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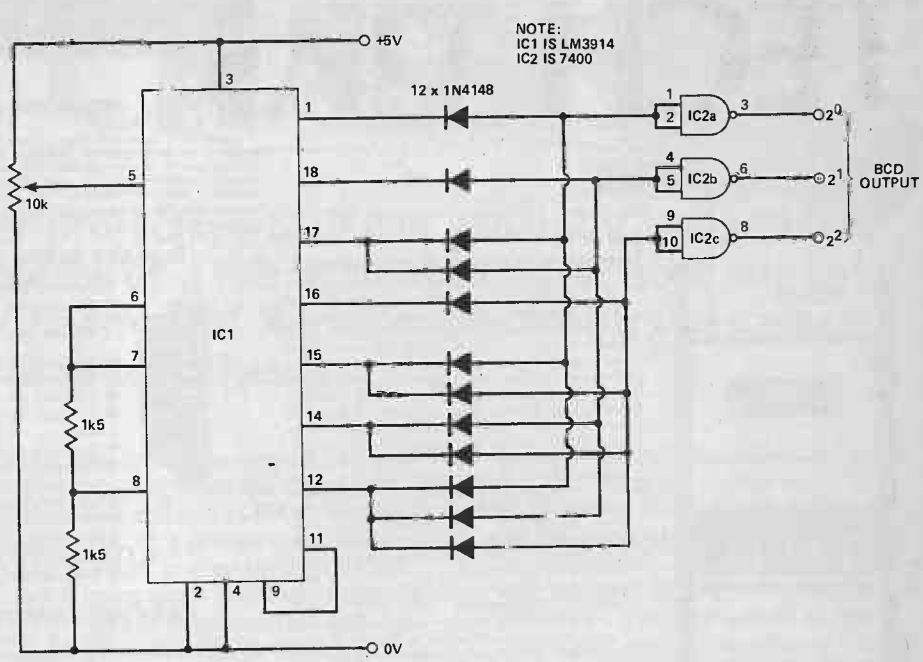
Low Resolution A-to-D Converter

Paul Hill, London

A simple, low-resolution analogue-to-digital converter (three-bit output) can be built using the LED bargraph driver IC, the LM3914. This ADC can be used in applications where more usual (and more expensive) eight-bit or greater ADCs would give unnecessary accuracy.

The bargraph IC converts the input voltage into a low signal on one of the 10 outputs, depending on its amplitude. The diodes code the outputs to binary-coded decimal, IC2 providing the ADC with TTL outputs.



CMOS Fuzz/tremolo

S.P. Giles, Edmonton

Fuzz and tremolo have been around for years now in one form or another, but this circuit gives the guitar player the option of either effect, depending on how hard he plucks the strings. When a string is plucked hard, fuzz is heard at the output, but softer plucking produces tremolo.

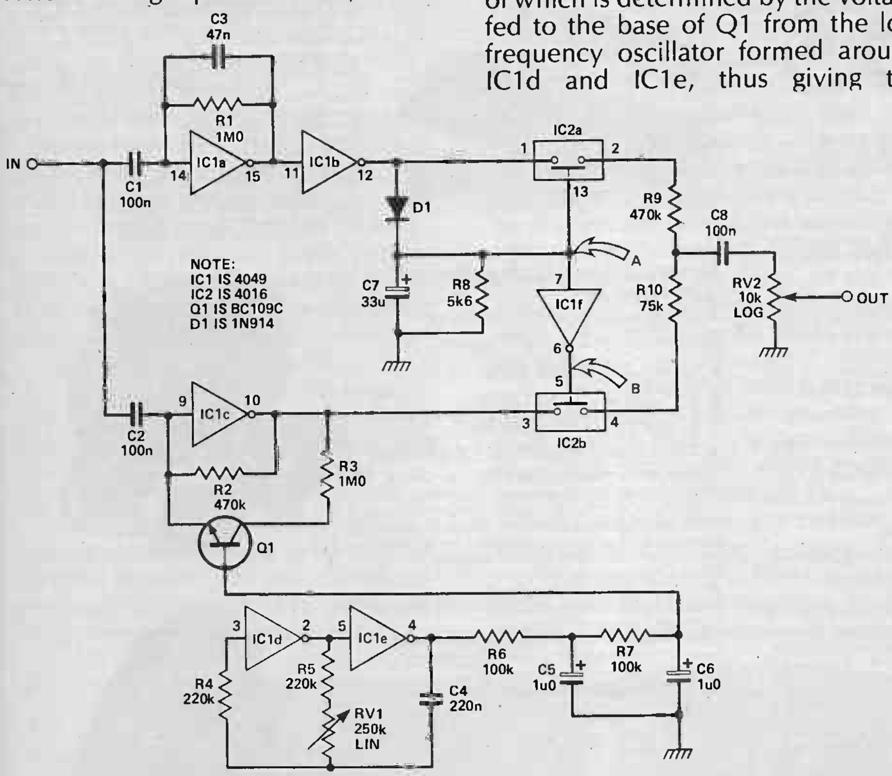
IC1a and IC1b form the fuzz circuit, which is not so harsh as the usual op-amp or transistor variety. C3 eliminates a lot of high-frequency rubbish from the output. IC1c is a voltage-controlled amplifier, the gain of which is determined by the voltage fed to the base of Q1 from the low frequency oscillator formed around IC1d and IC1e, thus giving the required tremolo effect.

The fuzz output at IC1b is rectified by D1 and smoothed by C7 and R8. Part of this is fed direct to IC2a as the switch control, and the other part is inverted by IC1f before being fed as the switch control to IC2b. The DC voltage at point A will follow the strength of the input signal as follows:

Low level input;
A is low (IC2a is off)
B is high (IC2b is on)
Tremolo output.

High level input;
A is high (IC2a is on)
B is low (IC2b is off)
Fuzz output.

If a chord is played hard and allowed to decay naturally it will start as fuzz and change to tremolo as IC2a and IC2b alter state. There is no reason why this principle cannot be applied to any two effects chosen by the constructor.



MINIMUM PAYMENT £20

Tech-Tips is an ideas forum and is not aimed at the beginner. We regret we cannot answer queries on these items. ITI is prepared to consider circuits or ideas submitted by readers for this page. All items used will be paid for at a competitive rate.

Drawings should be as clear as possible and the text should be typed. Text and drawings must be on separate sheets. Circuits must not be subject to copyright. Items for consideration should be sent to ETI TECH-TIPS, Electronics Today International, 145 Charing Cross Road, London WC2H 0EE.

ROBOT CHASSIS

Unlike most of our other projects, which can generally 'stand alone', the electronic bits and pieces of a robot are useless without suitable mechanics. Here Geoff Chapman of Remcon Electronics describes the construction of the Mobile 2 chassis.

This article describes the construction of the chosen chassis for the ETI Mobile 2 (or Robot Tractor Unit RTU2, as Remcon describe it). Included in the kit from Remcon are all the aluminium body sections (ready punched and drilled), motors, gearboxes and injection moulded components for the tracks — the complete price is £125 including VAT and postage. A more detailed specification of the mechanics was included in last month's article — alternatively you can contact Remcon at 1 Church Road, Bexleyheath, Kent (telephone 01-304 2055).

Chassis Assembly

Taking the base and one of the sides, the side is introduced into the base so that it fits snugly into the notches at the top of this unit. It should be noted how the chassis fits in this position and it is aligned all the way round the base unit to maintain a parallel line. The chassis may be held in position by either the use of simple G cramps while the holes are transferred from the base plate into the chassis side using the special drill provided in the kit. It is recommended that this be carried out at this stage so that any swarf produced can be cleaned away before other assembly takes place. The second side is drilled in a like manner.

We suggest that, although self-tapping screws are going to be used for the final assembly, the two side pieces previously drilled should be 'pre-tapped' using one of the self-tapping screws. This avoids the problem of the screw-driver slipping when the chassis is being assembled and makes for a generally easier job. It is further recommended that the burr pushed through by the self-tapping screw is left on the inside of the chassis side since this assists in locking the screw. The three pieces of the chassis are left disassembled until required at a later stage.

Assembly Of The Track

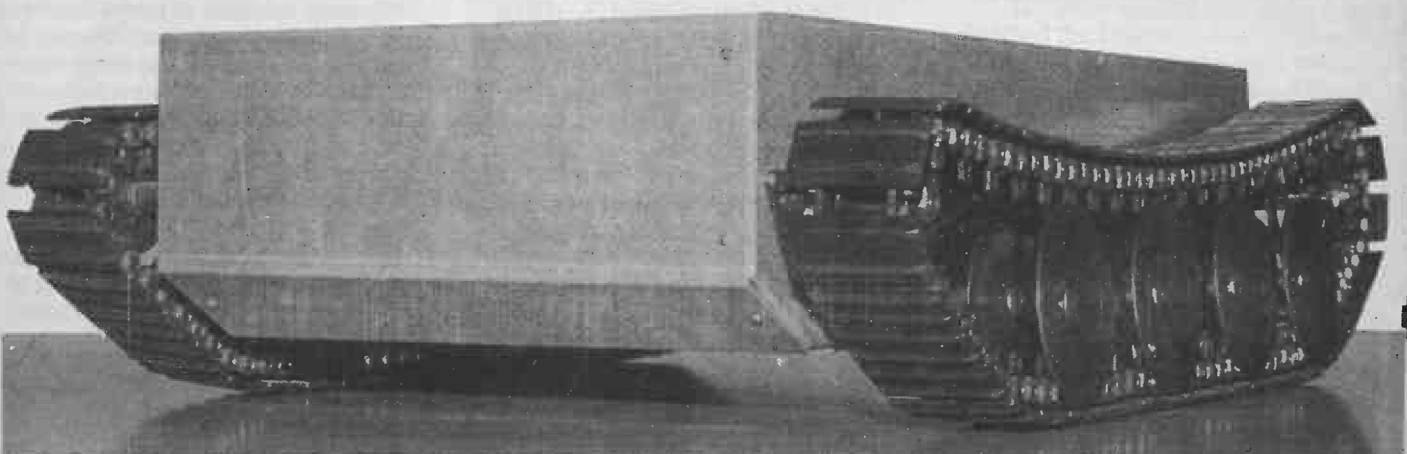
First separate the moulding from the sprues, using a sharp knife or side-cutters, then count out 47 track plates and 94 side connectors for each track. The track is assembled by slipping the side plates over the circular protrusions on the track plates and tightening up the nuts which are pushed onto the plastic and driven home with the spanner provided (see Fig. 2). Care should be exercised when screwing on the nuts; first to ensure the thread is cut square to the axis of the circular protrusion, and second that the nut is tightened sufficiently to remove any end-play between the side-plate and the track plate. At the same time, it should allow the plates to swing freely, ie it should make a good hinge.

Having assembled both tracks in this fashion, they are then put aside until required later. It is a good idea to spray each of the joints with a small amount of Rocket WD40 to keep friction to a minimum, and to allow the lubricant to 'soak' into the plastic.

The mouldings for the suspension units are a four-piece moulding on one sprue. These pieces should be separated by cutting with side-cutters as before, but kept together in their sets. The suspension blocks of each set are fitted to the side chassis members so that the bulk of the moulding goes through the side member into what may be termed the 'tray'. The clamp plate of the suspension unit is then slipped over the suspension block and the three self-tapping screws inserted through the holes, the clamp plate being pulled up tight by the screws to the inside face of the side-plate. The suspension units are fitted in eight positions; four on each of the side-plates.

The cranked bogey spindles are then inserted through the suspension blocks, and located in position by fitting a 6BA washer and a 6BA stiff nut. The stiff nuts are tightened so that the spindles may rotate without any end-play. The suspension springs are fitted to the spring retaining spacers — two spacers per spring. Note that there is a slot partially moulded across the retaining spacer just above its boring. The slot is blind at one end and it is through this blind portion that the end of the spring is pushed so that it is held in position while assembly takes place (see Fig. 6). The second spring retaining spacer is treated in a similar fashion.

With the two retaining spacers fitted to the spring, the spring is then fitted to the suspension unit over the two spindles which carry the bogies; the centre of the spring fits onto the centre stub of the suspension block. The centre coil of the spring is held into position with a 4BA nut and washer which is screwed over the stub and



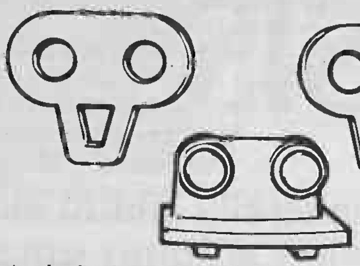


Fig. 1 The track plates and side connectors.

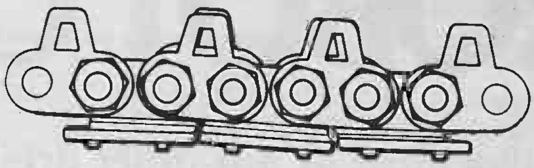


Fig. 2 The caterpillar track is assembled like this.

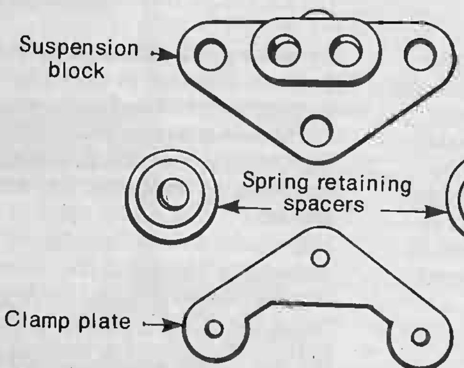


Fig. 3 Each suspension unit has a set of four mouldings.

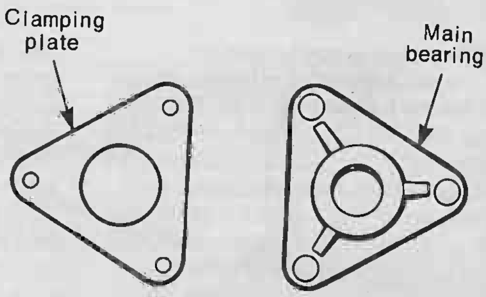


Fig. 4 The mouldings for the bearing unit.

tightened securely, but at the same time allowing the spring to move freely.

Assembly Of The Suspension Units

To retain the assembled smaller bogey on its spindle, the starlock washer is fitted to the shaft. This starlock washer is covered with a chrome button to give an attractive appearance. The other larger-diameter bogey wheel is completed in a similar manner together with the fitting of the star washer. This process is then repeated for all eight suspension units. This completes the suspension unit assembly.

The smaller diameter bogey is fitted to the other axle, again with the rivet side to the side of the chassis. The second smallest bogey is fitted to the same axle, this time with the simulated rivets to the outside.

The bogey wheels are fitted to the suspension units after having separated them from their sprues; note how each bogey wheel is moulded into two pieces. The bogey wheels are fitted to the suspension units by taking the larger diameter bogey wheel first and inserting it over one of the axles. The raised simulated rivets should be against the side member of the chassis.

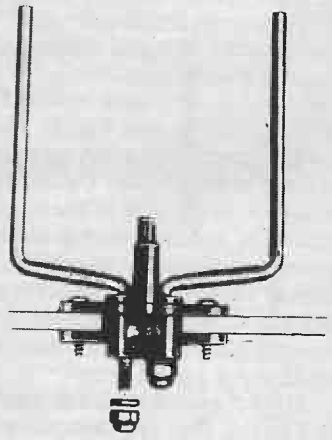


Fig. 5 Fitting the suspension units.

Assembly Of The Take-up And Shaft Encoder Brackets

There are two brackets in the kit together with two bearing units. Again, the bearing units are separated from their runners by cutting through the sprues with sharp side-cutters. Looking at the take-up brackets, you'll see that on one face there are five holes — three arranged on a pitch circle around a large diameter central hole. It is through these holes that the main bearing unit is fitted and retained by self-tapping screws onto the clamping plate. This is fitted inside the bearing bracket angle.

The two remaining holes are for the 4BA cap screws which retain the take-up bracket on the chassis. If the infra-red shaft encoder detector (not supplied in the kit) is to be fitted, it is recommended that it is fitted at this time; it locates through the three holes on the other face of the take-up bracket using the 6BA x ¼" screws and nuts provided.

With both take-up brackets thus assembled, they are fitted to the chassis side-plates through the triangular holes which appear at one end. The bulk of the bracket is on the inside of the chassis, the bulk of the bearing is on the outside. The two 4BA Allen cap screws are inserted through the chassis into the brackets and are left untightened so that the brackets can slide freely along the slots located above and below the triangular aperture.

The ¼" diameter spindles are next located through the main bearing. The spindle is threaded at each end; the longer portion of thread projects outside the chassis while the shorter remains inside. The shaft is located longitudinally within the bearing by the fitting of a nut at each end, these being tightened against the bearing face with a washer interposed to again allow rotation with a minimum of end-play. Two further nuts are fitted on the inside of the spindle; these are to retain the encoder disc which should be fitted now.

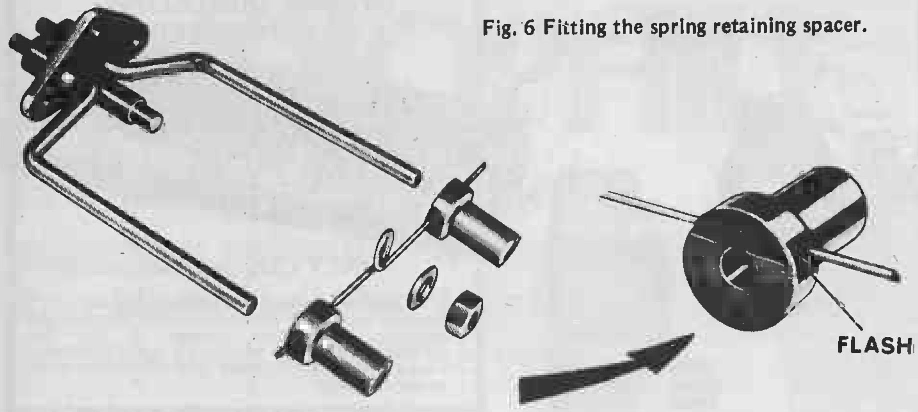


Fig. 6 Fitting the spring retaining spacer.

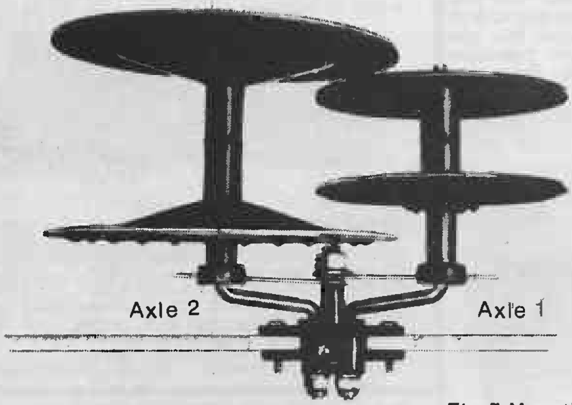


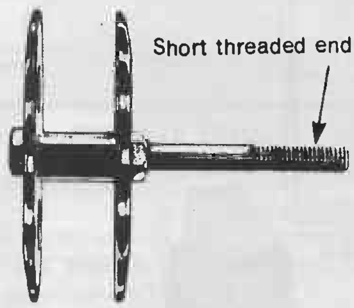
Fig. 7 Mounting the bogey wheels.

The two idler sprocket wheels are those with the smaller diameter holes in their centre. They locate over the spindle and are arranged so that the two dog-couplings at the centre are engaged. The two sprocket wheels thus formed, with spokes and sprocket locations aligned, are tightened by a nut fitted on the outside of the spindle. At this stage, the nut should only be tightened lightly. It is recommended that all these nuts be finally locked into place with the use of one of the locking adhesives available, such as Screwlock. Do not use the more permanent adhesives since removal is then virtually impossible.

Setting Up The Gear Ratio

As supplied by Remcon, the drive units are set up with a ratio of 360:1. This we found to be too high and used a ratio of 60:1 for our Mobile 2. It is therefore necessary to carry out the simple operation of changing the ratio. Before doing this, or indeed opening up the gearbox for any particular reason, please ensure that the area in which you are working is clean and free from dust. The gears are pre-coated with grease and so easily pick up any pieces of swarf or dust which will cause irreparable damage if left in the gearbox.

The four long screws retaining the gearbox plates must first be removed, and the end cap with the output shaft taken off. Inside there are a series of planetary gear platforms, each one being



stamped with a number. The numbers are 3, 4, 5 and 6 and refer to the actual ratio of the platform.

Thus to set up the gearbox to a particular ratio, it is simply a question of choosing the multiples of the number to give that ratio. For example, to set up the required ratio of 60:1, then gear platforms 3, 4 and 5 are used, since 3 x 4 x 5 equals 60:1. In the interests of quietness and operating efficiency it is advisable to put the highest step down, namely the largest numbered platform, nearest the motor drive unit.

The gearbox is reassembled but the screws are not fitted, the gear platforms being carefully held in place while they are fitted to the inside of the chassis member, at the opposite end to the take-up sprocket wheel. The long screws are then inserted through the four holes of the chassis and through the gearbox sections, and the nuts fitted on the inside and tightened.

It will be noted that the screws now protrude further through the nuts than previously; this is because the number 6 section has been omitted. However, it is not recommended that the screws be shortened since at some later stage you may wish to use this particular section of the gearbox.

Having thus fitted the screws through the gearbox and retained them, the sprocket wheel with the larger boring is fitted over the spindle which now protrudes through the outside of the chassis. A back nut is first fitted to the spindle and tightened up against the shoulder of the thread. The two halves of the sprocket wheel are then aligned and fitted on to the main drive spindle. The sprocket is retained in position by the fitting of the 8 mm nut on the outside.

Final Assembly

With both chassis side plates thus assembled, they can be introduced into the chassis pan and the 11 self-tapping screws used to hold each side plate to the pan. The previously assembled tracks are now wrapped around each set of sprockets and bogies and finally closed by fitting the closing nuts to the side plates and track plates.

It will be noted that the track tension is sloppy at this stage — this is adjusted by pulling on the take-up bracket sufficiently to take out the slop from the top of the track and, at the same time, holding the take-up bracket in this taut position while the cap head screws are tightened through the spokes of the idler sprocket.

Do not take all the slack out of the track; experience and the first few runs will show exactly how tight it should be. The tension should be sufficient to give a good drive without too much noise and without too much 'flailing'.

It will also be found that when the track has been run for a few hours it will require further adjustment. This may happen on two or three occasions during its early life.

General Comments

It will be noticed that when you've finished the assembly, four holes remain at one end of the chassis pan. These are the holes which locate the custom box to carry the manipulator, which may be purchased as a separate item from Remcon. There are also some pieces of aluminium strip left over in the kit. These are for making suitable brackets to retain batteries, speed control units, and any other item that the constructor may wish to locate positively within the chassis. ETI

AUDIOPHILE

Ever mindful that people lust after low priced(?) hi-fi sound, Ron Harris has put together a system that will rattle your windows, not your bank manager.

There has been a distinct tendency to nag lately. I've noticed this rising tide of grumbling discourse, via the mail, sweep up the sides of my desk in the past month. When it gets to the top I'll have to open a few more, I suppose. Takes the surprise out of life, though, once you know what's in the envelopes. Far better to leave them closed and mysterious. Besides, it's good exercise climbing over them in the mornings to reach my chair.

Believe it or not this is my way of apologizing to readers for taking so long answering their letters recently. There has been something of a panic on around our offices of late and yours truly has been a little tardy in putting the letter-opener to work.

One clear request that has emerged from the SAE- laden wrappers, on more than one occasion, is for a reasonably priced system to play 'rock' music. I can only interpret that, being of soundly warped mind, as meaning a system NOT intended for classical.

The requirement would seem to be for a 'forward' presentation with a good dynamics and tight punchy bass(?). Detail is to be preserved wherever possible and the sound is not to be 'bright' or hard at high volume levels, at which the system should excel.

Doesn't sound much to ask, just the Earth and half the known Universe, is all. Anyway, never being one to refuse a challenge, here goes. Five hundred quids worth of window-rattling, cat-killing, neighbour-baiting, divorce-inducing rock. Loud, clean and detailed. I've tried to avoid the obvious few brand names that are consistently trotted out, on the grounds that it's high time we admitted that not *all* systems begin in Scotland and end up the same.

Lining Up For Lines

Our line-up is: Thorens TD166 II (with either Shure M97EJ or Coral MC88E cartridges), Trio KA-50 amplifier and JBL Radiance R82 loudspeakers. The option on cartridges is to allow a little 'fine-tuning' to individual taste. The two cartridges sound totally different, but both offer excellent compatibility with the JBLs.

The Shure provides a more forward and exciting presentation, while the Coral is better at retrieving mid-range detail and has a slightly better bass control. I was happy with both. Overall the system will cost you just under £500 if you shop prudently. Annoy a few retailers and see what discounts you can get on the whole system.

As both the cartridges in use have been previously reviewed in these pages little comment is needed upon their various merits, save to quote test results and system balance findings.

All Decked Out

On the basis of starting at the beginning and going on to the end, let's look at the Thorens TD166 II first. This is a new addition to the '160' range of turntables and retails at around £100, complete with TP11 II pickup arm.

Mechanically, the deck has a lot in common with its big brother. The basic turntable assembly is sprung on a sub-chassis in a manner identical to that of the 160. I can find few differences beneath the base plate at all, in fact.

It would appear that the 166 II may well be in for a career of being purchased, only to have its arm amputated! Thorens decks in the past have often suffered this fate, right back to the faithful TD150, which performed sterling service in many a system (and still is in some). From my tests on the bench, and from listening to the deck in both this and a reference system, it seems that the 166 II would be excellent value even if it did not come complete with arm. The sound quality is very good, noise is low and unobtrusive and the floating chassis confers good isolation upon the pickup system as a whole.

The arm arrives packaged in a plastic holder, which turns out to be a mounting gauge as well. Two months ago no manufacturer did anything remotely like this, and now both Shure and Thorens have produced products which use the packaging to align the cartridge. To all things under heaven there is a season . . . etc.

The Thorens gauge works fairly well, but being a cynic I'd still advise the use of an alignment protractor to check setting up on the turntable. These adjustments are so critical to the final sound that anything that remotely resembles a compromise should be viewed with curled lip and raised eyebrows.

Mat Matters

The TD166 is very well constructed, with a good 'feel' to it. The lift/lower control operates smoothly, if a little slowly. The arm appears to be a rigid enough structure to handle medium or high compliance cartridges and exhibited commendably low friction in both vertical and horizontal planes.

The headshell has an ingenious slotted arrangement to allow adjustment of the cartridge, but I'd like to see more contact area between cartridge body and arm, to allow better control of energy transmitted back from the stylus. This is unlikely to be problem with anything other than low compliance cartridges, however, and as you shouldn't be using these anyway, serve you right if you hit trouble.

Mustn't forget the standard gripe about the ribbed Thorens turntable mat. *Surely* it can't be easier to make these fancy indented things than a nice simple solid piece of rubber with *one* indent in the centre to accept the label?

After listening for a while with the standard mat, I replaced it with a 'Dumpa' from my own TD160S, to be returned an improved mid-range and cleaner top end. The Coral MC88E picked out the difference very clearly and suggests that if you go for this pickup, it might be worth raiding the piggy bank to get a better mat.

A de-grilled Radiance R82 — the acceptable face of hi-fi. On the backside, so to speak, you'll find the three-position treble control and some unusual rotary-locking speaker terminals.

Silk Purses Again?

The TD166 II is excellent value and is probably the best turntable and arm available in its price range. I wouldn't be suggesting all these little changes if it wasn't. No good fiddling with the listening apparatus of live pork in an attempt to generate natural fibre cash storage equipment, after all.

The arm is capable of carrying pickups of up to around £50 in value with distinction and the turntable is easily up to being used with a better arm, once the owner begins to aspire to a fi of greater elevation. The obvious candidate that comes to mind is the SME Series IIIS.

Overall a very good product indeed and a worthy addition to the Thorens range.

Briefs On Pickups

Just quickly past the two pickups, then: Shure's M97EJ features much of the technology of the V15 IV line, with its stylus damper and 'side-guard' protection system. An elliptical stylus is employed and tracking proved best at around 1.75 g (at the stylus — add 0.5 g for the stabiliser assembly). Compliance is fairly high (over 20 cu anyway) and gave no resonance problems with the Thorens.

Coral's MC88E is a low-cost high-output moving coil device, having a lot in common with the MC81, their best known pickup. The 88E also uses an elliptical stylus, but needed 2.2 g to track cleanly in the Thorens arm. Compliance is lower than that of the Shure and again produced no problems with bass resonance. Both are a good match for the deck.

There is little to choose between them in respect of output voltage either, and the Trio amplifier coped with both easily. Speaking of which . . .

Trio KA-50

With the high efficiency of the R82s, high volume levels need not demand hundreds of watts. Peak delivery still needs to be high, though, to ensure clean transients and that good bass control demanded as part of the intitial specification.

A power of around 50-60 W was though to be adequate, provided the PSU of the chosen unit did not pack up and go home when asked to play peaks.

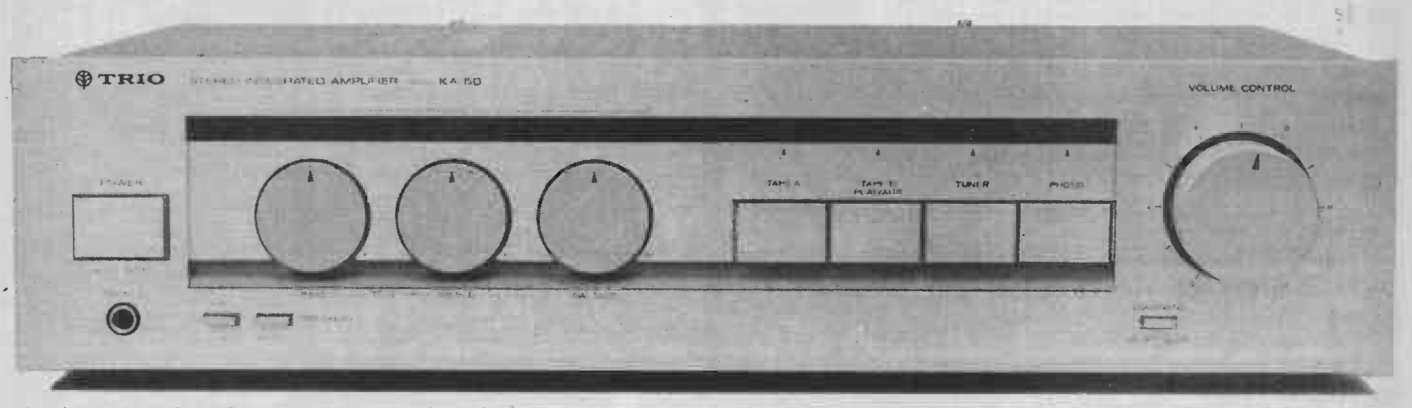
The eventual choice was the Trio KA-50, rated at 45 W RMS, but easily exceeding that on all counts. There is no shortage of good amplifiers in this price range and the Trio is a solid, well-made product with a creditable performance and sound quality. As you can see from the photos, the appearance is a little different and you will either love it or hate it. The seemingly inevitable peak meters (LEDs) are at least fairly well integrated to the front panel.

