## TOPDROJECTS



I

## NOTHING ELSE COMES CLOSE



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ELECTRONICS TODAY INTERNATIONAL

# DIGITAL DOORBELL 

John Jameson's front door greets visitors with a blood curdling scream, a fanfare and the chimes of Big Ben thanks to his ingenious sampler doorbell

This project provides you with the means to have a totally unique doorbell. Almost any sound effect lasting up to three seconds can be emitted at considerable power whenever a visitor calls. If you have several doors, you could build several units and have them announce Front Door, Back Door or you could sample the neighbours' bloodhound and frighten the wits out of your postman.

The possibilities for such a project are limited only by your imagination. Slightly modified, the unit could even find uses in electronic music, as a simple but versatile drum effect.

To make things easy for you, the author has arranged to provide a range of sound effect ROMs, and a tape to EPROM service (see Buylines) so you won't need a sampler and an EPROM programmer to build the project.


Fig. 1 Block diagram of the Digital Doorbell


## Construction

The project is constructed on a $4 \times 4$ in double sided PCB with only the loudspeaker and the power and bell push sockets mounted off the board. This design is simple enough to build up completely before testing, provided you have the proper PCB.

Start by making the through-hole connections or 'vias' first. Don't forget the ones under the EPROM. Next insert then solder in the six pins used to make the external connections and add the passive components. The ICs should go in last.

Only the EPROM need be mounted in a socket. IC3,4 and 5 will have to be soldered in as their legs make further through-hole connections. These are CMOS devices so take the usual anti-static precautions of earthing yourself and everything else in sight when doing this.

If you house the unit in the recommended boos, the PCB will be glued in, so it is highly advisable to test it thoroughly and get it working on the bench. A 15V DC power supply will do just as well as $A C$ for testing so you can use your bench supply. Initially, power the unit up without an EPROM inserted. Several simple tests can now be made.

First and foremost, check the 5 V supply is present. The output from the DAC, IC6, should be at around 2.5 V (it is seeing \&0FF on its inputs) and the output of the LM380, pin 9, should be at about $7-8 \mathrm{~V}$. Switch off, insert a programmed EPROM, set the DIP switches as detailed in Table 1, set RVI atmid range and switch on.

If all is well, the unit should emit the sample and go quiet. Triggering the unit by shorting the bell push terminals should cause it to emit the sample once and stop.

If you get nothing, use a crystal earpiece or a scope to trace the fault back down the analogue path. If nothing is coming out of the DAC, see if data is coming out of the EPROM. Failing that, check around the address counter and the clock. Highly distorted sound might simply be due to shorted address or data connections around the ROM.

Once you have a working doorbell and assuming you are using the recommended case, drill a grid of holes in the top of the case for the loudspeaker and bolt it to the top (or glue it using Araldite or something similar). Drill two more holes at one end of the box for the two jack sockets. The prototype has the 2.5 mm connector for power and the 3.5 mm one for the be! push, though this is not at all crucial.

If you plan to mount the unit on a wall, drill a couple of holes in the bottom of the case, and file them into keyhole shapes so you will be able to hang the unit on a couple of screws in the wall.

Put the PCB in the bottom of the box at the other end from the mounting holes (so as to avoid the screws shorting things on the underside of the board) and measure up connecting wires to go from the PCB pins to the sockets and the loudspeaker. Wire up these connections, check everything and then glue the PCB onto the bottom of the box using Araldite. Finally screw the lid onto the box and wire up cables to your bell transformer and bell push.

You will need a bell transformer with a 12 V tap. Also, bear in mind that sticking a jack plug into a socket causes a momentary short, so turn the power off to the transformer while setting everything up.

## Samples

For the benefit of those readers who want to program their own EPROMs, here is the procedure you will
need to follow for the doorbell of your dreams.
Firstly, follow the normal procedure for sampling a sound on your sampler. If you want it to play back

## HOW IT WORKS

The unit is in effect a self-contained, play-only sampler. The sampie data is stored on a standard EPROM and the rest of the design consists of a clock, a trigger circuit, an address counter, a DAC, a 4 -pole filter and a power amp. The arrangement is shown in the block diagram (Fig. 1).

The clock is (almost inevitably) a 555 timer - in this case the CMOS version. The component values have been chosen to make it oscillate at around 10 KHz giving up to 3.2 s sample length with a 32 K EPROM (27C256).

As with all sampling systems the band width is restricted to half the clock frequency and is therefore about 5 kHz . This may seem low to hi-fi purists but it is equivalent to that of AM radio and quite adequate, given that the sound will emanate from a cheap 4in loudspeaker.

Pressing the bell push (SW6) triggers the monostable iC5 which resets the address counter made from IC3,4. The monostable is configured to be edge triggered and non-retriggerable which means that however long the bell push is pressed, the pulse from the monostable will only last about 20 ms , after which its Q output will return low.

Once the reset pulse has ended, the address counter will start counting and data will be fed to DAC IC6 and the analogue stages causing the doorbell to burp, scream, bark or what you will.

If an 8K EPROM is in use, SW1 will be closed and SW2,3 open. This means that A13 will be connected to the clock enable of the address counter. It should be clear that when a count of 8192 is reached, A13 will go high disabling the counter and stopping the system until the counter is reset.

SW5 selects retriggerable or non-retriggerable operation. If it is open, then re-pressing the bell push will re-start the sample from the beginning. If SW 5 is closed, then the monostable will be held reset until the sample has finished, thus preventing the sample being retriggered while it is playing.

The chosen filter is a fixed cut-off 4 -pole Chebyshev low pass filter and is a good example of economic design. The output from the DAC ranges from 0 to 2.5 V , and the overall gain of the two filter stages is just over five. It follows that if the output of the DAC were fed directly into the filter, an output swing of over 12.5 V would be required. Normally you would achieve this using $\pm 12 \mathrm{~V}$ or 15 V rails but by choosing op-amps capabie of single rail, low voltage operation and by doing a few simple calculations, it is possible to power the filter from the regulated 5 V rail required for the EPROM.

R11 and R12 attenuate the signal from IC6 so it will not drive the second op-amp, IC8, into overload. C4 removes the DC content from the output of the DAC and R21 in combination with R12 sets a new DC bias level for both op-amps. Bearing in mind that the DAC has an output impedance of 10 k , application of simple resistors in series/paralle theory to the network shows the impedance at the node (without the filter connected) would be 8 k 1 . R13 then makes this up to near enough 39 k as required for the filter.

C9 couples the filter to the output stage, RV1 sets the volume and the LM380 provides power ampilfication sufficient for the purpose.

A standard 78 L 05 regulator circuit provides the supply for the logic and the filter while the LM380 is run off the unregulated supply, which should be about 15 V from a 12 V AC input.


Fig. 2 The full circuit diagram of the doorbell

PARTS LIST


Fig. 3 The component overlay for the doorbell
at its natural pitch, try to sample it at a rate of 10 kHz , or at least a multiple of that. You will then need to extract a binary dump of the sample. This is easiest if your sampler is home made and based on a home computer.

Those of you who have built my Amstrad Sampler (ETI, September 1987) and have the full software, will find that the SMP file is a direct binary dump of the sample and is in the same 8 -bit linear format used by the doorbell.

The Spectrum Sampler (ETI, November 1985 to July 1986) uses logarithmic ADC and DACs, so you will need to write a program to convert the data to linear form.

MIDI sampler owners with a home computer based MIDI interface should be able to extract samples from their machines using the MIDI sample dump protocol.

Having got a linear sample dump, the next job is to adjust it for the 10 kHz playback. If you sampled at 10 kHz , you can omit this step. If you sampled at 20 kHz , then you need a program to average succes-

| ROM | SW1 | SW2 | SW3 | SW4 |  |
| :--- | :---: | :---: | :---: | :---: | :---: |
| 27C64 | ON | OFF | OFF | OFF |  |
| 27C128 | OFF | ON | OFF | OFF |  |
| 27C256 | OFF | OFF | ON | ON |  |
| SW5 OFF | - retriggerable mode |  |  |  |  |
| SW5 | ON | - non-retriggerable mode |  |  |  |
|  |  |  |  |  |  |
| Table 1 The DIP switch sertings |  |  |  |  |  |


\section*{$\begin{array}{ll}\text { RESISTORS (all 1/W } \mathbf{W} & \text { 5\% unless specified) } \\ \text { R1,7,8,9,12,16,20 } & \text { 10k } \\ \text { R2 } & 8 \mathrm{k} 2 \\ \text { R3,15 } & 18 \mathrm{k} \\ \text { R4 } & 390 \mathrm{R} \\ \text { R5 } & 100 \mathrm{R} \\ \text { R6 } & 470 \text { R } \\ \text { R10 } & 220 \mathrm{k} \\ \text { R11 } & 47 \mathrm{k} \\ \text { R13 } & 30 \mathrm{k} 1 \% \\ \text { R14 } & 39 \mathrm{k} \\ \text { R17,18 } & 82 \mathrm{k} \\ \text { R19 } & 9 \mathrm{k} 11 \% \\ \text { R21 } & 180 \mathrm{k} \\ \text { RV1 } & 10 \mathrm{k} \text { horiz preset }\end{array}$ <br> | CAPACITORS |  |
| :---: | :---: |
|  | 3 3 3 ceramic disc |
| C2,3 | $10 \mu 25 \mathrm{~V}$ radial electroivtic |
| C4 | 100 n polyester |
| C5,6,7,8 | InO ceramic disc |
| C9 | 24263 V radial electrolytic |
| C10, 13 | 44763 V radial electrolytic |
| C11 | $470 \mu 16 \mathrm{~V}$ racial electrolytic |
| C12 | $1000 \mu 16 \mathrm{~V}$ radial electrolytic |
| C14,15,16,17,18 | 100 n dipped multiayer | <br> | SEMICONDUCTORS |  |
| :---: | :---: |
| 1 Cl | 2764 or 27128 (see text) |
| IC2 | 7555 |
| IC3,4 | 4520 |
| IC5 | 4098 |
| 1 C 6 | ZN426 |
| $1 \mathrm{C} 7,8$ | CA3140 |
| $1 \mathrm{C9}$ | LM380 |
| IC10 | 78.05 |
| 201 | 5V1 zener |
| BR1 | W005 |
| MISCELLANEOUS |  |
| LSI | 4in 4R 2 W speaker |
| PL1 | 2.5 mm jack plug |
| PL2 | 3.5 mm jack plug |
| SKT1 | 2.5 mm jack socket |
| SKT2 | 3.5 mm jack socket |
| SW1-5 | 6.way DIP swich |
| SW6 | single pole push switch | <br> PCB. Case $190 \times 110 \times 74 \mathrm{~mm}$. IC sockers. Wire. Glue. Nuts and botts.}

## BUYLINES

All the electronic components should be readily available. The recommended case is sold by Rapid Electronics (Tei. (0206) 272730 ) as order code 30.0420 . Alternatively, a kit including the PCB , all components, one standard sound effects ROM (8K or 16 K ) and the case, but not including a bell transformer is available from Labcenter Electronics, 14 Marriner's Drive, Bradford BD9 4NT. The cost is $£ 29,95$ plus 50 p postage inc VAT. The PCB is available for $\mathrm{£7.20}$ plus 30 p postage.

Write to Labcenter for details of the Tape-to-Sample service and for a list of standard sound effect EPROMS.
sive pairs of bytes, and for 30 kHz , your program should average groups of three bytes.

Once this is achieved, you now need to ensure the data is represented such that $\& 000$ represents the most negative value, $\& 080$ represents quiescent conditions, and $\& 0 \mathrm{FF}$ represents the most positive

value. Again, Amstrad Sampler owners need do nothing here.

Choose an EPROM type large enough to hold the data $(8 \mathrm{~K}, 16 \mathrm{~K}$ or 32 K ) and then pad out the sample dump with $\& 080$ so as to fill the ROM. The
first byte of the ROM should be $\& 080$ as well, so as to prevent a sharp click occurring at the end of the sound.

Finally, transfer the data to your EPROM programmer to blow the device.


