction of bit shifting required and also the byte storage into RAM.

Heading For Use

Once we have the two ends of the header record defined we can fill up the rest of the header with useful data, eg

Indent or SOR code.	x'5A'
Forward Read code.	x'01'
Record Number	Single byte number of this record in file
Start Address.	Address (2 byte) at which to store record, can be overridden.
Length of record	2 byte length, in a tape header this would be max size.
Record Name.	any 6 byte name, eg. HEADER, INDEXb, PAYROL, etc.
Record type code.	X'48' (= H for Header), x'54' (T=Trailer), x'44' (D=Data)
Transfer address.	A 2 byte address to which execution should be transferred at EOR, can be over-ridden as an option.
Chain from code	A single byte record number which is compared with the 'last record' in RAM, if x'00' ignore.
Chain to code.	Single byte record number of record which should follow this in chaining sequence, x'FF' = end of chain, x'00' ignore.
Header data.	User header data, eg Index, etc. Length as defined above.
Checksum code.	A checksum byte which is compared to that computed during the read operation, a difference could indicate a read error.
Length of record.	Duplicate for Reverse read.
End address.	IE. Start address for reverse read, can be used for checking during forward read.
Record Number Two	Duplicate for reverse read.
Reverse read code.	x'FF'
End of Record code.	x'5A'

Note that the data above is repeated for each record in both fast and slow modes, except that in a fast mode the record length would be zero to indicate a fast header and that the 'Record number' field would indicate the number of the next data record in slow mode which will be encountered in the current direction of travel.

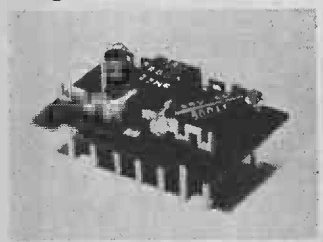
. . . news digest . .

junk calls

From the land that brought us Muzak and MPUs comes the Junk call — the same as Junk mail but verbal! A machine is being used to dial up to 1,000 numbers a day and make a pre-recorded sales pitch, unlike junk mail there is no way of knowing when the call will be junk or not. By dialing up numbers from 0001 to 9999 the machine annoys everybody who answers on a particular exchange, even if you hang up it holds the line open until the pitch is finished — this has caused emergency calls to be delayed in some cases.

Ten states are considering legislation to curtail the activities of the machines. However they intend to exempt charities, pollsters and politicians. Some people want an electronic 'no thanks' sign to be developed, although nobody is quite sure how it would work. What next?!

diy dil



A new dil package is available from Erg Components, designed to house "numerous" components the pack has two rows of 7 linked terminals. The links can be easily broken with wire cutters if required. Uses suggested include hybrid circuits, passive networks and board to board coupling (using ribbon cable out the top). Two versions of the snap on cover are available one 5.7mm high, the other 8.9mm, connection links and pins are hard gold plated. Erg Industrial Corporation Ltd, Luton Road, Dunstable, Beds. LU5 4LJ.

bulble memories

AND IBM said 'Let there be light' and there was — but it moved! Boffins at the IBM research labs in San Jose have been investigating microscopic sources of light in a certain electroluminescent thin film, and have discovered that they move about and repulse each other. The effect starts when a high frequency voltage is applied across the films, and reaches a peak of activity at about 50kHz.

The anology with magnetic bubbles has given the researches the idea that they should try and find a way of controlling the light bubbles. They still don't know exactly what causes the effect, one suggestion is that the materials are riddled with microscopic defects in crystalline structure. Wonder if they are feeling 'light headed' with their discovery?