On test the KA-50 delivered 56 W into 8 R from 20 Hz to 20 kHz, both channels driven. Peak power output was around 85 W using my normal tests, although I did go through a few fuses getting that reading! Delivery is well maintained into low impedance loads and distortion is suitably non-existent ($<$ 0.06% THD; $<$ 0.03% IM). Output impedance wandered about a little with frequency, but the damping factor never fell below 30, even down into the bass (50 Hz).

Technically impeccable then — but since when have the Japs made something that wasn't?

Sound Stuff

In use the amp controlled the R82s well and contributed little to the overall sound of the system, as a good amp should. The mid-range, judged against absolute (and far more expensive) standards is a little muddy at times, and highly complex sounds can get a little



The front panel of the Trio amp has clean lines and a sort of classical simplicity. The photo below shows how tidy the inside is, too.

confused. For the money, though, a very good performance is offered, comparable to those NAD boxes which are (rightly) so highly rated.

There is little more to say about the KA-50 really; it did its job well, gave me no trouble and drove the speakers to budgie-killing levels. What more can one ask?

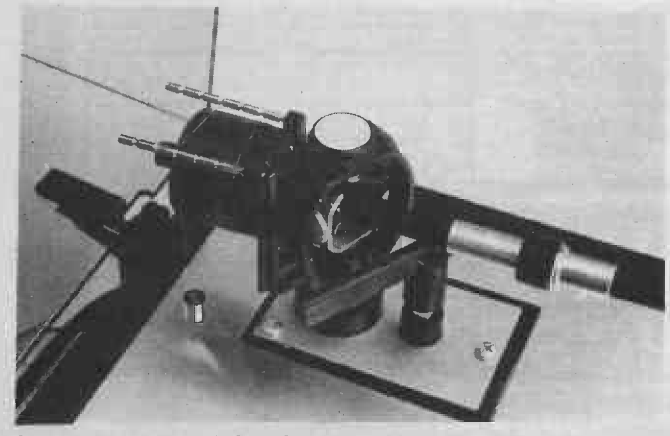
Speakers To Speak Up

JBL's Radiance series is well known by now, especially amongst those who like their music loud. The units chosen for this system, the R82s, are of reasonable proportion ($21\frac{1}{4}$" high by $13\frac{1}{2}$" wide by 11" deep), and are a two unit design filled with three position treble control. It was the JBL's I tried first, disconnecting my more stately KEF 105 IIs to wire them up. Rather like replacing Lord Denning with Perry Mason really — one can be relied on for the truth, the whole truth and nothing but. The other makes life interesting, if only at the expense of absolute honesty!

The R82 proved a good all-round speaker, with a very distinctive character. It is forward and a little inclined to aggression if fed low-quality material. The bass is outstanding for the size of enclosure and was cleaned-up considerably by employing open stands of around 12" height.

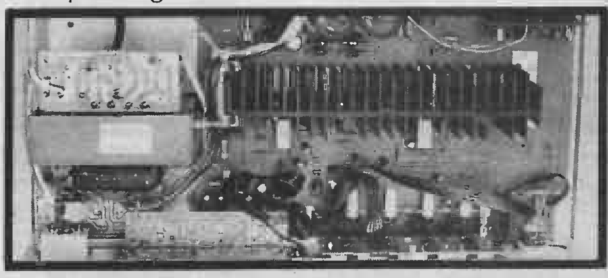
The treble control I never used anywhere except in 'normal' setting — it is only a marginal adjustment and makes little real difference to the final sound.

Being of the ported or bass reflex design, the R82 needs an amplifier which is capable of controlling it well. This normally means a high damping factor, to absorb the back EMF generated by the bass cone. The Trio coped well, although as would be expected, the vastly more expensive Denon PA 3000 showed that the JBLs are capable of truly outstanding bass response with an amp of very rigid control.



The pivot system of the Thorens arm — which you may wish to amputate!

In summary, the JBL can be described as an exciting speaker to listen to with good dynamics, well extended bass response and a clean top-end. The stereo image does seem to move about a bit unless you're exactly on-axis to the tweeter, however. Whether you consider this a problem will really depend on the room in which the R82 will be operating.



Systematically

Taken together I feel this system has a lot to offer for that £500. Lovers of 'middle-of-the-road' or chamber music are unlikely to get on with the presentation, which is difficult to ignore, to put it mildly! Anyone who *listens* to records will find a wealth of detail pushed out from the speakers in a very entertaining manner, which is neither bright nor hard (as some skeptics might suspect).

Ample Amplifiers

Next month it's back up the scale again, to look at some high-power (high-cost) amplifiers. We have been working on a comparison of some of the bigger beasts around for a while now and I hope to report in detail next time out.

The units concerned at present — and I will add more if Mission and others deliver in time — are: Hitachi 9500 II (a 150 W per side N-channel MOSFET design), Denon PA 3000 (a 180 W Class A monster that is huge and magnificently constructed), Crimson CK1000 (100 W of low-cost power), Carver's Cube (200 W and too small to believe) and hopefully our own System A amps for comparison. Enough people have asked why we don't review them for me to insert them into the blind listening test to see where they come out!

The same preamp is to be used for all the tests and I've chosen the Denon PRA 2000, which is my opinion is the best on the market, bar none. A detailed review of that next month, too.

The idea behind this assembly of high priced watts is is to see just how far amplifier technology has come and to compare the different methods of obtaining high power.

God knows what's going to happen; I don't! I can only promise it will be interesting. The potential disasters are numerous and the chances of everything going smoothly, negligible! **ETI**

DESIGNING MICRO SYSTEMS PART 2

Bemused by those funny-looking groups of hexadecimal numbers that microprocessors use for a language? From voltages to binary, binary to hex, and hex to English — Owen Bishop does ,the translating.

In describing the many and complex circuits which go to make up the CPU and its peripheral devices we often refer to **information** being transferred from one to the other. Last month, it was explained that such information consists of either **instructions** or **data**. In order to understand how the several parts of a computer interact to form an operating system we need to go into more detail about these instructions and the data. In this month's instalment we pause from considering electronic aspects in order to discuss the nature of the information and the form in which it is transferred.

Information Technology?

The MPU and other integrated circuits composing the micro respond to only one kind of information. This is an electrical signal, a voltage level present on one of its input lines. To the input circuit of the IC, this level is either 0V (or so close to 0V that it counts as 0V), or +5V (or high enough to count as +5V). An instruction given to a Z80 MPU is received as a set of such voltages on each of the eight lines of the data bus. For example, the Z80 might receive a set of signals as set out in Table 1. When (and only when) these signals are present, internal logic within the Z80 is set and causes every flip-flop of the accumulator register to change state. Every 'set' flip-flop is reset, every 'reset' flip-flop is set.

The description above is in purely electronic terms, which is reasonable enough, for a Z80 MPU is a purely electronic device. When we use it in a computer and communicate with it through the agency of a keyboard, using a high-level language such as BASIC, we tend to forget that it is only a rather complicated electronic circuit. In order to make it do anything at all we have to communicate with it by sending it information (ie what we want it to do) in terms of electrical signals.

Coding For Clarity

Having looked at this from the MPU's point of view, let us look at it from ours. We are somewhat cleverer than CPUs and can take advantage of this to make things simpler for ourselves. For example, it is a cumbersome procedure to state: "To make each 'set' flip-flop become reset, and each 'reset' flip-flop become set, place +5V on line D0, +5V on line D1, . . ., and 0V on line D8." We need a way of symbolising the actions we want performed on the flip-flops, and we need a way of symbolising the signal levels to be applied to the data bus. In short, we need **codes** which will relate our requirements to the electronic activities of the computer. Mathematics provides us with ways of coding operations performed on the accumulator or other registers. The information sent on the data bus is coded in another way, known as **machine code**. These codes make it much easier for us to follow the workings of the computer, to tell it what to do and to discover what it has done. They make it easier for us to discuss computer activities among ourselves. But keep in mind that these codes are used for *our* convenience to symbolise events which are essentially electronic.

TABLE 1

DATA LINE	VOLTAGE LEVEL
D0	+5 V
D1	+5 V
D2	+5 V
D3	+5 V
D4	0 V
D5	+5 V
D6	0 V
D7	0 V

Bits and Bytes

As mentioned last month, the actions of the electronic circuits of a computer are binary in nature. Voltage levels are 0V or +5V; intermediate voltages are not recognised. Flip-flops are either set or reset; there are no other stable states. Consequently, the simplest way of symbolising voltage levels is to represent them in binary form.

Conventionally, 0V is represented by the numeral '0', while +5V is represented by '1'. Using this convention, we only have to write out a row of eight such binary digits to represent the eight voltage levels on the data bus. In doing this we use another convention, that the digits refer to lines D0 to D7, in numerical order, written from right to left. We usually split the eight binary digits (or **bits**, a shortened form of the words '**BI**nary digi**T**') into two groups of four bits each. A group of eight bits is referred to as a **byte**. We can write out the voltage levels of Table 1 as a single byte:

0010 1111

A set of bits representing a set of voltage levels which are interpreted by the MPU as an instruction is called an **op code.**

The action of flip-flops is binary too — we can represent the state of the flip-flops of the accumulator or other registers as a set of bits. The accumulator of the commonly-used MPUs has eight flip-flops, so its contents may be represented by a byte. The Z80 and 6502 are of this type. If the contents of the accumulator are, say, '0101 1100', the op code 0010 1111 instructs the Z80 to change every '0' into a '1' and every '1' into a '0', so that the contents become '1010 0011'.

Such an operation has a mathematical name; we call it **complementing**. If we symbolise the accumulator contents as A, and the inverse of its contents as $\overline{A}$, the whole operation can be represented by the equation:

$$A \leftarrow \overline{A}$$

(A is replaced by $\overline{A}$).
We can now state what happens in much simpler (to us!) terms:

Op code 0010 1111 causes operation $A \leftarrow \overline{A}$

To the Z80, it is still a set of voltages causing a change of flip-flops.

Helpful Hexadecimal

Putting op codes into binary form is certainly simpler than referring to voltages in terms of their actual values, but it still has features which are inconvenient. For instance, writing out strings of 1s and 0s is tedious, and readily subject to error. It is not easy to notice the difference between 1001 1101 and 1001 1001, and the situation becomes worse when we are dealing with 16-bit codes. Although many of the early systems used switches to feed in a series of bits to the data lines, we prefer not to have to key in each bit separately. There is a simpler way of doing this.

TABLE 2

BINARY	HEXADECIMAL	DECIMAL
0	0	0
1	1	1
10	2	2
11	3	3
100	4	4
101	5	5
110	6	6
111	7	7
1000	8	8
1001	9	9
1010	A	10
1011	B	11
1100	C	12
1101	D	13
1110	E	14
1111	F	15

The hexadecimal code is a short-hand way of representing the binary code. More than this, it is a number scale in its own right. Whereas the binary scale is based on powers of two, and has only two kinds of figure (0 and 1), the hexadecimal scale is based on powers of 16 and has 16 kinds of figure. We already have 10 kinds of figure (0 to 9) available in our common decimal scale and, rather than invent six new symbols, we have adopted the first six letters of the alphabet for use as figures. Table 2 shows how this is done.

By using hexadecimal coding we are able to save time and confusion in writing out lists of instructions for the computer (programs). We are also able to make use of a 16-key keyboard for entering these instructions quickly. Against these advantages there is the disadvantage that we begin to lose sight of the binary nature of the operation of the computer. The individual 0s and 1s no longer appear in our reckoning and we are one stage further removed from the working procedures of the MPU.

One source of confusion in using several number scales is that we must be careful to state which scale we are using. For example, the digits '11' may represent 'three', if they are in the binary scale, 'eleven' if they are in the decimal scale, or 'seventeen' if they are in the hexadecimal scale. In this article we will write all binary numbers in blocks of four bits, and begin them with at least one zero. All hexadecimal numbers will by followed by the letter H. Thus '0011' represents 'three', '11' represents 'eleven', while '11H' represents 'seventeen'.

Giving Instructions

A byte can represent any number from zero (0000) up to 255 (0000 1111 1111). It is therefore possible to specify up to 256 different instructions for the MPU, using just one byte. Such a set of op codes is called the **instruction set**. Examples of the instruction set of the Z80 are listed in Table 3. The Z80 can respond to more op codes than listed there; in fact, its instruction set contains over 500 different instructions. This is more than 256, so some of these are coded by using two bytes. Such a large instruction set makes it a very powerful MPU.

On the other hand, the 6502 has a relatively restricted set, consisting of only 148 instructions, yet remains a popular MPU in spite of this. There is no doubt that the limited set makes it easier for a programmer to get to know how to use the capabilities of the 6502 to the full. Needless to say, the op codes used by different types of MPU do not mean the same thing. When a Z80 receives the op code 25H it subtracts 1 from the contents of its H register; in other words, it decrements H. If you present this same op code to a 6502 it performs the logical AND operation on all bits in the accumulator.

Taking Several Bytes

The previous example raises the next point to be considered. An operation such as 'complement' or 'decrement' involves only the register concerned. When we ask the 6502 to AND its accumulator, the MPU must be given something to AND it with. The same applies for instructions such as 'add'. If something is to be added to the accumulator, we must tell the MPU what to add.

TABLE 3

Z80 OP CODE	INTERPRETATION
00	Do nothing
04	Increment register B
05	Decrement register B
25	Decrement register H
2F	Complement accumulator
5A	Load register E with contents of register D
5E	Load register E with the contents of a memory register, the address of which is stored in registers H and L.
76	Halt
FB	Enable interrupts

The instructions which have been mentioned earlier have consisted of a single op code. Most instructions require the op code to be followed by one or more bytes to supply the data upon which the MPU is to operate.

As an example, consider the op code 25H, which tells the 6502 to AND its accumulator with something. When the 6502 has received this instruction, the next byte fed to its data bus inputs must be a byte which specifies what to AND the accumulator with. The 25H op code is what is known as a **zero-page** op code in that the single byte which follows is to be taken as an address located in page zero of memory. Since the address bus has 16 bits, it requires a double-byte to specify any given address. Zero-page addresses in a 6502 system are those which begin with 00H, for example, the address 00 32H. With zero-page addressing we do not need to send the MPU the first byte (00H) for it already knows from the op code that a zero-page address is forthcoming. We need only send the second byte (32H). Thus the full instruction is '25 32', consisting of op code (25H) followed by a single **operand** (32H). On receiving the second byte the 6502 fetches the contents of address 00 32H from RAM, and ANDs it with the contents of the accumulator, leaving the result of this logical operation in the accumulator.

Using The (Post?) Code

Zero-page addressing is a feature of the 6502 which allows a certain range of addresses to be passed to the MPU in an economical way. As a second example, suppose that the value to be ANDed with accumulator was stored not in page zero (00 32H), but at an address higher in memory, for example, 2D 32H. This address needs *two* bytes to specify it, so after the MPU has received the ANDing op code, it must wait for the next two bytes before proceeding to carry out the instruction. The op code for this procedure is now 2DH (0010 1101) instead of 25H (0010 0101) as before. Looking at the binary code (which is what the MPU is looking at), we see that data line 3 is at +5V now instead of at 0V, as before. The effect of this is to make the MPU wait for two bytes and put them together to make up the full 16-bit address. Then it addresses that unit of memory, fetches the value stored there and ANDs it with the value present in the accumulator.

This second example is a triple-byte instruction, and would be written out as '2D 2D 32'. The first byte (2DH) is interpreted by the MPU as an op code. The next, though it has the same hexadecimal value, is not taken as an op code but as the first byte of a double-byte address. The MPU is able to distinguish between instructions (op codes) and data (eg addresses) by their context. It is rather like our ability to distinguish between the meanings of the word 'lead' in these two sentences:

1) He mended the lead pipe from the kitchen sink.
2) He played the lead guitar in the local pop group.

We know which way to interpret the word 'lead' by its context; by its relation to the other words which occur with it in the same sentence. Similarly, the op code 2DH and the partial address 2DH give rise to precisely the same set of voltages on the data bus, but they are interpreted differently by the MPU according to what has gone before. Information on the data bus can therefore be an instruction (op code) *or* data (eg an address).

For Immediate Attention

An address is one kind of data, but this is not the only kind. We have mentioned that the MPU fetches a value from a location in memory in order to AND it with the accumulator. This value is transferred from that location to the MPU as a byte on the data bus. Data can include values of this kind, to be used in arithmetical or logical operations, or it might be a value which is part of an instruction. For example, the op code 29H is yet another instruction to perform the AND operation, but this one ANDs the accumulator with the next value to appear on the data bus. This is often called the **intermediate mode**. Thus the instruction '29 16' tells the MPU to AND the accumulator with the value 16H. The MPU knows that 16H is a value, not part of an address, because of its context: it follows the op code 29H.

The op code D0H causes the value which follows it to be added to the value held in the program counter register. The effect of this is to make the MPU jump from one part of its program to another. This op code is only obeyed if a given condition holds true. If the result of the most recent operation has left 00H in the accumulator, the jump is effected. If not, the instruction is ignored.

Talking To The Chips . . .

Ultimately, the only way of sending information to the MPU is by means of voltages on the data lines. The only way that the MPU can send information to other parts of the micro, and to the world outside (including us) is the same. In the simpler micro systems, including most of those specialised systems used in control applications (eg automatic washing machines and other robots), all communication is at this level.

If the system has a keyboard, it is often a 16-key hexadecimal one with perhaps a few additional function keys. Readers may remember the Sinclair MK-14, the popular but primitive forerunner of the ZX80 and ZX81. This had a 16-key keypad and a reset button. Visual output was by means of an eight-digit seven-segment LED display, which displayed the figures 0 to 9 and the letters A to F. Unless you were using the tape-recorder input, one had to load a program by laboriously keying in machine code. To write one's own programs it was essential to master the machine code of is MPU, the National SC/MP 8060.

. . . In Machine Code . . .

Nowadays such systems are mainly used for industrial control applications (eg Acorn System 1) and for development of programs to run on other such systems (eg Softy). Most people prefer to be able to instruct the MPU by using a higher level language, such as BASIC, and to receive its output in graphical form on a monitor screen.

At the input and output to the MPU only machine code can be used. Since it is directly understandable by the MPU, a machine code program runs faster than one in a higher level language; it also takes up less storage space in memory. The monitor program stored in ROM (see next month's article) is in machine code. It is common for complex and lengthy software to be in code, simply to make it possible to cram it into RAM. An example is the SCRIPSIT word-processing program which is being used in preparing this article.

Many games programs such as Adventure games and the several versions of Chess are in code. Great complexity can be packed into a reasonable amount of memory and they do not take long to respond to the player's commands. Many utility programs (eg renumbering programs) are written in code because, although they are short and simple, they must be made as compact as possible in order

As an example of the MPU as a general purpose logic chip, we'll consider one operation. The logical operation called AND can be summarised like this:

INPUTS A	INPUTS B	OUTPUT Z
0	0	0
0	1	0
1	0	0
1	1	1

Output Z is 1 if (and only if) both input A AND input B are 1, otherwise Z is 0. Many readers will be familiar with the TTL or CMOS logic gates which perform this operation for two (or more) inputs. The 74LS08 has four two-input AND gates; the CD4081 is its CMOS equivalent. When the Z80 CPU receives the instruction 29H, its logic circuits in the arithmetic logic unit (ALU, see last month) are configured as eight two-input AND gates. Each gate deals with one bit. The eight bits are ANDed simultaneously: if the accumulator holds

0010 1101

And the next byte on the data bus is

1011 0100

then the result of ANDing the corresponding bits is

0010 0100

This result then replaces the value previously held in the accumulator.

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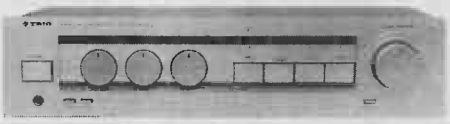

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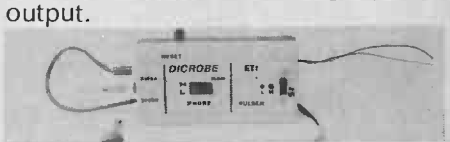

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to leave plenty of vacant space in memory for the programs on which they are to operate.

. . . With An Interpreter . . .

One factor in the rise in popularity of the microcomputer is the ease with which people unskilled in (or even totally ignorant of!) machine code are nevertheless able to program the MPU and make it do their bidding. This involves the use of a high-level language, the most popular of which is BASIC. When a BASIC program line is typed in, it is stored in memory. To save space, the words are usually coded so that each word or variable requires only one byte; the handbook usually lists the system of coding used. This gives yet another possible interpretation of a byte on the data line. For example, in the ZX81, the code FBH may be interpreted as:

- An opcode — enable interrupts
- A value — equivalent to 251 in decimal
- A code for the BASIC command 'CLS' (clear screen)

The code is interpreted according to context, which includes the mode in which the computer is operating at that time.

Once a BASIC program has been loaded and the RUN command given, the MPU reads the program from memory one byte at a time. It cannot operate directly on the bytes as it reads them. If the byte is FBH, for example, there is no way in which the MPU can directly clear the screen. At this stage the MPU is under the control of a special program called a BASIC **interpreter.** The interpreter contains the complete set of machine code instructions for the operation of clearing the screen. When FBH has been read, the MPU goes back to the interpreter program to find out what to do and how to do it.

Although the MPU is working just as fast as ever, it has to go back to the interpreter program at every step in order to find out what to do. This is why a program takes so much longer to run when it is written using a BASIC interpreter. The interpreting is done line by line and, moreover, has to be repeated every time the program is run.

. . . Or By More Basic Means

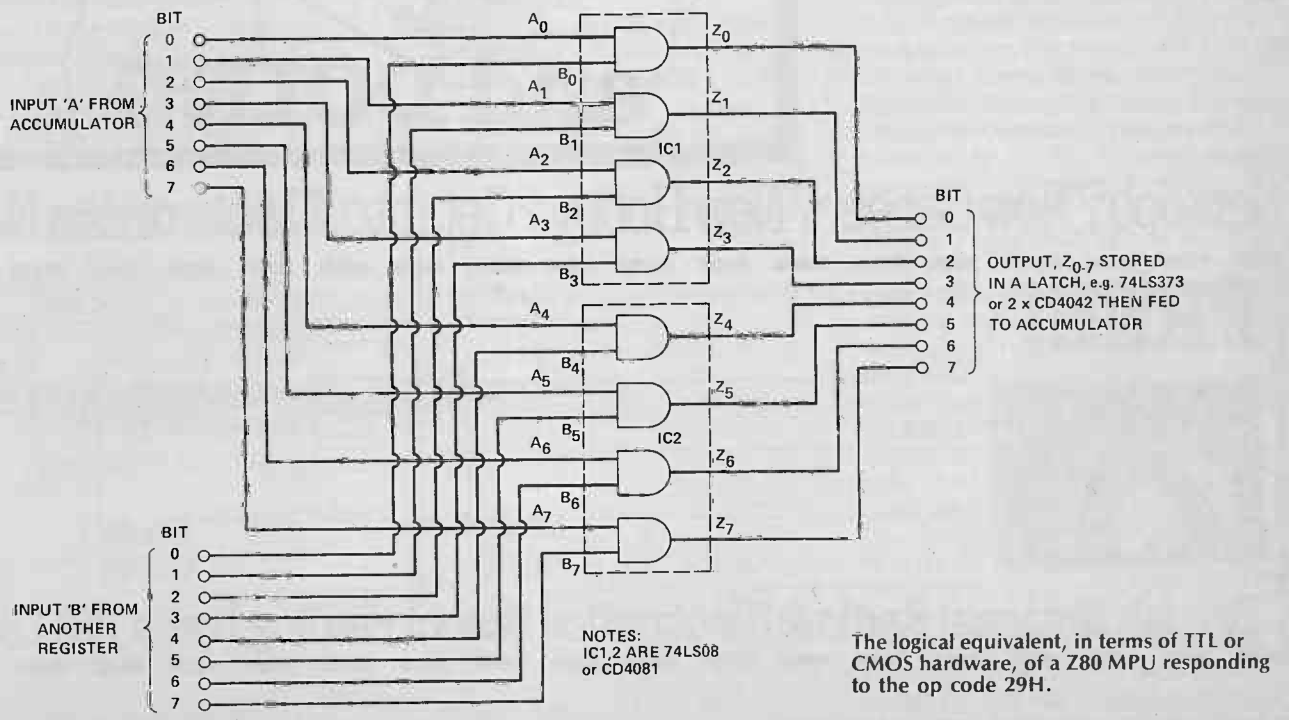
A faster method is to use a BASIC **compiler** program. Once the program has been written, the compiler goes through it and converts it into machine code. This is done once and for all, after which only the machine code program is used. The difficulty with this approach is that the program must be recompiled if even a small change is to be made. For this reason most people prefer to put up with the slowness of an interpreter.

Many microcomputers have the BASIC interpreter already stored in ROM, so that it is available immediately the micro is switched on. Other micros have no resident language program. If you want to use a high level language, you have to load the interpreter or compiler program into RAM before you are able to enter your own programs in the high level language.

Those who prefer, or are forced for various reasons, to program in machine code can make use of yet another type of program called an **assembler.** Instructions to the MPU are typed out in a symbolic form, consisting mainly of abbreviations of the operations which are to be performed. These abbreviations are generally known as **mnemonics,** which means 'to help the memory'. For example, 'decrement register H' is written as 'DEC H'. When the assembler is run, the op code 25H is assembled into the machine code program. The assembler can convert decimal numbers to hexadecimal, removing a constant source of headaches for the programmer, and can calculate the hexadecimal values required in jumping from one part of memory to another. Provided it is well written, an assembler can be of great assistance, yet it is not too far removed from machine code. The programmer is dealing with particular registers within the MPU and specifying each step of operation of the MPU. If it is *not* well written (and some assemblers are not), then one might just as well learn the commonly-used op codes, have a table of op codes handy, and work out the hexadecimal values on a scrap of paper. Once you have gained a little insight into its peculiarities there is really nothing quite as satisfying as talking direct to your MPU in its own language!

ETI

LOGICAL OPERATIONS IN THE MPU



The logical equivalent, in terms of TTL or CMOS hardware, of a Z80 MPU responding to the op code 29H.

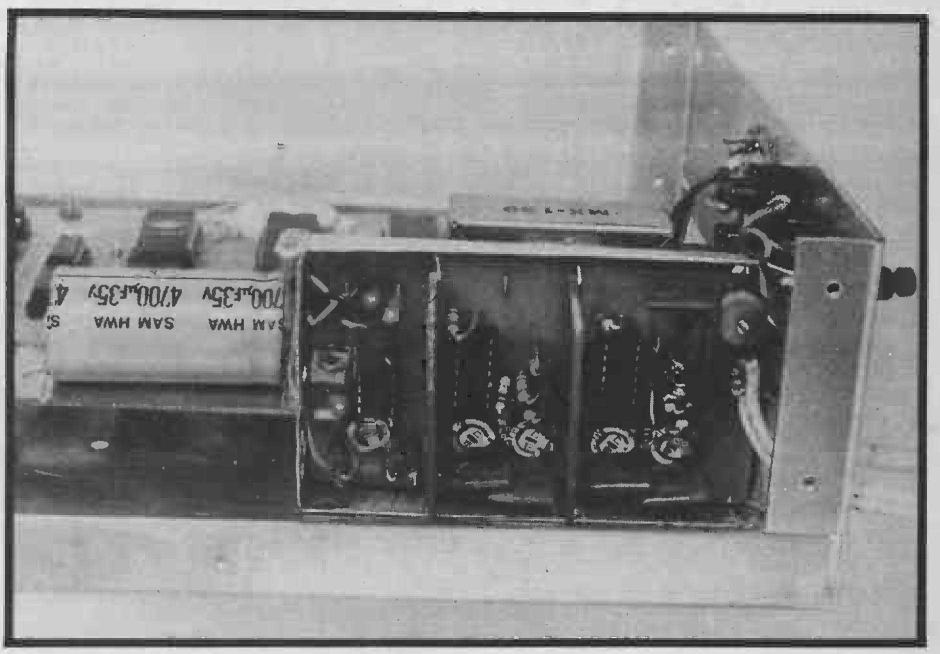
RUGBY CLOCK PART 2

MSF Rugby is your personal atomic clock/calendar. We conclude this project with the full circuit and construction details. Design and development by Stephen Makumbi.

Construction should be straightforward for a reasonably experienced constructor, but if you haven't constructed a kit before, it is recommended that you buy the ready-built version. The PCBs are double-sided and plated-through so you'll have to buy those even if you want to assemble the components yourself (see Buylines).

It is essential to use a low power soldering iron (15 W). When soldering plated-through holes it is not necessary to have small mountains on top of the joint; sometimes a little crater is satisfactory. The way to do this is to apply the soldering iron on the joint to be soldered, leave it there for about 3 seconds then apply a very small amount of solder. The solder will be drawn through the hole (capillary action), and provided the hole looks filled, the joint should be strong enough.

Starting with the main board, solder all IC sockets then the rest of the components, with the crystal last. Fit the transformer and attach the mains lead but do not insert the ICs yet. Note that one of the transformer fixing screws is attached to the chassis so you will only fit the other screw at this stage. Fit a mains plug with a 3 amp fuse and connect the earth lead to the wide earth track on the PCB round the transformer. Connect the mains and switch on. Check that all the voltages from the power supply are correct.



The receiver board, mounted in its screened enclosure.

On Display

Next the display board. Follow Fig. 1a to fit the edge connector. Count off 25 soldercon pins and cut the strip; do not remove the top connecting bar yet. Solder the pins into one row and do the same for the other. Now, using long nosed pliers carefully twist the top connecting bar of the pins from side to side until they snap off.

Follow Fig. 1b to fit the switches. They remain hanging in the air until fixed to the front panel of the enclosure.

Next comes the receiver. This should also be quite straightforward. Fit all components following the component overlay diagram, again starting off with the IC sockets.

The aerial is wound on a standard long wave ferrite rod; the long wave winding is wound round a cardboard former. Remove the winding and on the same cardboard wind 400 turns (over the entire length) using 26 swg wire. Use the metal tags of one end to secure the ends of the winding. Attach to this at least a one metre length of double core insulated cable, and solder the other end to the receiver board (secure the screen to an earth track). It is highly recommended that the receiver is fitted inside a metal box for screening purposes. Also two partitions made out of PCB board should be fitted inside (see overlay diagram). If a box is used then these should be soldered on to the sides of

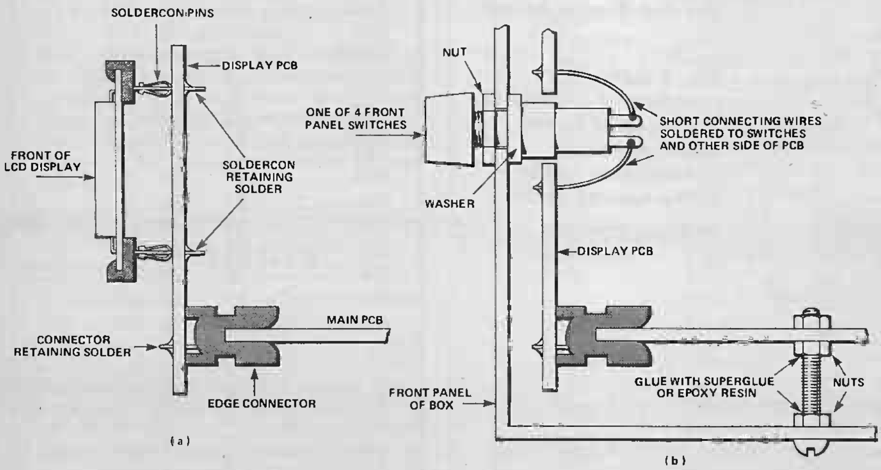


Fig. 1 How to mount the LCD display (a) and the switches (b).