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Jeff Macauley gets out<br>the glue and reports on the ports of his bass reflex loudspeaker design

# REFLEX ACTION MICRO MONITOR SPEAKERS 

Last month I described in some detail the basic principles behind the design of reflex speakers. This month I've taken the opportunity to get my hands dirty and actually build one. There's nothing to beat the practical application of theory and so I present the following design which I hope will prove both instructive and a spur to like-minded readers to have a go!

Despite the fact that the theory behind speaker design is now fairly well explored, building a speaker will always throw up interesting challenges. For example this project started with running the characteristics of some woofers through the Optibox computer program I detailed last month.

Although I live in a flat where


Fig. 1 The circuit diagram of the crossover space for audio equipment is somewhat limited I like my fair share of bass so l'm always on the lookout for small systems with good low end extension. I came across the woofer used in this project in the Tandy catalogue. Although it's only four inches in diameter and is primarily intended for in-car use its characteristics make it ideal for a micro monitor application. The unit, Tandy catalogue number 40-1022A, has the following specifications.

| Resonant frequency | fo | $=55 \mathrm{~Hz}$ |
| :--- | :---: | :--- |
| Total Q | Qts | $=0.35$ |
| Equivalent volume | Vas | $=0.23 \mathrm{cu} \mathrm{ft}$ |

If we plug these values into the equations given last month we come up with the following information. Optimum volume $V_{0}=0.169$ cubic feet, cutoff frequency 62 Hz !! 've seen enclosures of over a cubic foot which cannot reach that low.

This obviously merited further investigation so I went down to my local Tandy store and bought a pair.
 I then embarked on the process of testing them to make sure the parameters were as stated. I was not disappointed. Both units measured up very closely to the published spec and in addition the enclosed data sheet promised a very smooth midrange response and a top end rolloff of about 5 kHz .

There were only two flies in the ointment. Firstly the quoted sensitivity of $84 \mathrm{~dB} / \mathrm{W}$ is on the low side as is the published power rating of 10 W rms. However 1 have been driving mine at neighbour-disturbing levels from my $35 \mathrm{~W} / \mathrm{ch}$ amp for over a month with no signs of distress to the drivers. My initial reservations as to the spl obtainable and the woofer's durability have proved unfounded.

I finally chose, after some deliberation, a cabinet volume of 0.166 cuft . The reason for the slight change in volume was rounding the dimensions to whole numbers: 3 dB cut off at 64 Hz , ripple 0.1 dB . The final external dimensions of the enclosure are 9.5 in by 5.75 in by 9 in ( $\mathrm{h} \times \mathrm{w} \times \mathrm{d}$ ). A cabinet this small has the advantage that the panels.are more rigid and hence
less liable to resonate and colour the sound. I eschewed the normal .75 in thick high density chip board for the more cosmetically pleasing white melamine covered variety.

Any constructor that cares to can choose the high density stuff in preference as long as the internal dimensions are retained. An 8 ft length of 9 in wide board will amply provide two cabinets for less than $£ 7$. Unless one actually enjoys woodwork or has masochistic tendencies it is worth getting your local timber yard to cut the board for you. It is absolutely essential to have the cuts made accurately (this will prevent the possibility of the air turning blue as you fit the case together).

To complement the woofer a high quality tweeter is required. As with woofers there is a large variety on the market. I finally chose a Philips polycarbonate dome type which is available from Electromail, catalogue number 249-435. The main reason for this choice is that I have had wide experience of this unit and know its strengths. It has a good flat response from $3-20 \mathrm{kHz}$, amplitude variations being kept to within a couple of dB . It is also very sensitive and capable of wide dispersion.

## Considering Crossovers

Having decided on our drivers and the cabinet size, the next task is to decide on a crossover. To understand how well or dismally a crossover works it is first necessary to consider the tasks it is required to perform.

The obvious function is to divide the incoming signal into the high and low components. Unfortunately this is complicated by the fact that the speaker units don't present pure resistive loads. Another complication is that back emf is generated by the units as they operate and this behaviour is both amplitude and frequency dependent. Add to this a compensation for response deviations in the driver and you end up with a very complex network that evades straightforward analysis.

The other alternative is to accept that some variations are inevitable and minimise these by choosing drivers that are inherently flat in their operating range. A simple network can then be designed which gives good results. Luckily both the units chosen here are fairly flat (to within a couple of dB of their operating range) and therefore the simple approach can pay dividends.

Getting back to the immediate problem of designing a crossover for the speaker in hand I made the following observations. Firstly the frequency response of the woofer is flat within $\pm 2 \mathrm{~dB}$ and has a well damped hf rolloff, 3 dB down at 5.5 kHz . Secondly the tweeter is far more efficient than the woofer and has a well controlled 'bass' resonance at 2 kHz . It is also sensibly flat between 5 and 20 kHz .

It followed from these observations that the easiest solution was to leave the woofer response as it is and precede the tweeter with a first order network. It means an asymmetric crossover and implies a careful choice of turnover frequency to avoid response
anomalities. This was the design path chosen and the resulting circuit is shown in Fig. 1. The tweeter is several dB 'hotter' than the woofer which means its signal must be attenuated. This is the function of R1. C 1 is chosen to produce a -3 dB point at 5.5 kHz in conjunction with the series resistance of R1 and the tweeter's impedance. Note that the tweeter is phase inverted with respect to the woofer due to the drive signal phase differences between them.

Despite its simplicity this network does the job it was intended to do and has a couple of advantages compared to more conventional designs. It's extremely easy to drive, the impedance doesn't fall below 7R. It uses no coils and because of the high value of R 1 a normal value polyester cap can be used for coupling

Even so, I always find the advantages of an active crossover boost performance further, and a suitable design for the Micro Monitors will follow next month.

## Construction

The required wood cuts are shown in Fig. 2. While dimensions here are in millimetres, the Parts List quotes in inches since many timber yards have yet to find their way in to the '80s!

Mark out the panels for the drivers and port. The tuning duct is a length of plastic piping with an intemal diameter of 1.25 in ( 31 mm ) which has an overall external diameter of $1.5 \mathrm{in}(37.5 \mathrm{~mm})$. The internal area is thus $3.14 \mathrm{r}^{2}=1.22 \mathrm{in}^{2}$. Plugging this value into the port equation gives a length of $4.1 \mathrm{in}, 104 \mathrm{~mm}$. The piping used is available at plumbers supply merchants and a metre length cost me all of 84p.

Note that the port is mounted on the rear baffle. This is mainly because a small frontal area is required to give good stereo imaging. It has no adverse effects on the sound because at the low frequencies at which the port operates the speaker's radiation pattern is omnidirectional anyway. Cutting the pipe accurately is more difficult. A mitre box is almost essential. If you don't have one the easiest way is to wrap a piece of card tightly around the tube to obtain a straight edge which can be slid up and down. Mark off the length you need by drawing around the card. Using a hacksaw carefully make a cut around the circumference of the tube before cutting it through.

Having got to this stage, final assembly of the enclosures can commence. Everyone is entitled to their own foibles when it comes to adhesives. Mine is to use a contact adhesive Thixofix which is widely


Fig. 2 Cutting diagram for a single Micro Monitor and enclosure construction


Fig. 3 Recess dish drilling detail
available. One smears the surfaces to be joined with glue, then leaves them for ten minutes or so for the glue to cure. The panels can then be slid together and positioned precisely. Firm pressure then completes the bond.

Whatever adhesive is used it is as well to use it liberally as this will stop air leaks.

Once the case has been glued it needs to be screwed together. I used lin no8 self tappers for this job. Drill pilot holes $1 / \mathrm{sin}(3 \mathrm{~mm})$ diameter to take the screws and countersink these. I have detailed the screw positions on Fig. 2. These should be adhered to.

Having built the case, attention can be turned to sealing it to prevent air leaks. This is most easily done with 'polyfiller'. Mix it up into a stiff paste and work it along the panel seams with your finger. Wipe the excess away with a damp cloth. The hole for the port is best cut with a hole cutter. A nest of these, suitable for cutting various diameter holes between one and two inches, can be obtained from your local tool shop for a few pounds. These are a useful addition to anyone's tool box. To fit the port into position I used araldite rapid. The other apertures are larger and are


Fig. 4 Wiring diagram

## PARTS LIST

| ONE CHANNEL ONLY |  |
| :---: | :---: |
| R1 | 47R 1W wirewound |
| Cl | 470n 100V polyester wkg |
| Woofer | Tandy 40-1022A |
| Tweeter | Electromail 249-435 |
| Recess Dish | Maplin FS34M |
| Wood | 15 mm thick Melamine covered chipboard |
|  | 2-of $91 / 2 \times 4 / 2 \mathrm{in}$ |
|  | 2 -off $91 / 2 \times$ in |
|  | 2 -off $41 / 2 \times 7 \%$ in |
| 104 mm length sockets. | $11 /$ in id plastic tubing. 3 -off 4 mm panel mounting |

best tackled with a jigsaw or jigsaw attachment. If this is not available then a coping saw can be used.

Reflex enclosures cannot be stuffed with sound absorbants like other forms of enclosure. To do so will wreck the carefully calculated relationship between driver and enclosure resonance. For this reason the interior is left unstuffed.

One nuisance with the woofer is that the môunting holes are slightly recessed from the frame. This means that in order to seal the driver against air leaks, further action must be taken. The best idea is probably to stick some self adhesive draught excluder tape around the hole before mounting the driver. A less strenuous alternative used on the prototypes was to seal around both units with polyfiller after mounting them. The tweeter hole should be similarly treated.

Before fitting the drivers into position it is as well to solder the leads to the terminals. Because of the simplicity of the crossover unit this is simply wired across two of the terminals as shown in Fig. 4 drawing. Nothing elaborate is required here -I used 12in lengths of 5 A twin speaker cable from Halfords. This has the advantage that one of the conductors is identified with a white strip along the insulation thus allowing easy phasing of the units.

For connections to the drivers I used three 4 mm banana sockets/channel. If these are connected as shown in the schematic then upgrading the speaker to active operation becomes easy. The drilling for the rear access dish is shown in Fig. 3.

Final assembly consists of mounting and wiring the crossover and terminals then screwing the rear panel into position. Remember to seal around the back plate with more polyfiller to ensure airtightness.

At this stage all that remains is to test the unit out and finish the cabinet as desired. If you've never used melamine faced board before you will find that edging strip is available to cover the bare chipboard. This is first trimmed to the correct width and applied to the edge by ironing over it. The backing adhesive bonds with the heat.

In conclusion this pair of speakers can be built for a total expenditure of less than $£ 40$ the pair. They have a better bass response for their size than any others I have heard. What's more important is that this response is nicely damped with no audible ripple in the response. They also offer a good transient response and a 3-dimensional stereo image. They can also be simply and fairly cheaply upgraded to active use as detailed in a forthcoming article.


# John Linsley Hood's AUDIO DESIGN 

 80W MOSFET AUDIO AMPLIFIER

There is an old adage which says that li's nice to be clever but ti's a lot more clever to be nice. Translated into audio amplifier terms this could read that it is nice if equipment has an excellent technical specification, but it is even better if it sounds nice.

Of course it is gratifying if gear scores top marks on both counts but the former, sadly, does not guarantee the latter. It is salutary, from time to time, to visit one of the more run-of-the mill High Street hhifi' shops to be reminded that some quite horrible noises can emerge from combinations of reasonably prestigious seeming kit.

This lack of agreement between conventional measurements and perceived sound quality poses a problem for any designer in the audio field. Account must be taken, however reluctantly, of the claims of those who base their judgement of equipment performance on listening trials, and checks made to see whether these claims have foundation in fact.

I admit to reservations about many of the effects attributed to the characteristics of minor amplifier components in amplifier circuitry, particularly where these are clearly peripheral to its function. However in places such as the negative feedback loop (which defines the whole performance of the amplifier) and in the supply line decoupling (which establishes the relationship of the system to the 0 V line) there are good technical reasons why component type may affect sound quality and there are small but measurable (and reproducible) effects which can be demonstrated. Having looked at a wide range of these marginal aspects of audio quality, I incorporated those I thought to be sensible into the design of my 80W amplifier featured in ETI from' June to September 1984.