odds & ends

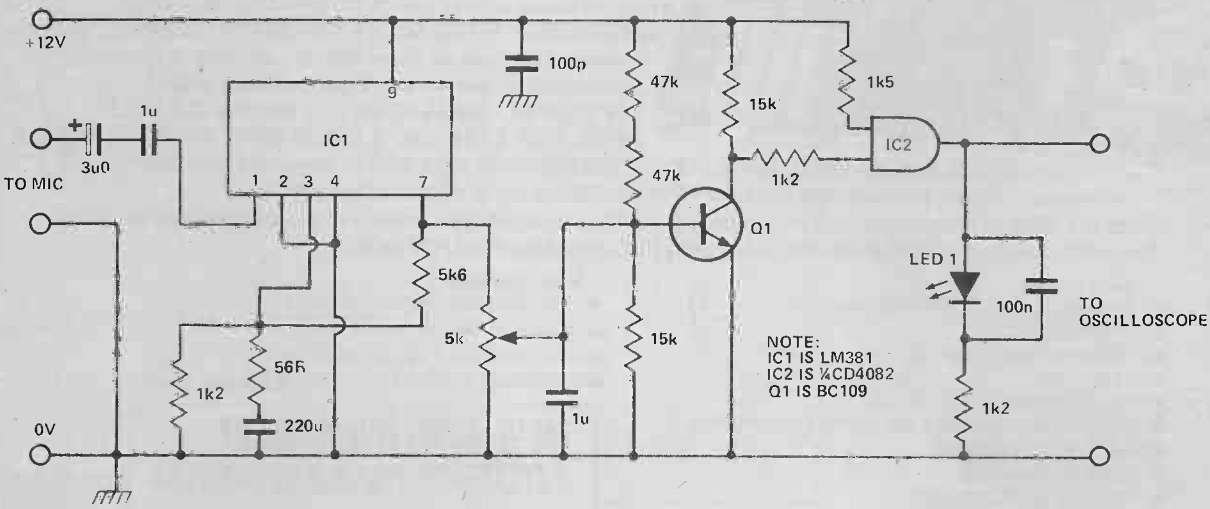
* Polaroid are about to release an automatic focusing camera that uses an ultra-sonic transducer to measure distance.

* Computers stores in the US are opening up literally every day — we have just heard that 700 have been identified by someone preparing an exhibition! In addition to those dedicated to Home computers, office equipment suppliers and camera shops are at the forefront when it comes to jumping on the bandwagon; even Macey's stores have now got a computer department in some of their stores.

* Sanyo have demonstrated a 6 mm thin solid state green and black television. The display is made out of 6,144 green LEDs in an area only 50 mm by 75 mm. They hope to have a commercial set by 1981.

* A radar based overspeed detector is in use in the U.S. of A, the unit measures your speed and lights up a neon sign saying YOUR SPEED IS . . . . REDUCE SPEED. The unit is very effective, only problem was the local hot-rodders using it to check their top speed! Problem solved by limiting display to 75 instead of 99.

tech tips



Morse Code On The Oscilloscope

S. J. Stamps

This circuit enables morse code to be displayed as dots and dashes on an oscilloscope screen. By speaking into a microphone, saying 'dit' and 'dah' as appropriate, short and long pulses appear on the screen in a format similar to that of written morse.

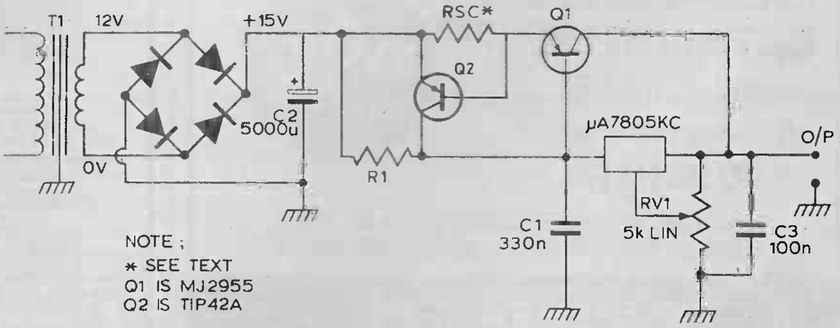
One half of an LM381 and a BC109 are used to amplify the signal from the microphone, which is then clipped into digital form by the AND gate. The output from the circuitry is fed to an oscilloscope set to 2V/cm and 5ms/cm, set to trigger on the start of a 'dit'.

Input to the circuit can be from a microphone, or tape recorder. If words are recorded onto the tape with the microphone and then played back via the circuit, practice at reading morse is possible.

High Current Regulator

N. Gray

This circuit can supply 10A at 5V which falls to about 8A at 15V, — (make sure your transformer can take it!). The circuit is fairly straightforward. Most of the output current flows through Rsc and Q1 (less than 1A flows through the regulator), the current being regulated by the current flowing through the e-b junction of Q1. Voltage is regulated by the µA7805 and controlled by RV1, giving a variation from 5V to 15V.