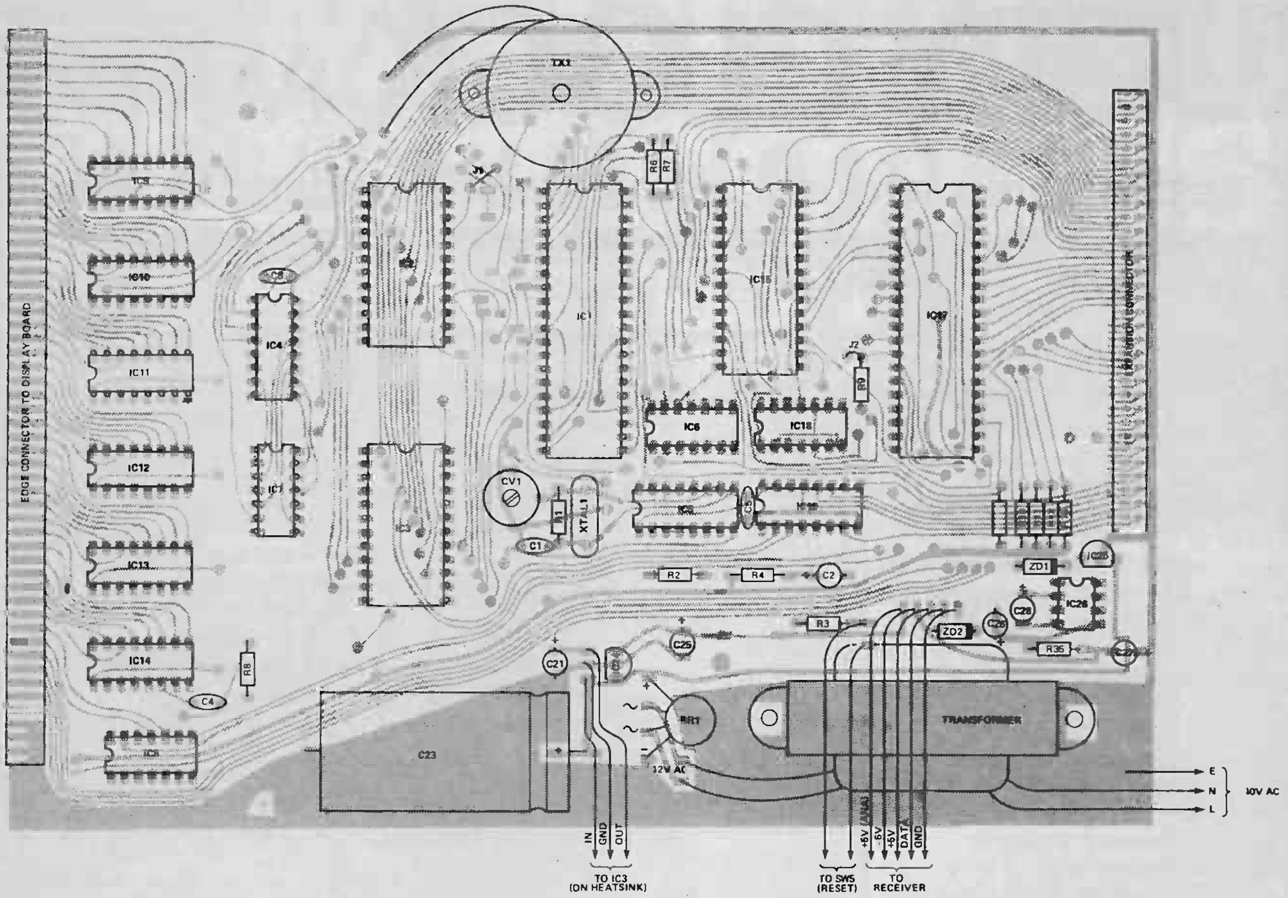


Fig. 2 Component overlay for the main board of the ETI Rugby Clock.

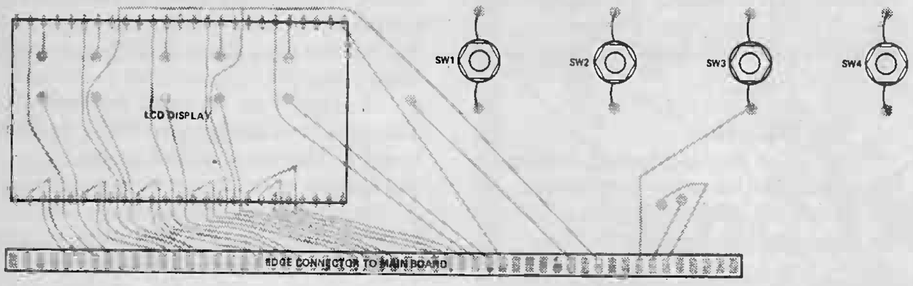


Fig. 3 (Above) Overlay for the display board.

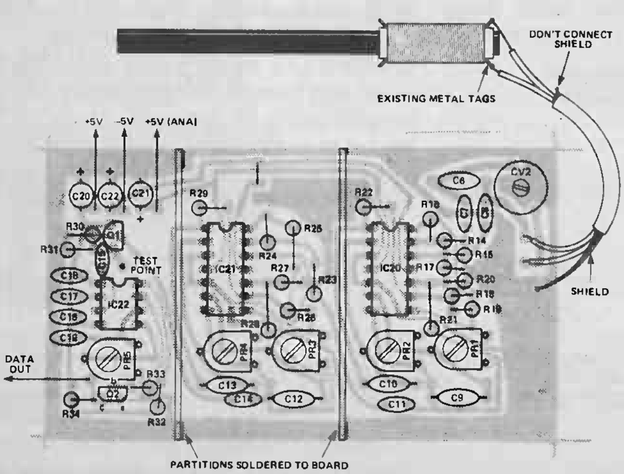


Fig. 4 (Left) The component overlay for the receiver board, showing the connections to the home-wound aerial.

EXPANSION SOCKET PINOUT

$\overline{\text{CS}}$ '0C'	63	64	$\overline{\text{CS}}$ '08'
$\overline{\text{CS}}$ '14'	61	62	$\overline{\text{CS}}$ '10'
$\overline{\text{CS}}$ '1C'	59	60	$\overline{\text{CS}}$ '18'
	57	58	
	55	56	
	53	54	+V (UNREG)
GND	51	52	
	49	50	
	47	48	$\overline{\text{RESET}}$
B $\overline{\text{RDY}}$	45	46	$\overline{\text{INT}}$ OUT
$\overline{\text{INT}}$	43	44	B $\overline{\text{STB}}$
PA3	41	42	PA4
PA5	39	40	PA6
$\overline{\text{IORQ}}$	37	38	$\overline{\text{M1}}$
$\overline{\text{RD}}$	35	36	$\overline{\text{MREQ}}$
$\overline{\text{NMI}}$	33	34	$\overline{\text{WAIT}}$
PB0	31	32	PB1
PB2	29	30	PB3
PB4	27	28	PB5
PB6	25	26	PB7
D0	23	24	D1
D2	21	22	D3
D4	19	20	D5
D6	17	18	D7
A0	15	16	A1
A2	13	14	A3
A4	11	12	A5
A6	9	10	A7
A8	7	8	A9
A10	5	6	A11
A12	3	4	A13
A14	1	2	A15

PARTS LIST

Resistors (all ¼ W, 5%)	
R1,2	1M0
R3,31,33	1k0
R4-7,9-13, 18-20,22, 25-27,29	10k
R8,14-17	100k
R21	300k
R23,24,30	2M2
R28	470k
R32	2k2
R34	4k7
R35	10R
Potentiometers	
PR1-4	47k miniature horizontal preset
PR5	10k miniature horizontal preset
Capacitors	
C1	30p ceramic
C2,20-22, 24,25	2u2 16 V tantalum
C3,5,14-16	22n ceramic
C4,18	10n ceramic
C6,9,10,11-13	100p silvered mica
C7,8	50p ceramic
C17,19	4n7 ceramic
C23	4700u 35 V axial electrolytic
C26	100u 16 V tantalum
C27,28	47u 16 V tantalum
CV1	30p trimmer
CV2	100p trimmer
Semiconductors	
IC1	Z80 CPU
IC2	2732
IC3	4118
IC4,19	74LS138
IC5	4049B
IC6	74C00
IC7	74C04
IC8	4070B
IC9-14	4056B
IC15	Z80 CTC
IC17	Z80 PIO
IC18	74C90
IC20,21	TL074
IC22	NE567
IC23	7805 (TO3)
IC24	78L05 (TO92)
IC25	78L12 (TO92)
IC26	ICL7660CPA
ZD1	2V7 400 mW zener
ZD2	5 V 400 mW zener
BR1	1A5 bridge rectifier
Miscellaneous	
XTAL1	2 MHz crystal
SW1-5	miniature push-buttons
TX1	PB-2720 buzzer

PCBs (see Buylines); LCD six-digit display; 12 V 500 mA mains transformer; single 54-way 0.1" edge connector; dual 32-way edge connector; 26 swg enamelled copper wire; long wave ferrite rod assembly (3/8" diameter); IC sockets and soldercon pins; case to suit.

BUYLINES

A complete kit of parts including the PCBs and case will be available from Technomatic Ltd, 17 Burnley Road, London NW10 (phone 01-452 1500). Technomatic will also be supplying ready-built units. Prices are £120 plus VAT for kit, £145 plus VAT for the ready-built version. People who do not wish to buy the complete kit can get the EPROM for £12 and the set of three PCBs for £25.

the box, otherwise they should be connected to the edges of the PCB (earth track) with short pieces of solid core wire. Finally, make all the interconnections between receiver and main board.

Setting Up

You will need an oscilloscope and possibly an accurate signal generator. Insert all ICs, connect the display board and turn off your television. Move the aerial winding about 3 mm from the end of the ferrite core, and extend the cable fully. Turn PR1-4 to about "5 minutes past", then turn PR5 to about ¼ turn from left (see layout diagram).

Switch on the mains. If the buzzer sounds press the reset button. The display should be filled with 6 dashes and two colons should be in the middle.

If you have an accurate signal generator, set it to precisely 60 kHz and wrap the output probe round the aerial. A very small amplitude indeed is all that is required. Then adjust PR1-4 for maximum amplitude with the probe at the test point. If you do not have access to a signal generator then put the oscilloscope probe at the test point and adjust PR1-4 until a pulsating signal appears. Adjust the position of the aerial winding for maximum output, and then continue adjusting the presets for maximum amplitude.

The minimum signal should be about 200 mV peak-to-peak on the test point. Maximum is limited by the point at which the receiver starts oscillating. If this is the case at the point of optimum tuning then reduce the value of R21 slightly. On the other hand if the signal is below 200 mV after being tuned properly, then increase the value of R21 until it is above 200 mV peak-to-peak, say 300. When using a properly screened box amplitudes of above 3 V peak-to-peak are possible! The higher the amplitude the better the AGC effect.

Now carefully turn PR5 clockwise until clear blips are heard from the buzzer every time the carrier is turned off. If you have reached this far well done! — sit back and watch the display. The CPU is now searching for the synchronising pulse. When it thinks it has found one, it will generate a higher tone and an attempt will then be made to read the fast serial code. If it is validated then the right time will occur and the clock will be in Mode 1. If it's not OK (parity error, etc) then the lower tone blip will resume, meaning that it was a false synchronisation pulse (possibly noise).

SPECIFICATIONS

MODE 1 (TIME)

Face 1:	Hours, minutes, seconds (12/24 hour format), default is jumper-selectable.
Face 2:	Date, month, year.
Face 3;	Day of week, alarm tune on/off, AM/PM indicator.
Face 4;	Elapsed time since transition to automatic crystal back-up.
Face 5:	Time difference (GMT/UTC).
Back-up:	Automatic crystal clock back-up in the absence of a readable signal.
Error checking:	parity, comparison of atomic time with crystal clock, validation of max/min value of each unit where applicable.
Outputs:	All the above information is available on six decoded, unlatched ports.

MODE 2 (ALARM)

General:	Eight independent alarms with tune; eight programmable output lines in conjunction with the alarms.
Phase 1:	Display/set alarm times (eight faces).
Phase 2:	Display/set port lines (eight faces)
Phase 3:	Display/enable/disable alarms or ports (eight faces).
Outputs:	Eight TTL-compatible lines, each individually programmable; separate strobe logic available to enable multiplexing into maximum of 127 ports.
Tune:	12 bars of designer's composition, if alarm tone enabled.

MODE 3 (STOPCLOCK)

General:	Stack-orientated up-counter; up to 240 lap times may be stored; laps viewable while count is in progress.
Phase 1:	Advance top of stack and reset new top of stack (start/stop).
Phase 2:	View next time on stack (view laps).
Phase 3:	Advance top of stack and copy the previous one (latch lap).
Phase 4:	Display top of stack (view current count).
Resolution:	1/100th second
Maximum count:	99 minutes 59.99 seconds.
Output:	Current count is output on three decoded, unlatched ports every 10 mS.

COMPUTER CONTROL

It is possible to disable the switch debounce software, enabling the unit to be controlled by external logic or computer.

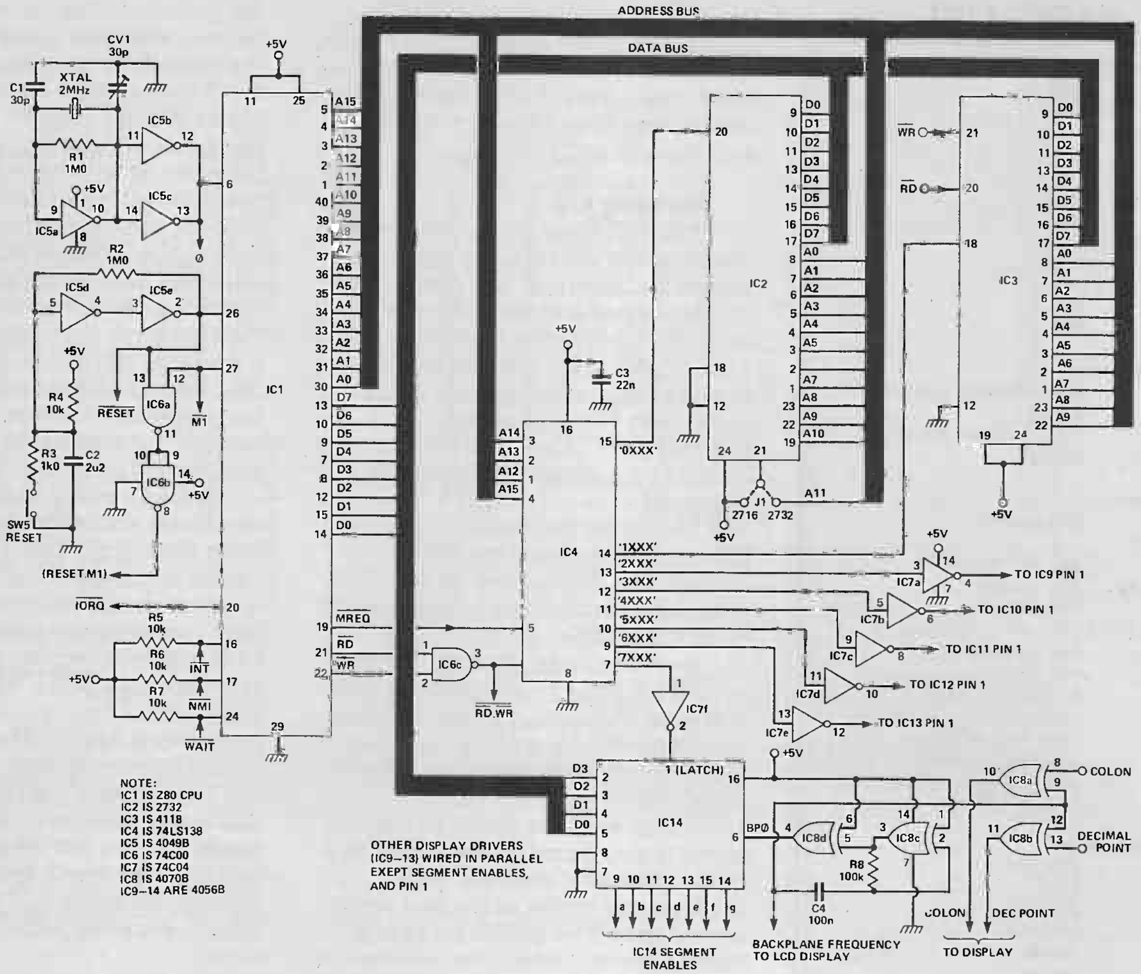


Fig. 5 Circuit diagram of the main logic section, with the master clock, address decoding and display driver circuitry.

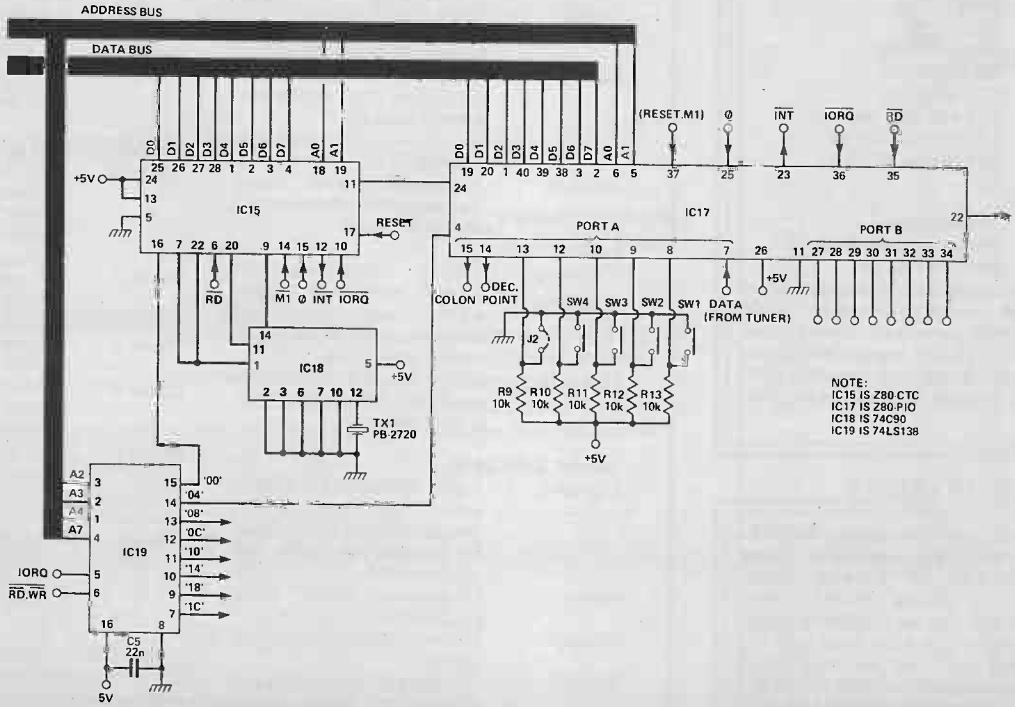


Fig. 6 The CTC and PIO section of the main board.

HOW IT WORKS

The first CTC in IC15 (CTC 0) generates pulses at rate of 1 kHz on its output pin (7). This is used by the unit as the main time-base. This 1 kHz is fed to the input of the second CTC, which generates an interrupt in multiples of 1 millisecond depending on the initial time constant loaded in its counter (1 count for 1 millisecond).

During the initial synchronization period, this varies considerably depending on the required delay. After synchronisation the delay is set permanently to 50 mS (50 counts).

From now on every event occurs on a 50 mS boundary ie, the entire second is divided into 20 equal slots, each of 50 mS. Each slot has its own task. Some tasks are carried out at certain slot numbers in a certain second. For instance, in each second between 17 and 51, at the third slot (150 mS) the input data is sampled to get the time and date information. Since this slot is not shared by an other task, the relevant software takes its time about it by implementing a software filter to reject all but the highest noise levels in the signal. This is done by taking samples over a period, counting separately the number of 1s (carrier on) and 0s (carrier off) and at the end using the greater of these two counts to decide if the bit is 1 or a 0.

To explain how the back-up function works we will start off when the clock synchronises on tne fast time code on second 0 as described in the text. After verification (parity, BCD validity etc) the time is displayed and the seconds counter (and thus display) advanced every 20th slot (1 second). As it advances through the minute, one of the clock's tasks is to pick up the slow time signal and store it.

At the end of the 59th second, this saved time should equal to the current time plus 1 minute (remember it is still the 59th second). If this is so, then CTC 1 will be stopped. At this time the carrier level should be high, ready for a low which starts second '0'. The software loops until the carrier goes to a low level and initiates the 50 mS CTC, thereby synchronising with the radio time. The new time is then once again latched into the display.

If the two times don't match due to a noisy or non-existent signal, then CTC 1 won't be stopped and continues on counting as a normal crystal-controlled clock, until the next 59th second, when once again a comparison is made.

The CPU will respond to a switch request (a) if a time slot does not have a task delegated to it, (b) if a task is finished inside the 50 mS of allocated slot time, since in those two instances the CPU will be executing one of its only two instructions (!) in the whole of the main program, namely

```
MAIN: EN        ;Enable interrupt
JR, MAIN        ;Jump back to Main
```

The rest of the program is comprised of interrupt routines which would interrupt this two instruction program.

When the alarm buzzer is active then every 50 mS is utilized by the tone generating software even if the next sound is silence! CTC2 times out at a rate which will be twice the required frequency of the tone, the pulses being output at pin 9 of IC15. These pulses are very short and would produce a harsh sound in the ceramic resonator; thus they are fed to IC18 pin 14 where they are divided by 2 to get the proper frequency as well as giving them an equal mark-space ratio. The other half of IC18 is a divide-by-5 counter with input from IC15 pin 7 (1 mS pulses) and output at pin 11 to the input of CTC 3. This division is not implemented in the present application.

CTC

The CTC contains four independent down-counters. The initial count is software-programmable and is automatically reloaded on reaching zero. The counters are decremented by one of two signals:

- The system clock after being divided by either 16 or 256
- An external clock applied on the clock/trigger input.

In either mode each channel (except channel 3) generates a short pulse on its output pin when it reaches zero. At the same time it can generate an interrupt request if programmed to do so (including channel 3).

PIO

The PIO provides 16 I/O pins divided into ports A and B, each having eight bits. If programmed to run under 'bit mode', a lot of wonderful tricks can be performed. For instance, each bit within a port can be programmed as an input or output; furthermore, all the bits that are programmed as inputs can be programmed to generate an interrupt request to the CPU, on either a negative or positive transition of the input signal. This proved handy in this application because it enabled the same port to be used for monitoring the condition of the switches as well as outputting control levels for the decimal point and colon on the display. This leaves port B completely free for mode 2 programming.

POWER SUPPLY

The 12 V across T1 secondary is rectified by BR1 and smoothed by C23. IC23 produces 5 V for all the logic and part of the receiver while IC24 produces 5 V for the receiver's bandpass ICs.

IC26 is a DC-to-DC converter which produces an output voltage at pin 5 of opposite polarity to that present on pin 8. This IC was used instead of a conventional centre-tapped transformer splitter in order to reduce the size of the transformer, and save space which would have been taken up by another largish smoothing capacitor.

This IC has poor regulation and to overcome this problem, a higher voltage than is required is input at pin 8 (9V3) producing approximately 9 V. This is then regulated by ZD2, a 5 V zener diode. If the output of IC25 was connected directly to pin 8 of IC26, IC26 would be damaged since the maximum input voltage is 10 V. ZD1 is added to drop this by 2V7 volts to a safe level.

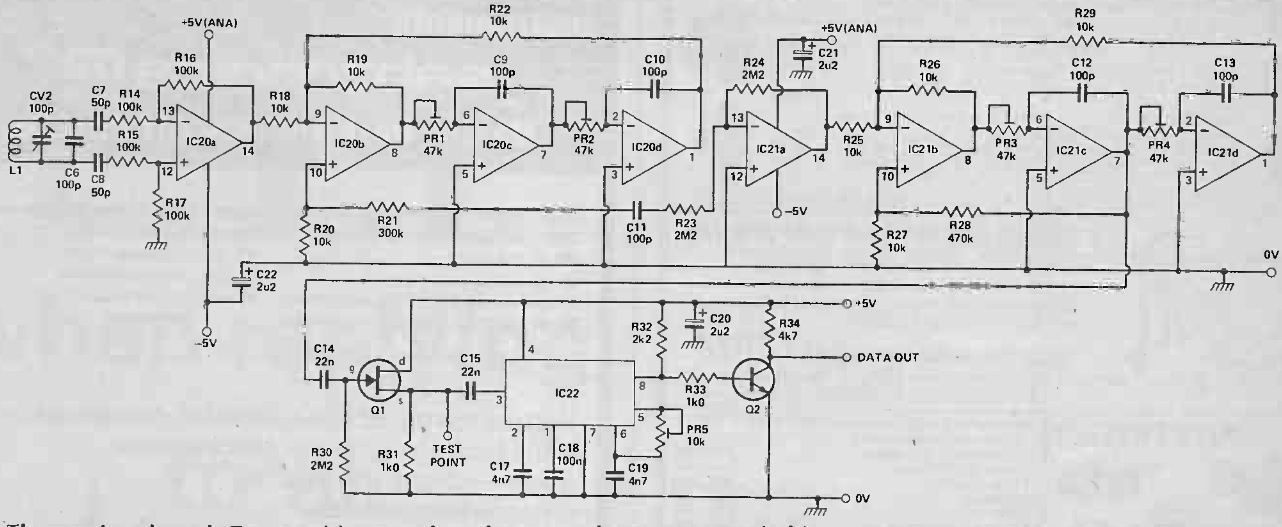


Fig. 7 The receiver board. Two positive supply rails are used to prevent switching spikes on the op-amp supply.

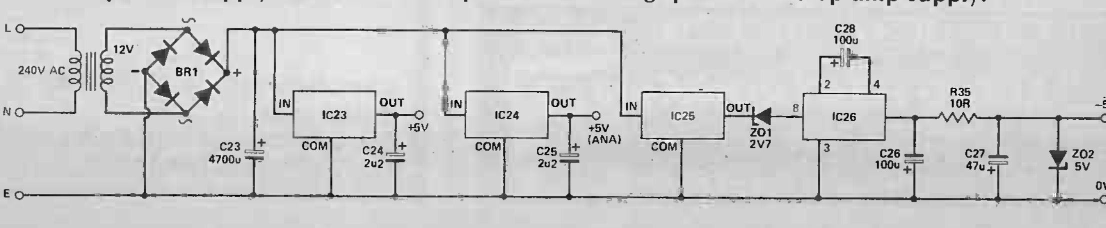


Fig. 8 The power supply section.

ETI

ACTIVE SPEAKER

Once upon a time ETI published a design for the most advanced DIY loudspeaker available. Now, history repeats itself . . . Design by Barry Hughes and Paul Hynes.

Although it is now some six or seven years since ETI published the ER II high quality loudspeaker system, Badger Sound Services, as suppliers of the more specialised bits for that design, still receives enquiries. The reason is not hard to find — ER II's combination of 8″ bextrene coned bass drive units, 1″ soft dome tweeter and carefully tailored crossover network were not only capable of seeing off commercially made speakers costing three times as much but the design was then bang up to date. The ER II was one of the earliest systems to exploit the 8″ bextrene-plus-1″ soft dome approach and the intervening years saw a plethora of commercially made speakers using that combination until it became almost an industry standard!

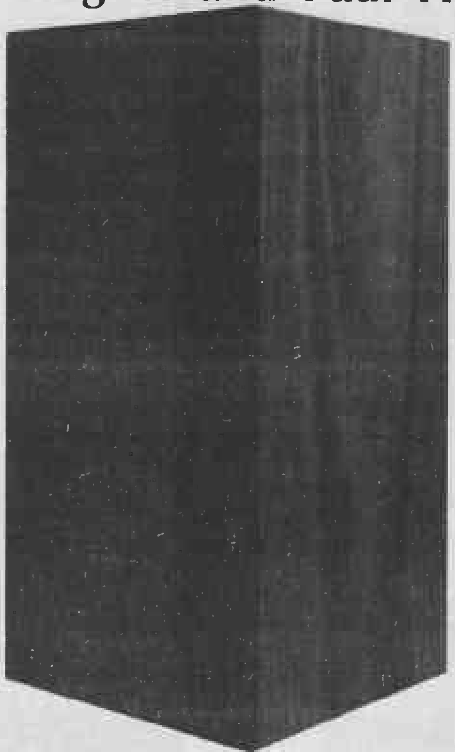
This adequately demonstrates ETI's policy of keeping that little bit ahead of up-to-date, and if we are to maintain our readers in that happy state then the time has come for an ER II replacement.

Design Philosophy

There can be little doubt that in balancing performance, domestic acceptability, simplicity of woodworking and cost, the medium size, two-way reflex system wins hands down. This particular format has been employed, for example, on several BBC monitors and if carefully designed can produce a very accurate performance.

The construction of a reflex enclosure is illustrated in Fig. 1, and it will be observed that a tubular duct extends into the enclosure. In operation the mass of air in the duct is acoustically coupled to the rear of the bass driver cone, and at low frequencies it becomes virtually a second radiator, reinforcing the output from the bass driver. Above the bass region this effect vanishes and the system behaves as a closed box. The benefits of this system are:

- slightly higher efficiency at the bottom end
- lower bass distortion since cone excursion is reduced



- ease of construction when compared with more exotic methods of bass loading.

If reflexing is to work properly, the system must carefully balance enclosure volume, port characteristics and certain drive unit properties, notably Q, free air resonance and the compliance or 'stiffness' of the suspension. It is not, as commonly thought, just a matter of tuning any old box to a given frequency and then sticking in a drive unit. Some drive units are incapable of satisfactory reflex operation and some would need enormous enclosures.

Advanced Driving

The 2040 II's bass driver is a new high performance unit by IMF Electronics, the 28/30. The 28/30 has a ribbed cone of engineered polymer (for rigid behaviour and reduced colouration), a high temperature voice coil, powerful magnet assembly and a free air resonance of typically 28 Hz. Having the right compliance characteristics, it reflexes nicely in our enclosure.

Our tweeter is also a new product by IMF Electronics, the TW1500 soft dome unit with a foam diffraction ring. In the authors' view the TW1500 is capable of a subjective performance in advance of previous similar types, offering a smooth, detailed response to beyond audibility without the 'tizziness' of some older competitors. The new drive units are a welcome addition to the amateur's armoury and this design is the first magazine project to employ them.

On Active Service

The only item now left to consider is the crossover network. At the time of the ER II the passive crossover was universally used but the modern trend is towards the active system (known to our American friends as bi-amping).