In general, I am very well pleased with the performance of this amplifier which is, I think, the best design I have done so far. In the view of some experienced 'audiophiles' whose opinlons I respect,
it is significantly better than other amplifiers of their acquaintance. It has also attracted some very flattering comment from constructors in the correspondence columns of the hi-fit press.

I share the general regret that a kit of parts for this design did not become available at the time of the original publication. Now that a kit specialist, Hart Electronic Kits, has produced the necessary components, PCBs and metalwork for the power amplifier, I have taken this opportunity to do a little tidying up in areas generally related to convenience rather than performance.

The pre-amplifier has also been given a few amendments, notably the addition of a moving coil head amplifier stage and new tone controls. Hart is


Fig. 1 Circult diegram of the input swhehing, volume and belance controls

already working on the kit for the pre-amplifier and hopes to have it available later this year. It will be featured in ETI at that time.

In both the power and pre-amplifiers my aim with this revised design remains the same - to provide a system which will be equal to or better than the best of the contemporary market offerings, without making the whole system too expensive for the average potential constructor.

## Input Switching

It was an oversight on the original design not to provide an input selector switch to accompany the
'gain' and 'balance' controls on the input of the power amplifier which would have increased the versatility of the power amp. I should have thought of this earlier since I had specifically intended that the power amplifier should be capable of use on its own, by those who were only going to use LS outputs, and whose sources (for example tape, CD or radio) would not need frequency response adjustment.

The small circuit modifications involved are shown in Fig. 1 and the component overlay in Fig. 2.

## Output Power Indication

The power output display which I had provided for

## COMPONENT CHARACTERISTICS

I have read much, and have been given much advice by some of my experimentally minded friends, on the way in which the nature of a component (as distinct from its actual electrical valua) can influence the quality of the sound of an amplifier. I have made a lot of tests to see il lagree with these findings. Mainly the differences are too small for me to be able to say definitely whether there is a change and if so whether it is better or worse.

I do not say that these differences do not exist or that when a lot of very small changes are added together the cumulative effect will not be greater. What I think is that, in most cases, the changes are very small in comparison with those atterations in sound quality which arise from different circuit approaches. I feel that I am better employed in trying to optimise these.

There is one exception to my reluctance here and this concerns capacitors, particularly those in the feedback loop of a feedback amplifier and to a lesser extent those in the signal line.

This is a complex business but there appear to be a few general rules. The extent to which a capacitor will influence the sound depends on the signal voltage which could appear across it, upon its impedance in relation to the other circuit parameters, and upon its dielectric hysteresis and dieiectric loss. In a feedback loop, a low leakage aluminium electrolytic is greatly preferable to a tantalum type but must operate with a polarising voltage present across it. Electrolytics will usually benefit from the presence of a paralle. connected non-polar type.

Ceramic capacitors can cause some curious effects due to the voltage dependence of their capacitance and should be left to FF circuitry. Silvered mica types seem quite harmless but there is nothing which they can do which polystyrene ones will not do better.

Inlevel low level audio circuitry, such as moving coil head amps,
aluminium electrolytics appear to work quite well (which is a good thing since the alternative ways of amassing the necessarily large values of capacitance would be extremely bulky - not to mention the cost).

At higher signal voltage levels, polypropylene, polystyrene, polycarbonate and polyester types, in that order of preference, are better than electrolytics - if one has to use a capacitor at all. If one can rearrange the circuit to avoid or lessen the number of capacitors in the signal line, this is worthwhile.

There is not, Ithink, much difference between polystyrene and polypropylene types and the latter tend to be more variable in quality - perhaps due to some manufacturers saving pennies by using a packaging grade polypropylene film instead of the more expensive electrical quality material, or perhaps by an inadvertent substitution of film types.

I have tried, therefore, particularly in the revised version of the preamplifier, to recast the circuitry in a form where the capacitors used are the optimum choice for their circuit function in terms of sound quality, practicability and cost. The same considerations apply to the other components used - none of which has escaped scrutiny.

The other major areas in the choice of components are in order of impertance, the switches and connectors /which should have a contact area appropriate to their fikely current flow and should be gold plated in low level circuitry) and the integrated circuits such as op amps and voltage regulators. These are now very widely available from a range of sources, but the performance of notionally identical devices from some of the minor manufacturers can sometimes be very poor due to lack of adequate quality control. It pays, therefora, to spend a little extra for a component from a major producer.
the -original system used a pair of moving coll microammeters, connected to provide peak-reading linear scale power meters. I thought that the choice of a linear power scale would be useful since tt would give the user a good idea of how close he was to the output power overload region - which would, in practice, probably be a bit above 100 W per channel.

However, the meter pointers were nearly always at the bottom end of the scale. I know that in my own home, and wth my own 'BBC monitor' type LS units, I very seldom use more than about 3 W peak, but I had assumed, incorrectly, that there were out there in the great beyond more stalwart audiophiles whose eardrums and nelghbours would allow them to push their amplifiers to much higher power levels.

Since it seems that I had misjudged this, I have opted instead for a fairly conventional twin LED bargraph circuit (Fig. 3). This gives a near instantaneous peak-reading, log-scale display, covering the output power range $0.2-100 \mathrm{~W}$. This design provides a module which would be usable as an output power display for other audio amps.

I appreciate that some people like to listen to their music in near darkness, where the flickering bar-graph power display might be tresome. Others would resent the continuous presence of such a visual intrusion, even in full lighting. I have therefore arranged that the display can be switched off when not needed.

The same LED display panels also carry the power 'on' and power supply overload trip' warning LEDs. The overlay is shown in Fig. 4. In order to power this bargraph display circuit, a pair of $\pm 15 \mathrm{~V}$ lines is also derived from the power supply unit by way of a small additional piece of circuitry, shown later.

## Power Supplies

Experience has indicated that a few changes would be useful here. The circuit is shown in Fig. 5. In particular, fault reports from constructors suggest that Q17, an MJ2501 pnp 'Darlington' transistor (in the main $\pm 55 \mathrm{~V}$ power supply line), is more vulnerable to damage on PSU output short-ctrcults than I had hoped. In some cases, when this caused the transistor
to go open ctrcuit, Q7 and R31 would try to pass the output current instead and this wasn't good for them. I have included two diodes, D23 and D31 (together wth D24 and D30 in the -ve channel) to prevent this undesired current flow through the over-current protection ctrcuit.

However, the main improvement in the PSU principally to ensure that the output lines were truly s/c proof (but which also seems to have given a slight sonic bonus by way of a small lift to the already high level of audio 'transparency') - is that the series 'pass' transistors in both lines are replaced by power MOSFETs, since these are both faster in action and also much more rugged devices. The ctrcult changes required to allow this substitution are relatively slight, and are shown in the amended PSU circuit diagram (Fig. 5).

The use of MOSFETs here would also allow the HF stabilisation capacitors (C7 and C8) in the loop feedback circult to be omitted, since with MOSFETs, which have a very good HF response, the feedback loop is quite stable without them. I would however recommend their retention at about the 3 n 0 value, since this has no ill effect and an increase in stability margin is always useful.

A point Id like to make here is that there is a current vogue for using greatly oversized mains transformers because it is thought by their users that this modification somehow improves the solidity' of the sound. So it might, in a simple transformer-rectifier-reservoir capacitor PSU, provided that the reservoir capacitors were adequately large and the conducting impedance of both the rectifiers and the wiring joining them to the transformer and to the reservoir capacitors was adequately low.

However, a competently designed electronically stabilised PSU can have an output impedance of a small fraction of an ohm, even down to subsonic frequencies and to match this impedance characteristic with conventional capacitors would require values in excess of 1 Farad.

I do not recall ever seeing a 1 F 80 V working capacitor but were such components available they would be both dear and bulky - an inelegant and


Fig. 3 Circuit diagram for the dot bargraph display


Fig. 4 Component overlay for the bargraph power display
wasteful way of approaching the problem of PSU design. An additional advantage of an electronically stabilised PSU is that it also effectively isolates the power amplifier from the mains transformer, whose characteristics will then be relatively unimportant so long as it is adequately rated to deliver the required input voltage without overheating. Also it doesn't contribute any annoying mechanical 'hum' or unwanted stray magnetic fields.

The power supply for the LED bar-graph display needs only to be stable enough to give a constant LED brightness and to avoid exceeding the voltage or dissipation ratings of the bar-graph IC. The circuit used employs a single transistor emitter follower (Q23 or Q24) whose base is supplied with a $\pm 16 \mathrm{~V}$ potential from a simple zener diode voltage regulator.

The power supply unit has an output voltage which is capable of adjustment and could stand as a quite versatile independent module, to upgrade other amplifiers designed with rather cruder PSU arrangements. It has therefore been designed as a monoblock unit on a heavy gauge aluminium subchassis, with the PCB shaped to fit around the toroidal mains transformer.

In the 80W design, this sub-chassis also carries the LS output connections (via 4 mm binding posts), the mains inlet and output sockets and provides heat sinking for the PSU MOSFETs which are mounted in holders and fitted with insulating covers to obviate inadvertent case-to-chassis short circuits.

The new component layout is shown in Fig. 6.

## DC Offset Protection

In order to avoid the need for relay contacts in the LS output lines (in the event of a DC offset arising) I had included a protection circuit in the PSU system $(\mathrm{Q} 9, \mathrm{Q} 11, \mathrm{Q} 13, \mathrm{Q} 14$ in the + ve line, and $\mathrm{Q} 10, \mathrm{Q} 12$, Q15,Q16 in the -ve line), which would shut down the PSU in the event of an offset.

I had, however, erred on the side of excessive caution, and this could cause the PSU to shut down on quite legal low frequency excursions - especially from compact discs, which have a very good subsonic response. I propose therefore that two 470 k resistors, R 42 and R 43 , should be added between the bases of Q9 and Q11 to the 0 V line, to reduce the 'trip' sensitivity of the circuit.

Please note that there was a misprint on the onginal diagram which showed D11/D18 joined to the OV line. Most constructors spotted this but if it was left uncorrected, it rendered inoperative the second shutdown mechanism in the PSU, which is designed to be triggered if an excessive difference occurs between the PSU $\pm$ output voltage levels.

## The Power Amplifier

1
The circuit (Fig. 7) of this is unchanged from that shown in ETI in July 1984, except for an amendment to the values of the output gate stopper resistors R18,19,21 and 22, and the inclusion of the small damped inductor L1/R31 in the output line, to ensure complete compatibility with a wider range of $L S$ units or strange speaker cables.


The added input switching arrangements have already been noted but there are also some changes added in the evolution of the kir' design which concern the balance and stereo/mono switching. This last facility was included to save difficulty if the amp was driven from a mono input source, such as a TV sound pick-off socket. The recommended PSU component overlay is shown in Fig. 6.

## Physical Layout And Kit Evolution

Looking at the efforts of friends and acquaintances who had made their own PCBs and assembled this amplifier from scratch made me aware, not for the first time, just how many things it was possible to do wrong. Even the unwise connection of earth points may destroy the purity of the output signal, let alone
the effect of allowing a pair of unscreened input wires to trail across the chasis underneath the power amp and PSU boards.

I was, therefore, very happy to find that Hart was prepared to invest the quite considerable amount of effort needed to work out a neat and fully debugged kit for this design, and 1 am grateful that they were prepared to make and submit prototypes for me to test and to make layout amendments with very good grace when. I felt that some aspect might be improved.

Unfortunately, because their works is some hundreds of miles away from my own lab., the task of evolving a fully 'designer approved' collection of hardware has taken rather longer than either of us had hoped.