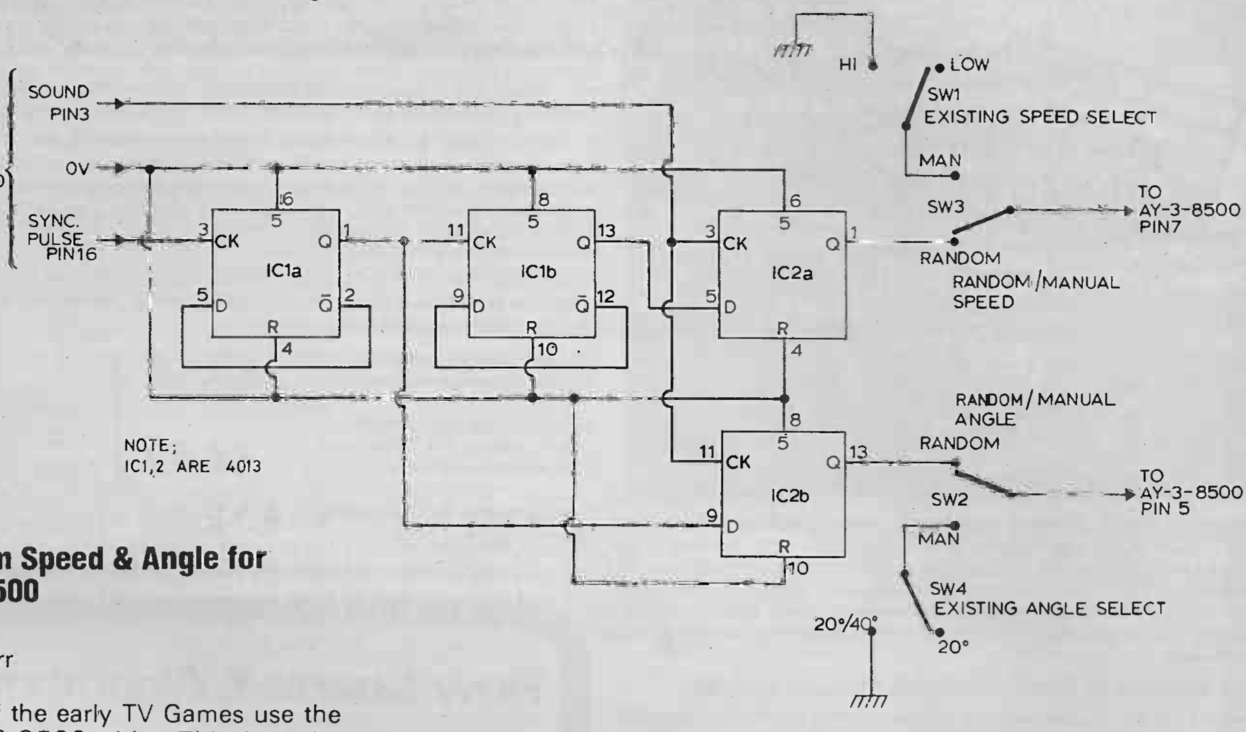
Output current is limited by Rsc and can be calculated from

$$Rsc = \frac{0.9}{I\ max}$$

For currents greater than 5A, Q1 should be mounted on a heatsink. Q2 and the regulator should run cold (if not there's something wrong!).

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

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



Random Speed & Angle for AY-3-8500

E. A. Parr

Many of the early TV Games use the GI AY-3-8500 chip. This has the facility to switch ball speed pin 7, and the angle of rebound on pin 5. These two pins are usually brought out to a switch for selection by the users.

The games can be made more exciting and realistic if the speed and rebound angle vary randomly at each bounce or when a player hits the ball. This can be simply achieved with the addition of two 4013s (Dual D type).

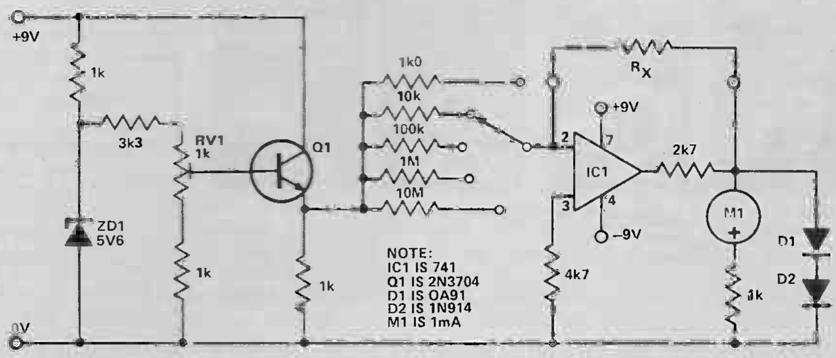
SW1 and SW2 are the existing manual select switches. ICI forms a two bit counter, clocked on by the sync pulses from pin 16 on the AY-3-8500 chip. This counter will assume a random state from bounce to bounce.