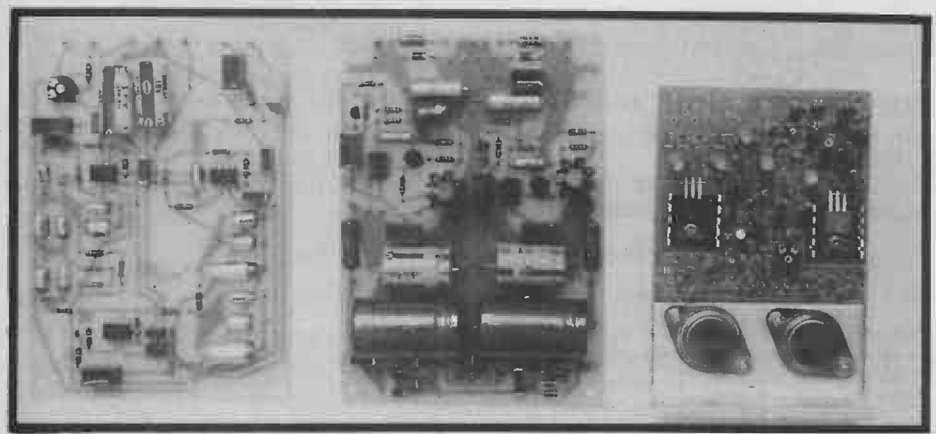
As with a passive system, a good active crossover is dedicated to the speaker system for which it was designed. The response of the 2040 II active crossover is *not* flat and its benefits are not transferrable to other speaker systems. Sadly, therefore, and without wishing to be unhelpful, the authors cannot enter into correspondence regarding its application to other systems, the answers to which would involve extensive and expensive measurements.

Speaker Construction

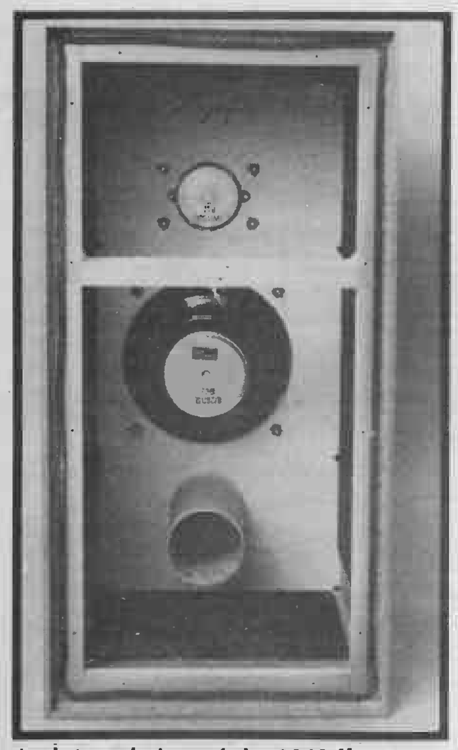
The essential enclosure details and dimensions appear in Fig. 1. For two main reasons, we have not gone into very great detail: first, the precise method of construction is not critical so long as the three guiding principles set out below are observed, and second, every constructor has his own preferred way of going about things.

Every construction question which might arise can be answered by applying three guiding principles:

- The enclosure must be to the correct dimensions. Small mistakes in overall size will not matter, however, providing the internal volume does not vary more than 10% either way.



From left to right we have a crossover, regulated PSU and Crimson Elektrik CE1008 power amp.



An internal view of the 2040 II.

- The enclosure must be rigid. Use only the specified timber and do not omit the bracing shelf.
- Despite the presence of a duct tube, the enclosure *must be airtight*, calling for straight cutting and plenty of glue. 90% of the battle here is to get the panels cut accurately to size by the supplier.

The timber used should be either good quality ply, chipboard of 600 or 750 grade (if offered 'roofing' grade or 'flooring' grade, take flooring grade), or solid timber if well seasoned. Chipboard will be the most readily available and the cheapest. Veneered chipboard such as 'Contiboard' is not suitable but that apart, none of the three timbers has proven sonic advantages.

The method of corner jointing is not important and can be decided by the constructor's skill. The panels can be butt jointed, dovetailed, screwed-and-glued-onto-corner-battens, pinned-and-glued — or any other method can be used which would produce rigid and airtight results. If corner battens are used, these should be of about ¾ square section. The presence of such battening has no practical effect on performance.

Adhering To Principles

Oddly enough, a domestic loudspeaker does not need to be *strong* (as opposed to *rigid*). Many commercially made loudspeakers have glued butt-jointed or mitred corners, with no extra reinforcing, and if properly done this is adequate. Use only water-based adhesives such as Evostik Resin W or similar PVA types. Solvent-based adhesives should **not** be used since the vapours they release into the enclosure can attack the drivers. The warranty on drivers is void if other than the recommended adhesives are used.

The enclosure should be constructed as a sealed box which helps rigidity and airtightness. There is no need for either front or back to be detachable since all drivers are mounted from the front with wood screws for the tweeter, captive 'T nuts' and bolts for the bass driver. In the unlikely event of service attention being needed in the future, either driver can be removed without dismantling the enclosure.

Rebating the drivers is desirable but not essential. The treble driver will be fitted using small wood screws and appropriate pilot holes should be drilled. The bass driver will be installed with bolts and captive T nuts (supplied with the driver). To fit the T nuts, drill each fixing hole ¼ diameter. The T nut is fitted from the inside of the cabinet shell as shown in Fig. 3, the teeth being engaged in the wood and holding the nut in place. It is most easily fitted by pulling into place with a bolt and washer.

Fit two pairs of terminals (one pair for each drive unit) in the speaker back. Colour-coded screw terminals are probably best, but any type which could cause air leaks (eg jack sockets) must be avoided.

Dead Ends

To reduce bass colouration arising through enclosure wall vibration, self-adhesive damping (sound deadening) pads are fitted. Working through the bass driver hole, *carefully clean any dust or grease from inside the enclosure walls.* Prepare the self-adhesive damping pads by cutting to size (do not remove the backing paper yet). Cut to such sizes as will allow the

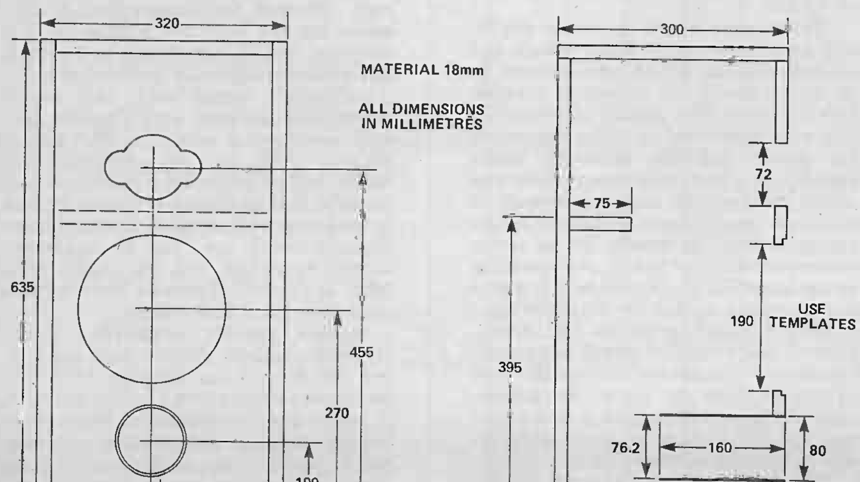


Fig. 1 Speaker dimensions. The front panel template is for the right hand speaker; it should be reversed for the left hand one. If a speaker grille is desired, the frame should be made from timber no larger than 1½" by ½".

pads to be distributed as follows (double thicknesses are used, ie one piece of pad stuck on top of another, so 'one pad' means a full pad cut into two giving two layers):

One and a half pads on the back panel

One and a half pads on each side panel

Half a pad on the top panel

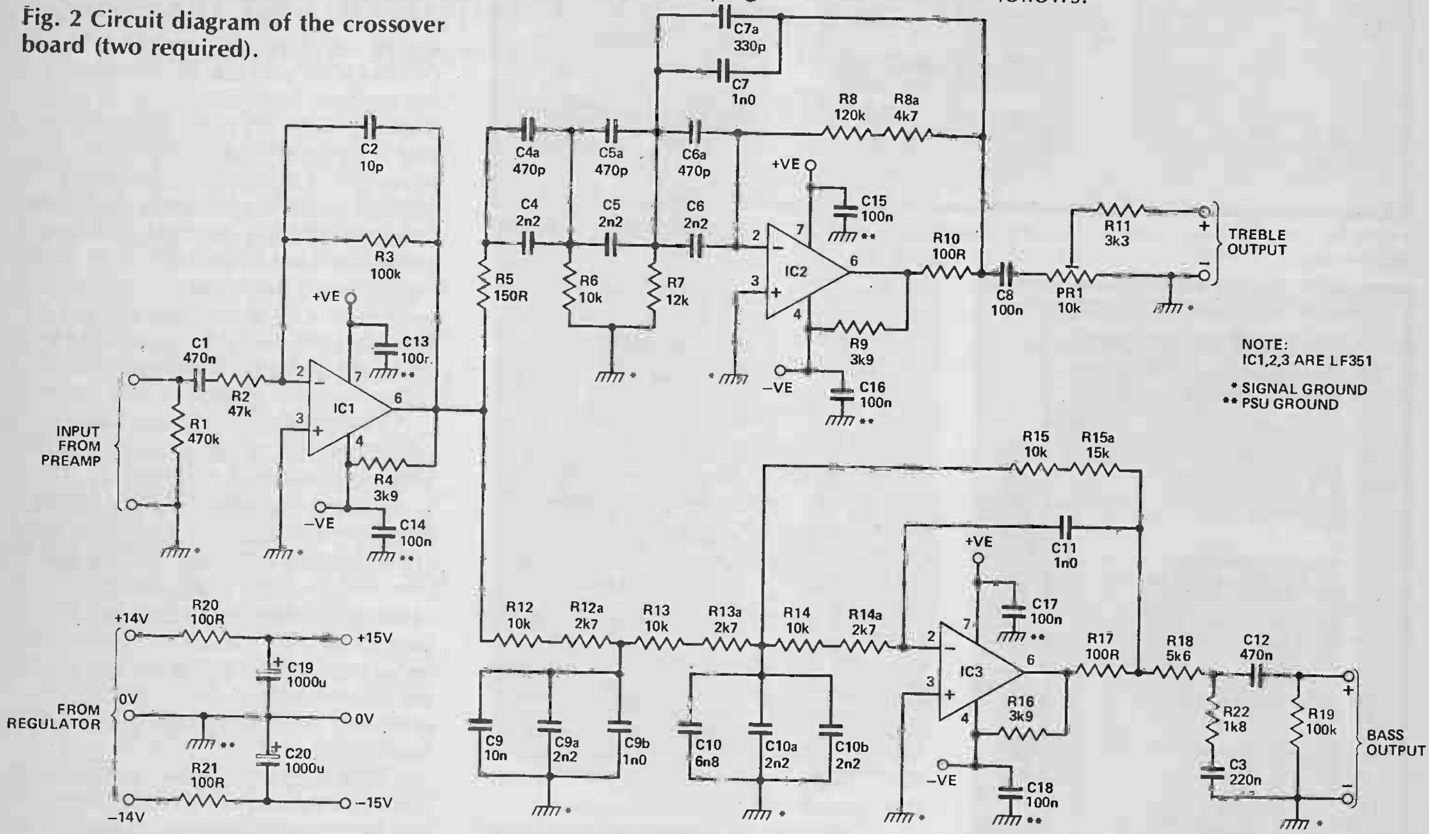
Half a pad on the bottom panel

Neither exact size nor exact positioning are critical. Carefully fix the pads in place.

Using 6 amp colour-coded cable, connect flying leads to the input terminals.

24" wide by 1" thick BAF wadding (bonded acetate fibre) is employed in the enclosures for acoustic damping. Place the wadding in position, working through the bass driver hole, as follows.

Fig. 2 Circuit diagram of the crossover board (two required).



HOW IT WORKS

THE ACTIVE CROSSOVER

The crossover network is usually a passive circuit (see Fig. 4), living in the loudspeaker, which takes the signal from the power amplifier and divides it into frequency bands (a crossover is sometimes called a dividing network). It passes to each speaker drive unit the appropriate band of frequencies; low to the bass driver, high to the treble. The crossover, however, is a much misunderstood animal which does more than just divide up the signals. It carries out all the compensation needed to obtain flat response from the drivers, adjusts for enclosure matters and generally integrates everything to produce optimum balance and performance.

Consequently there can be no such thing as a 'general purpose' crossover. It has to be dedicated to its particular speaker system and crossover design is very much an art in itself.

It will be observed that the passive crossover comes between the power amplifier and the drivers, so that what the crossover divides up is the *power output* of the amplifier. Although this relatively cheap and effective method has stood the test of time, and has been refined to remarkable precision, there are some disadvantages. Not only can the passive crossover present the amplifier with a sometimes difficult load but interaction can arise between the sections. More importantly, however, it has to use inductors and capacitors of large value which produces energy storage. This is a form of electrical intertia, removing a lot of the immediacy of the music. At the risk of oversimplification, it is rather as though the driver cones are being moved by a piece of rubber!

The modern way to overcome this effect (and some other lesser effects not mentioned) is to use an active system. In an active system the crossover network comes before the power amplifier so that what is divided up is not the outgoing power but the incoming audio signal. Using low value components and active devices, an active crossover is under no power stress and there are no energy storage problems. In an active system, each driver has its own personal power amplifier stage with which it is in direct contact, so that the amplifier has a much improved 'grip' on the driver. Control and transient attack are greatly enhanced, producing a speed and definition hard to equal by passive means. The only drawback is that a two-way speaker system in stereo calls for two stereo power amplifiers.

The circuit of one channel of the crossover appears in Fig. 2. Basically we have a fairly textbook filter, operating at 18 dB/octave with a crossover point (— 3 dB) of 2.5 kHz. The response shape has, however, been tailored to the 2040 II's characteristics in the mid/bass area and so the filter alone does not produce flat results.

POWER SUPPLY

The crossover network needs a regulated dual rail supply of ± 12 to 15 V at 40 mA total (20 mA per channel). Although the constructor can certainly use his own pet supply for this purpose, the circuit shown in Fig. 6 is advisable for optimum performance.

Although complicated, our power supply is in keeping with the best current commercial practice. The power supply, even for preamplifiers and filters, is now regarded as an important factor in the resolution of fine detail and in securing wide dynamic range. Some manufacturers go to extraordinary lengths to ensure that the supply is as 'stiff' as possible (Crimson Elektrik have even gone back to batteries).

Simple circuits employing three-terminal regulator devices have not proved suitable for our purpose, since the source impedance can become too high at upper audio frequencies. Our circuit has the desirable properties of low output impedance across the audio bandwidth, high power supply rejection and low noise on supply lines. A separate regulator for each channel is specified although as an economy measure, one alone could power both channels if some crosstalk degradation is acceptable.

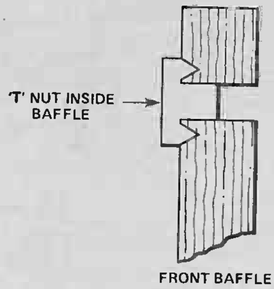


Fig. 3 Fitting the 'T' nuts to the baffle.

- Take a 6 ft length for each speaker.
- Cut into a 5 ft and a 1 ft length.
- Fold this 5 ft length into a 2½ ft × 2 ft piece.
- Place this piece into the bottom of the enclosure, laying it so that it covers the floor, lower sides and lower back.
- Take the remaining 1 ft × 2 ft piece and fold it into 1 ft × 1 ft. Roll this into a sausage and place vertically behind the treble driver hole. Ensure that the end of the duct tube has not been closed by the wadding — there should be a little free space around the inner end of the tube and the end should not be rammed into the wadding.

Making Connections

Connect the bass driver to the flying leads from the bass input terminals, ensuring that the positive terminal connects to the red marked solder tag on the bass driver. Check that a foam gasket is in position on the inside face of the bass driver rim and bolt the driver evenly into place, ensuring that the connecting lead has not become trapped under the driver frame. Airtightness is just as important here as elsewhere. Repeat the process with the tweeter, again observing polarity and airtightness.

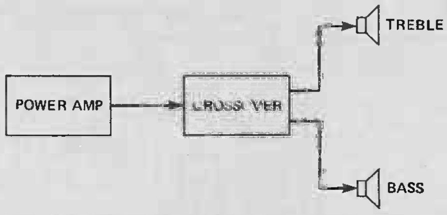


Fig. 4 Block diagram of a passive crossover.

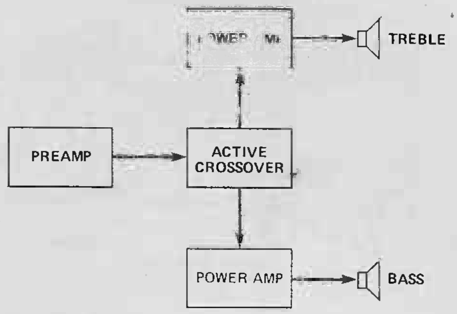


Fig. 5 Block diagram of an active crossover.

The Electronics

It is now generally accepted that high quality amplifiers can have individual sonic properties quite apart from paper specifications. A major consideration with the 2040 II's electronics has therefore been to ensure the best sonic performance *from the electronics,* not only as an end in itself but to ensure that we can interface the crossover with high performance pre and power amplifiers without subjectively degrading their performance. A suggested circuit board pattern appears in Fig. 7 — this layout is strongly advised if instability is to be avoided.

We have tried to specify generally available components and substitution is made at your peril! In particular, C12, C13, C14 and C15 MUST be solid tubular tantalum (not bead types) and IC1, IC2 must not be replaced by 741 types. R3, R4, R5 and R6 must be matched as specified. We know solid tantalum capacitors are expensive (sorry!) but they offer low impedance at high frequencies which low cost electrolytics and tantalum beads don't.

The mains transformer should be a good quality type providing 15-0-15 V secondaries at 10 VA with good regulation and an interwinding screen. If a small toroid can be

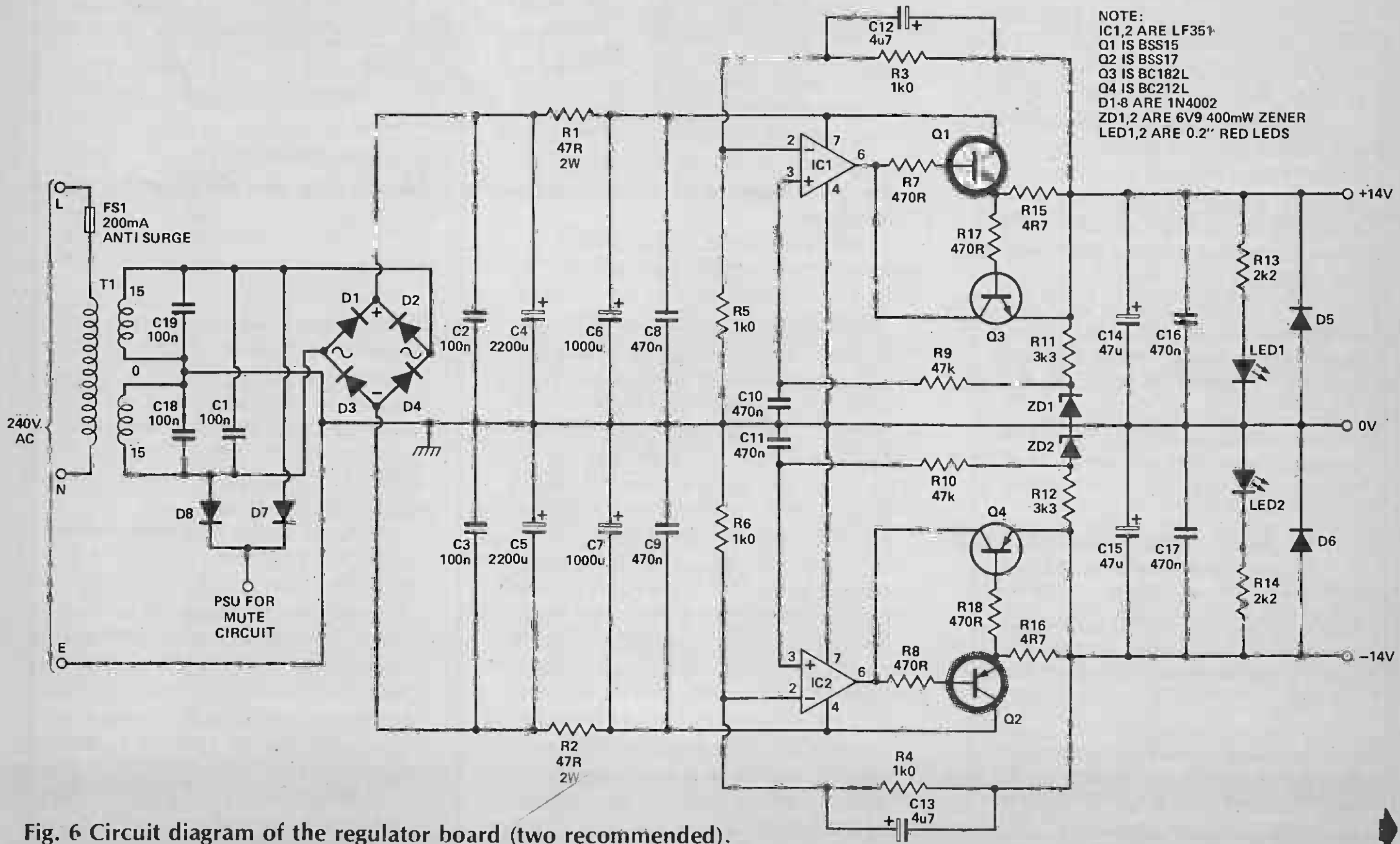


Fig. 6 Circuit diagram of the regulator board (two recommended).

DIGEST

Watching TV!

We like to think we're fairly hardened journalists who can take the rapid advances in the electronics field in our stride, but a few days ago the office was echoing to the sound of jaws bouncing off the carpet and cries of "I don't believe it!" The cause of all this excitement? A press release from Seiko announcing the world's first wristwatch TV (was that a strangled cry we heard from Clive Sinclair?). The watch face has a 17 × 25 mm LCD screen made up of approximately 32,000 dots, plus an LSI integrated circuit on the same silicon wafer. The rest of the electronics is contained in a pocket receiver, Walkman style, while the TV aerial is built into the headphones. As well as all VHF and UHF TV channels, the wearer can also listen to FM stereo.

Despite all this, the watch still has room inside for — well, the watch. There's a normal 24 hour display, (hours, minutes, seconds), day/date calendar, alarm and stopwatch facilities. The battery that drives the chronograph section has a life of approximately two years, while the TV screen can be expected to operate for more than seven years. The pocket receiver runs off two AA batteries, measures just 145 × 76 × 18 mm and weighs 190 g.

Still pinching ourselves to see if we were awake, we hurried down the corridor to the *Which Video* office, where we were told that such a device was not, in fact, a belated April fool, but does exist, although it's going to be something like a year before we'll be able to get our hands on a production model. The price isn't yet decided but it's estimated to be just over £200. The thing that really amazed us was the casual way Seiko presented the release for such an incredible piece of technology — no glossy brochure or folders full of advertising hype, just a couple of photocopied sheets and two indifferent black and white photos. We've had better-presented information on cable grommets.

Roadrunner Boards

In our review of the Roadrunner wiring system in the Breadboards article (ETI June 82), we described the boards as 'specially designed'. The manufacturers have asked us to point out that this is a bit ambiguous, and implies that their boards are only suitable for use with the Roadrunner system. In fact they are eminently suitable for circuit-building no matter what method you're using. Sorry about the possible misunderstanding.

Lascar Left Out

Our apologies are due to Lascar Electronics Ltd of Unit 1, Thomasin Road, Burnt Mills, Basildon, Essex SS13 1LK — we forgot to mention them as suppliers of the DPM200 LCD panel meter used in the Kitchen Scales project and Designer's Notebook last month. Now we've put the record straight, we're off to bed without our suppers . . .

Breadboard '82

In November the 5th Breadboard Exhibition hits London once again. This, the most successful show of its type, will offer the electronics enthusiast an opportunity to see for themselves what they've been reading about in their favourite magazines. There's also a chance to examine the wares of the mail order companies — in the silicon, as it were — and the latest hardware and software on offer from the companies that are a household name to the electronics buff.

This year a series of lectures and demonstrations are planned that will give the visitor the chance to learn at first hand how some of those projects and circuits work and how they may be modified to suit their own personal needs. A number of exciting competitions are also planned that should fire the interest, so don't miss BREADBOARD '82.

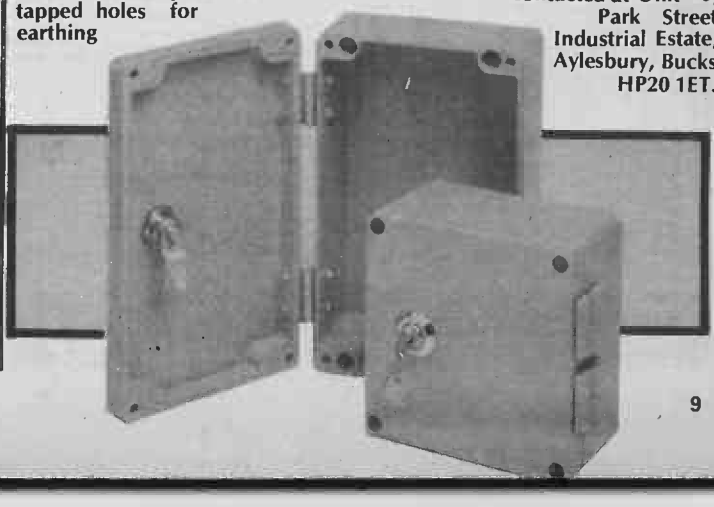
Prettier Daisy

CPU Peripherals Limited are introducing an upgraded version of their Daisywriter printer — the model 2000. Main differences between this and its little sister the Daisywriter 1000 are a larger buffer, greater baud range and more versatile print. Connectable to any of the usual serial or parallel interfaces, the Daisywriter 2000 has a 16 KByte or 48 KByte buffer memory, whichever the user feels he needs, to avoid any chance of slowing down his computer or word processor's own operations. A '99 per cent full' signal is fitted to the buffer — just in case. Baud rates are selected automatically or by switch override from 50 to 19,200. The printer can be used with all the popular word processing packages — Wordstar, Applewriter, Easywriter, Spellbinder, Wordmaster and so on. A thumbwheel selects page length and print settings of 10, 12 or 15 pitch are available, all with automatic proportional spacing.

Daisywriters are the first printers to use print wheels enclosed in their own dust-free plastic cassettes. These are available in over 100 different character sets and typestyles covering 15 different language alphabets ad they have a minimum life of 10 million impressions. Reliability of the printer itself is enhanced by a three-chip microprocessor, reducing the mechanical parts, and a linear motor for the drive mechanism. Daisywriter 2000 is available from CPU's network of dealers in the UK, France and Germany at one-off recommended end-user prices starting around £995 plus VAT. The lady in the picture seems to be indicating you need your head examined if you don't get one.

Lock-up Boxes

West Hyde Developments have a new range of housings for applications where security of installation is important, coupled with 'protection to IP65', whatever that is. The hinged lids are fitted with cylinder locks as standard (see the photograph); however, square key locks or slotted-head screws can be supplied to special order should the fancy take you. The enclosure will remain dustproof and hoseproof in temperatures up to 90° C due to a neoprene gasket. On the outside the alloy cases have a smart blue/grey stove enamel finish; inside, pads with tapped holes for earthing and PCB- mounting purposes are provided at each end. West Hyde can supply the enclosures ex-stock in a range of six sizes, and can be contacted at Unit 9, Park Street Industrial Estate, Aylesbury, Bucks HP20 1ET.



found, so much the better. Q1 and Q2 require heatsinks and the usual T05 push-fit type will serve.

Cross Over Components?

Construction on the specified board is straightforward but as with the power supply, certain component types should not be substituted. All ICs are FET op-amps which in combining wide bandwidth, low distortion, high slew rate and adequate noise performance with modest cost have proved admirably suited to our purpose.

Items asterisked in the parts list are frequency-determining components and must be close tolerance types. Although metal oxide resistors are not strictly essential they do represent one of the easier ways of getting hold of 2% resistors.

Mechanical Assembly

The two regulator boards, two crossover boards and mains transformer can be mounted in a single screened case. Having the mains transformer in the same case produced no hum problems in the prototype. We have deliberately avoided specifying a case since in our experience, case cost and availability are a major headache for the constructor. Considerable latitude is possible here and the constructor is free to adopt a case style appropriate to his needs and pocket; but screening is essential.

Providing sensible precautions are taken (eg not putting the transformer next to the input sockets) and overcrowding is avoided, board disposition within the case is not critical. Alternatively the advanced constructor could consider mounting the boards in the amplification and in the interests of such flexibility we have adopted separate channel boards. As with all audio equipment, earthing arrangements are critical and the arrangement shown in Fig. 8 is hum free and stable.

Picking Your Power Amp

Since the active crossover is connected between pre and power amplifiers, the system cannot be used with integrated amplifiers. The 2040 II system requires four power amplifier channels and so two stereo power amplifiers are called for. In many cases the constructor will be extending an existing system and so we have not specified a particular power amplifier. Essentially the chosen power

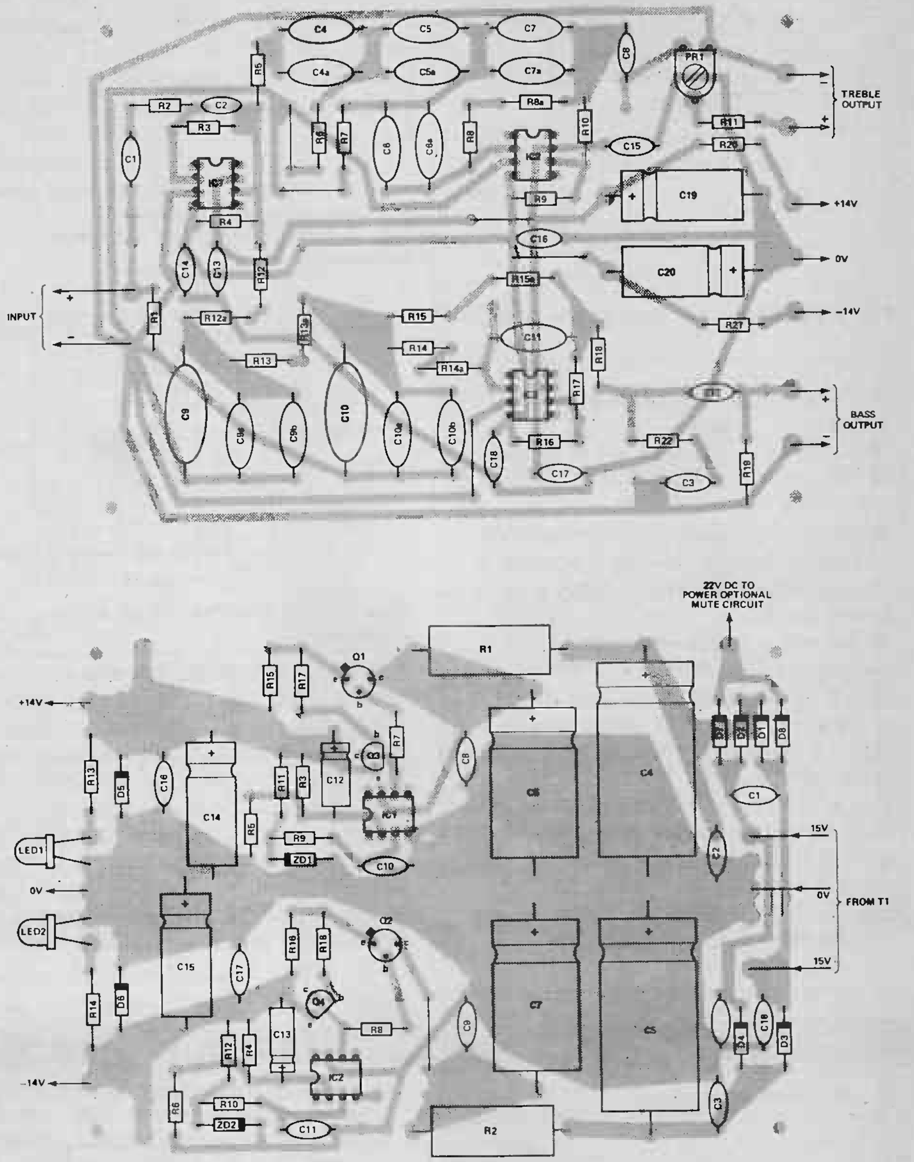


Fig. 7 Component overlays for the active crossover (top) and the regulator board (bottom).

amplifier must be of the highest quality. The benefits of active operation would be masked by an inferior amplifier and that germanium job you put together in 1963 will not do. It should be capable of being driven by the preamplifier (the active crossover has little more than unity gain), and of adequate power. The minimum would be 30 W RMS per channel into 8 ohms and a figure of 50-60 W or so would be desirable. 100 W could be used with care — see below.