1 had initially underestimated the amount of space which would be taken up by the PA and PSU boards and the toroidal mains transformer. There is




| $01 \quad$ BC182 (0 pad) | 82,3,47,48 <br> R5-10,23-28,41 <br> R11,12 <br> R13,4 <br> 815,16 <br> R17,18 <br> R19,20 | 10 K |
| :---: | :---: | :---: |
| $02 \quad$ BC212 (0 pod) |  | 4.7 |
| D1-4,45 1 N4148 |  | 337.5W |
| D5-14 $\quad$ LED bargraph display assembly |  | 154. 5 W |
| LEDI Ted LED, upper PCB only |  | OR222.5W w/w |
| LED2 green LED, lowee PCB only |  | 1208 |
|  |  | 1 mo |
| miscellaveous | R21,22,37,38 | $10 \times 0.5 \mathrm{~W} \mathrm{w} / \mathrm{m}$ |
| SK2 10way 2.5 mm pitch socket (Lppes PCB | R29,30 | 12\% |
| only | R31,32,40 | 12K.5.5 |
| PCR. KC sockess. $90^{\circ}$ LED mounting. PC8 2 -pin hroder, 2.4 mm . | R33,34 R35,36 | 15k |
| Jumper socket. 3-Way header and prewired socket for 5 Wm . | R35,36 | 68\% |
| Power AmplifierRESISTORS (all 3 W metal film $1 \%$ unloss specified) |  |  |
|  | R42,3 R44,45 | 470. 820.5 W |
|  | R46 | $3 \times 30.5 \mathrm{~W}$ |
| $\begin{array}{ll}\text { R1,14 } & \text { 150k }\end{array}$ | $R 49$ | 2k20.5W |
| $\begin{array}{ll}\text { R2 } & 467\end{array}$ | R50 | $270 \mathrm{R} 2.5 \mathrm{~W} \mathrm{~W} / \mathrm{w}$ |
| R3 1 k 2 | R51,52 | 2208 |
| R46 | RN1,2 | 10k preset horiz |
| $R 7$ 47 k <br> RO  <br> 000  | R 3,4 | 22 kpreset horiz |
| $\begin{array}{ll}\text { R8 } & 820 R\end{array}$ |  |  |
| R10 10 470k | capmerois |  |
| $\mathrm{R11}$  <br> $\mathrm{R12}$ 470R <br> 13  | C1,2 | roou 63V olec radial 5 mm |
| $\begin{array}{ll}\text { R12 } & \text { 56R } \\ \text { R13 } & \\ \text { P15 }\end{array}$ | C3,4 | 220 COV elec radial 5 mm |
| R13 398 <br> R15 22k | C5, 6 | 20263 V pohester radial 15 mm |
| $\begin{array}{ll}\text { Ri8,19 } & \text { 330R }\end{array}$ | C7,8 | 370 polystyene axial |
| R20 8R2 2.5 W w/w | C11,12 | 220163 V elec radial 517.5 mm |
| R21,22 270R | C13,14 | 22 v 25 V elec radial 12 mm |
| R23-27 OR2225W w/w |  |  |
| R28 10R | seuconouctors |  |
| $\begin{array}{ll}\text { R29 } & \text { 10k } \\ \text { P30 }\end{array}$ | 01,24 | TP428B0538 (T0220 heatsink) |
| R30 39 cos | 02,3,8,19 | BC447 |
| R31 8 R2 \%W (matrix for Ll) | 04,5,7,20 | BC448 |
| RV2 $\quad 1 \mathrm{kO}$ lin cermet presst | 06,23 | TP418,80537 (10220 heatsink) |
| $\begin{array}{ll}\text { RV4 } \\ \text { R9 } & 2 \times 2 \text { balance preset }\end{array}$ | 09,11,14,16,21 | BC184 |
| R9 CAPACTIORS $2 \times 2$ | 010,12,13,15,22 | BC214 |
| CAPACTIORS | 017 | 2SJ49/50 fitted with insulated covers and |
| C1,6,10,12,14,16 470n polycarbonate film radial 15 mm | 018 | 2 2K13445 holders on power supply chassis |
| C2 330 p polystyrene film axial | 01-16,23,24,30,31 | 1N448 |
| C3 $\quad 100 \mathrm{p}$ polystyrene film axial | D17,18 | 1 10003 |
| $\mathrm{C4,15} \quad 100 \mathrm{n}$ polycarbonate 10 mm | D19222 | B7X49 600R |
| C5 $\quad 100$ polystrene fim axial | D25-28D29 | 10 V zener 400mW |
| C7 $\quad 2 \times 407$ polycarbonate fimm radias 27.5 mm |  | $24 / 30 \mathrm{~V}$ zener 400 mW |
| C8 $\quad 2 \times 10 p$ polystyrene film axial in series | D32,33 | *V7 zener 400nw |
| C9,11,17,18 $\quad 220063 \mathrm{~V}$ electrolvtic radial 7.5 mm |  | , |
| C13 $\quad 220 \mathrm{n}$ polycarbonate film redial 10 mm | miscrianeeus |  |
| $\begin{array}{ll}\text { C20 } & \text { 100u1 } 25 \mathrm{~V} \text { radial electrolvtic } 7.5 \mathrm{~mm} \\ \text { C21 } & \text { 10n polycarbonate film radial } 5 \mathrm{~mm}\end{array}$ | SK1 10-way 2.5 mm pitch verical |  |
|  |  | $50-50300 \mathrm{VA}$ toroidal |
| SEMCONOUCTORS | PCR Chassis. PCB tabs. Diode cradles. Heatsinks. Speaker terminals. Mains swich. Mains inlet and outtet sockets + cover boots. |  |
|  |  |  |
| 03,5,7,8 8C184 |  |  |
| 04,11 | Input selector section |  |
| $\begin{array}{ll}09 & \text { BF870 } \\ 010 & \text { V412104 }\end{array}$ | RESISTORS (1).3W 5\%) |  |
| $\begin{array}{ll}010 & \text { VN1210M } \\ 01214 & \text { 2SK135 }\end{array}$ | R1,3 |  |
| $\begin{array}{ll}012,14 & \text { 2SK135 } \\ 01315 & \text { 2S150 }\end{array}$ | $\begin{aligned} & \mathrm{R} 1,3 \\ & \mathrm{R} 2,4 \end{aligned}$ |  |
| $016 \quad 2 N 5459$ | $\begin{aligned} & R 24 \\ & \text { R5,6 } \end{aligned}$ | 3*3 RN1 100k log, 2 -gang 2k2 AN3 1 kO lin 2 -gang |
| $\begin{array}{ll}\text { D3,4 } & \text { 1N4148 }\end{array}$ |  |  |
|  | C1,2 | 100 polyerbonate rad 15 mm |
| miscauneous | msceluneous |  |
| PCB. Tronsistor pads (12). Gold phono sockets (22). Heatsink $\mathbf{T} \mathbf{0 3}$ engle | SW, 2 |  |
|  | SW4 | $2 P$ push switch |
| (4). PCB male tabs. 250 horizontal (8). Diode cradles (41). nsulating disess (5). |  | 4 P push switch 3W |
| Power Supply |  |  |
|  |  |  |  |  |
| RESSISTORS \{idll. $3 \mathrm{~W} 5 \%$ a better) R1,4 43 3 | Winchasis Howsink $190 \times 75 \times 50 \mathrm{~mm}$. 1 chassis cover. Bockplate. Froontlate Dias plate for display. Knobs. |  |
|  |  |  |  |



Fig. 7 Circuit diagram of the complete power amplifier
also the electrically desirable, but physically impossible, condition that the two power amp PCBs should occupy the same space, so that there would be no unwanted 'loops' in the wiring between them and the inputs, outputs and power transistors, and so that all the input and output wiring should be of the same length.

A further desirable aspect of the layout is that the connecting leads to the power MOSFETs should be

carefully arranged since, with these devices, if one doesn't watch one's step it is very easy to find that one has unwittingly constructed a VHF oscillator.

Hart solved the problem of the MOSFET connections very neatly by attaching an angle bracket (see photo) in thermal contact with the main heat sink to the PCBs and then directly mounting the MOSFETs through this bracket on to the tracks of the PCB. This also solves the problem of accidental mis-connection to the pins.

Their solution to the requirement of identical position of both boards was to mount the two power amp boards flat, one above the other, separated by spacers between the boards. Preset adjustments are made either from above or through access holes in the metal chassis.

In order to make it easy to remove the power amp boards, the input and 'balance' control connections are made via gold plated phono connectors and the heavy current power supply and output connections by means of $63 / 0.2 \mathrm{~mm}$ stranded wire and flat tab push-on connectors.

The critical requirements of the earth return connections are satisfied by care in the layout of the PCBs and by joining the two amplifier output $0 V$ lines, 'E2', by identical length, heavy duty cables to a common earth point on the PSU board, together with the lower current 'El' points. The earthy side of the LS output is then taken directly to the PA PCBs from their respective LS output binding posts.

In order to avoid input earth loops, the six goldplated phono input sockets (CD, aux. and preamp.) are mounted on a small separate PCB, which commons the earth returns so that this can be taken by one only of the six screened wires to the input selector board. This input socket panel also isolates the input earths from the chassis.

The input selector board has its four outputs via further gold plated phono sockets which take the
stereo pairs of signal and balance control connections through screened cables to the two power amp boards via the mono/stereo and mute switches.

## Performance

My intention in evolving this design was to offer a circuit which would outperform by a significant margin all but the very best audio amplifiers available today. I do not think that this aim is quite as vain as it might seem, because it is my belief that many apparently near-perfect designs are spoiled by their pursuit of an 'over kill' in aspects where further improvements bring little benefit, while neglecting such things as 'settling time' or loop stability margins, which are very important in terms of sound quality though never specified.

The technical specification for this design, in terms of output power, power bandwidth, harmonic and intermodulation distortion, is fully in line with modern expectations and the transient response is, I believe, a good bit better than average since this is one of the areas which tend to be neglected in commercial hardware.

However, there are aspects of design which affect sound quality but which are not easily related to technical measurements. This is not, I am sure, because they cannot be measured but because we (and I think I speak for most of the designers working in this field) cannot be sure what it is we need to measure. Where I know that one approach in terms
of circuit architecture or component choice gives a better sound quality, or even seems better, than another which is electrically apparently identical, I have deliberately opted for that which gives the best sound.

Since the success of the project depends to a large extent on layout and wiring detail, I have been surprised and pleased by the occasional very enthusiastic letters which have appeared in the "hi-fi' press, from constructors who have overcome the fairly major problems involved in the construction of this circuit.

I am, therefore, very grateful both to Hart Electronic Kits who have put so much effort into making this design a practicable reality rather than just a beguiling chimera, and to ETI for allowing me a second bite at this cherry. It is now my hope that this amplifier may provide the constructor with a unit which will not be outclassed in any company.

## BUYLINES

Everything in this amplifier design is available from Hart Electronic Kits Ltd, 4 Penylan Mill, Oswestry, Shropshire SY10 9AF. Tet: (0691) 652894.

A complete price list for each section can be obtained from Hart. The special price for a complete kit is $£ 318.25+$ VAT, Hart code K 1100 . A slave version without input selector and bargraph costs £269.07 + VAT, code K1100s.

Add $£ 2.00 p+p$ or $£ 9$ for next day delivery.


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Fig. 8 Component overlay of the complete power amplifier

## REV-RIDER

> Stuart Richards converts his Raleigh into a Kawasaki 1300 with the help of this motorcycle simulator


This project was inspired by my childhood memories of making my bicycle sound like a motor bike by fixing a piece of cardboard near the rear wheel so that it flapped against the rotating spokes. The customary items to use were a piece of cigarette packet from the gutter and a wooden clothes peg stolen from Mum.

The Rev-Rider described here is a solid state version complete with ignition key, adjustable tickover, push-button revving control, flashing LED and last but not least a siren to annoy the neighbours.

The project is simple to construct, will give hours of pleasure from a small 9 V battery and can be fitted to any vehicle (even an exercise bike!)

## Construction

The prototype was built in a surface mounting car speaker case with a piece of paxolin as the back baffle to the case (Fig. 2). The speaker was left in situ and the PCB designed to fit to one side of the case (which explains the lop sided shape of the board!).

Cheap car speaker units are easily obtainable for little more than the cost of a drive unit alone. A high quality drive unit is not required as the Rev-Rider is hardly hi-fi.