The two D type flip flops in IC2 are connected to pins 5 and 7 on the AY-3-8500 chip via the random select switches. To ensure that these only change at a bounce, these two D types are clocked by the sound output (which consists of a 32ms pulse train at each bounce). This pulse train will, of course, overlap several sync pulses, but the effect of the angle and speed changing rapidly for 32ms is not noticeable and the ball speed and angle stays constant after leaving the bat or boundary until the next interception.

Because the two ICs are of CMOS construction they will have little effect on battery consumption, and the circuit can be easily incorporated into TV games units.

Linear Scale Ohmmeter

M. Roberts



This circuit has several advantages over other linear scale ohmmeters.

Only one preset resistor is used for all the ranges, simplifying the setting up and reducing the cost. Diode clamping is included to prevent damage to the meter if the unknown resistor is higher than the range selected. The use of a FET 741 Op-amp reduces any zero error and makes offsetting unnecessary.

When the meter has been assembled, a 10k precision resistor is placed in the test position, Rx, the meter is set to the 10k range and RV1 adjusted for full scale deflection.

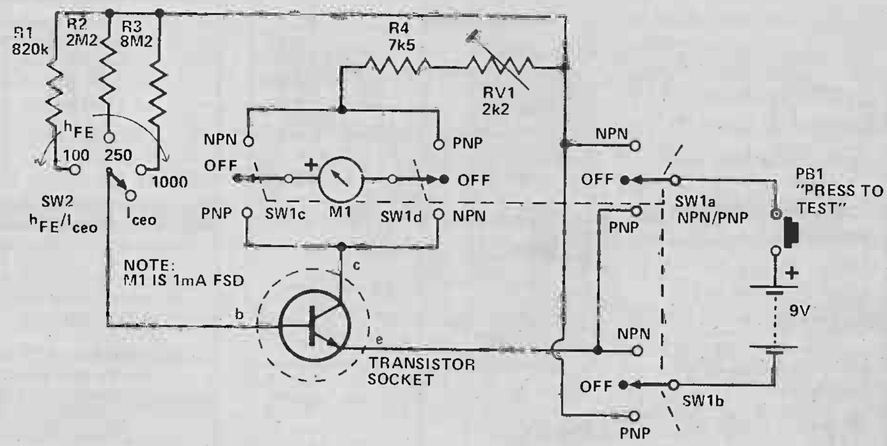
Transistor Tester

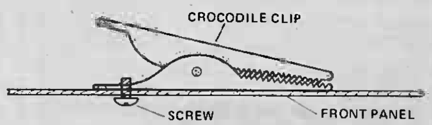
G. Smith

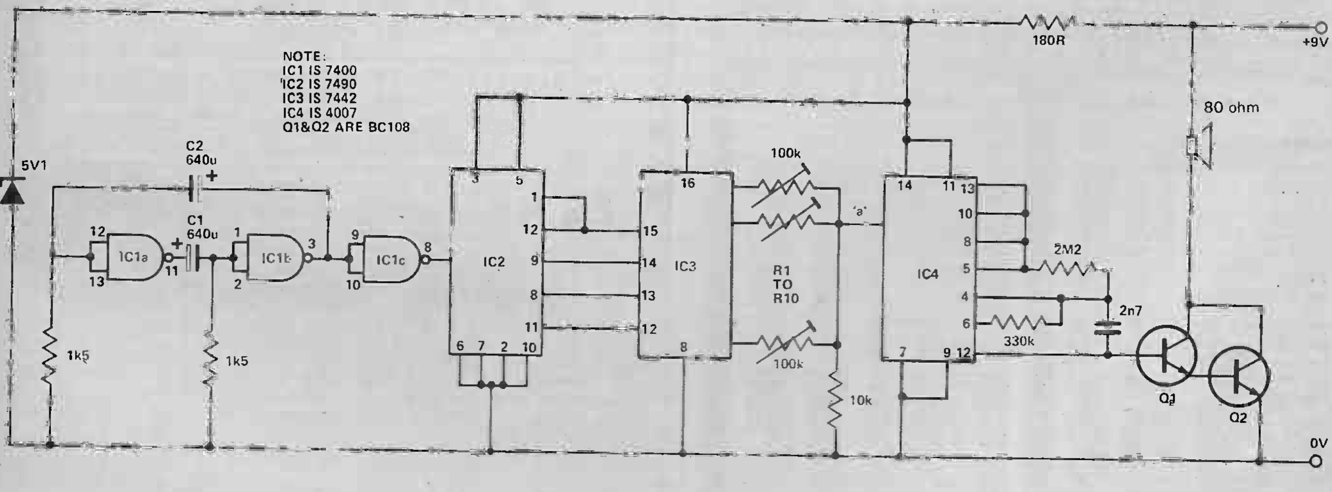
This transistor tester works by injecting a known current into the base of the transistor under test, and measuring the collector current. The values of R1, R2 and R3 give a base current of 10, 4 and 1uA which gives a FSD on the meter for transistors with a gain of 100, 250, and 1000 respectively. Since the gain of the transistor is proportional to its gain, the gain can be easily deduced from the reading on the meter. Leakage current is measured by leaving the base open circuit.

SW2 reverses the polarity of the battery and the meter to allow the testing of both NPN and PNP transistors. R4 and RV1 protect the meter from excessive currents, and do not affect the reading on the meter. RV1 should be adjusted so that the meter needle just touches the end stop when the collector and emitter terminals are connected together.

A simple transistor socket can be made by mounting three crocodile clips as shown in the diagram.







Musical Tone Generator

P. Reynolds

This circuit provides a means of generating a series of up to ten musical notes.