Damage is more likely to result from a low-powered amplifier than one of higher output since the heating effect of a clipped signal is considerably in excess of a clean audio output. Too small an amplifier is the surest way to destroy a tweeter. Unfortunately it is impossible to lay down wattage figures to guarantee against speaker damage. Speakers are killed by people, not by amplifiers, and a little commonsense in operation is the only safeguard.

Clean audio signals are not usually a problem. Usually it is extremes that do the damage — clipping, excess bass lift or treble boost, for example, concentrating power into one region rather than over the music spectrum. Within the figures specified, sensible domestic use on speech and music signals should present no problems. If in doubt, a 500 mA quick blow fuse in the tweeter line could help but does not offer a full guarantee.

PARTS LIST

CROSSOVER

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

R1, 3, 19	100k
R2	47k
R4, 9, 16	3k9
R5	150R
R6, 12, 13, 14, 15*	10k 2% metal oxide
R7*	12k 2% metal oxide
R8*	120k 2% metal oxide
R8a*	4k7 2% metal oxide
R10, 17, 20, 21	100R
R11	3k3
R12a, 13a, 14a*	2k7 2% metal oxide
R15a*	15k 2% metal oxide
R18*	5k6 2% metal oxide
R22*	1k8 2% metal oxide

Potentiometer

PR1	10k miniature horizontal preset

Capacitors (all 2½% polystyrene except where stated)

C1, 12	470n polyester
C2	10p disc ceramic
C3	220n polyester
C4, 5, 6, 9a, 10a, 10b*	2n2
C4a, 5a, 6a*	470p
C7, 9b, 11*	1n0
C7a*	330p
C8	100n polyester
C9*	10n
C10*	6n8
C13-18	100n disc ceramic
C19, 20	1000u 16 V axial electrolytic

Semiconductors

IC1-3	LF351

Miscellaneous

PCB (see Buylines)

REGULATOR

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

R1, 2	47R 2 W
R3-6	1k0 2% metal oxide
R7, 8, 17, 18	470R
R9, 10	47k
R11, 12	3k3
R13, 14	2k2 ½ W
R15, 16	4R7

Capacitors

C1-3	100n disc ceramic
C4, 5	2200u 25 V axial electrolytic
C6, 7	1000u 25 V axial electrolytic
C8-11, 16, 17	470n polyester
C12, 13	4u7 25 V solid tantalum
C14, 15	47u 25 V solid tantalum

C12-15 must not be substituted with other types

Semiconductors

IC1, 2	LF351
Q1	BSS15
Q2	BSS17
Q3	BC182L
Q4	BC212L
D1-8	1N4002
ZD1, 2	6V9 400 mW zener
LED1, 2	0.2" red LEDs

Miscellaneous

PCB (see Buylines)

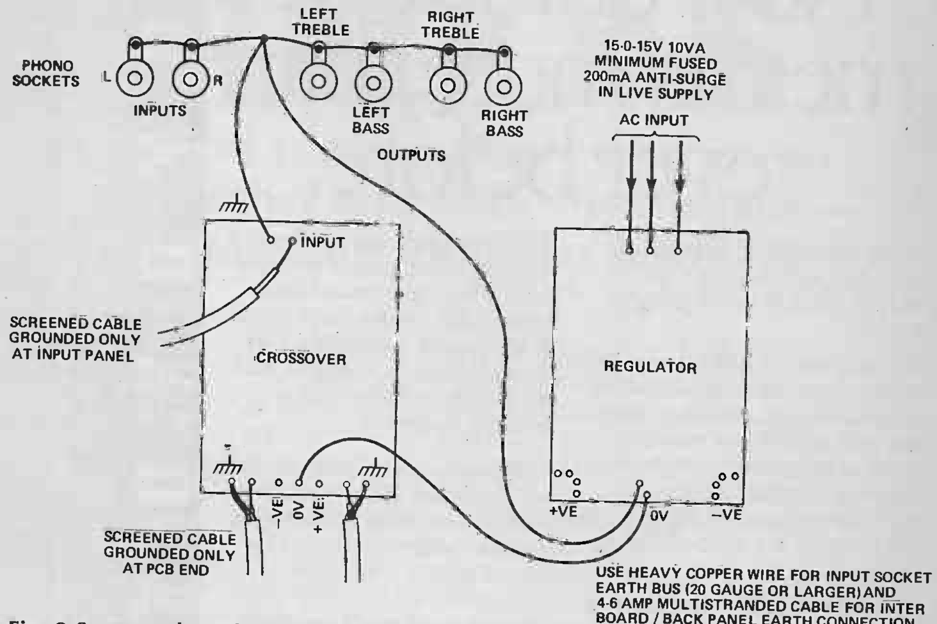


Fig. 8 Suggested earthing arrangement for the boards.

Unequal Partners

Oddly enough, the bass and treble amplifiers need to be of equal power. While the tweeter may not need as much power in wattage terms as the bass driver, it does need the same voltage swing availability. However, the two amps should be of equal gain. Small variations in power output between bass and treble amplifier would not matter but they must have the same gain, in terms of watts out for millivolts in. The safest course is to use the same amplifier throughout.

Finally the chosen power amps must be relatively free from switch-on thumps which could damage the tweeter. Aurak (listed among the suppliers) can supply a muting circuit.

Crimson Commended

To the constructor starting from scratch, we strongly advise the use of Crimson Elektrik components. Their amplifier kits are superb value for money (as was evidenced by the review in the February 1982 ETI), and offer a performance superior to most commercially made amplifiers. Those constructors who wisely have built a Crimson Elektrik kit amplifier need now only build a second power amplifier identical with the first. The constructor 'rolling his own' could well consider a project based on the CE1008 power amplifier module and Crimson Elektrik would be happy to advise.

For Your Listening Pleasure . . .

The speakers should be mounted clear of the floor, preferably on conventional stands about 10" high. Corner positions should be avoided. PR1 is a preset allowing adjustment of treble output, enabling compensation for room characteristics and so on. With a two-way system it is a straightforward matter to set balance in this way in the light of personal preference. Normal position is likely to be about one-third of the way round its travel. The speakers will do justice to the very best signal sources but are fully capable of revealing shortcomings in a poor "front end". The computer dictum of "garbage in — garbage out" holds equally true in audio!!

ETI

BUYLINES

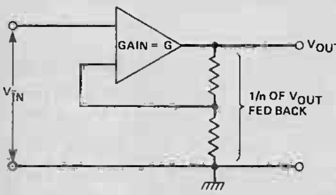
This project is a gestalt — the whole is greater than the sum of its parts — and the whole thing comes in various parts! The drive units, BAF wadding, damping pads, and Crimson Elektrik kits are available from Badger Sound Services Ltd, 46 Wood Street, Lytham St. Annes, Lancs FY8 1QG (telephone 0253 729247). The printed circuit boards and electronic components can be obtained from Aurak, 27 Derbe Road, Lytham St. Annes, Lancs FY8 1NJ (telephone 0253 728407 ext. 2). Crimson Elektrik components can be obtained from Crimson Elektrik, 9 Claymill Road, Leicester LE4 7JJ (telephone 0533 761920). Any enquiries regarding IMF Electronics drive units should please be made to Badger Sound Services and NOT direct to IMF Electronics. Continental customers could approach either Audio Sound I/S, Bygvaenget, 8, 4900 Nakskov, Denmark or A & O Electronics, Lenbachstrasse 14, 8130 Starnberg, West Germany. The design is offered as a project for home construction and no licence for commercial manufacture is implied.

CONFIGURATIONS

Our topic this month is twins and triplets. Don't worry, you haven't picked up a copy of 'Mother and Baby' by mistake — it's just Ian Sinclair explaining transistor amplifier feedback loops.

One of the major problems about the straightforward transistor amplifier is the calculation of gain. As we saw last month, the gain of the transistor itself is easily calculated, but if we are to allow for the effect of input and output resistance, we have to know a lot about the particular transistor we are using. The task becomes even more difficult when the amplifier operates at high frequencies, so that we have to deal with input and output impedances rather than with pure resistances.

Negative feedback is the main method that we use to get around this problem. To start with, imagine an amplifier whose voltage gain is G — this is sometimes called the 'open loop gain', the gain with no feedback loop connected (Fig. 1). The amplifier has an input connection and a feedback connection, and 1/n of the output signal is fed back to the input. Since the output signal is V_{OUT}, the amount fed back is V_{OUT}/n, and so the actual signal available to amplify is $V_{IN} - V_{OUT}/n$, the input voltage minus the feedback signal. This is the signal which appears at the inputs of the amplifier to be amplified, so that with gain G, the output from this must be $V_{OUT} = G(V_{IN} - V_{OUT}/n)$.



TOTAL INPUT = $V_{IN} - (1/n).V_{OUT}$
OUTPUT = G x INPUT
SO $V_{OUT} = G(V_{IN} - V_{OUT}/n)$
THUS $V_{OUT}(1 + G/n) = G.V_{IN}$
AND $V_{OUT}/V_{IN} = G/(1 + G/n)$
IF $G/n \gg 1$, THEN $V_{OUT}/V_{IN} \approx n$

Fig. 1 Principle of negative feedback.

An Open And Closed Case

When we collect terms in this equation and rearrange it to get V_{OUT}/V_{IN} (which is the gain with the feedback connected), we get this figure as $G/(1 + G/n)$. This is the gain-with-feedback or closed-loop gain, and this is the equation that should be used in calculations if G is small or n is large. If the ratio G/n is large compared with 1, however, we can ignore the 1 and rewrite the equation as $G/(G/n)$, which is simply n.

This important result shows that if the open-loop gain is large enough the gain when feedback is used is independent of anything except the potential divider that produces the 1/n of the signal which is fed back. This is the result that is often quoted in textbooks, but with a slightly different proof. One important point is that feedback is useful only if there is gain — if an amplifier has zero gain under certain circumstances, no amount of feedback can produce gain.

Looking On The Negative Side

Given that negative feedback can be useful, let's look at some circuits that make use of it. Since gain is important, there's not too much point in spending time looking at single-transistor stages, though both of the circuits in Fig. 2 are important and are widely used. The equations are approximations which apply to silicon transistors; no bias components have been shown.

Negative feedback really comes into its own when two or more transistor stages are used, and if the feedback path is a DC one then both gain and bias will be controlled by the feedback. This is seldom completely convenient, so that the basic circuits have to be modified to allow for separating bias and signal feedback paths. One of the most-widely-used of these circuits is the feedback pair circuit of Fig. 3. In this circuit, the transistors are direct-coupled, and the negative feedback path is from the emitter of Q2 to the base of Q1. Because of the bypass capacitor C2, there is no signal voltage across R6, and the signal voltage that is fed back is the voltage across R5 only. This value, along with the value of feedback resistor R7, can be chosen to produce the amount of gain that is

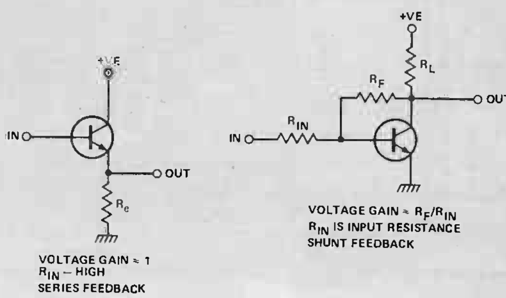
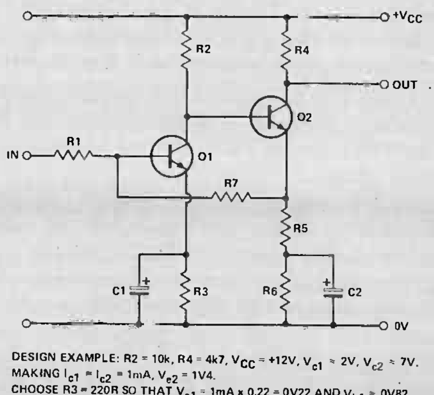


Fig. 2 Feedback over a single stage gives rise to these well-known circuits.



DESIGN EXAMPLE: R2 = 10k, R4 = 4k7, V_{CC} = +12V, $V_{c1} \approx$ 2V, $V_{c2} \approx$ 7V.
MAKING $I_{c1} = I_{c2}$ = 1mA, V_{e2} = 1V4.
CHOOSE R3 = 220R SO THAT V_{e1} = 1mA x 0.22 = 0V22 AND $V_{b1} \approx$ 0V82.
VOLTAGE ACROSS R7 = 1.4 – 0.82 = 0V58.
ASSUME h_{fe} = 100, THEN 1/100mA FLOWS THROUGH R7 TO BASE OF Q1.
VALUE OF R7 MUST BE 0.58/(1/100) = 58k; A 56k SHOULD DO NICELY!
R5 + R6 HAS TO PROVIDE 1V4 AT 1mA = 1k4.
IF WE MAKE R5 = 390R, R6 = 1k0, THEN GAIN = 56/R1 x 4.7/0.39 = 675/R1 (R1 MEASURED IN KILOHMS). WE CAN NOW SELECT THE GAIN WE NEED BY SPECIFYING A VALUE OF R1 — WHICH WILL INCLUDE THE OUTPUT RESISTANCE OF THE PREVIOUS STAGE.

Fig. 3 A two-transistor negative feedback circuit using emitter-to-base feedback.

desired; providing of course, that the value is small compared to the open-loop gain which will be the product of the open-loop gain figure for each transistor multiplied by the loss for the potential divider effects of input and output resistances.

Getting Good Value(s)

Design of the circuit should start with the load and bias components. The values of the two load resistors R2 and R4 are fixed first; these should be fixed at values which will permit the transistor to operate at suitable currents. Values in the range 1k0 to 12k are usually suitable, though R2 can be in the range 15k to 47k if Q1 is operated at low current to reduce noise signals. If you prefer it, write down the desired collector voltages (Q2 must have a higher collector voltage than Q1) and choose current values, then use Ohm's law to calculate the load resistors.

Once the load resistor values, the currents and the voltage levels have been fixed, the bias components can be calculated. The voltage at the emitter of Q2 will be 0V6 less than the base voltage of Q2, and the voltage at the base of Q1 will be 0V6 higher than the emitter voltage of Q1. Subtracting these gives the voltage across R7, and we then have to pick a value which will allow the correct amount of base current to flow. Once this has been done, we can split the total resistance of R5 and R6 (which has to be calculated to produce the correct bias voltage) so as to allow for enough gain for Q2(R4/R6). We can then choose the total figure of gain by selecting R1, which is the sum of a resistor and the output resistance of the circuit that is providing the signal.

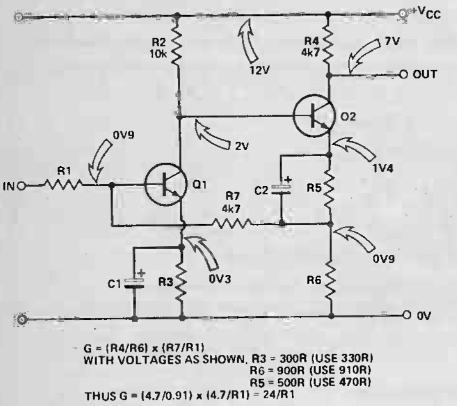


Fig. 4 A modified feedback circuit which is easier to design — sample voltages are shown.

Though this circuit works well, it is never easy to get the bias spot on by calculation unless you know what base current Q2 requires, and the total gain is rather dependent on the source resistance value. The bias problem can be eased by a modification of the circuit shown in Fig. 4, where R7 has a fairly low value, around 4k7, so that the base voltage at Q1 is practically equal to the voltage across R6. This allows the h_{ie} value of the transistors to be neglected, but the penalty to be paid is that the gain is lower unless R1 is also small.

Further Feedback

Figure 5 is another direct-coupled two-transistor circuit which uses negative feedback, but this time from the collector of Q2 to the emitter of Q1. As shown the circuit is impractical (because there are no bias components) and a very useful compromise is to combine this circuit for AC feedback with the previous circuit for DC feedback (Fig. 6). The gain of this example is approximately (R7 + R3)/R3, and the bias is set by the DC network R3, R6, R5. The design procedure is to decide on operating currents and voltage levels as before, calculate load

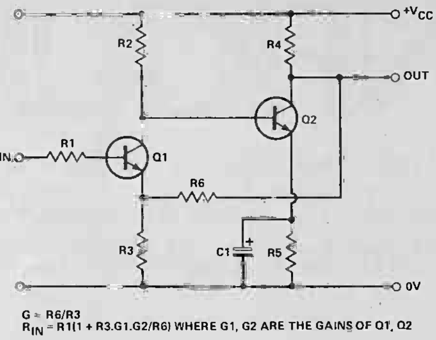


Fig. 5 An alternative two-transistor circuit, using collector-to-emitter feedback.

resistor values, and then assuming that V_{C2} will be 0V6 less than V_{C1}, find a value for R5. Make R6 a size that will suit its use as an input load (around 4k7) and calculate a value for R3 which will make the emitter voltage of Q1 0V6 less than the emitter voltage of Q2. You can then pick a value for R7 which will give you the gain you want, but remember that R1 (the source resistance of the signal supply stage) will form a potential divider with R6, so that the overall gain will be lower than the calculated gain from the stage.

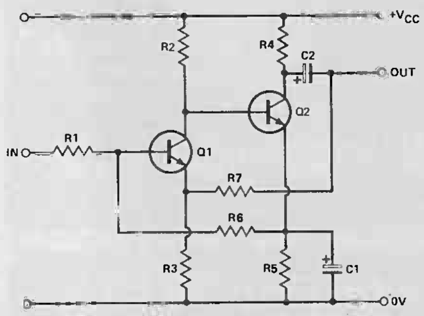


Fig. 6 Combining feedback systems, using one for DC and the other for AC.

After Twins, Triplets!

On the basis that negative feedback works best when there is plenty of gain, three-transistor amplifiers should perform better than twins. There's a catch, though, concerned with time constants. Each time constant in a signal path will cause phase shift at one of the extremes of frequency, high or low. Coupling capacitors will cause phase shift at low frequencies, as will emitter decoupling capacitors, and stray capacitances will cause phase shifts at high frequencies — usually beyond the audio limits. This phase shift in each time constant will cause a total phase shift over the whole amplifier circuit that can be quite large — and when it approaches 180° your negative feedback is in danger of turning into the opposite type — positive feedback. This converts your amplifier into an oscillator if the gain in the amplifying part is greater than the losses in the time constants at the frequency where the 180° phase shift occurs. Calculations on this are far from simple unless you use a computer (and if you ordered the BBC Model B, then it's quicker to work it out on an abacus anyway). Two useful points are (a) to use direct coupling as far as possible, and (b) if you must use time constants, don't use a set of identical time constants.

A Compromising Situation

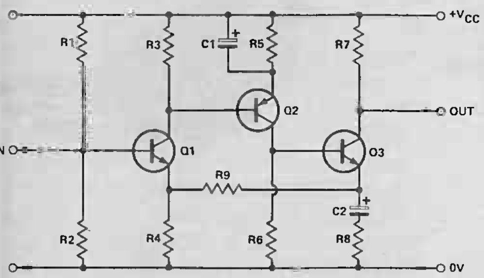
A circuit which can be a useful compromise is shown in Fig. 7. It uses direct coupling, made possible by having a PNP transistor as the intermediate stage, and the negative feedback is used to stabilise the bias as well as the gain. Looking at bias first, the base voltage of Q1 is set by R1 and R2, and the feedback of bias is achieved by passing all the DC current from Q2 through the emitter resistor of Q1.

This would cause the AC gain to be very low, but by using R8, which is coupled to the emitter of Q3 through C2, the amount of AC signal at the emitter of Q3 can be reduced, allowing the amount of AC signal fed back to be controlled. The gain of the stage is given by (R7/R8) × (R9/R4).

Design starts as usual with biasing. Assuming, as is reasonable, that the first transistor will be run at a lower current than the others to reduce noise, we can choose current levels of 100 uA for Q1 and 1 mA each for the other two. We can then choose voltage levels, making the voltage of the collector of Q1 reasonably high, the collector voltage of Q2 reasonably low, and the collector voltage of Q3 about midway between V_{CC} and the collector voltage of Q2. These figures and our selected currents allow us to calculate values for load resistors and for R5.

With this done, we can start to look at R9 and R4. We have to allocate the total voltage at the emitter of Q3 between these resistors, and a ratio of something like 5:1 is reasonable here — the voltage across R9 being about five times as much as the voltage across R4. I've used 6 in the example. Since R9 carries the allocated current of Q3, its value can be calculated right away, and the value of R4, which carries the current of Q1 as well as that of Q3 can be calculated.

The only remaining factor is R8, whose value then decides the total gain. Calculate the gain contribution of R9/R4, and divide this into the total gain required. This latter figure will then be the ratio R7/R8, and since you have already decided on a value for R7, a value of R8 can be calculated, followed by R1 and R2. C1 and C2 should have values which allow the gain to be maintained at the lowest frequencies at which the amplifier will be used, and the time constants R5-C1 and R8-C2 should be quite different.



EXAMPLE: V_{CC} = 9V, V_{c1} = 7V, V_{c2} = 3V, V_{c3} = 7V.
MAKE I_{c1} = 0.1mA, I_{c2} = I_{c3} = 1mA.
THEN R3 = 5k0, R5 = 1k4, R6 = 3k0, R7 = 5k0 (USE NEAREST PREFERRED VALUES)
R9 CARRIES 1mA, SO R9 + R4 HAS A 2V4 DROP.
ALLOW 0V4 AT THE EMITTER OF Q1 AND 2V ACROSS R9, SO THAT R9 = 2k0.
0V4 ACROSS R4 FOR 1.1mA MEANS R4 = 363R.
V_{b1} ≈ 1V SO THAT R1 = 8k0, R2 = 1k0.
R9/R4 = 6 SO FOR A GAIN OF 100, R7/R8 = 100/6 ≈ 16.
THUS R8 = 5k0/16 ≈ 300R.

Fig. 7 A three-transistor circuit, making use of a PNP transistor to change DC voltage levels and contribute gain.

Which Do You Prefer?

Finally, an important point. The resistor values which are obtained by calculation are seldom preferred values, so obviously you have to substitute the nearest preferred values. This is not a major worry, because feedback amplifiers are self-correcting, but there will be one resistor in each circuit whose value sets the gain and which will have to be adjusted — R8 is such a value in Fig 7. The other point which is often raised is — why bother about running Q1 at low current to reduce noise? Since this is a feedback amplifier, the feedback will do a pretty good job of noise reduction anyhow. The answer is that you use negative feedback to stabilise gain and operating conditions. It doesn't make a lousy amplifier into a good one; all it can do is to make a sound design easy to analyse and reliable to mass-produce, so you aim to start with a good amplifier and add negative feedback to it. **ETI**

DESIGNER'S NOTEBOOK

Tim Orr has taken time off from musical circuits to produce this article on instrumentation techniques. If you're one of those people who isn't quiet sure exactly what a decibel is, you'll find this feature particularly helpful.

The following sections describe several electronic measurement techniques, and by adopting a modular approach it will be possible to construct a wide range of electronic measurement systems. First of all, though, we'll look at some of the units involved and try to clear up some of the confusion about them.

The Decibel

The decibel (dB) is a convenient way of expressing signal gain or loss in a system. When a signal passes through several amplifiers or filters the signal level may alter; the overall system gain is the product of all the individual gains. However, it is easier to add a series of numbers than to multiply them and so it is easier to convert the signal gains into a logarithmic equivalent which can then just be added or subtracted. This is equivalent to multiplying and dividing in the linear world. The calculation of gain in dBs is shown Table 1. The chart shows a range of dB values and their equivalent multiplier. These multipliers are often abbreviated by engineers (see rule of thumb), just to make life easy.

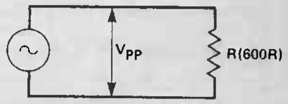
TABLE 1

dB	MULTIPLIER	RULE OF THUMB MULTIPLIER
+80	×10,000	10,000
+70	×3162	3,000
+60	×1000	1,000
+50	×316.2	300
+40	×100	100
+30	×31.6	30
+20	×10	10
+18	×7.94	8
+12	×3.98	4
+10	×3.16	3
+ 6	×1.99	2
+ 3	×1.41	1.4
0	×1.00	1.0
− 3	×0.708	0.7
− 6	×0.501	0.5
−10	×0.316	0.3
−12	×0.251	0.25
−18	×0.126	0.125
−20	×0.100	0.10
−30	×0.032	0.03
−40	×0.010	0.01
−50	×0.0032	0.003
−60	×0.001	0.001
−70	×0.00032	0.0003
−80	×0.0001	0.0001

$A_V = \frac{V_{OUT}}{V_{IN}}$, OR IN dBs = $20LOG\left(\frac{V_{OUT}}{V_{IN}}\right)$

0dBm IS USED AS A VOLTAGE MEASUREMENT.

0dBm IS A SINEWAVE LEVEL THAT DISSIPATES 1 MILLIWATT OF POWER INTO A 600R LOAD



THE POWER DISSIPATED = $\frac{V_{PP}^2}{8 \times R}$

∴ FOR 1mW, $V_{PP}^2 = 8 \times R \times 10^{-3}$

$V_{PP} = \sqrt{8 \times 600 \times 10^{-3}} = \sqrt{4.8}$

$V_{PP} = 2.1909V$

∴ 0dB IS EQUIVALENT TO 2.1909 V_{PP} or 0.7746 V_{RMS}

THIS IS USUALLY ROUNDED OFF TO 2.2 V_{PP} or 0.775 V_{RMS}

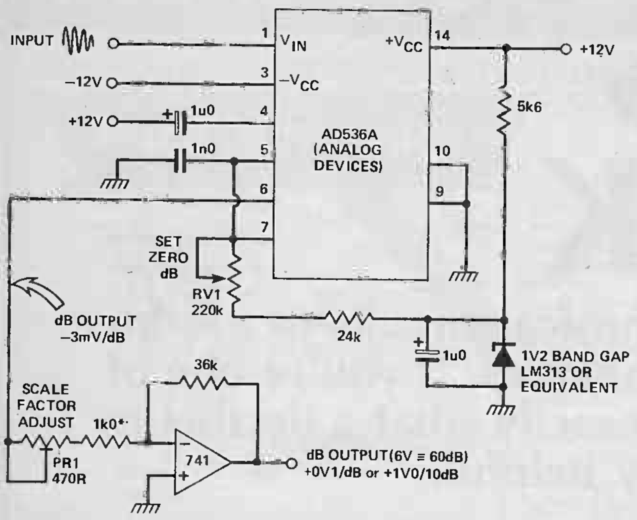
Fig. 1 Everything you ever wanted to know about the dBm!

The dBm

The dBm is used to describe a signal level and not signal gain and loss as for the dB (Fig. 1); it is widely used in audio and telecommunication engineering. For example, microphone amplifiers often have their equivalent input noise voltage expressed in dBm. If, say, this noise is −124 dBm and a microphone is being used that delivers a signal level of −40 dBm, then the signal-to-noise ratio at the output of the amplifier is 124 − 40 = 84 dB. Not exactly a difficult calculation! Table 2 shows a chart of dBm signal levels. The dBm is meant to be used for 600 ohm impedances; however the same signal level across any load is referred to as the dBU.

TABLE 2

dBm	V RMS	V_{p-p}
+20	7V75	22V
+18	6V16	17V47
+12	3V08	8V76
+10	2V45	6V95
+6	1V55	4V39
+3	1V094	3V11
0	775 mV	2V2
−3	549 mV	1V56
−6	388 mV	1V1
−10	245 mV	695 mV
−12	197 mV	553 mV
−18	97.6 mV	277 mV
−20	77.5 mV	220 mV
−30	24.5 mV	69.5 mV
−40	7.75 mV	22 mV
−50	2.45 mV	6.95 mV
−60	775 uV	2.2 mV
−70	245 uV	695 uV
−80	77.5 uV	220 uV
−90	24.5 uV	69.5 uV
−100	7.75 uV	22 uV
−110	2.45 uV	6.95 uV
−120	775 nV	2.2 uV
−130	245 nV	695 nV



*USE A TEL LABS Q81 IF POSSIBLE. THIS HAS A T_C OF +3500PPM/°C. IT COMPENSATES FOR DRIFT IN THE AD536A

Fig. 2 Circuit for a dB meter using an Analog Devices IC.

Measuring Decibels

The AD536A is an RMS-to-DC converter chip, which is made by Analog Devices. It contains an absolute value circuit, a squarer/divider and a current mirror. With these elements it can compute a true RMS value or a log (dB) equivalent of the input signal. The circuit in Fig. 2 has a useful operating range of 60 dB. PR1 sets the scale factor of the meter. Insert a 1 V peak-to-peak sine wave and then change it to 100 mV peak-to-peak; this is to a change of 20 dB and so the preset PR1 should be adjusted for a 2 V change in the output voltage. RV1 sets 0 dB by inserting a current that cancels the log output current from the squarer/divider. This can be used to measure absolute voltage in dBm by setting zero at an input voltage of 775 mV RMS. Alternatively it can be used to compute a dB ratio. Insert a signal, set zero and then change the input signal level — the resulting output change will represent the dB change in input level. Note that the input of the AD536A can accept DC signals, but a small DC offset on the input signal will add a large error to low level readings, For the computation of dBs for AC signals it is best to AC-couple the input with a 4u7 capacitor.