Before assembling the board, familiarise yourself with Fig. 3, checking orientation of all polarised components.

Start with smallest components first - resistors, diodes and links - then progress through to ICs and capacitors finishing with presets and the power Darlington, Q1.

I recommend fitting IC sockets. They prove invaluable if you need to change an IC.

Insert single sided pins at the off-board connections. This willallow wires to be connected and disconnected from the component side of the board. To fit these, push them through from the copper side as far as you are able then carefully push them the rest of the way with a hot iron, then solder to the track. This completes assembly of the board.

For convenience, the 'Ignition Switch' is simply a 3.5 mm jack socket, with a short-circuited plug for the key. Drill a 2 mm hole through the plastic plug cover to accept a key-ring and fob. This key will prove easy to replace if lost!

Drill holes in the case to accept jack socket, LED ( 5 mm diameter) and push buttons if fitted. The LED will be a tight fit but should be secured with epoxy adhesive. Install the jack socket and push buttons, connect these together following the diagram leaving sufficient length of wire to enable the board to be released if necessary. Mount a 9 V alkaline battery to the side of the case using a double sided adhesive pad. You shouldn't need to change this very often!

The two push buttons SW1,2 can either be fitted in the speaker case or mounted near the handlebar grip and connected to the board using thin 3 -core mains cable taken through a hole on the case.

A simple way to mount the switches this way is by using a metal bicycle bell, first removing the striker and other internal bits. One or two holes are drilled in the side to accept SW1 and/or SW2 and the bell clamped to the handlebars so they can be operated by the thumb.

A back panel for the case can be cut from a sheet of plastic or hardboard using the speaker case as a template. Drill the two holes in the rear panel and
mount to the speaker case using M3 nuts and bolts. Parts from two bells were again used to clamp the case to the handlebars of the bicycle but suitable clips can be obtained from any hardware/cycle shop.

Check again all wiring, battery polarity and correct insertion of all ICs. Insert ear plugs and switch on!

## Kick Start

You should hear an engine type noise somewhere between a slow tickover and a high-pitched scream. Adjust the tickover control RV2 to the desired tickover speed. If you get no sound at this stage, check first that 9 V is getting to the board. The VCO input of IC3, Pin 9 should be at $1.9-2.6 \mathrm{~V}$ according to the charge on C2. If this seems in order, look for a square wave output on Pin 4 of IC3.

If you have no scope, a multimeter connected from 0 V to Pin 4 should read an average of half supply volts ( 4.5 V ). The voltage on the reservoir capacitor

C 9 will be almost 9 V at tickover and will decrease as the revs increase, as more energy is used by the speaker.

Press SW1, revs should increase slowly up to a maximum which can be set by adjusting RV1. Release SW1, revs should decrease again to the set tickover



## HOW IT WORKS

The heart of this project is a voltage to frequency converter. This gives out a square wave pulse train, the frequency of which is proportional to a DC voltage that we apply to its input. For convenience, we use a 4046B phase lock loop IC which includes the voltage to frequency converter.

By applying an increasing voltage input to IC3, the 'revs' will increase and vice-versa. Capacitor C2 gives the circuit inertia, acting like a flywheel in an engine. Pressing the Rev button SW1, charges C 2 via R1, R3 and D2, the maximum revs being set by RV1. Releasing SW1 allows C2 to discharge via RV1, R1, R2 and D1.

To achieve a steady tickover speed, independant of battery condition, the VCO input requires a positive offset of about 1.9V. To do this, a constant current of 0.56 mA from regulatordiode D 3 , passes through R4 and 'jacks up' the bottom end of C2. RV2 allows this voltage to be adjusted by shunting R4 and R5. D4 prevents C 2 from being reversed biased. C3.ensures the circuit starts up slowly at switch on.

To give a high volume output with low battery consumption, the loudspeaker is fed with short duration pulses at 4.6 V in amplitude. Q1 drives the speaker in emitter follower mode, drawing peak currents of over 1 A from the reservoir capacitor C . Resistor R15 limits the current drawn from the battery.

IC4 is used as two monostables triggering on alternate positive and negative edges of the VCO square wave output. To achieve a more reallistic sound, these monostables are given different time constants, (typically 50 and 150 microseconds) resulting in alternate short and long pulses to the speaker.

Capacitor C8 slows the edges of the pulses making the noise more 'user-friendly'.

The resulting output provides ample volume while the drain on the battery is less than 5 mA at tickover, rising to about 20 mA at high revs.

The siren effect is obtained by utilising a 4066B quad analogue switch IC to change over the input of the VCO IC3, from the revving circuit previously described to triangular waveform generator $I C 2 b$.

The triangular signal from IC2 pins 1 and 2 passes through the switch ICla and is AC coupled via R 6 and $\mathrm{C1}$ to the cathode of current regulator diode D 3 , modulating the VCO input signal to IC 3 via R 7 and closed switch IClb .

Pressing the siren button SW2, brings this into action and also effectively shorts R15 allowing greater volume in the Siren mode. Two paralleled spare gates, IC2c, drive the flashing LED directly from the square wave output of $\operatorname{IC} 2 \mathrm{~b}$.


Fig. 1 The circuit diagram of the Rev-Rider

speed. Press Siren button SW2, the circuit should give loud wailing noise.

You can increase or decrease wailing speed by reducing or increasing the value of R9. For test purposes (or for indoor use) it may be desirable to insert a resistor ( $10-47 \mathrm{R} \frac{1 / 4 \mathrm{~W}) \text { in series with the }}{}$ loudspeaker to reduce the volume to a tolerable level.

## Further Thoughts

The project outlined here suggests only one means of varying the speed of the output. Readers with mechanical flair might like to fit a proper throttle using a potentiometer connected either to a twist-grip or lever operated throttle.
'Gear changes' can be effectively achieved by switching in different values of charge resistor instead of R 3 to give differing rates of charge/discharge of C 2 .

A slightly unstable tickover speed, as in many two-strokes can add further realism by slowly changing the voltage at the junction of R4/R5.

Types of speakers and enclosures will change the sound dramatically - a speaker mounted inside a cardboard tube for example will add resonance and give a nice popping sound like many $50-100 \mathrm{cc}$ machines.

With small changes the circuit could be adapted to other toys such as lawn mowers, pedal cars and so

PARTS LIST on. I wish you hours of fun.

| RESISTORS (all $\% \mathrm{~W}$ 5\% 5 |  |
| :--- | :--- |
| R1,9,10 | 47 k |
| R2 | 1 MO |
| R3 | 220 k |
| R4 | 3 k 3 |
| R5 | 10 k |
| R6,12 | 15 k |
| R7,13 | 33 k |
| R8 | 100 k |
| R11 | 1 kO |
| R14 | 470 R |
| R15 | 220 R |
| RV1 | 47 k preset |
| RV2 | 22 k preset |


| CAPACITORS |  |
| :--- | :--- |
| C1,4 | $10 \mu 25 \mathrm{~V}$ radial electrolytic |
| C 2 | $22 \mu 16 \mathrm{~V}$ radial electrolytic |
| C 3 | $100 \mu 16 \mathrm{~V}$ radial electrolytic |
| $\mathrm{C} 5,6,7$ | 10 n ceramic |
| C8 | 100 n polyester |
| C | $1000 \mu 16 \mathrm{~V}$ radial electrolytic |


| SEMICONDUCTORS |  |  |
| :--- | :--- | :---: |
| IC1 | $4066 B$ |  |
| IC2,4 | $4093 B$ |  |
| IC3 | $4046 B$ |  |
| 01 | TIP122 |  |
| O1 |  |  |
| D1,2,4 | IN4148 |  |
| D3 | J503 |  |
| D5 | IN4004 |  |
| LEDI | 5mm Red LED |  |


| MISCEL |  |
| :---: | :---: |
| $B 1$ | 9V battery |
| LS1 | 4R 10W loudspeaker |
| SK1 | 3.5 mm jack socket |
| PLI | 3.5 mm jack plug |
| SW1,2 | SPST push switch |

PCB. Case. Battery clips. Connecting wire Mounting clips. Nuts and bolts.



## BUYLINES

Most of the parts for the Rev-Rider should pose no particular probiems. A complete kit of parts for the Rev-Rider excluding a battery and the case back baffle is available from Technova Developments, Grange Walk, Wroxham, NR18 8 RX (Tel: 106053 ) 2215) for $£ 16.50$ plus VAT plus $£ 1.75$ postage.

A suitable speaker and case can be purchased at a variety of
shops including Tandy, motorist accessory shops and your Jocal component stockists. The case and speaker used in the prototype is available from Technova for $\mathrm{f} 3.95+$ VAT f 1.75 postage. The J 503 constant current diode (D3) is available from Electromail as part number $283-463$ or from Technova for $£ 1.20+$ VAT $+75 p$ postage. The PCB is available from Technova for $£ 2.25+$ VAT +75 p postage.

## 41

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# AERIAL AMPLIFIER 



Paul Chappell, ETI's illustrious Projects Editor, adds even more class to a former First Class project



What do project editors do in their spare time? Build other peoples' projects, of course! The last one I had a go at was Keith Brindley's Traveller's Aerial .Amp from the September 1988 issue of ETI. Although Keith designed it for mobile TV addicts - caravanners and the like - my reason for building it.was to get decent pictures from the portable TV and indoor aerial in my London flat. I felt that a smarter looking box was called for, and a mains power supply of course, but first I took a good look at the PCB layout.

Laying out PCBs for UHF circuits can be a very tricky business. A few millimetres of component lead or PCB track can look like a whopping great inductor at TV frequencies (not great in value, which will be the same at any old frequency, but great in its effects!). Two conductors side by side will make a pretty good capacitor.

When you think that 10 pF at 500 MHz (the kind of frequency we're dealing with) has the same effect as $1 \mu \mathrm{~F}$ at 5 kHz or $100 \mu \mathrm{~F}$ at 50 Hz (and bearing in mind that you'll be hard pressed to lay out a PCB without introducing capacitances of this kind of size), it's amazing that any UHF circuit ever works. Imagine trying to make an op-amp circuit work with $1 \mu \mathrm{~F}$ capacitors connected here and there between adjacent pins and 10 mH inductors in series with all the leads. Rather you than me!

For audio frequency circuits, we're used to thinking in terms of lumped components. Two pieces of wire running side by side will have a certain capacitance, bring them closer together and the capacitance will increase, put certain materials between them and the capacitance will be increased still further. But a home-made two-wire capacitor will' never have anything like the capacitance needed for an audio coupling capacitor, let alone a power supply smoothing cap. So we buy specially made parts where the two wires have been spread out into a huge area placed very close together and interleaved with a suitable dielectric material to intensify the effects. Since the capacitance effect that goes on inside the capacitor is perhaps a million times larger than the effect that occurs between two adjacent tracks on the PCB, we think of the capacitance as being concentrated in the
component and for the most part ignore any that goes on elsewhere.

For UHF circuits, the capacitance and inductance effects that turn up whether you want them or not are comparable in size to the components you're using and certainly can't be be ignored. Every single PCB ${ }^{\prime}$ track, piece of wire and component lead has to be regarded as an extra component in the circuit. The capacitance and inductance of these new components is not conceñtrated at a certain point but is distributed along their length, so two adjacent PCB tracks will look to the circuit like the combined inductor and capacitor of Fig. 1.


Fig. 1 The electrical appearance of PCB tracks at (a) low frequencies and (b) UHF frequencies

A first guess at what this circuit does would probably be that it acts as a low-pass filter. However it has one very interesting characteristic: if you balance up the ratio of inductance to capacitance just so and terminate the circuit with a certain resistance, you can fool the drive circuit into thinking it's driving 'a resistance and the receiving circuit into thinking that the signal comes from a resistive source.

On a PCB, you trim the inductance to capacitance ratio by adjusting the width of the track. This leads to the track dimensions chosen by Keith in his design. An extension of the idea can lead to all kinds of reactive components being made just by the size. and shape of the PCB tracks: inductors, capacitors, tuned circuits and so on.