The 7400 oscillator produces pulses at about 1 second intervals. These pulses, after being buffered are fed to a decade counter which produces a BCD output. The output is fed to the 7442 which produces a decimal output. Each output is taken to a preset forming a potential divider. The VCO senses the voltage at point 'a' and changes the frequency of the output tone. Careful adjustment of the presets can give a reasonable range of notes. The length of each note as well as the time between notes can be varied by changing the timing components in the 7400 oscillator.

ROCK SOUND

The last couple of years have brough bigger and better equipment to the concert stage . . . here Howard Page of Jands Ltd describes the equipment used in presenting artists like Rod Stewart and Abba to Australian audiences exceeding 30,000 and this illustrates the techniques in use today.

CONCERT SOUND SYSTEMS come in many sizes, shapes and forms and I don't think I've ever heard two systems that sound identical in the same hall. The sound engineers have different design philosophies although they share a common objective.

Expressions such as 4 way crossovers, front loaded horns, radials, dispersion angles, etc, are bandied about when the crews get together on tour but what really makes a good "state of the art" sound system? A system that, given the hundreds of variables such as hall acoustics, mood of audience, time available for set-up and tuning, road damage (that must be taken into account at every concert), will consistently deliver the best possible sound to the audience.

For some of the answers let's look at a system I designed for the Australian tours of Rod Stewart and Abba. The 'Jands No. 1 Touring System' weighs 28 tonnes and delivers a power output of 24 000 Watts RMS.

Let's follow the sound from its source looking first at microphones. The majority of these are made by Shure — type SM 58 for vocals and SM 57 for instruments. On the drums I use some other favourites such as Sennheiser MD 421 or AKG D12. The actual set-up depends on taste and the way the kit is tuned. The mics plug into 20-way multi-core cables leading to the mixer in the hall. The multi-core input box also has splitting outputs to feed any mic to the stage monitor mixer located on one side of the stage. The house mixing console is custom designed by myself and Jands consultant 'electronic genius' Phillip Storey. This is a 24 track in, 16 track out, studio style board made super-rugged for the 'road'. It has many facilities not normally needed on a PA mixer, such as the ability to do a stereo house mix, a separate stereo recording mix, a mono TV mix and an all-up 16 track output all at one time.

Why such extravagance? It is because in Australia (due to the limited audio facilities in TV OB vans) we often get asked if we can do all the above — for a live TV show with an album to be released later, so the extra features can be readily justified.