RMS To DC Conversion

The AD536A calculates the true RMS value of the input signal by converting it to an absolute value. It is then squared, averaged and then square rooted. The result is a DC voltage that is equal to the RMS value of the input signal (Fig. 3). Most moving coil AC voltmeters are

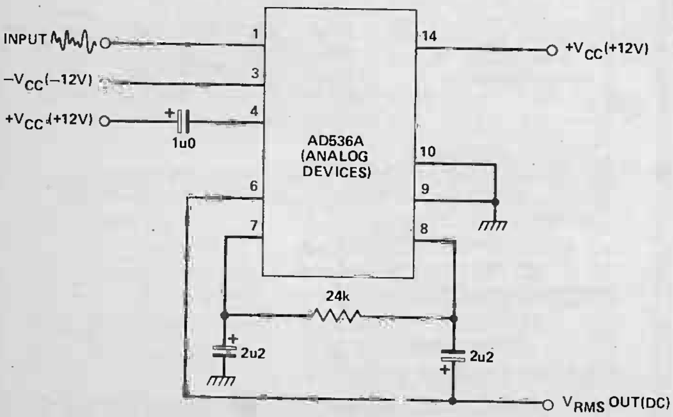
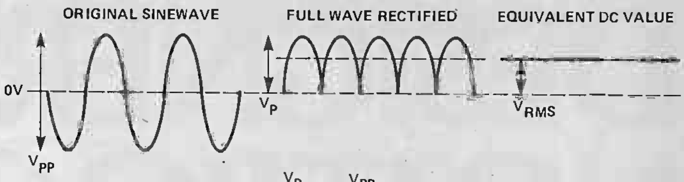


Fig. 3 The same chip can be used for an RMS to DC converter.



$$V_{RMS} = \frac{V_P}{\sqrt{2}} = \frac{V_{PP}}{2 \times \sqrt{2}}$$

$$V_{RMS} = 0.707\ V_P = 0.353\ V_{PP}$$

Fig. 4 Sine wave level measurement.

calibrated for sine wave operation only; the peak voltage of the sine wave is multiplied by 0.707 to produce the RMS value (Fig. 4). This type of meter will give wrong readings if noise, square waves, triangle waves or any non-sinusoidal signals are measured, as in Fig. 5. To measure any non-sinusoidal waveform a true RMS converter is needed, otherwise the reading could be wrong by a factor of 5!

WAVEFORM	RMS	CREST FACTOR*
V_P SINEWAVE	$0.707\ V_P$	$\sqrt{2} = 1.414$
V_P SYMMETRICAL SQUAREWAVE	$1.0\ V_P$	1.0
V_P TRIANGLE WAVE	$\frac{V_P}{\sqrt{3}} = 0.577\ V_P$	$\sqrt{3} = 1.732$
$2\ V_P$ NOISE	TRUE RMS	3.0 TO 4.0 DEPENDS ON THE SPECTRUM
V_P SPIKE	TRUE RMS	10.0 DEPENDS ON SHAPE!

$$\text{*THE CREST FACTOR} = \frac{V_P}{V_{RMS}}$$

Fig. 5 Measuring different waveforms.

Noise Annoys

All electronic components generate noise, and the combined effect of all the noise components in a system gives rise to the phenomena of a noise floor. This is a background noise signal that is forever present. The measurement that is often used to describe the noise performance of a system is called the signal-to-noise ratio which is expressed in dB. This is the ratio between the normal operating signal level and the residual noise level with the signal removed. Measurement of the signal is relatively straightforward; a sine wave signal can easily be measured with an AC voltmeter, to give the RMS value. The noise measurement is more difficult. First, the noise level will probably be very low and so it will need amplification. Second, we will only be interested in noise within a certain bandwidth. Noise outside our selected bandwidth will make the noise reading seem larger than it really is. Therefore we will need to band-limit our amplifier. For audio applications a bandwidth from 20 Hz to 20 kHz is often used.

Finally, noise is a random process and it has a constantly varying ratio between its peak and RMS voltage. A simple rectifying and smoothing process will give an erroneous reading and a *true* RMS-to-DC converter is need-

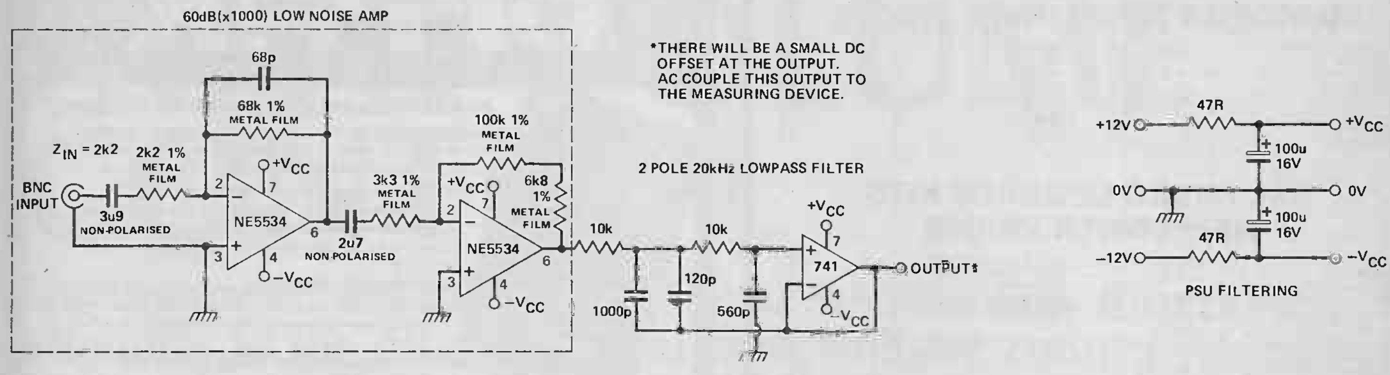
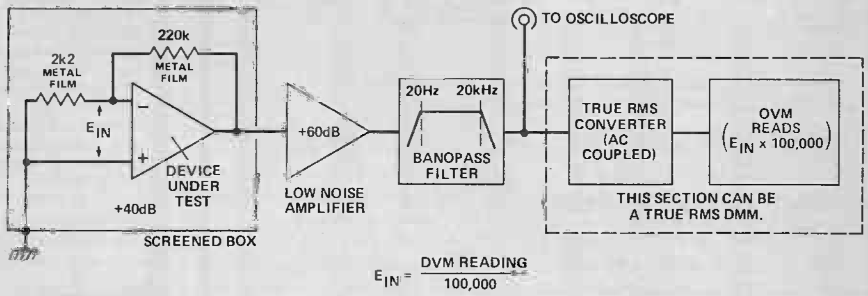


Fig. 6 (Above) A low noise 60 dB amplifier and 20 Hz to 20 kHz filter.

Fig. 7 (Right) Measuring noise in op-amps.



ed for correct readings. Thus by using a band-limited amplifier with a true RMS converter it is possible to accurately measure the residual noise level. A suitable amplifier and filter for audio work is shown in Fig. 6. The NE5534 op-amps are low noise devices, so they will contribute very little noise to the output signal. The amplifier has a voltage gain of 60 dB and so the output reading must be attenuated by the same amount; Table 1 shows the equivalent is ×1000, so millivolts turn into microvolts. This calculation assumes a flat noise spectrum.

Figure 7 is a circuit that measures the equivalent input noise of op-amps. The circuit has an overall gain of 100 dB (× 100,000) and so the DVM reading must be attenuated by this factor; additional filtering may be included to remove hum or to produce an A-weighting.

The noise often has an A-weighting frequency response applied to it before measurement (Fig. 8). This curve is similar to the frequency sensitivity of the human ear and so by A-weighting the noise measurement it becomes a subjective quantity. It also improves the apparent noise performance figure. Usually when you make a noise reading there is a mains hum component that is actually bigger than the noise. This can be removed with a 400 Hz high pass filter as in Fig. 10. This filter also removes some of the energy of the noise below 400 Hz, but even so the noise reading is still useful. Sophisticated digital multimeters often give you the choice of 400 Hz high-pass, 30 kHz and 80 kHz low-pass and A-weighting filters; sometimes they also have the facility for true RMS, dBm and dB ratio measurements.

Noise in op-amps is specified in nV/√Hz. The equivalent input noise voltage is:

$$E_{IN} = E_n \times \sqrt{\text{BANDWIDTH}}$$
$$= (E_n \times 141)\ \text{nV for a 20 kHz bandwidth}$$

where E_{IN} is the equivalent input noise voltage in V RMS
E_n is the input noise in nV/√Hz.

The calculation assumes a flat noise spectrum.

Next month Tim Orr looks at the measurement of voltage, current, resistance and temperature.

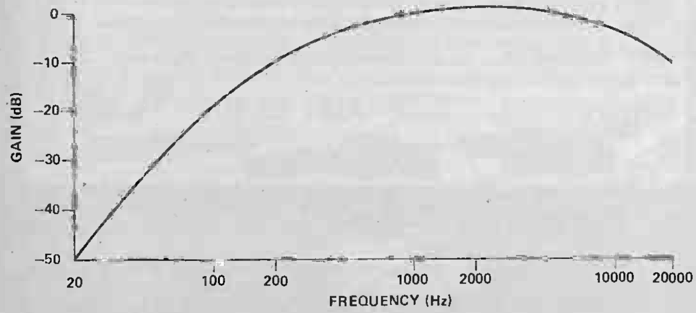


Fig. 8 The IEC 'A' weighted curve.

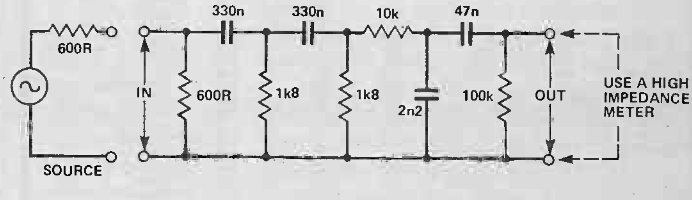


Fig. 9 An IEC 'A' weighting circuit.

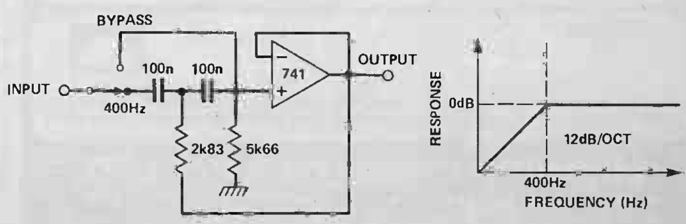


Fig. 10 A 400 Hz two pole high pass filter.

ETI

AUTO-VOLUME CONTROL

Having sorted out mike switching in July, we now turn our attention to the other PA nightmare, volume control. This clever little gadget will sort out your problems — automatically. Design by Vivian Capel. Development by Peter Green.

Automatic volume control has been used for years in broadcast radio receivers — at least it has been termed automatic volume control although it was actually automatic gain control. AGC responded not to changes in the audio frequency output, but fluctuations in the level of the RF carrier. Part of the carrier appearing at the last IF stage was rectified, applied to a circuit having a time constant sufficient to remove audio modulations, then fed as a DC control potential to the grid circuit of the first IF stage. Thus fading was reduced and strong local carriers did not produce overloading of later stages. This illustrates the rather confused manner in which the term is used.

Expanding On Compression

Another confusion often encountered is between automatic volume control and volume compression. Although both are concerned with audio levels nd their control, there is an important difference that dictates which feature should be used in a particular application.

With volume compression, the dynamic range is reduced and the gain of the circuit varies according to the amplitude of the signal. This reduction may take place over the whole range of signal levels, or it may start at some predetermined threshold below which the circuit behaves normally with linear gain. Above it, the signal produces an output which is proportionally less as the amplitude increases.

Volume compression is used where the dynamic range the range between the loudest and softest signals) is too great for the equipment or system to safely handle. Either the loud signals would overload the circuit, overmodulate the carrier, or introduce unacceptable distortion, or the soft signals would approach the noise level so that a poor signal-to-noise ratio would result. Broadcast radio transmissions are an example of the use of compression, and the compressed result is generally considered satisfactory for domestic listening.

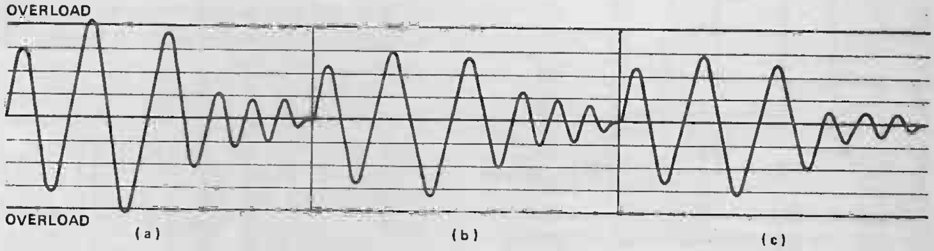


Fig. 1 A signal comprising successive loud and soft tones. The loud tone exceeds the overload level of the equipment (a). With volume compression, the loud tone is reduced more than the soft one, reducing dynamic range (b). With automatic volume control, all the signals are equally reduced (c).

Orchestral Manoeuvres

It may be worth mentioning at this point that there is often a considerable 'overkill' in the range stipulated in hi-fi circles. Some quote figures around 70 dB as the range between the quietest and loudest sounds produced by a symphony orchestra. Measurements of sound levels made by the author at public symphony concerts failed to produce anything like this. The difference between the quietest passages (whispering strings) and full-blooded climaxes with percussion, brass and everything else playing flat out was little more than 45 dBA.

The opportunity has not arisen for measurements during large scale choral and orchestral works in which a somewhat wider span could be anticipated, but from theoretical calculation it would not be a lot more. (For example if a choir and orchestra each generated 45 dB, together the combined sound level would be only 48 dB.)

Levelling Off

Automatic volume control does not reduce dynamic range by lessening the amplitude of loud signals more than that of quiet ones. It reduces the overall level (so that both loud and quiet sounds are affected equally) when the level is high, and increases it when the overall level is low. The operation is the same as that of a manual volume control, except it is automatically actuated by the signal.

One common use is the automatic recording level control that is incorporated in most portable cassette recorders. This permits recordings to be made without trial runs and careful watching of the recording level meter. Though very useful for spontaneous recordings it does have its drawbacks in being too inflexible in the form usually used. Fades, either up or down, cannot be executed as there is normally no recording level manual control. The effect is also less than satisfactory when recording music which has sudden changes of volume. The manual override which accompanied the first automatic level circuits in recorders now seems to have disappeared completely.

PA Problems

Public address systems are another application for which auto level control is ideal. Many platform orators are notoriously bad microphone users; they turn away and often step away from the rostrum, while at other times some lean forward and shout into the microphone to emphasise a point. As the level varies according to the square of the distance between the lips of the speaker and the microphone, some pretty hefty volume changes take place. With interviews or forums there may be a variety of participants, some soft-spoken and some loud. Any of these may shift his position during the programme and wind up nearer or further from the microphone. The poor PA operator struggles manfully to compensate by rapid adjustment of the sliders, sometimes the wrong ones, and in the wrong direction. No wonder they sometimes emerge from a session as nervous wrecks!

It was for the purpose of PA control that our unit was designed, and very useful it has proved too. However, there are no doubt other applications for which readers may find it practical. Being comparatively small it could be built into most existing amplifiers or mixers, drawing only a small amount of current from the power supply. However, if desired it could be mounted in a separate box with a battery supply.

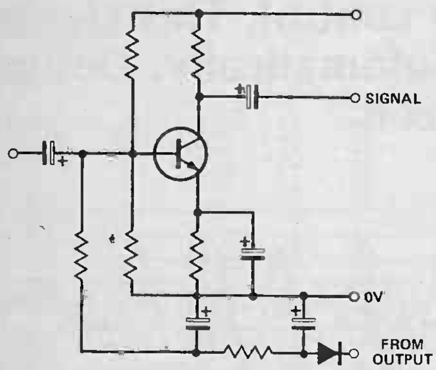


Fig. 2 A simple AVC circuit as used in some cassette recorders for recording level control. It can be overloaded as the forward bias is reduced by a strong signal.

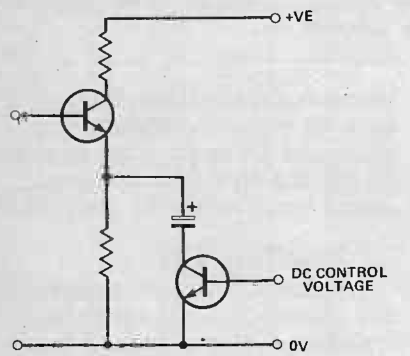


Fig. 3 Experimental circuit for controlling the gain of a stage by varying the negative feedback.

Alternative Technology

Several methods were tried (and rejected) in order to achieve satisfactory auto-control. The simplest method (which hs been frequently used for cassette recorder ALC circuits) is to rectify part of the output and apply it as negative bias to an early NPN transistor (or positive bias to a PNP device). Thus the forward bias is reduced, reducing the collector current and also the gain of the circuit (Fig. 2).

There are several snags with this arrangement. One is that with the reduction of base current on which the signal current is superimposed, the capability of the circuit to handle large signals is also reduced. Now this occurs precisely at the time when large signals are present at the base, which are not affected by the gain of the following circuit. Hence there is a strong possibility of overloading and distortion, as the

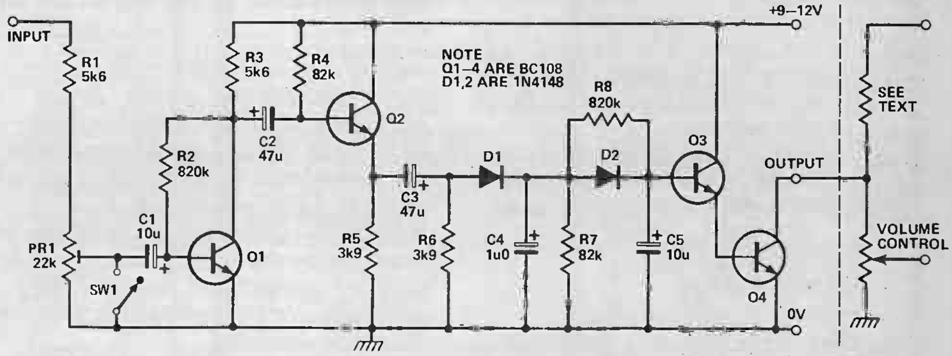


Fig. 4 The circuit of the AVC unit, showing the connections required when it's built into existing equipment.

HOW IT WORKS

The signal is sampled from the output of the mixer or some other convenient high-level point, and is applied to the potentiometer PR1. This permits the amount of AVC to be preset. SW1 shorts the preset wiper to earth, which allows the circuit to be switched out without disturbing the control setting.

Q1 amplifies the control signal and the output is applied to Q2; this is an emitter follower with a low impedance output and high current sourcing. The purpose of this is to ensure that the capacitors C4 and C5 charge rapidly at the onset of a sudden loud sound and so achieve a quick reaction. The capacitance of these components has been limited for the same reason.

After rectification by D1, the DC control signal charges C5 through D2 which offers a low impedance path. However, the capacitor cannot discharge back through D2 because of reverse polarity, but does so through R8 and R7. Thus the discharge is considerably slower than the charge, so the control signal doesn't fluctuate with every change in amplitude of the original signal (which would result in volume compression). Instead it maintains a level which corresponds with the average level of the ncoming signal; thus if a quiet signal follows immediately after a loud one, the same dynamic range is maintained and the gain drifts back only slowly to the previous level. Small changes in signal amplitude as in normal speech have little effect.

The rapid initial response greatly reduces the devastating effect of the explosive consonants P and B when spoken too close to the microphone. But because the general overall level is governed by the average input signal, the circuit evens out longer variations which arise when a speaker turns his head away from the microphone to address one section of his audience.

Transistor Q3 drives the control transistor Q4, which behaves as a variable resistor and is connected as the bottom leg of a potentiometer across the signal source. The upper leg is a fixed resistor, and the tap between them is the variable point for feeding the following amplifier. A shunt resistor is connected across Q4 to maintain the load on the source when the transistor is cut off.

When a large signal is received, Q4 is made conductive and the resistance of the bottom leg reduced, changing the ratio between it and the top leg. When the signal is small the opposite happens and Q4 becomes less conductive, increasing the lower leg resistance. Thus the circuit operates in the same manner as a manual volume control, by reducing the signal applied to the controlled stage rather than changing the gain of the stage itself. Therefore it can't be overloaded and self-generated noise remains the same. Noise from earlier stages is reduced along with the signal, so the drawbacks of other control circuits are eliminated.

Making A Beeline

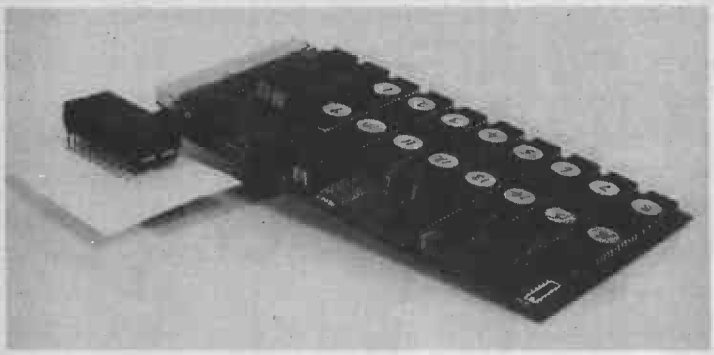
Expand a standard VIC computer from 5K to 32K memory, with 80 column display and word processing capability and the benefits are enormous. Beelines Ltd, the Bolton-based computer research and development specialists have done just that with the Beelines BeeBox System. The BeeBox unit also has RS232 capability, so it interfaces with printers or telephone modems, and it can be fitted with no loss to all the standard VIC features. Thus it will still support disk drives, games and business ROM cartridges, cassette deck, domestic TV output, printer and not impair use as a standard VIC.

Beelines sees the BeeBox as the answer to realising the full business potential of the VIC, with a package designed to give anyone a small, entirely portable viewdata business system at a sensible and competitive price. For around £650 plus VAT, you get the VIC-20, a BeeBox 40, modem and RS232 card, power supply, cassette deck and carrying case. If you don't want to buy everything at one go, you can buy in stages and build up the system to its full capacity — this also applies to current owners of the BeeBox 40. Beelines is the research and development wing of B & B Computers, one of the leading computer retailers in the North West. It has a mail order division on Freepost, Bolton BL3 6YZ and a 24-hour answer phone service on 0204 385299.

The Stripper

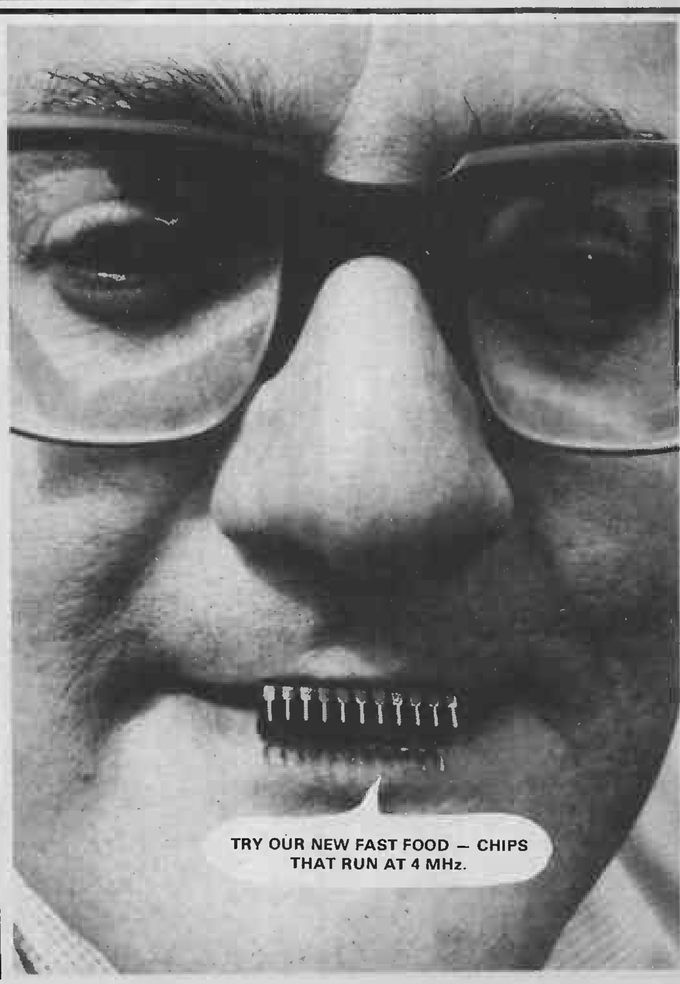
Wire strippers are wonderful things; every hobbyist should have one. Until you've actually used the things you don't realise how interwiring, though never actually being fun, can at least be made tolerable. Good news, then, from AB Engineering — their new wire stripper, the Micro-strip, is self-adjusting for depth of incision and gripping pressure, and can cope with all PVC insulations and some thermo-resisting insulation. A patented self-adjusting floating cam adjusts the stripper's jaws to the correct stripping depth and at the same time sets the gripping pressure exerted by the jaws to prevent damage to the insulation. Using the tool is simplicity itself; a stop is set to the required stripping length, then the wire is fed through the front of the tool to the stop position and the handles squeezed. That's all — sheer luxury!

The Micro-strip will deal with all sizes of wire up to a maximum diameter of 1.5 mm without adjustment, although an adjustment mechanism is provided for very thick or thin insulation. It doesn't matter if the wire is single, stranded, flat or round. Blades can be replaced in less than two minutes; a self-sharpening wire cutter is also built into the tool at an ergonomic point, which roughly translated means you get maximum leverage. People who know what's good for them can find out more from AB Engineering, Timber Lane, Woburn, Milton Keynes MK17 9PL.



TUGging Away

Products designed for the Microtan 65 system are constantly emerging from the Tangerine Users Group, otherwise known as TUG, and the latest one is an EPROM Storage Card, otherwise known as ESC. This is built on a high quality PCB and has been specifically designed for mass EPROM storage. Don't let them run around your home messing up the carpets, keep them on this board where you know they're safe. The module accepts 32/64K of 2716 or 2732 EPROMs respectively, allowing very fast transfer of mass data or stored programmes into user RAM. Programmes can be accessed manually or automatically under software control, allowing repetitive calls to the data store for multiple programming menus. The initial design of the module allows for the banking of four of the cards on the system rack. Group members will pay £37-38 plus VAT due to their discount, but we weren't told the price for non-members. If you want to buy an ESC (or join TUG) the address is 16 Iddlesleigh Road, Charminster, Bournemouth, Dorset BH3 7JR. Incidentally there's a PS on the press release concerning the photo — it says "the young fellow looking on is Eric, he's not for sale". The first person to tell us what on earth they're talking about doesn't win a prize.



PARTS LIST

Resistors (all ¼W, 5%)	
R1,3	5k6
R2,8	820k
R4,7	82k
R5,6	3k9
Potentiometer	
PR1	22k
Capacitors	
C1,5	10u 16 V PCB electrolytic
C2,3	47u 16 V PCB electrolytic
C4	1u0 16 V PCB electolytic
Semiconductors	
Q1-4	BC108
D1,2	1N4148
Miscellaneous	
SW1	SPST or SPDT switch (see text)
PCB (see Buylines)	

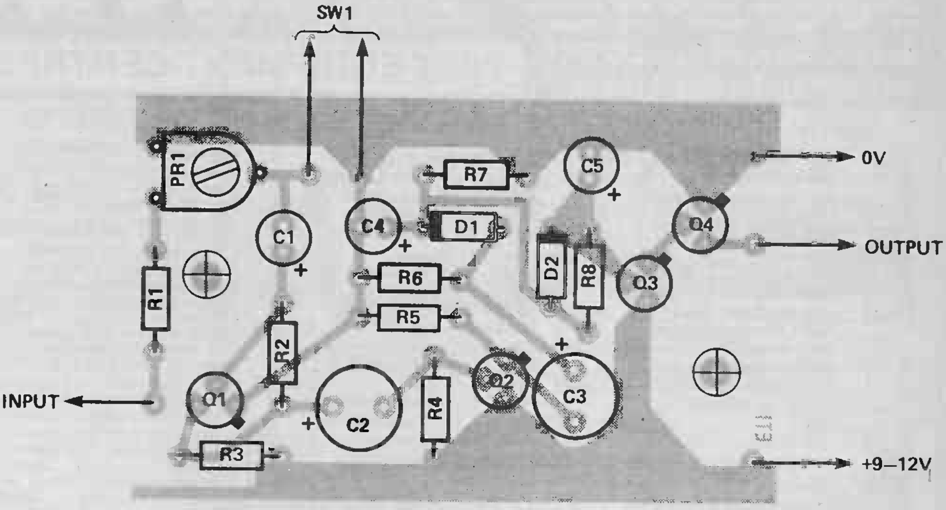


Fig. 5 Component overlay for the AVC unit.

stronger the signals are, the smaller the base current becomes.

Overloading is less likely in a cassette recorder when the internal microphone is being used, as close-talking is not usual under these circumstances. But it could occur if a high sensitivity external microphone or auxiliary input were being recorded. It could certainly occur when used as a general automatic volume control for PA or other purposes.

The other drawback is that noise in a transistor is dependent on the current in the base and collector circuits; the larger the current, the greater the noise. As the current varies with the operation of the automatic control, noise will also vary. However, the current, and hence noise, is greatest in the absence of a signal, and least when the signal is at a maximum. This of course is the opposite to what it should be, as noise is relatively unimportant in the presence of large signals which mask it.

Variable Variation

Another circuit which at first seemed promising (Fig. 3) consisted of a transistor used as a variable resistor controlled by a rectified voltage derived from the output, which was connected in series with an emitter bypass capacitor. The signal developed over the emitter resistor by the collector current is in opposition the the applied signal between base and emitter, hence it provides a form of negative feedback. When bypassed with a capacitor having a low reactance compared to the emitter resistor, the feedback disappears.

A variable resistor in series with the capacitor enables the gain of the stage to be varied in sympathy with the signal amplitude by changing the total impedance across the emitter resistor and hence the amount of negative feedback.

Unfortunately, the circuit proved unsatisfactory in practice because the DC charge in the capacitor changed according to the value of the series resistance.

Design Developed

Finally, the circuit here given was developed, and with some minor modifications over a period of use has proved to be very effective. When the device is built into existing mixers or amplifiers, the top leg resistor of Fig. 4 can be inserted in series with the main volume control at its 'high' end. The transistor is connected across the control which therefore serves as the shunt described in the 'How It Works' section. The value of the top leg resistor should be about a tenth of that of the volume control, so a 25k control would need a 2k2 resistor. This will reduce the gain of the amplifier by an eleventh or less than 1 dB. If the unit is constructed in a separate control box, the shunt resistor can be 47k and the upper one rather less than a tenth at 3k3.

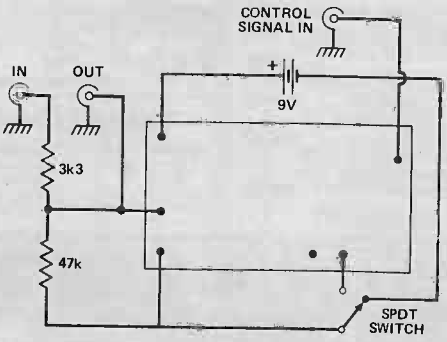


Fig. 6 Board connections to peripheral components when used as a separate control box. PR1 can be replaced by a manual control on the panel if desired.

Manual Mode

The AVC control is shown as a preset mounted on the circuit board, but if desired it could be an ordinary manual control fitted either to the control panel of the mixer or on the case of a separate unit.