This sorts out the main signal connections. The next consideration is to prevent signals being fed back from output to input, which could very easily cause
the circuit to oscillate. The first line of defence here is to use a ground plane (a large area of copper connected to ground) on the component side of the PCB and even better to have the same on the track side too. You can think of this as making the capacitance to ground at any point of the circuit large in comparison with any other stray capacitances, taking away energy which might otherwise end up in the wrong place.

Ground planes were included in the original design but being a cautious old projects editor I decided to take the technique one stage further. At low frequences, if two areas of copper are connected together, they'll be at the same voltage and that's all there is to say about the matter. With UHF signals, the changes are going on so fast that on a TV downlead (for instance) there will be several complete cycles of the signal along its length. If you could freeze the wire at one moment in time and then measure the voltage along the wire, you'd find a peak positive voltage at one point, zero at perhaps 150 cm further along, peak negative voltage after another 150 cm and so on. If you imagine connecting a 500 MHz signal generator to a length of cable which is shorted at the other end, the generator would go through several cycles before
bottom edge of the screening piece is soldered all the way along to the top ground foil of the main PCB.

## Construction

The circuit hasn't been changed at all - it's shown in Fig. 2. If it is to be permanently connected to its power supply, you can miss out D1 altogether - just connect


Fig. 2 The amplifier circuit diagram


Fig. 3(a) Component overlay for the aerial amplifier
the energy even reached the other end of the wire, so a short circuit at UHF doesn't mean quite the same thing as it does at low frequences. Eventually you'd get standing waves set up along the length of the cable - not at all what you'd expect if you fed the output of your audio amp into the same shorted length of cable. You might find the idea hard to swallow but if you're having any truck with UHF circuits, you better believe it!

The very same thing applies to the PCB - if you have a ground plane, you can't just think of it as an area of copper at 0 V . The voltage will actually vary at different parts of it and a voltage difference can exist between the top and bottom ground foils of the board. One thing that helps to keep the circuit in order is to connect the top and bottom foils of the PCB together in several places, particularly around the IC where the signals originate. If you've already built Keith's circuit, it's an easy enough matter to drill a few extra holes in the PCB and put in through links, just to be on the safe side!

One more thing I did was to include a screening piece made from an offcut of PCB material to further isolate the input of the circuit from the output. This straddles the IC like a cowboy on his horse. The


Fig. 3(b) the screening arrangement
a piece of wire across its position on the PCB.
The component overlay is shown in Fig. 3. In mounting the OM335 there are two conflicting requirements: one is that the $I C$ should be as close as possible to the PCB , the other that pins 2,3,5 and 6 should be soldered to the top foil of the PCB. The trick is to leave a gap just wide enough for the solder to be applied, without having the IC standing on long
stalks (each one of which will become an unwelcome inductor). If the IC stands with 1 mm of the leads showing above the board, this will be about right.

The capacitor should be mounted with the lead which connects to pin 4 of the IC as short as possible, and the other lead soldered on both sides of the board. The diode you just put in as usual. All the other holes, apart from the ones at the edges of the board (which will later connect to the sockets) should be used for through-links: use offcuts from the diode and capacitor leads, post them through the holes and solder to the top and bottom foils of the PCB.


The dimensions of the screening piece are shown in Fig. 4. If you intend to use the specified case, it would be just as well to follow them exactly. If not, just cut a rectangle of PCB material and make a slot in one edge to fit the IC. The screen slips over the body of the IC between pins 4 and 5 and is soldered to the top foil of the PCB all along the edge. The best method is to solder at one or two points to hold the screen in place, then to run solder all the way along the join.


The drilling positions for the front and rear panels of the case are shown in Fig. 5. The TV sockets are mounted just above the top surface of the top side of the PCB using the shortest possible lengths of wire. To see how much you need, screw one of the sockets to a panel, slide the PCB into the lowest slot in the box, fix the panel to the box, then sight along the top of the board to gauge the distance from board to socket terminals.

Attach both TV sockets and the phono (power inlet) socket to the PCB, soldering all earth wires to both foils of the board. Then fix the output and power sockets to the front panel, with the TV socket mounted behind the panel. To the rear of the input socket, glue a 4BA nut behind each flange.


Fig. 5 Construction detalls for the amplifier case"
Slide the PCB into the lowest slot in the case until the front panel meets the body of the case, then screw the panel in place. Screw the rear panel to the input socket, then screw that to the box too. The aerial amp is now complete!

If you're using the rubber feet, they come in the form of a strip which should be cut into two lengths, each the length of the body of the case. Before putting on the rear panel, you simply slide them into the grooves on the underside of the case. The panel holds them in place.


PARTS LIST



## Power Supply

For caravan and camping use the amp can be run from a car battery or dry batteries but for my purposes I wanted a mains power supply. The circuit is shown in Fig. 6 and the component overlay in Fig. 7. There's nothing complicated about the construction.

1 used a 24 V regulator, since this is the voltage at which the IC gives its maximum gain but if you can't get hold of one a 12 V or 15 V regulator will do. I used the 1 A version since the ' L ' version doesn't seem to be available at 24 V . There's no need for a heat sink - the amp takes very little current.

The PCB is attached to the bottom of the case by means of self-adhesive supports, avoiding any drilling. The fuseholder and 'on' indicator go right at the very top of the case - be careful to put them high enough so they don't foul the transformer. To take the power to the amp, a length of bell wire with a phono plug at the end is fine. Make sure to connect the plug so that the positive side of the supply goes to the inner

## BUYLINES

Various parts sets for this project are available from Specialist Semiconductors Ltd. See their ad. for details.

The OM335 on its own can be had from Electromail (tel: (0536) 204555) or from Highgrade Components, 8 Woburn Road, Eastville, Bristol BS5 6TT.

PCBs are available from our PCB service - details on the centre pages. For DIY PCB etchers, the patterns are printed on page 61.
pin (perhaps that diode is a good idea after all!). The wire should be knotted just behind the exit hole in the case to prevent any strain on the PCB connections, as should the mains wire for the same reason.

After checking with a multi-meter to make sure that the power supply is indeed giving 24 V , it's time to plug everything together. The aerial plugs into the input socket of the amp and a length of TV co-ax connects the output to your receiver. The power supply plugs into the power inlet on the amp and that's about it.

Oh yes, you have to turn the TV on too! Happy viewing.


Fig. 6 Circuit diagram for the power supply


Rig. 7 Component overlay for the power supply

## Rashid Adat has a simple circuit to ward off thieves

 MOVEMENT DETECTOR ere is a simple design for a movement detector with possibilities when used as a security device. It could protect expensive electronic apparatus, personal documents or indeed anything in a container that might be carried away by an unauthorised person. To protect documents in a filing cabinet, the detector could simply be placed in the drawer and when the drawer is opened, a high intensity sound results.

The main sections of the 'device are: motion detector (a mercury tilt switch), control logic and a high intensity audible buzzer. All parts are mounted in a small plastic box. The box has an LED to show the setting time and battery condition, a keyswitch socket to enable and disable the detector and several holes to allow the sound to leave.

So, to set our sentinel, here is what you do. Place or attach the box to the piece of apparatus you want to protect (preferably out of sight) and withdraw the jack plug. The jack plug functions as part of the switching device that enables the circuit when it is removed from the socket. When this happens, the LED turns on for approximately 10 seconds. After this time, the LED goes out and the circuit enters standby mode.

Thereafter, any motion imparted to the box is sensed by the mercury tilt switch. The angle of the switch to the circuit board was set at about 15 degrees (Fig. 1). This provides detection when the box is tilted slightly. The angle can be adjusted for maximum sensitivity. If movement in the horizontal plane is required, the angle should be made as small as possible. The mercury switch could also be pointed at a corner of the box thus providing a detection of tilt in two planes. Any tilt or movement results in the triggering of the buzzer. The circuit is designed to sound a 90 dB tone continuously and can only be silenced by inserting the key (jack plug) into the socket. This cuts power to the circuit.

If you wanted extra security you could use a keyswitch. With the jackplug it is just possible to shove a paper clip into the socket, short it out and reset the circuit. However this takes a fair amount of time to work out (as the ETI cleaning staff have discovered much to their chagrin), particularly if you're an alarmed burglar. In the meantime, the high intensity buzzer really is very loud.


## Construction

Construction is fairly simple for this project. When using the PCB, mount and solder a 14 -pin dual-inline socket first. Then solder the two resistors, capacitor and mercury switch as shown in Fig 4. Next solder the wires in for the connections to the battery, keyswitch, buzzer and LED. The wires to the LED and switch must be long so that the two halves of the box can be separated easily.

As the IC is a CMOS device, care must be taken when handling and inserting it into the socket. Make sure your body is earthed when carrying out this operation

All the circuitry is then housed in a plastic box of suitable size. The one used in the prototype measured $114 \times 76 \times 30 \mathrm{~mm}$. The PCB, buzzer and
battery can be mounted by simply sticking them inside the box with double-sided foam tape.

When constructing the movement detector on stripboard (Fig. 5), cut the copper lines first using a sharp drill bit. Then mount and solder the socket. Place the wire links in next and solder the components in as before. Finally, a matrix of holes was drilled in the plastic box to let the sound out.

## BUYLINES

All components are easily available. The buzzer is available from Maplin. cat no. FK84F and the mercury tilt switch also available from Maplin. cat no. FE11M. The PCB is available at £ 2.50 or the complete kit at $£ 11.99$ from the author at 20 Highview St, Botton BL3 400 .

## HOW IT WORKS

Once the key is removed, the LED is switched on for a period of about 10 seconds. This length is determined by the values of Cl and R 2 (see Fig. 1). When the voltage on C 1 reaches the threshold value of 3.6 V , the input to NAND gate c (connected as an inverter) goes high resulting in a low output from pin 10 . The LED switches off.

ICa and IC are connected as an R/S flip flop and when the key switch is first removed, S sets to logic 0 (low) and $R$ to logic 1 (high) because the mercury tilt switch is normally closed in its relaxed
position. The output from ICb (pin 4) adopts a logic 1 state. This output remains at logic 1 even when Cl has passed its threshold voltage making S go high. The circuit is now in its standby mode.

If the box is moved, MSI will open. R will now go low (logic 0 ) and the bistable will flip making ICb output go low. ICd serves as an inverter and its output will change at this point supplying the voltage to trigger the buzzer. The circuit can then only be switched off by reinserting the jack plug in the key switch

## PARTS LIST



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# PRIORITY QUIZ SWITCH 

> Ken Blackwell presents his fair fare to adjudicate over the most cantankerous contestants

This simple priority switch was designed for use as a quiz controller, with four contestants answering questions on a firstpress first-served basis.

Such a situation provides obvious design criteria for consideration. The push switches should be remotely positioned from the main unit and clear indication of the winning switch should be available for all to see. There should also be a simple and instantaneous reset facility and of course the circuit has to be scrupulously fair, otherwise you'll have a panel of contentious contestants before the game is out!

So this unit switches the first received signal from any number of inputs (four in this particular case but it is trivial to expand the unit), latching this signal and


Fig. 1 Circuit diagram for the priority switch
illuminating the relevant LED. If it were desired the LED could be replaced by a relay or opto-isolator to trigger a much bigger light design - or even to light contestants' names in front of them. Bob Holness eat your heart out!

## Construction

The circuit with four switches has been designed to fit into small handheld case from the company listed in Buylines. To give the clearest indication of the 'winner', different colour LEDs are used for each contestant and the switches and the wires to them are of the same colour as these LEDs. The coloured wires in the prototype were removed from a multi-strand cable after stripping the screen and outer cover.

The PCB layout is very straightforward and is shown in Fig. 2. The switches SW1-4 are obviously outside the case. Stripboard could be used as an alternative to the PCB.

The LEDs are positioned so that they are already in alignment to poke through holes in the box lid. The leads of the LEDs should be left long enough for the board to sit in the bottom of the case with the tops of the LEDs just above the height of the case.

Note that LED4 is a tricoloured LED. The centre leg is the cathode (negative bias) and goes to the common rail. Join the two outside legs of LED4 together to the positive position.