Tuning Up

The stereo 'house mix' outputs of the board feed to a set of one-third-octave stereo graphic equalisers. These are set up using pink noise and real time analysis to accurately 'tune' the sys- ▶

This set-up shows the speakers used at the Sydney showground for the Rod Stewart concert.

tem for both the hall and, in some cases, the type of sound required. The stereo signals then feed a set of stereo DBX 160s (Compressor/Limiter) which are set as a final safeguard on the system to ensure the amplifiers are not driven into consistent square waves, one of the primary causes of speaker system failure.

Having been tuned and compressed as necessary the signals feed into a custom-built switchable 3, 4 or 5 way stereo electronic cross-over unit, the design of which is classified information. Also feeding in and out of the mixer are what we call FX devices, ie, echo unit, flanging units, extra compressors for various instruments, digital delay devices, etc, these are used as required.

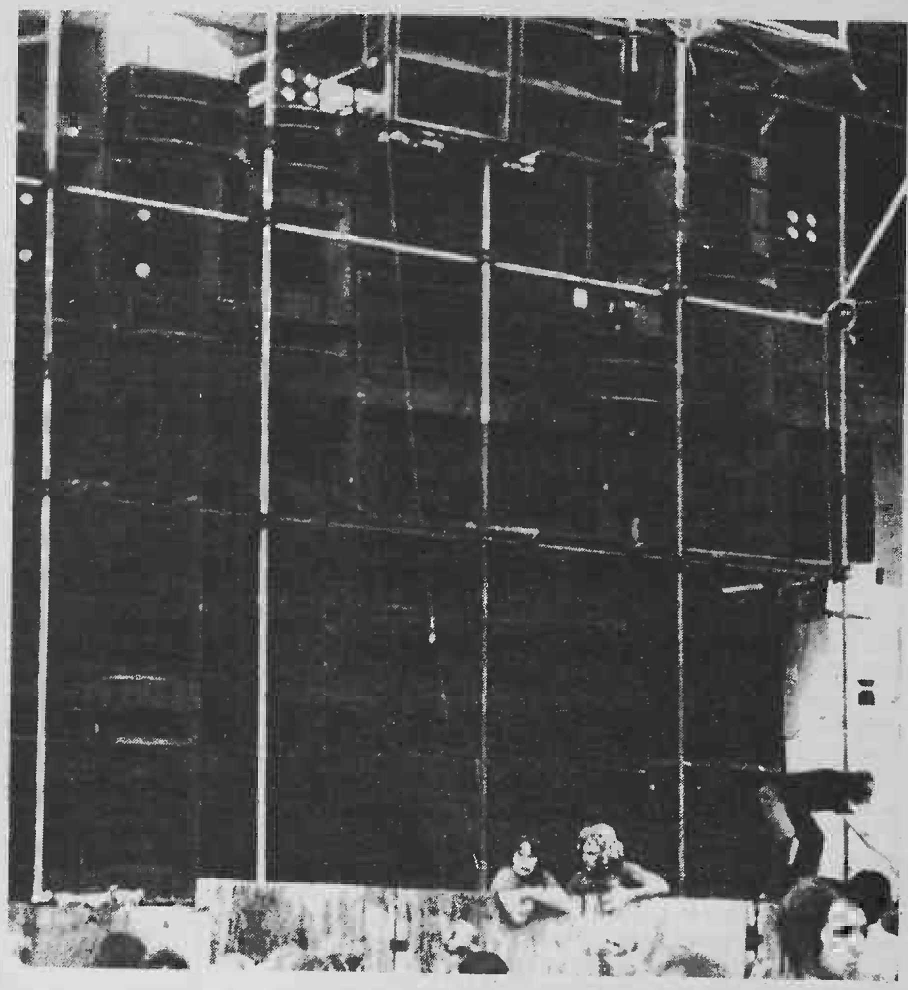
Once the sound has been divided it is sent down a separate multi-core cable called a system feeder which plugs into the amplifiers on stage behind the speaker stacks. The amplifiers we use are the finest available 'state of the art' units: Phaser Linear 700B, Crown DC 300A, SAE 17K111CM, and a new unit we're especially proud of, our own Jands J600S which is proving equal, if not superior to, anything available from overseas.