A manual control is useful as different conditions call for different degrees of control. In the lowest positions, operation of the control automatic circuit is not noticeable to an audience, but the operator can see from his output meters that the loudest signals are restrained and kept within the range of the equipment. At the other extreme, if turned fully up, the loud sounds come out quieter than the soft ones! This degree of control would obviously never be used, and the setting will be somewhere well below that.

Modifications

A switch can be incorporated, if desired, to short the wiper of the control to earth. The same effect is obtained by turning down the control to zero, but the separate switch enables a particular control setting to be preserved when the circuit is switched out. Another optional feature is that the switch can be a double pole type with the second pole arranged to switch a LED on and off as an indication that the automatic circuit is in operation.

BUYLINES

All the components for this project are the common-or-garden variety so you should be able to get them from any of the companies advertising components in this issue like Rapid Electronics, Bi-Pak, Ace Mailtronix or Electrovalue. People unable to etch their own PCB can find our PCB Service order form on page 87.

ETI

DUAL LOGIC PROBE

With all the logic probe designs that have been published, you didn't think we could come up with an original design, did you? Oh ye of little faith . . . this one's cheap, compact, and clever. Design and development by Phil Walker.

This month's 'project par excellence' from the ETI workshop is a very useful dual purpose logic tester. It is designed with CMOS in mind and uses mostly logic ICs of that family. Our prototype gives results with pulses down to 200 nS wide at frequencies from DC to over 2 MHz.

The ETI Dicrobe is designed as a dual purpose test instrument. One half of it is a reasonably conventional logic probe giving indications for logic high, low, and pulsing states while also having a transition memory to store those events you might otherwise miss. All these conditions are displayed on a seven-segment LED display to give a practical representation of conditions at the probe tip.

Probing For Pulses

The other half of the unit is a logic pulser which automatically senses the logic level of the secondary probe and gives it a short pulse to the opposite state. This can either be a square wave drive at about 5 Hz or 2 uS pulses, again repeated five times per second.

Two LEDs indicate the logic state on this probe but no indication of pulse conditions is given. The two parts of the instrument can be used independently or together. This enables a considerable amount of useful testing to be carried out with this unit alone.

For instance, the pulser could be applied to a clock input while the primary probe looks at the outputs to see if anything happens. Both probes could be applied to a part of the circuit to see if it is shorted to 0 V or the positive supply; in this case pulses will be seen if everything is OK (even if there is an output from a gate driving it).

On TTL circuits the probe may not operate so quickly due to the low supply voltage but a useful indication of functions should be possible. Bear in mind, however, that the logic high and low levels for TTL are much lower relative to the supplies than for CMOS.

A Tour Round The Circuit

The primary probe circuit uses a 4049 to sense the input, with the advantage of high speed, high input impedance and CMOS logic thresholds. The output from three sections of this drives the edge detector and two segment drivers. The edge detector drives a set-reset latch (to store the fact that a transition occurred) and a monostable (to give indication of previous conditions).

The secondary probe circuit is a little simpler than the primary probe as it is not designed for high speed. Part of a hex Schmitt trigger senses the input and the result of this is stored on a set-reset latch. Another part of the Schmitt package works as an oscillator at about 5 Hz and drives two other sections via CR networks to provide reset pulses for the set-reset latch and enable pulses for the output circuit.

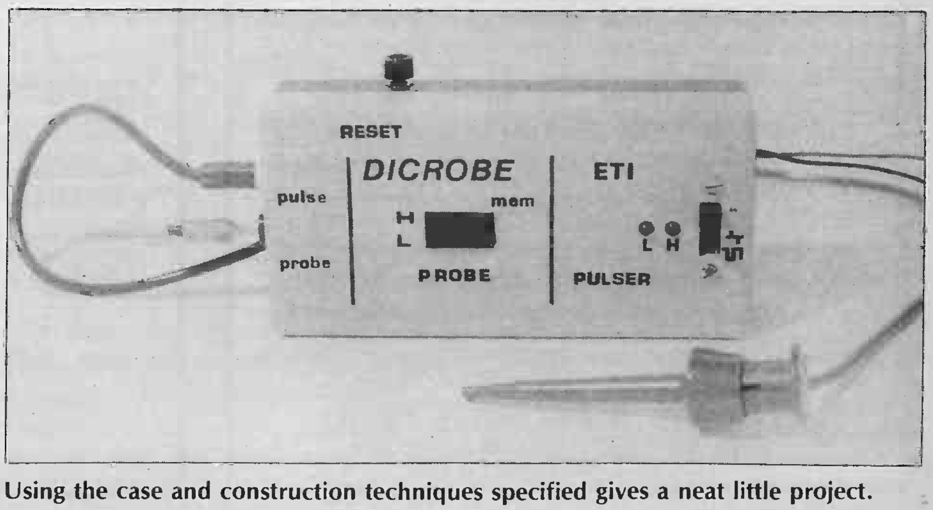
The output transistors are connected so that they are normally off. When a pulse is to be applied to the output the logic drivers apply base drive to one of the transistors. The particular transistor driven is the one which gives a pulse of the opposite polarity to the normal state of the circuit point to which the probe is connected. After the pulse the state of the circuit is sensed ready for the next pulse.

The three-position switch allows the secondary probe to act as a straight logic probe or an automatic pulser with short duration or square wave pulsing.

Construction

The PCB for this project has been designed to fit inside a small plastic box made by BICC-Vero Packaging Ltd. There is very little room to spare inside the box so great care must be taken to use small components and make the connecting wires as thin and short as possible.

A fine tipped soldering iron is essential for this project and great care must be taken to avoid solder splashes between tracks. The ICs can be mounted in sockets of the low profile type and the seven-segment display should be mounted in a socket as well. For the display we used some Soldercon connectors; these are taller than the low profile sockets on the other ICs and held it higher above the board to project through the case lid when assembled. Apart from the LEDs and two resistors, all the other components mount normally on the PCB. Take care to get the diodes,



Using the case and construction techniques specified gives a neat little project.

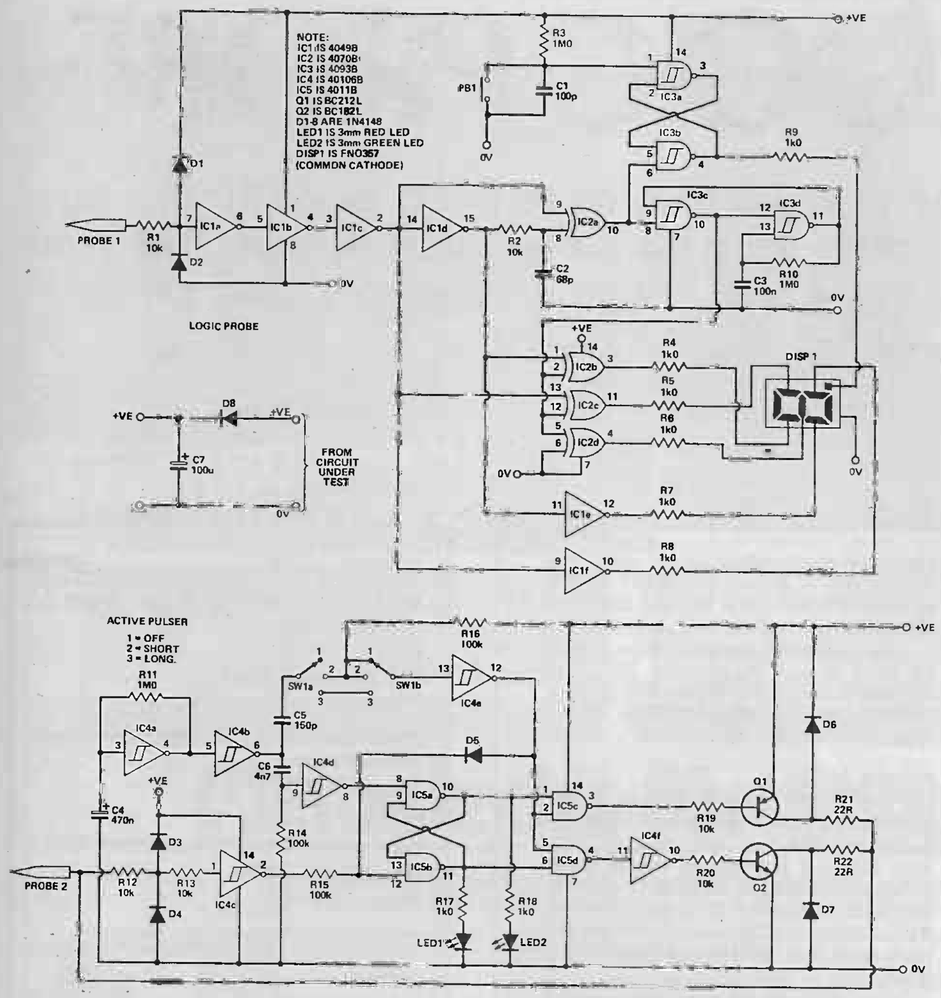
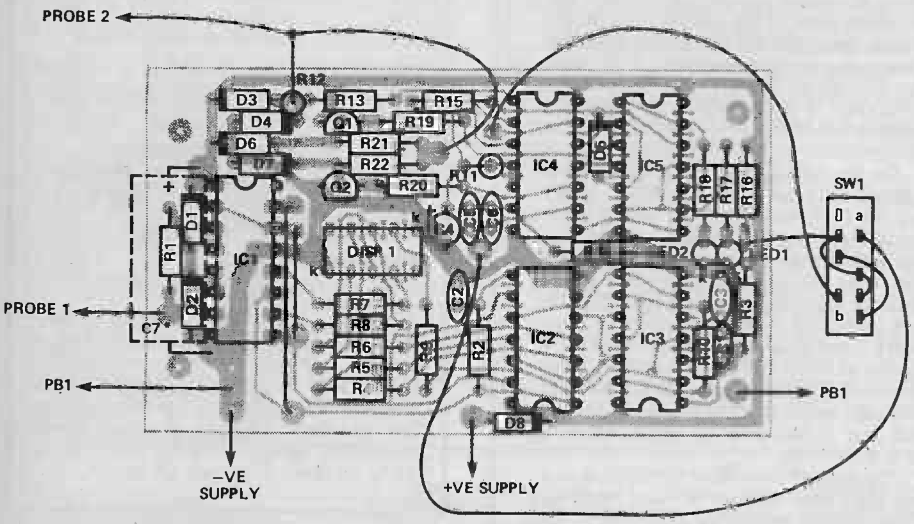


Fig. 1 Complete circuit diagram of the logic probe.



NOTE: k = CATHODE
*C7 IS MOUNTED ABOVE R1,D1,D2

Fig. 2 Component overlay. Note that one end of R12 is soldered to two flying leads, not the board.

transistors, ICs and capacitor the correct way round. The LEDs are mounted so that they will project through the lid.

Two resistors have to be mounted on end so refer to the overlay to see where they should go. Two wire links are needed near IC1 so don't forget these either. The links are close together and should be made with insulated wire. When wiring up, connect lengths of 6" or so to all the connection points indicated except the power supply, which should be 18" or so.

Fit the sockets for the probes into the end of the box where the board mounting pillars are closest together. They must be as near to the bottom of the box as possible. Also fit the reset switch into the side of the box in a similar manner. This switch must be very small to fit in the space provided. If a suitable component cannot be obtained, increase the value of R3 to 10M and use two small bolts as a touch switch. This is suggested only as a last resort!

A small grommet should be inserted in the opposite end of the box to the probe connectors to take the power supply wires. The box lid must have holes cut or drilled in it to clear the display, LEDs and mode switch SW1. SW1 is bolted to the lid and the interpin connections made before wiring to the board.

When fitting the PCB into the box the wires from it must all pass over the ends as there is not enough clearance at the side for them. The probe and switch wires should be short or they will get in the way of the lid. Our prototype used small grommets cut in half as spacers between the PCB and the mounting pillars to get the correct height.

Probing Deeper

The probes themselves were very simple. The main one was a piece of brass threaded rod turned to a point in a drill with a file. This was soldered into a metal tube taken from a piece of plastic connector block. A small piece of thick wire was soldered into the 2 mm plug and then this was soldered into the other end of the metal tube. You may care to experiment with a sewing machine needle instead of the brass rod as these form very durable probe tips.

The other one was simply a piece of wire with a spring probe on the end terminated in another 1 mm plug.

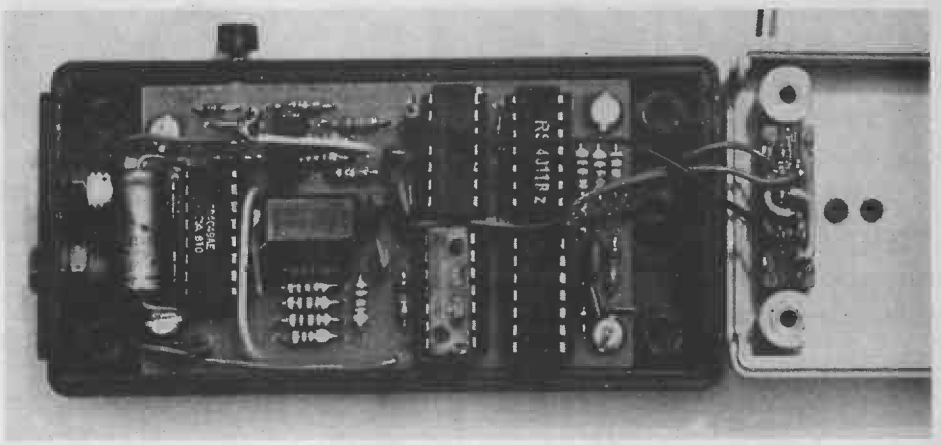
The two probes are interchangeable as required. Power supply wires can be terminated in crocodile clips or anything convenient.

What's The Diagnosis?

The probe is intended for diagnostic work on CMOS circuits where the main part is used to look for the effects caused by the pulser probe being applied to an earlier part of the circuit. This let's us find faults in components such as gates and many counters and other devices. By connecting both probes to the same point, a check can be made to see whether a short circuit exists to either power rail (no pulses detected). It can also be used as a crude signal source if nothing else is available.

The pulser works best if the point to which it is attached is static, as it has a fairly slow response time. Note that when it is in the active pulsing mode, the LED display does not give a true picture of the logic level at the pulse probe.

The main probe will catch pulses down to 200 nS with a 9 V supply which should be adequate for a lot of test work.



PARTS LIST

MAIN LOGIC PROBE
This part of the circuit is based around IC1, 2 and 3. The input from the probe tip passes via a simple protection network R1, D1 and D2 to IC1a. This device senses the logic level at the probe tip while also presenting a very high impedance to the circuit under test. Two more sections of the device, IC1b and IC1c, are used to speed up the transition time when the logic state changes at the input. The output from IC1c is used by IC1f to drive one of the segments of the display. IC1c's output is inverted by IC1d and used by IC1e to drive another segment. These two display segments form the immediate logic state indicator.

The outputs from IC1c and IC1d go to IC2b and IC2c. IC2 is a quad exclusive-OR gate which in this case is being used as a controlled inverter. The outputs from these two sections drive two more segments of the display but these are the ones which indicate the previous logic state.

To detect a transition from one state to another, the output from IC1c is applied to one input of IC2a while the output from IC1d is applied to the other via R2 and C2. These components cause the inputs to IC2a to be slightly out of phase with each other. In their normal rest state the inputs to IC2a would be at opposite logic levels causing the output to be high, but for a short time after a transition the inputs to IC2a will be at the same level and the output will be low during this period.

This low pulse has two effects. The first is to set the latch formed by IC3a and IC3b such that the decimal point LED in the display is lit, indicating that a transition has occurred. The second is to force the output of IC3c high, enabling the astable oscillator formed around IC3d. In fact IC3c and IC3d form a monostable to effectively stretch input pulses and transitions so that they can be seen. The low pulse on IC3c input puts a high on IC3d input which, since C3 has been resting at a high level, will make the output of IC3d go low immediately. This forces IC3c output to remain high via its second input until C3 discharges enough to allow IC3d output to go high again.

The output of IC3c is connected to IC2d which acts as a buffer to drive the centre segment of the display. This flashes to indicate that a transition has occurred. IC3c output also drives IC2b and IC2c, causing them to invert the signal on their other input while the monostable is active. The effect of this controlled inversion is that the last transition is mimicked on the display.

The transition memory, IC3a,b, can be reset at will by pressing PB1. The decimal point on the display will go dark until another transition occurs at the input.

PULSER PROBE
The input from the probe tip is sensed by IC4c, part of a hex Schmitt trigger. Some protection is provided by R12, R13, D3 and D4 against excess input voltage. The output from IC4c goes to the latch formed by IC5a and IC5b via R15. A low on IC4c output will set the latch such that LED1 is alight and LED2 is off.

IC4a, R11 and C4 form a slow speed oscillator which, via a buffer IC4b, drives the pulse generation circuits. On the rising edge of the output from IC4b, IC4d is driven by C6 and R14 to produce a low pulse of short duration. This pulse tries to reset the IC5a,b latch but will only succeed if IC4c output is high. This updates the latch every cycle of the slow clock.

On the falling edge of the slow clock a signal may be passed to IC4e via C5 and SW1. Position 1 of the switch does not allow the pulse to pass and the circuit acts as a slow logic probe only. Position 2 of the switch allows the signal through but connects R16 into the circuit such that it forms a differentiator with C5 and makes the output from IC4e appear as short pulses. In position 3 the output of IC4b is coupled to IC4e via C5 virtually without change. This means that the pulses will be approximately 50% duty cycle.

The output from IC4e (consisting of a low logic level with or without positive-going pulses) passes to IC5c and IC5d. The other inputs to these gates are taken from the outputs of the input sense latch, IC5a and IC5b. If the output of IC4e goes high, then one or other of the outputs of IC5c or IC5d will go low. A low level on IC5c will turn Q1 on via R19 and pull the probe to a high level via R21. If IC5d goes low instead, IC4f output will go high and turn Q2 on via R20. This will pull the pulser probe to the negative supply via R22. D6 and D7 provide a small amount of protection for this part of the circuit while D8 provides overall polarity protection for the probe supply.

HOW IT WORKS

Resistors (all ¼W, 5%)

R1,2,12,13, 19,20	10k
R3,10,11	1M0
R4-9,17,18	1k0
R14-16	100k
R21,22	22R

Capacitors (all miniature ceramic except where stated)

C1	100p
C2	68p
C3	100n
C4	470n 35 V tantalum bead
C5	150p
C6	4n7
C7	100u 25 V axial electrolytic

Semiconductors

IC1	4049B
IC2	4070B
IC3	4093B
IC4	40106B
IC5	4011B
Q1	BC212L
Q2	BC182L
D1-8	1N4148
LED1	3 mm red LED
LED2	3 mm green LED
DISP1	0.4" seven-segment common cathode display (FND357 or similar)

Miscellaneous

SW1	3-position miniature slide switch
PB1	subminiature push-button

PCB (see Buylines); two off 2 mm sockets and plugs for probes; piece of brass rod or studding for probe; crocodile clips; 10-way Soldercon strip; low profile IC sockets (if used); small grommets, wire, screws etc; case (100 × 50 × 25 mm), Vero ref. 65-2514F.

BUYLINES

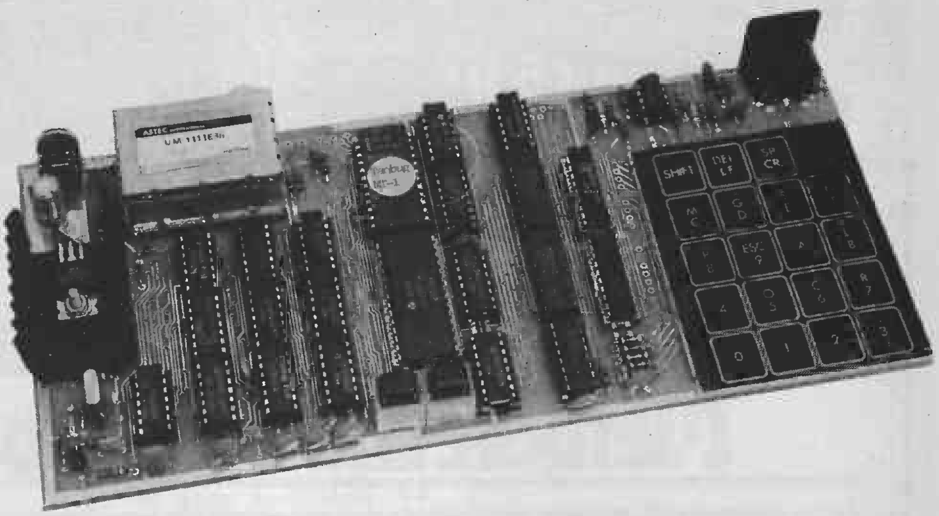
No unusual components on this one, but make sure capacitors, IC sockets and so on are the smallest types you can get if you want to fit the board into the case specified. The PCB can be obtained from the PCB Service — see page 87 for details.

ETI

MICROTUTOR PART 2

This month we look at the remaining commands provided by the TANBUG monitor on our machine code trainer for easy programming. Design and development by Tangerine Computer Systems.

Having shown how to enter and run machine code programs last month, we now consider the commands which aid in debugging and program writing, and save/load operations using the cassette port.



Single Instruction Mode S

Single instruction mode is a very powerful debugging aid. When set, TANBUG executes the user program one instruction at a time, re-entering the monitor between each instruction and printing out the status of all the microprocessor's internal registers as they were after the last instruction executed in the user program. The S command is used in conjunction with the proceed command P and the normal mode command N.

Normal Mode Command N

The N command is the complement of the S command and is used to cancel the S command so that the microprocessor executes the user program in the normal manner without returning to the monitor between each instruction. RESET automatically sets the normal mode of operation.

Proceed Command P

The P command is used to instruct TANBUG to execute the next instruction in the user program when in single instruction mode. Pseudo register contents are transferred into the microprocessor's internal registers and the next instruction in the user's program is executed with the monitor. P may also be used with an argument, thus: P <NUMBER> <CR> where NUMBER is less than or equal to FF. In this case the program executes the specified number of instructions before returning to the monitor.

Each time the monitor is re-entered after execution of an instruction or instructions, the status of the microprocessor internal registers as they were in the user program are printed across the screen in the following order:

- Address of next instruction to be executed.
- Processor status word.
- Stack pointer.
- Index register X.
- Index register Y.
- Accumulator.

Note that these are in the same order as the pseudo registers described last month.

Whenever the user program is entered, the cursor is removed from the display. Whenever the monitor is entered, the cursor returns to the display as a user prompt. While in the monitor between user instructions, any monitor command can be typed. A program must always be started by the G command, then P used if in single instruction mode. A P command used before a G command is issued is likely to cause a program 'crash' and should not be attempted.

As an example consider the simple program which repeatedly adds 1 to the accumulator.

Address	Data	Mnemonic	Comment
100	69	ADC 1	: add 1 to acc.
101	01		
102	4C	JMP 100	
103	00		
104	01		

Set the single instruction mode and start the program. (The user may wish to initially set the accumulator to 00 by using the M command).

```
S
G100
0102    20    FF    00    01
```

TANBUG then responds with the characters shown above.

0102	is the address of the next instruction to be executed.
20	is the processor status word value.
FF	is the low byte value of the stack pointer. The high byte is always set to 1, the stack is therefore pointing at location 1FF.
00	is the value of the index X register.
00	is the value of the index Y register.
01	is the value of the accumulator. It is a 1 as 1 has been added to the accumulator and it is assumed that the user cleared the accumulator before starting the program.

Since the cursor has re-appeared, TANBUG is ready for any monitor command. For example, registers or memory locations can be modified, or the program may be re-started from scratch by typing G100<CR> again. If the user wishes to continue

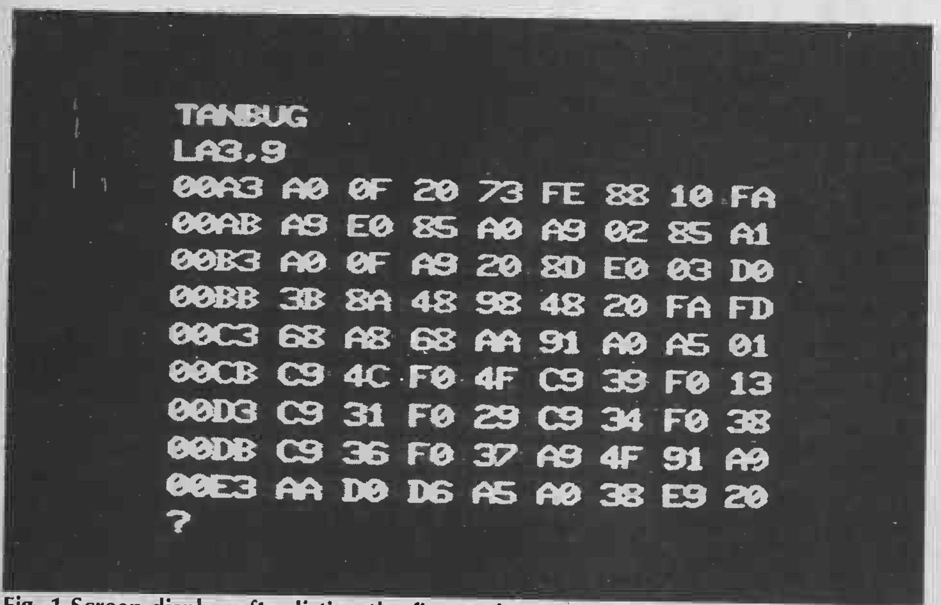


Fig. 1 Screen display after listing the first 72 bytes of a program which starts at address 00A3.

then type P<CR>. The resulting display is

S
G100
0102 20 FF 00 00 01
P
0100 20 FF 00 00 01
■

Since the instruction at location 102 was 'Jump to 100', the status printout shows that this has indeed occurred. Registers, since they were not modified by any program instruction, remain unchanged. To proceed further type P<CR> again:

S
G100
0102 20 FF 00 00 01
P
0100 20 FF 00 00 01
P
0102 20 FF 00 00 02
■

The add instruction has been executed again, so the accumulator has incremented by 1 to become 2. Now typing P4<CR> gives this display:

S
G100
0102 20 FF 00 00 01
P
0100 20 FF 00 00 01
P
0102 20 FF 00 00 02
P4
0102 20 FF 00 00 04
■

TANBUG allowed execution of four instructions before again returning to the monitor. The four instructions were two add instructions and two jump instructions thus giving the accumulator the value 4.

Typing N<CR> then P<CR> removes the single instruction mode and causes the program to proceed. It now does not return to the monitor but continues to race around this small program loop continually adding and jumping back. To re-enter the monitor, either press the reset button or the interrupt button. The interrupt will cause the processor to break into the program and enter the monitor with the display as described above.

It can be seen that the S and P commands are particularly useful when tracing a program which contains instructions that transfer program control eg jumps, branches and subroutines, since these commands allow the user to interrogate the order of execution of his/her program.

Breakpoint Command B

A breakpoint is a complementary debugging aid to single instruction mode. Instead of stepping singly through all instructions in a program, the breakpoint facility allows the user to specify the address at which he requires the monitor to be re-entered from his/her program. As an example, consider a long program in which a fault is suspected to exist near the end. It would be very tedious and time consuming to single step through the program to the problem area. A breakpoint can be set just previous to where the fault is suspected to exist and the program started with the G command. Normal execution occurs until the breakpoint is reached, then the monitor is re-entered with the same status print-out as for single instruction mode. Any monitor commands can then be used and the program continued.

The format of the breakpoint command is

B <ADDR>, <NUMBER> <CR>

where <ADDR> is the address of any instruction op code (but not argument), <NUMBER> is any number from 0-7 defining one of eight breakpoints. B<CR> removes all breakpoints. As an example consider the following program:

```
100 E8        LOOP: INX
101 C8              INY
102 69 01           ADC#1
104 4C 00 01        JMP LOOP
```

Firstly set index X, index Y and the accumulator to 00 using the R command. To set breakpoint 0 at the jump instruction and start the program type B104,0<CR>G100 <CR> . The display will then be

B104,0
G100
0104 20 FF 01 01 01
■

The jump instruction was reached and the breakpoint re-directed control back to TANBUG. If it was required, single instruction mode could be set for further debugging. However, assume that we wish to execute the loop again by typing P<CR>.

B104,0
G100
0102 20 FF 01 01 01
P
0104 20 FF 02 02 02
■

The proceed command P has gone once through the breakpoint and then re-entered the monitor. If P<NUMBER> <CR> was typed it would have proceeded through the breakpoint <NUMBER> times.

Up to eight breakpoints can be set at eight different locations. The B<CR> command removes all

Fig. 2 Setting the first breakpoint at the start address and running the program gives this register printout.

breakpoints. A single breakpoint can be removed by setting its address to 0. For example, imagine a breakpoint is set as follows; B102,2, and it is subsequently wished to remove it but leave any others unaltered; type B0,2<CR> to remove it.

Caution: The breakpoint system works by replacing the user's instruction with a special instruction (BRK) whose op code is 00. Replacement is carried out when G or P is typed. On return to the monitor the original op code is replaced. It is therefore possible to corrupt the user program under some circumstances. The following points should therefore be observed:

- Breakpoints must only be set at the op code part of a user instruction and nowhere else.
- If the user program utilises the BRK instruction as part of the user code, then this will cause a break in the normal manner, but note that the BRK instruction will be interpreted by the monitor as a 1-byte no operation.
- If breakpoints are set in the user program and a reset is issued while the microprocessor is executing the user program rather than the monitor, the breakpoints are lost and those locations at which breakpoints were set in the user program will be corrupted. These locations must be re-entered using the M command before restarting the user program.
- Setting more than one breakpoint at the same address causes the user program to be corrupted.
- To use breakpoints, the user must not have modified the interrupt link, ie the interrupt code within TANBUG must be executed.

The status of breakpoints may be inspected by using the M command to examine the breakpoint status table. This is located in RAM locations 20-2F and are as in Table 1.

TABLE 1

Address	Contents
20	PCL B0
21	PCH B0
22	PCL B1
23	PCH B1
24	PCL B2
25	PCH B2
26	PCL B3
27	PCH B3
28	PCL B4
29	PCH B4
2A	PCL B5
2B	PCH B5
2C	PCL B6
2D	PCH B6
2E	PCL B7
2F	PCH B7

For example, typing M20<CR> followed by <LF> gives
M20,00,
M0021,01,■
This indicates that breakpoint 0 is set to location 100 by taking the contents of location 20 as PCL and of location 21 as PCH. If the breakpoint is set at location 0 then this particular breakpoint is disabled.

Offset Command O

The offset command O is a program writing aid. It calculates branch offsets for the user for incorporation as arguments in branch instructions. Consider the example:

```
100  E8   LOOP: INX
101  C8         INY
102  69         ADC#1
103  01
                .
                .
120  D0         BNE LOOP
121             (branch argument)
```

To calculate the number to enter into location 121 is quite tedious without a facility such as the O command. It is used with the following format.

O<ADDR OF BRANCH OP CODE>
<ADDR OF DEST><CR>

and in this case it would be necessary to type 0120,100<CR>. The display would be
0120,100 = F0
■
F0 is the number that should be entered into location 121 such that if the BNE instruction is true the program counter will jump to the label LOOP.
Note that the maximum branch range is 7F forwards and backwards. If the range is exceeded a ? is displayed.