The four LEDs will mount through the top of the

box, so use the PCB or stripboard as a template to mark the positions for four 6.35 mm holes into which you can insert 5 mm plastic clips and bezels for a tidy finish.

The wires to the trigger switches leave through the front template. Drill four evenly spaced 5 mm exit holes across the centre width of the blank and fit 3 mm plastic bezels. Through each hole pass one of the pairs of coloured wires. It might be a good idea to ensure that the wire lengths are the same for each switch although this will make about as much difference to the fairness of the circuit's action as the different lengths of PCB track or the different actions of almost identical switches, it is the first thing a punter will identify as the cause of his slow reactions!

The reset switch can be mounted anywhere you like, remotely or on the case itself. A 7 mm hole is required to mount the recommended switch on the case.

With all the switches and LEDs wired up it remains only to connect the battery and to practice your catchphrase as quiz show host!

## HOW IT WORKS

The circuit consists of thyristor switches each comprising a normally open push button switch, a thyristor with a high resistance gate resistor and an LED.

Any number of these can be placed parallel to each other - the circuit illustrated in Fig. 1 uses four.

The circuit is restricted to supply only enough current for one module to operate, about 13 mA for an LED supply. To restrict power resistor $\mathrm{R1}$ is placed in the positive supply and all connections are made below this.

When one switch is closed, power is passed through that gate resistor, activating that module. As soon as it is conducting the remaining modules experience a voltage drop down to about 3 V . Although other switches are pressed there is insufficient power to switch a second module.

To reset the circuit, switch SW5 shorts the 13 mA to ground, unlatching the thyristor.

Different values of 11 would allow more or all modules to be lit simultaneously. This could be used for, say, four different teams to hit their button on completion of a task, in the style of the Krypton Factor or It's a Knockout.

Note that if different thyristors are used, the values of the gate resistors R2-5 may need to be lowered (between 470k and 1MO) to counter a higher base resistance in the thyristors.


Fig. 2 Component overlay on PCB for the priority switch

## PARTS LIST

| RESISTORS (all $1 / 4 \mathrm{~W} 5 \%$ ) |  |
| :---: | :---: |
| R1 | 680R |
| R2-5 | 1M0 |
| SEMICONDUCTORS |  |
| SCR1-4 | TIC106D |
| LED1 | 5 mm red LED |
| LED2 | 5 mm green LED |
| LED3 | 5 mm yellow LED |
| LED4 | 5 mm tri-colour LED |
| MISCELLANEOUS |  |
| BATTI | 9 VPP 3 battery |
| SW1 | push-to-make switch red |
| SW2 | push-to-make switch green |
| SW3 | push-to-make switch yellow |
| SW4 | push-to-make switch white |
| SW5 | push-to-make switch black |
| PCB or st match SW | LED bezels and clips. Wire and e. |

## BUYLINES

All the components in the prototype were obtained from Rapid Electronics, Hill Farm Industrial Estate, Boxted, Colchester, Essex CO4 5RD. Tel: (0206) 272730.

The PCB is available from the ETI PCB Service.

## PCB FOIL PATTERNS



The Aerial amplifier board


The Aerial amp power supply board


The Digital Doorbell solderside foil


The guitar tuner foil


The hearing booster for Granny


The Camera Trigger board


The Trembler board


The Twin loop metal locator board


The MIDI keyboard processor and power supply foil pattern solderside


The MIDI keyboard contact foil pattern



The MIDI keyboard processor and power supply foil pattern topside


The MIDI keyboard control panel foil pattern


## GUITAR TUNER



As a guitarist and bass player I have always considered that silent electronic guitar tuners, giving a visual indication when a string is in tune are an asset to any player.
Apart from poor performances, the best way to get your audience really agitated is to make them suffer fifteen minutes or so of random twangs as the band 'unes up' prior to the performance. Silent tuners eliminate completely this most unprofessional and annoying prelude to a performance.

In 1981 I was therefore keen to build my own tuner when a project for an inexpensive silent tuner appeared in one of the electronic hobbyists magazines. On completion of this project I tested it using a signal generator and frequency meter. The unit worked perfectly and I was impressed with its performance . . . for about three minutes.

When I tried to tune my guitar with it, the indicator LEDs jumped around so much that I assumed I had read the project title incorrectly and it was in fact a random lighting display.

To date I have built about half a dozen tuners published in various electronics hobbyists magazines - all bar one with the same result - they work fine with the almost pure sinusoids or square waves of a signal generator but are completely useless for tuning guitar strings.

The only one that did operate correctly used an
expensive top octave generator' IC, moving coil panel meter and two PP3 batteries. The cost of that project made a strong case for buying a commercially manufactured tuner.

My guess is that all of the designers of these projects suffered from the same handicap - they


Fig. 1 Block diagram of the guitar tuner

Laurie Barron keeps in tune with a guitar tuner that actually works. with guitars


Fig. 2 The circuit diagram of the guitar tuner
were not guitarists and therefore did not possess gultars. In consequence of this, they all produced designs which responded correctly to waveforms that were almost pure but did not realise the output from

| Guitar |  |  |  |  | Bass |  |  |  |
| :---: | :---: | :---: | :---: | :---: | :---: | :---: | :---: | :---: |
| Tone | Frequency | Volts | Tone | Frequency | Volts |  |  |  |
| E | 329.628 | 2.000 | G | 97.9989 | 2.000 |  |  |  |
| B | 246.942 | 1.515 | D | 73.4162 | 1.510 |  |  |  |
| G | 195.998 | 1.212 | A | 55.0000 | 1.137 |  |  |  |
| D | 146.832 | 0.916 | E | 41.2034 | 0.854 |  |  |  |
| A | 110.000 | 0.692 |  |  |  |  |  |  |
| E | 82.4069 | 0.522 |  |  |  |  |  |  |
| Table 1 Tones, frequencies and outputs of F-V converter |  |  |  |  |  |  |  |  |

a guitar is rich in harmonics and that these harmonics must be eliminated, leaving only the fundamental frequency before being compared with reference frequencies or voltages.


38

## Design Approach

The way I have approached this design is to look back over these failed designs, extracting the best ideas and adding efficient filtering to the input signal to effectively eliminate the unwanted harmonics.

The result is a tuner that uses a comparison of the output from a frequency to voltage converter with reference voltages derived from a precision voltage divider chain. The recent availability of cheap $1 \%$ resistors ( 3 p each) from suppliers such as Maplin makes this an extremely economic approach which has produced a guitar tuner that costs less than $£ 9,000$ to build (excluding PCB and case).

Referring to the block diagram of Fig. 1, the guitar signal first passes through a high pass filter with a cut off frequency of 72.34 Hz to remove mains hum.

Next follows a combined $x 100$ amplifier and 338 Hz low pass filter. From Table 1 it can be seen that the highest frequency required is about 330 Hz so it
makes sense to first remove all frequencies above that value. Because of the very high gain, the output from this stage is clipped, but this causes no problems in this circuit as the final waveform required is a square wave anyway.

The next stage is the switchable bandpass filter. This is probably the most important part of the tuner, (the one the others leave out!). It has a bandwidth of 36 Hz and provides a signal gain of $x 5$. The purpose of this stage is to effectively eliminate all frequencies not within 18 Hz either side of the frequency of the note to be tuned. The output from this stage is a crude square wave of the fundamental frequency only.

The fundamental frequency now passes to a Schmitt Trigger stage. This gives a clean square wave at the fundamental frequency. The components selected for this stage ensure that its output amplitude and frequency will remain constant for several seconds, almost until the string ceases to vibrate, the amplitude of the input signal can decay to about $10 \mu \mathrm{~V}$ before correct operation ceases and the string must be struck again.

The next stage is the heart of the unit. It is actually an adaption of Paul Chappell's Frequency Meter Module design, (ETI April 1986). This configuration provides the frequency to voltage converter block ( $\mathrm{F}-\mathrm{V}$ converter). Some component values are changed from the original design for use in this application but the principle remains the same.

The output does not exactly follow Mr Chappells rule of thumb' formula, although if voltage/frequency is plotted from the values given in Fig. 2 it will be seen the relationship between voltage and frequency is still linear.

The purpose of this stage is to convert the filtered fundamental frequency of each string into a voltage level and then to pass this voltage level to one input of a standard LM311 comparator. The other comparator input is fed inta a selected reference
voltage, derived from a voltage divider chain, consisting of a series of $1 \%$ tolerance resistors.

The divider chain is simply worked out using Ohm's law. The divider chain was calculated for a current of 1 mA , this value helps to keep the power


## HOW IT WORKS

The low level guitar signal, which is rich in harmonics passes through the simple high pass filter of R 1 and Cl with cut off frequency 72.4 Hz and then the low pass active filter network of IC la and associated components (cut off frequency 338.63 Hz ).

This combination from the tuner's input to the output of 1 C 1 a effectively gives us a simple bandpass filter of centre frequency 205.49 Hz and a bandwidth of 266.3 Hz . The low pass filter attenuates the unwanted frequencies above the top open string of the guitar and the high pass filter helps eliminate any mains hum pick up.

Frequencies within these passbands are amplified by a factor of x 100 which boosts the input signal to a usable value, which can be measured well into the decay time.

The amplified signal then passes through IC1b and associated components which constitute a selectable, narrow, bandpass filter of bandwidth 36 Hz . This configuration effectively removes all of the unwanted harmonics, leaving only the fundamental frequency

From the bandpass filter the signal passes through IC1c which with its associated componenets is configured as a Schmitt trigger. This serves the purpose of squaring up the signal fed to the frequency to voltage part of the circuit. The components for the Schmitt trigger are selected for very low upper and lower trigger points to activate the trigger almost until the string ceases to vibrate altogether.

ICId and associated components act to split the single 9 V supply into a dual 4.5 V supply required for use by IC1.

From the Schmitt trigger the signal passes to IC2a,b and associated components, configured as a frequency to voltage converter circuit. This circuit is really an adaption of the 'Frequency Meter Module' published in ETIApril 1986 and the reader is referred to this text for a detailed explanation of its operation.

At the bottom of the circuit diagram is the reference voltage divider chain of RV2 and R32-46. The value of each block of $1 \%$ resistors has been chosen to give a voltage division equal to the intervals between the notes of the open guitar strings after frequency to voltage conversion. TP1 is initially set up to 2.000 V with RV2. This sets up the reference voltages which will remain accurate until the battery voltage falls to below 7 V .

The outputs of the frequency to voltage converter circuit and the reference voltage divider chain are fed to the two inputs of comparator IC3 via resistors R28 and R29. This IC performs a comparison between the output of the V-F converter circuit and the selected output of the reference voltage divider chain.

If the note is flat, then the output of the comparator will be high, allowing Q 1 to conduct, switching on LED 2 fully and extinguishing LED1. If the note is sharp, the output of the comparator will be low and LED1 will be switched on and LED 2 extinguished. When the note produces a voltage equal to the reference voltage divider chains selected voltage, the comparators output will be half of its high level, thus allowing only partial conduction of Q1 and hence LED1 and LED2 will be equally illuminated.
consumption of the unit low and also makes for extremely easy calculation of the divider chain as the value of resistance between each reference point and 0 V is simply the $\mathrm{F}-\mathrm{V}$ output voltage $\times 1000$ which gives the required resistance in ohms.

The comparator stage simply compares the voltage level from the F-V converter stage with the selected reference voltage from the divider chain. If the F -V output is higher than the reference voltage (the note is 'sharp') then the comparator output will swing towards its negative supply rail ( 0 V ). This will light the 'sharp' LED indicator.

On the other hand if the voltage level from the $\mathrm{F}-\mathrm{V}$ converter is lower than the reference voltage (the note is 'flat') then the comparator will swing towards its positive rail (5V). This supplies base current to transistor Q1, lighting the 'flat' LED.

When both the reference voltage and the output voltage from the F-V converter are equal, the comparator output will be at half of its high level, thus only allowing partial conduction of Q1 and hence both LEDs will now be equally illuminated, indicating that the string in 'in tune'.