Each amplifier rack unit contains switching and matching systems to enable complete flexibility and access should a failure occur. Heavy duty speaker cables connect the amplifier outputs to the final link in the chain, the speaker units themselves. These, in the No 1 System, are for the 'Lo Boxes' custom-designed Super 'W's containing 4×15" JBL (all components in the system are JBL) speakers; for the 'Hi Bass' or 'Mid Bass' another custom-designed front loaded 2×12" speaker box tuned reflex porting (for use as the bass unit in a 3-way system); for the 'Mids' JBL 90° and 60° Radial horn units with high powered compression drivers; and for the 'Highs' 2402, JBL 075 radiator units.

Well, that's it, total cost approx. £150 000 but it represents where concert sound reinforcement is at now. Certainly a far cry from a column speaker on each side of stage but its worth it when I hear members of the audience muttering as they file out "They sound just like their record."

Ample Amperes

One of the biggest problems now facing Jands when operating a PA and lighting rig, such as that used on the Rod Stewart tour is to ensure



Above is the tower of speakers used at one of the smaller gigs on the tour! Below, the scene as seen from behind the main control desk — the diminutive figures on stage are ABBA.

JANDS CONCERT SOUND SYSTEM AS USED BY ABBA/ROD STEWART TOURS OF AUSTRALIA

MONITORS

Mixer: Twenty input and six output buses. Each mic can be mixed onto one or all of the six buses, with or without tone control. This gives up to six separate monitor mixes so that each musician can have the extra foldback mix he requires. Each feed then passes through a graphic equalizer and into a Jands J600S to feed a foldback system.

Foldback Speaker System:

Each Side	1 × JBL 4550 with two JBL 2220.
	2 × JBL 4560 with one JBL 2220.
	2 × JBL 90° horns.
	1 × JBL 2390 horn lens.
Back Monitor	4 × JBL 4560 bass bins.
	2 × JBL 90° horns.
Front	4 × wedge monitor housing one JBL 15 bass and one JBL horn and driver.

MAIN SYSTEM

2 × 20-way multicore cables feed the signal from forty microphones to the front of house mixer. A Jands 24 channel in and 16 channel out mixer.

The custom-designed 24 track, 16 track out mixer has the following facilities on each module.

1. Selectable Input Attenuation
2. Channel Mute
3. Mic Phase Reverse
4. Mic/Line Switch
5. High Pass Filter (250 cycles 18dB/octave)
6. Equalizer Bypass
7. Lo; Mid; High; 18 dB Boost/Cut at four selectable frequencies
8. Pan Pot
9. Eight Full Stereo Group Select Buttons
10. Solo Prefade Listen Button

There are eight stereo sub groups with two other sets of eight for making separate mixes of the sub group for recordings, TV, etc.

At the mixer are two 19" electronics racks.

The effects rack and the main system rack housing.

One third octave (27 band) stereo graphic DBX 160

2 × limiters DBX 160

2 × Jands 4-way crossover

The signal passes through each item then goes via a separate multicore to the stage to drive the amplifiers.

At each side of the stage are built the sound towers. These being 24' × 12' with three levels. Better dispersion is achieved by stacking high rather than wide. Each stack has the following

8 × Amplifier Racks each containing 3 amplifiers these being Crown DC300A Phase Linear 700B and Jands J600S.

The Speaker System:

12x4 130 (Jands designed W Bins with four JBL 15" speaker in each).

12xW cabinets containing two JBL 15" speakers.

24xJBL 4560 Bass cabinets with one JBL 15" speaker.

16xDouble 12" cabinets (Jands design) containing two JBL 12" speakers.

16xDouble 12" cabinets (Jands design) containing two JBL 12" speakers.

20xJBL 90° horns

16xJBL 60 horns.

8xJBL, long throw horns.

48xJBL 075 high frequency.

The total JBL count on the Rod Stewart/Abba main system Sydney Concert was.

80x15" speakers.

32x12" speakers.

44xHorns and drivers.

48xHigh frequency.

Total value at your local hi-fi shop approx. £150 000.

The entire system is equalized before each concert using a pink noise generator and a Real Time Analyzer.

adequate mains supply (240 V). Simple arithmetic gives power consumption: the PA has six amplifier racks per side, and each rack has three stereo amplifiers each drawing four amperes. Total consumption is 2×6×3×4=144 A. Stage equipment, including special effects, can easily draw 100 amperes. The lighting system comprises 100 lamps, each drawing 4 amperes. This adds another 400 amperes to the total requirement!