Copy Command C

The copy command allows copying of the contents of one block of memory to another. Its format is

C<START ADDR SOURCE>
<END ADDR SOURCE>
<START ADDR DEST><CR>

Suppose it is required to copy the block of data in locations C00–D00 into a block starting at location 200. This may be achieved by typing CC00,D00,200<CR>. The display will be
CC00,D00,200
■
As 200 is the starting address of the display memory the user will notice that the top half of the screen has been overwritten with all sorts of weird and wonderful characters. What this example has done is to take the first 256 bytes of TANBUG and copy them into the top half of the display. The display then scrolled leaving the top seven rows filled with these characters.

The Interrupt Button

Depression of the interrupt button will cause a re-entry into the monitor from the user program. For example, if the trivial program

```
100  69    LOOP: ADC#1
101  01
102  4C          JMP LOOP
103  00
104  01
```

has been started by typing the G command the program continues to loop around continuously with no exit path to the monitor, except by issuing a reset. Instead of a reset the user can press the interrupt button, TANBUG responding thus
0100 20 FF 01 01 01

■
Using the interrupt button has caused a breakpoint to be executed and the monitor invoked. The register print-out above is only typical, the value of each being that when interrupt was depressed. Any monitor command may now be typed, for example P causes the user program to proceed once again.

The interrupt facility is most useful in debugging where the user program gets into an unforseen loop where breakpoints have not been set. It enables the user to rejoin the monitor without using reset and losing the breakpoints that have been set. Notes:

- Interrupts must be enabled for the interrupt facility to operate. TANBUG enables interrupts when entering a user program, so don't disable interrupts if the interrupt facility is required.
- The user must not have modified the interrupt jump link. TANBUG's interrupt code must be executed.

ETI

PLAYMATE PART 2

For aspiring artistes we conclude our portable guitar amp with added ingredients fuzz and wah. Design and development by Phil Walker.

Before we continue with our analysis of the circuit theory we have to apologise for the manpower shortage last month which meant that the Playmate circuit diagram missed out on our usual thorough checking. Naturally the perversity of the universe ensured that this was the project that had several errors, so we're reprinting the circuit diagram with mistakes and duplicate component numbers corrected and all pots marked CW or CCW to tie up with the front panel wiring diagram. Meanwhile, back to the plot . . .

The State-variable Filter

Things start getting a bit heavy now! The following equations are the transfer equations for a state-variable filter such as that in Fig. 2:—

$$V_x = [(R1/R3) \times V_{in} + (R2/R3) \times V_y] + R5/(R4 + R5) \times [1 + R3(R1 + R3)/(R1R2)] \times V_o$$

If R1 = R2 = R3 = R5, then:

$$V_x = -(V_{in} + V_y) + R1(1+2)/(R4 + R1) \times V_o$$

$$V_x = \frac{3R1V_o}{R4 + R1} - (V_{in} + V_y) \qquad (1)$$

$$V_o = -1/sCR \times V_x,$$

$$V_x = -sCR\,V_o \qquad (2)$$

$$V_y = -1/sCR \times V_o \qquad (3)$$

Substitute (2) and (3) in (1):—

$$-sCR\,V_o = \frac{3R1\,V_o}{(R1 + R4)} - (V_{in} + (-1)/sCR\,V_o)$$

$$\frac{V_{in}}{V_o} = (sCR + 3R1/(R1 + R4) + 1/sCR)$$

$$= j\omega CR + \frac{3R1}{(R1 + R4)} + 1/j\omega CR$$

($s = j\omega$, $\omega = 2\pi f$, f = frequency)

Compare this with the equation for an LCR tuned circuit:—

$$V_{in} = (j\omega L + 1/(j\omega C) + R)\,I_{out} \qquad \text{(Fig. 3)}$$

$$\frac{V_{in}}{I_{out}} = j\omega L + R + 1/(j\omega C) \qquad \text{(LCR circuit)}$$

$$\frac{V_{in}}{V_{out}} = j\omega CR + \frac{3R1}{(R1 + R4)} + 1/(j\omega CR) \qquad \text{(State variable filter)}$$

From this it is apparent that these responses are similar except that the LCR circuit gives a current output rather than a voltage.

For this type of LCR circuit, the frequency of minimum attenuation or maximum gain (the resonant frequency) is given by:—

$$f = \frac{1}{2\pi\sqrt{LC}}$$

For our circuit this is:—

$$\frac{1}{2\pi\sqrt{CR.CR}} = \frac{1}{2\pi CR}$$

The 'Q' factor influencing the bandwidth is given by:—

$$Q = \frac{2\pi f L}{R} = \frac{2\pi \times (1/2\pi CR) \times CR}{3R1/(R1 + R4)}$$

$$= \frac{R1 + R4}{3R1}$$

Figure 4 shows the configuration necessary to use the LM13600 as a filter of this type. Last month

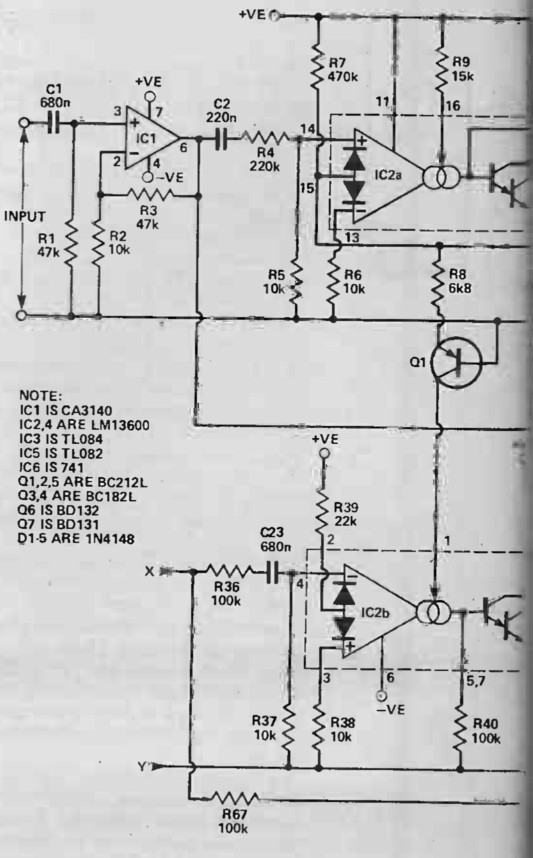


Fig. 1 The correct version of the circuit diagram — sorry . . .

we found that the transfer function of the LM13600 with no diode current is given by:—

$$I_{out} = \frac{I_{abc}}{26 \times 10^{-3}} \times V_{in}$$

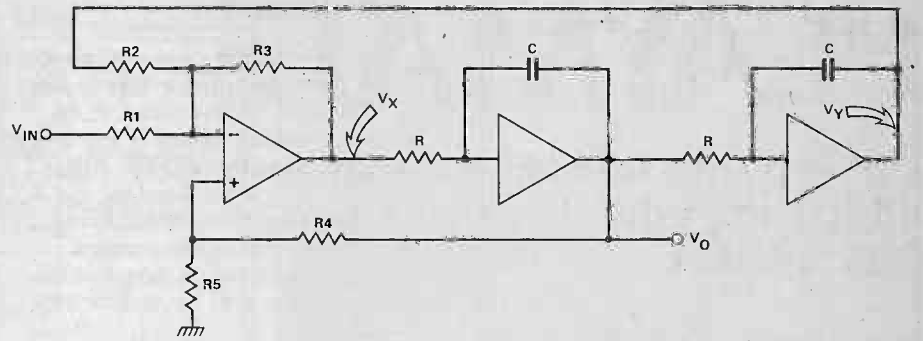
As we are using a capacitative load this output current will generate a voltage of:—

$$V_{out} = I_{out} \times \frac{1}{j\omega C}$$

$$= V \times \frac{I_{abc}}{26 \times 10^{-3}} \times \frac{1}{j\omega C}$$

$$= V_{in}\frac{R1}{R1 + R} \times \frac{I_{abc}}{26 \times 10^{-3}} \times \frac{1}{j\omega C}$$

Since we are not dealing with a normal type of op-amp, analysis of the circuit is not as easy as the normal filter but the result is an equation of much the same form. We do not show all the derivation here as it would occupy most of a page. However, the end result is that the centre frequncy of the filter pass band is proportional to the amplifier bias current. Therefore we have an easy way to control the wah wah effect.



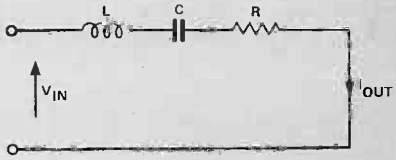
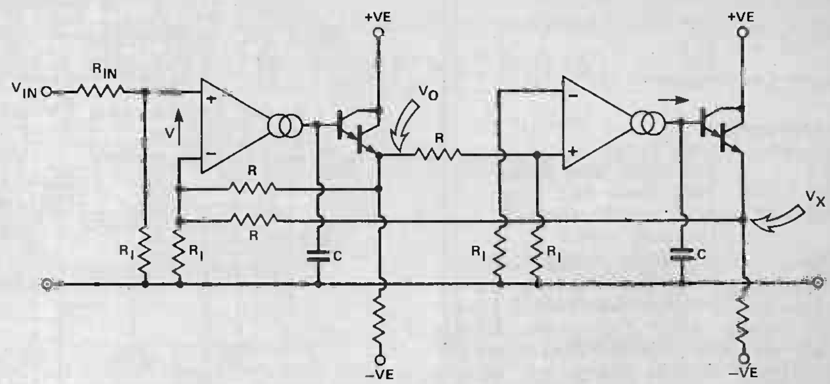
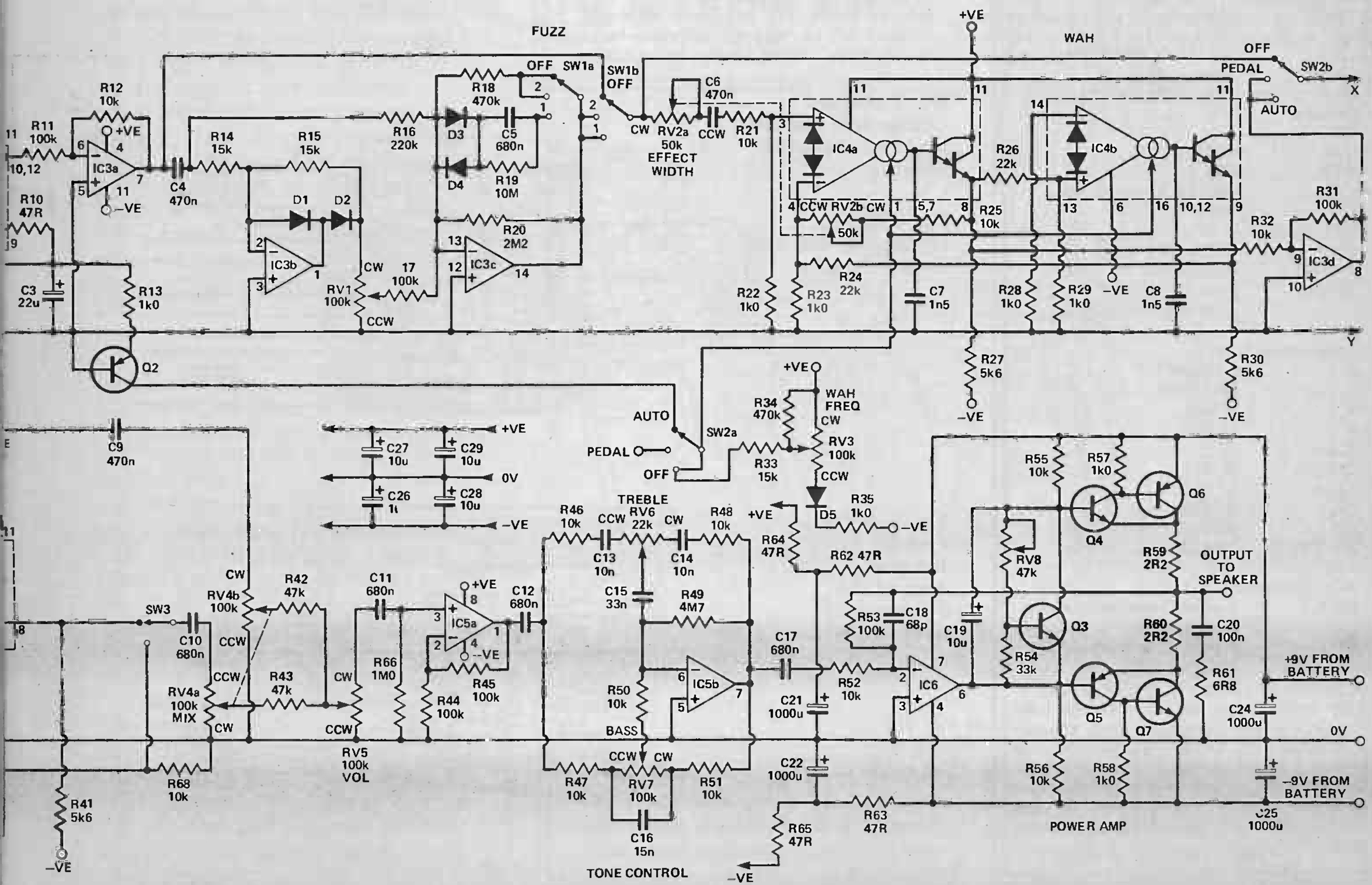


Fig. 2 (Above) Generalised state-variable filter.

Fig. 3 (Right) Generalised LCR tuned circuit.

Fig. 4 (Below) Using the LM13600 as a state-variable filter.





PARTS LIST

Resistors (all ¼W 5%)

R1,3,42,43	47k
R2,5,6,12, 21,25,32,37, 38,46-48, 50-52,55,56, 68	10k
R4,16	220k
R7,18,34	470k
R8	6k8
R9,14,15,33	15k
R10,62-65	47R
R11,17,31, 36,40,44,45, 53,67	100k
R13,22,23, 28,29,35,57, 58	1k0
R19	10M
R20	2M2
R24,26,39	22k
R27,30,41	5k6
R49	4M7
R54	33k
R59,60	2R2
R61	6R8
R66	1M0

Potentiometers

RV1,3,7	100k linear
RV2	50k linear dual gang
RV4	100k linear dual gang
RV5	100k logarithmic with two-pole switch
RV6	22k linear
RV8	47k miniature horizontal preset

Capacitors (all polycarbonate except where stated)

C1,5,10-12, 17,23	680n
C2	220n
C3	22u 16 V tantalum
C4,6,9	470n
C7,8	1n5
C13,14	10n
C15	33n
C16	15n
C18	68p
C19	10u 35 V tantalum
C20	100n
C21,22,24, 25	1000u 10V axial electrolytic
26-29	47u 10 V PCB electrolytic

Semiconductors

IC1	CA3140
IC2,4	LM13600
IC3	TL084
IC5	TL082
IC6	741
Q1,2,5	BC212L
Q3,4	BC182L
Q6	BD132
Q7	BD131
D1-5	1N4148

Miscellaneous

SW1,2	2-pole, 3-way miniature slide switch
SW3	1-pole, 2-way miniature slide switch

PCBs (see Buylines); case (220 × 105 × 230 mm), Vero ref. 75-2443A; wire (single, single screened and double screened); 4" or 5" loudspeaker (8 ohms), 5 W); standard ¼" jack socket for input; stereo jack socket for foot switch (if required); 75 mm of 12 × 12 mm aluminium angle; two PP9 batteries and clips.

Construction

Except for the controls, almost all the components for this project are mounted on the three PCBs. The preamplifier board is the largest and most densely packed. It is advisable to use sockets for the ICs and don't forget the links. The capacitors used in the project should be as small as possible otherwise you will have difficulty fitting them. A fine tipped soldering iron will be useful when assembling the boards and care should be taken to avoid creating short circuits between tracks with accidental solder splashes.

Ensure that all the diodes, transistors and other polarised components (especially IC3) are fitted the correct way round. On the power amplifier board the output transistors are mounted on top of a

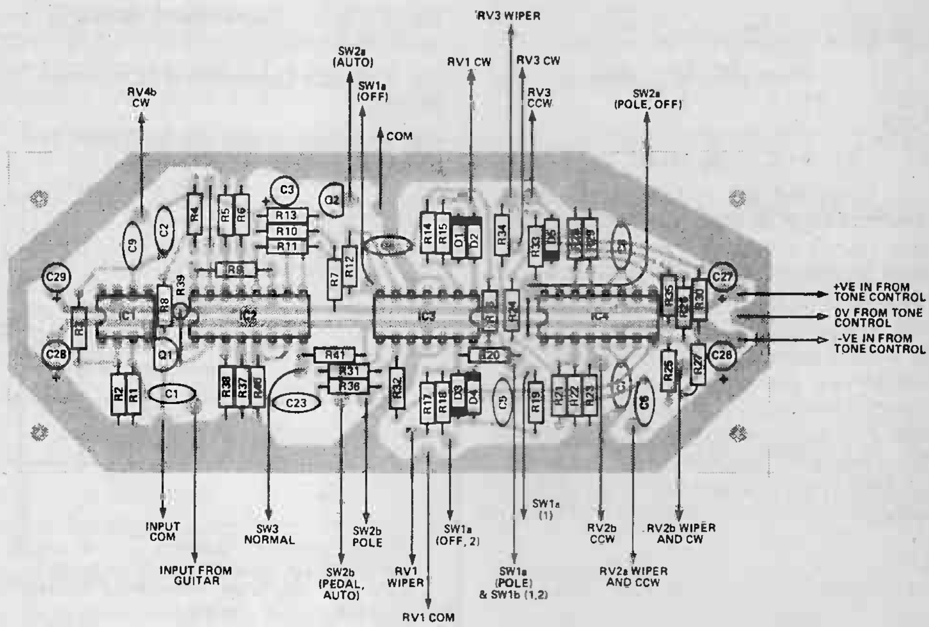


Fig. 5 (Above) Component overlay for the preamplifier board.

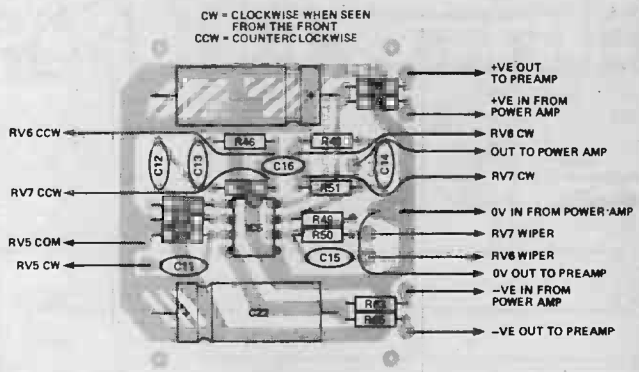


Fig. 6 (Right) Overlay for the tone control board.

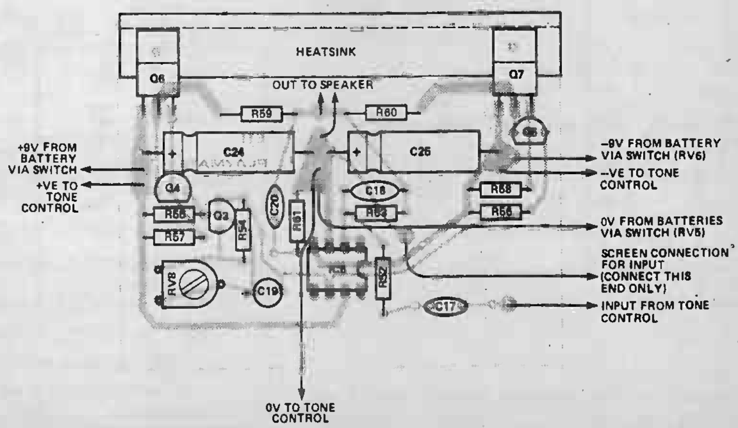
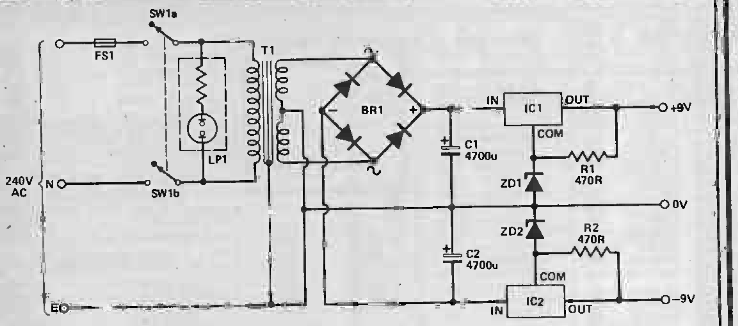


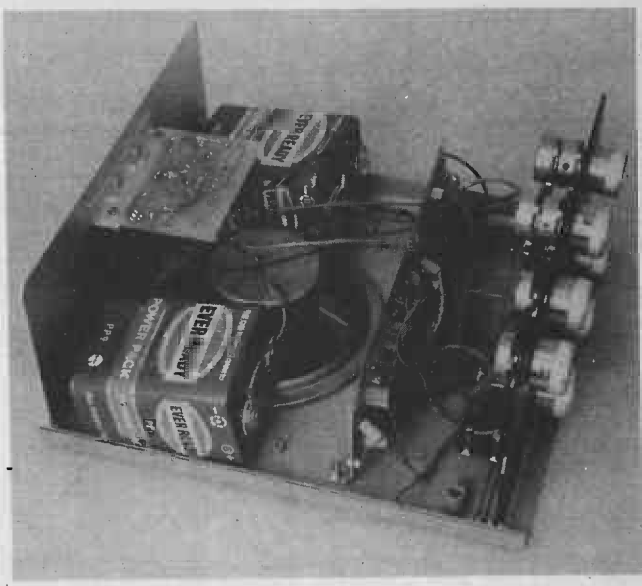
Fig. 7 (Right) Overlay for the power amp of the Playmate.

SUGGESTED MAINS UNIT



Parts List

R1,2	470R ¼W 5%	T1	9-0-9 12 VA transformer
C1,2	4700u 25 V electrolytic	SW1	DPDT miniature rocker or slide switch
IC1	7805	FS1	1 A 20 mm fuse and holder
IC2	7905	LP1	240 V panel neon with integral resistor
ZD1,2	3V9 400 mW zener		
BR1	W01 (1 amp bridge rectifier)		



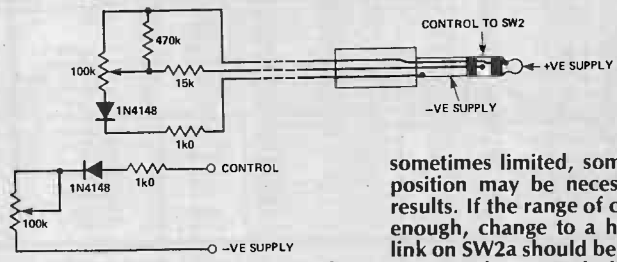
short length of ½" × ½" aluminium angle which acts as a heatsink. The transistors and the angle are held in position by the transistor mounting screws.

Mount the control switches and potentiometers on the front panel (see photo for our layout) and make the necessary interconnections and fit the three components needed around the balance control. The wiring from the front panel to the preamplifier/effects and the tone control boards should be carried out using thin flexible wire for control signals and miniature screened cable for any sound signals. These should be short but allow enough slack to be able to fit them in position easily. (It is probably easier to connect all the wires to the circuit boards first.) The power amplifier board was fitted to the metal base plate of our case using the angle on which the transistors were mounted. It was positioned so that it fitted neatly between the two batteries in the bottom of the case. Together with the loudspeaker the amplifier board holds the batteries without any further help.

The small loudspeaker for this project was mounted on the plastic case of the box through which a number of holes were drilled to let the sound out. If required a small mains power unit capable of 9-0-9 V at 100 mA or more may be used to power the unit.

A foot pedal control for controlling the wah wah effect could be plugged into a jack socket. This would need to be one of the three-connection or stereo type so that positive and negative supplies as well as the control signal could be connected.

SUGGESTED FOOT PEDAL



The first diagram shows the circuit used with a three-connection (stereo) jack. This will duplicate the effect of the panel-mounted control. As the angular rotation of foot pedal controls is sometimes limited, some adjustment of position may be necessary to get best results. If the range of control isn't wide enough, change to a higher value. The link on SW2a should be removed for this circuit. The second diagram shows a simple circuit for use with ordinary jack connectors which can be used if the link between SW2a pole and off position is retained.

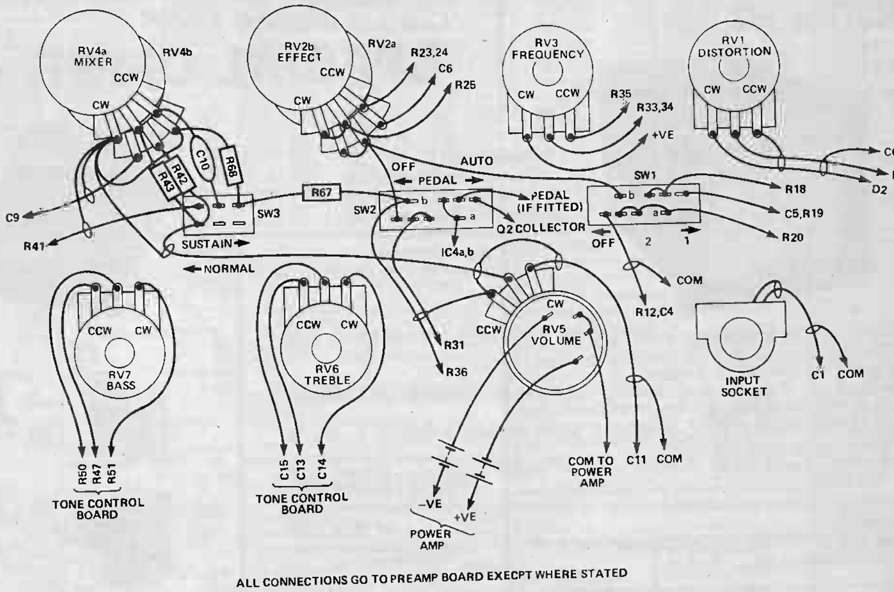


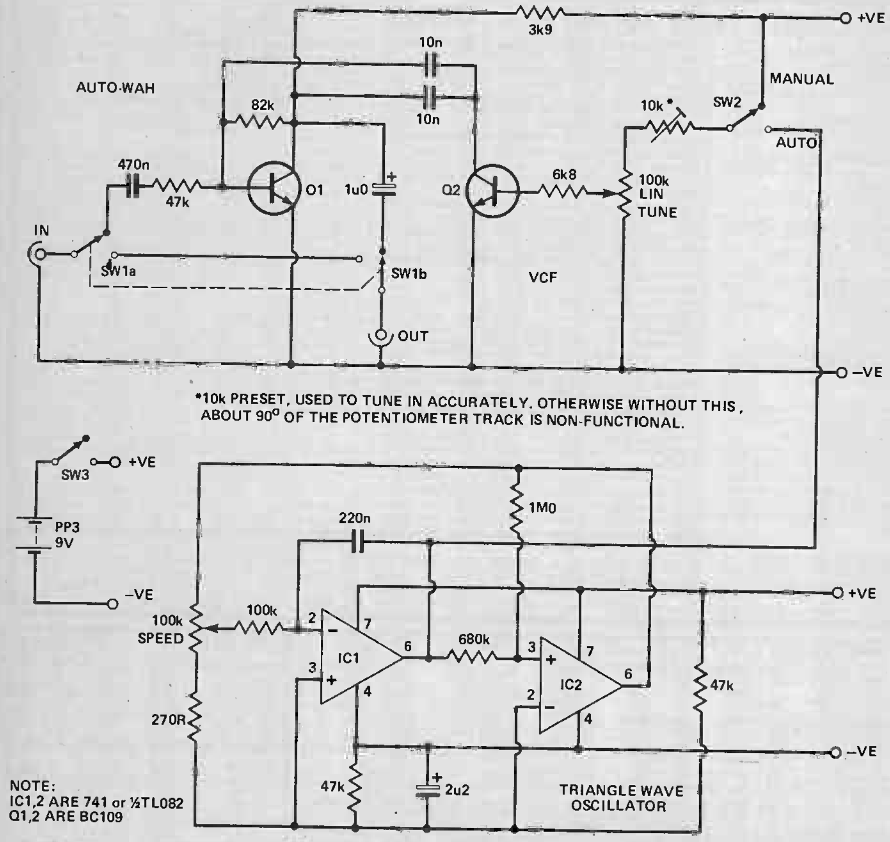
Fig. 8 How to wire the front panel. Compare with the front panel photo last month.

BUYLINES

There shouldn't be anything here to cause problems; most of the components are standard items and several mail order companies now stock the LM13600. Shop around in our advertisements for the best price. The set of three PCBs can be ordered from our PCB Service as advertised on page 87, should you not wish to etch your own.

ETI

TECH TIPS



Cheap Auto-Waa

Tony Allgood, Hornchurch

Although this unit is not capable of fulfilling the specifications of an expensive auto-waa, it has an advantage of being cheap and relatively simple. The basic idea is an expansion of ETI's Spot Design Waa-Waa unit from the November 1980 issue.

Q2 is used as a voltage controlled resistor connected to a simple filter network based around Q1. The controlling voltage is applied to the base of Q2 and can be varied manually or automatically through the use of a triangle wave generator consisting of a standard integrator and comparator oscillator. The frequency of the sweeps is determined by the 'speed' potentiometer and can be altered from 0.5 Hz to about 10 Hz.

A 9 V supply was used so the unit can be fully portable if a PP3 battery is used.

Tamper-proof Burglar Alarm

Iain S. Campbell, Paisley

Recently I have started to install a burglar alarm in my home and, while planning the installation, I realised that I would require some form of tamper-proof or 'self-activating' bell. In other words, I required the bell to ring if the wires leading to it were cut or short-circuited. I consequently designed the following circuit.

The circuit is a window comparator and works as follows. R4, R5 and R6 are voltage dividers with one-third of the supply voltage across each. R3 is chosen so that R1 > R3 > ½R1, noting that R1 = R2 (in practice, for R3 I used the next lowest standard value from R1, R2). Both non-inverting inputs of the comparators are thus kept lower than their respective inverting inputs.

If R2 is open-circuited, ie the leads to it are cut, the non-inverting input of IC1 is pulled high, thus taking the output high. If R2 is short-circuited the voltage across the non-inverting input of IC2 and ground is between one-half supply voltage and just over one-third of the supply (depending on the value of R3), thus taking the output of IC2 high. The two outputs may now be ORed using any general purpose diodes and the output used to drive a suitable load, in my case a TIP122 Darlington transistor and a 1½ amp bell.

I would recommend other users to put a monostable between the diodes and the load as well as employing a battery back-up, although these were not necessary in my case. A five minute monostable should be sufficient to keep the neighbours happy if the alarm is activated or tampered with. When installing this circuit, R2 should be situated inside the burglar alarm control unit across the existing relay contacts or an opto-coupler. When the alarm is activated, R2 is shorted. Tampering with the interconnecting cable can be detected and the bell enclosure may be protected with the appropriate normally-open or normally-closed microswitches.