Dim View Of Noise

To help eliminate dimmer noise in the PA system using the three phase supplies, the lights are placed across two phases with sound and stage equipment across the third phase.

The power supply Jands now insist on is 300 amperes per phase with a solid neutral. The electrical code permits a much lighter neutral than active in most installations, the assumption being the load can be expected to be balanced across three phases and hence little neutral current flows back to the sub board. With the lights full up and no PA (as occurs at the end of each song) there is a great strain to pull the neutral towards the lighting phases and with a soggy neutral it is possible to get over 300 volts appearing on the PA phase (the neutral drifting 50 volts above earth).

Earth At Stake

Power is run from the sub-board to the dimmer racks and audio equipment via 416/0178 glass-insulated rubber sheathed mining trailing cable (cable rating 320 amperes and the copper core being 14 mm diameter). Each cable is fitted with a 350 ampere connector imported from Switzerland.

Each lighting phase runs direct into a dimmer rack housing 35 2 kW dimmer modules. The sound phase runs into a 19" electronics rack containing two 150 A breakers, one to feed PA the other the stage gear. Each breaker is connected to an earth leakage detector set to trip when more than 20 mA flows to earth. The current required to cause a fatal electric shock is 50 mA Hence if any person comes in contact with a live wire on stage they cannot receive a fatal shock.

To avoid dimmer noise in the PA system it is often necessary to get a separate earth for the audio so Jands always carry a 6 foot solid copper earth stake and 10 kg of salt (for making a brine solution for better earth contact). ETI

FM TUNER

Bill Poel of Ambit has designed for ETI the International Mk3 FM tuner. Using a modular concept the performance of the Mk3 puts it in the top flight of tuners. With the digital tuning option the design is unequaled in specification at its price.

The Mk 3 will strike most potential constructors with one main feature — it has a digital frequency readout. This is a genuine count readout, and is included here as most constructors' big bugbear where radio construction is concerned, is the mechanics of the drive and its calibration. The unit is a complete RFI proof module, and although it is not cheap at around £45, it also incorporates an AM frequency option (fed from a plug at the rear of the unit in this case. Wait for the MW/LW add-on tuner) and the time. And since most listeners will want to know the time of the programmes, this is not an unnecessary extravagance. It further means that the tuner PSU is kept warm and running the whole time the unit is plugged in. Contrary to the beliefs of some, electronic devices left permanantly 'on' do not tend to explode or generally degenerate. In this case, leaving the 12V PSU running, permits the tunerhead local oscillator to be run constantly, and thus attain a steady state frequency stability that is very useful. For reasons of power economy, the mean amongst you may wish to disable everything but the clock/display module. But that's up to you, and really isn't warranted.

In case there are those amongst you not keen to lay out for the DT1200 module, an alternative circuit to drive an analogue frequency meter is offered as an alternative. And then the cheaper MA1012/1023 digital clock modules may be incorporated instead since the chances are that most of your friends will still think you have the very latest in digital FM tuners.

IF Stages

The IF and decoder systems are chosen for very low distortion and very wide separation. There are those in the HiFi fraternity who will insist that two six pole linear phase filters will narrow the bandwidth too severely for proper FM stereo to pass through. However, it can be shown that the 200kHz of this design is quite sufficient — especially since the HA1196 PLL decoder incorporates a bandwidth/separation optimizer circuit. Sceptical observers have been shown THD of less than 0.1%, and separations of 60dB at 1kHz in this system — which is really the ultimate justification anyway. To achieve these figures, it was necessary to build and align our own stereo encoder generator, using some of the spectrum analyzer exotica that doesn't usually find its way into consumer electronic designs. The system is optimized for about 50% modulation levels in the form shown here. This represents a more realistic approach in terms of UK broadcasting than full 75kHz, since programme dynamic range

SPECIFICATION

A correctly aligned unit will provide the following level of performance: (Measured at 50% modulation)

Mono sensitivity	50dB S/N2/3uV EMF
	30dB S/N 0.9uV
Stereo	50 dB S/N 9uV
	30dB S/N 5uV
Stero THD	0.1%
Mono THD	0.1%
Stereo separation at	
1kHz	40dB
10kHz	30dB
Image/spurious rejections	better than 90dB
adjacent channel	30dB
alternate channel	65dB
Ultrasonic rejection of 19/38kHz	60/85dB