For those of you who like to analyse each part of a circuit, so that individual blocks can be extracted for use in future designs, the diagrams and formulae for each part of the circuit up to the F-V converter are given in Fig. 3. Further information on the F-V converter part of the circuit can be found by reference to Paul Chappell's Capacitance Meter Module and Frequency Meter Module, in the March and April 1986 issues of ETI.


2nd ORDER BANDPASS FILTER
BANDWIDTH: $B=\frac{1}{\pi . R 2 . C}$

$$
\text { CENTRE FREQ: } \cdot \text { to }=\frac{1}{2 . \pi \cdot C} \sqrt{\frac{R 1+R 3}{R 1 . R 2 . R 3}}
$$

$$
\text { GAIN AT Io: } \cdot-A V=\frac{R 2}{2 . R 1}
$$


UPPER TRIGGER POINT:
$U T P=+\mathrm{Vec} \cdot \frac{\mathrm{R} 2}{\mathrm{R}_{1}+\mathrm{R} 2}$
LOWER TRIGGER POINT:-
$\mathbf{L T P}=-\mathrm{Vcc}, \frac{\mathbf{R 2}}{\mathrm{R} 1+\mathrm{R} 2}$
SYMMETRICAL SCHMITT TRIGGER

Fig. 3 Circuit diagrams and formulae for sections of the tuner

## Construction

Although $1 \%$ resistors are used in the critical positions (and ideally for the entire assembly) even this degree of accuracy can cause problems with some of the larger resistors in the divider chain. In the worst case, resistors R33 and R44 (470R) could vary from their marked value by as much as 4R7 each. This variation would affect the accuracy of the tuner and I would recommend that all of the resistors of 200R or over in the divider chain ( $\mathrm{R} 33,34,37,39,44$ ) are selected


Fig. 4 The component overlay for the guitar tuner PCB

## PARTS LIST


with an accurate digital multimeter or some other accurate device for measuring resistance, to select the values closest to those specified. The easiest way of achieving this is to order a pack of ten of each value from Maplins and select the best from each pack for each value.

The component overlay is shown in Fig. 4. Fit the wire links first, followed by the resistors, off-board connector pins, IC sockets, capacitors, presets, the
rotary switch and last the semiconductors. Do not fit the LEDs at this stage.

The 14 PCB holes for the rotary switch must be drilled slightly larger ( 1.2 mm ) than those in the rest of the board. The pins of the rotary switch must have each eyelet cut off as close to the eyelet as possible, to allow the switch to be fitted to the board

The next task is to drill the holes in the case for the rotary switch, LED's and jack socket in accordance
with the positions shown in Fig. 5. If you are not using the MB3 type box, pay special attention to the relative positions of the holes for the LEDs and the rotary switch.

Fit (but don't solder) the two LEDs to the PCB, ensuring that the anodes and cathodes are in the positions shown in Fig. 4. Remove the nut from the rotary switch and holding the PCB with the rotary switch uppermost, fit the rotary switch and PCB to the case. Fit the switch nut and tighten to finger tightness. Square the PCB to the case. Now push the LED into their respective holes in the case so that their entire dome areas are just proud of the case. Now solder the leads of the LEDs to the PCB, ensuring that their position does not move during this operation.

Fit the jack socket to the case and connect it. Screened cable is not necessary for this application. The battery connector wires should also be fitted to the PCB at this stage.

## Calibration

This operation is carried out with the PCB assembly out of the case. To calibrate the tuner connect a 9 V battery to the battery connector and insert a mono jack plug into the jack socket. Connect an accurate voltmeter between TP2 and OV and adjust RV2 until a reading of exactly 2 V is obtained.

Now connect a square or sine wave signal with a frequency of exactly 329.628 Hz and an amplitude of between 20 mV and 5 V to the input of the unit. Turn the rotary switch fully anticlockwise. An accurate voltmeter is now connected between TP1 and OV and RV1 is adjusted until a reading of exactly 2 V is obtained. The tuner is now fully calibrated.

If you are not fortunate enough to possess a signal generator and frequency meter then simply connect an accurately tuned guitar or keyboard to the input of the tuner, strike the open top Estring or play keyboard E of 329.628 Hz , allow a half a second for the frequency of the string to stabilise and adjust RV1 until a reading of 2 V is obtained with the voltmeter connected between TP1 and 0 V .

The reading will remain constant for several seconds but the string may have to be struck several times with this method until the potentiometer adjustment reaches 2 V .

The unit can now be replaced in its case and the respective switch positions of $\mathrm{E}, \mathrm{A}, \mathrm{D}, \mathrm{G}, \mathrm{B}, \mathrm{E}$, (eat all day great big eggs) in an clockwise direction marked around the switch.

## Using The Tuner

Power is applied to tuner by simply inserting the guitar jackplug into the socket.

Select the required note on the switch and strike the appropriate string, after a short period the LEDs will stop jumping and an indication of whether the string is sharp or flat will be given by the LED that remains illuminated. It is always best to tune from flat upwards, so if the string is sharp, loosen it, strike the string again and turn the appropriate machine head of the guitar until both LEDs are equally illuminated. The string is now 'in tune'.

After very little experience with this unit you will find the tuning of each string can be achieved with only striking the string once and then adjusting the machine head.

The current consumption of the tuner is less than 12 mA , and power is only applied to the unit when the guitar jack plug is actually inserted into the tuners socket. As the tuner is only used for the few minutes required to tune a guitar, a standard 9V PP3 type
battery should last for many months. The tuner will remain accurate until the battery voltage falls below 7 V .

## Bass Players

If you're not a budding Eric Clapton but you are a potential Jack Bruce, don't despair, you haven't been forgotten. The same PCB can also be used to make a bass guitar tuner.

Simply substitute the component values and links as shown below for those in the main component list and omit components as indicated.

In addition, use a frequency of 97.9989 Hz when adjusting RV1 or the top G string (or a G of 97.9989 Hz on the keyboard).

The switch positions should be labelled E, A, D, G.

BASS PARTS

| RESISTORS (all 0.6 W 1\%) |  |
| :---: | :---: |
| R9 | 2k4 |
| R10,36,42-44 | link |
| R11 | 4 k 7 |
| R12 | 300R |
| R13 | 13k |
| R14 | 91 k |
| R15 | 6 k 8 |
| R16-19 | omit |
| R27 | 750k |
| R32 | 20R |
| R34 | 330R |
| R35 | 43k |
| R37 | 270R |
| R38 | 13 R |
| R39 | 820R |
| R40 | 18R |
| R41 | 16R |
| CAPACITORS |  |
| C1 | 47 n polyester |
| C2 | 150p polystyrene |
| C3,4 | 100\% $5 \%$ polycarbonate |



Fig. 5 Case drilling details

## BUYLINES

All components can be obtained from Maplin or indeed many other suppliers. The PCB is available from the ETI PCB Service.


> Robert and David Crone present their snouted seeker that exposes the treasure other detectors cannot reach


## TWIN LOOP TREASURE SEEKER



Pulse metal detectors are powerful and versatile machines but in their basic form they suffer from ground effect and radio interference. However a very simple modification can almost entirely eliminate these two problems.

The principle of the pulse metal detector is very easy to understand. A large pulse of current is transmitted through a coil of wire and the resulting magnetic field induces eddy currents in nearby coins or metal objects. The eddy currents continue to flow after the transmitted pulse has ended and they in turn induce small voltages back into the coil. These voltages are amplified and detected in a receiver which operates an audio indication, usually a click generator.

A problem with this is that the transmitted pulse induces eddy currents in mineralised ground causing a ground effect signal Secondly the coil acts as a good
aerial for long and medium wave radio broadcasts, producing interference. So what can be done about these problems?

The ground effect comes from a large area and is almost constant over a flat surface like a wet sandy beach after the tide has gone out. If we were to position a second search coil about 100 mm from the original then it would pick up the same amount of ground effect. Now if we were to subtract the outputs of the two coils the ground effect from each would cancel out. However the system would still pick up coins because the distance between the coils is large compared with a coin. By similar reasoning, medium and long wave radio broadcasts will cancel out as the field strength of these signals does not change significantly in 100 mm and each coil will receive the same amount of interference.

So the second coil is a modification to the pulse


Fig. 1 Block diagram of detector


Fig. 2 Circuit diagram of transmitter
detection system. Figure 1 shows a block diagram of the unit. The central feature is the search coil assembly which in practice consists of two coils each of 200 mm diameter and overlapping by 100 mm .

## The Transmitter

Figure 2 shows the circuit diagram of the transmitter. IC1 is wired as an oscillator running at 100 Hz . IC2 is triggered 100 times per second from IC1 via the differentiating network of R3 and C3. Each time IC2 is triggered its output goes high for $165 \mu \mathrm{~s}$ and drives the two power transistors hard on into saturation. The full battery voltage is now applied across the coils and the current in each one builds up to about one amp.

## The Timing Circuit

Fig. 3 shows the circuit diagram of the timing circuit. IC3 is triggered from the transmitter at the end of the $165 \mu \mathrm{~s}$ current pulse. Its output goes high for $36 \mu \mathrm{~s}$ and then IC4 is triggered via C8 and R11. IC4 runs for $50 \mu$ s and its outputgoes to the receiver where it switches on the detector for $50 \mu \mathrm{~s}$.

## The Receiver

Fig. 4 shows the circuit diagram of the receiver. The outputs from the coils are fed to the inputs of the
difference amplifier IC5. Here the ground effect and interference cancel out but the coin signals are amplified and passed on to the next stage. The 709 is used in the IC5 position because its noise figure is good enough for the job. Diodes D1 to D6 protect the op-amp inputs and are configured so that IC5 does not go into an indeterminate state when the diodes are on. Q3 is switched on for $50 \mu \mathrm{~s}$ by the timing circuit and allows the coin signals to pass on to the detector and amplifier IC6. When constructed, set pin 6 of IC5



to $-1 V$ by adjusting RV1 and set the receiver output to -0.3 V by the front panel control RV2.

## The Click Generator

Fig. 5 shows the circuit of the click generator. With no input at all, Q4 is off and the circuit is inoperative. However with -0.3 V coming in from the receiver, Q4 starts to conduct very slightly and the circuit starts to click slowly. The clicks rapidly turn into a high pitched whistle as the search coil approaches a coin.

## Construction

The circuit is built on a single PCB and the components should be mounted according to the component overlay in Fig. 6. The usual precautions should be taken with the ICM7555s as these are CMOS devices. You need to keep yourself earthed when handling these chips.

Once all the components have been mounted on the PCB, the board can be drilled in the four
corners. The board is held firm in a plastic control box by four nylon cuts and bolts. Terminal pins were used on the PCB for external connections to the switches, potentiometers, sockets and battery connections.

Drill the required holes in the plastic control box. You will probably have to do a little additional filing for the volume, click control pots and the audio socket.

To make the search coils, first obtain a piece of scrap 25 mm chipboard and hammer into it a 200 mm diameter circle of nails, wind 30 turns of no 26 swg enamelled copper wire around the nails and secure the windings with string or cotton ties. Pull out a few nails, remove the coil and then wind a second coil. Then mount the coils, overlapping by 100 mm as in Fig. 7 on a suitable piece of 6 mm plywood and fasten them down with plastic cable clips and plastic screws. Connect the coils up to a few feet of 3-core cable terminated at the other end in 4 mm plugs. Alternatively you could use 2-core screened audio cable and use the screen for the common connection.

At this stage you would be advised to bench test the machine to check that you have wound the coils


Fig. 5 Diagram of click generator
correctly so that the current in each coil flows in the same circular direction. A method of testing the phasing or current direction in each coil, apart from inspection, would be to pass a small direct current through each coil and then detect the magnetic field produced with a small compass. The coils would need to be placed in the vertical plane with the compass positioned at the centre of each ring. If the currents are in the same direction, the compass will indicate that this is so.

## The Printed Circuit Board

Fig. 6 shows the component overlay. Make sure the components are placed in the correct positions. Once the $165 \mu$ s pulse has finished, the reservoir capacitor C 1 starts to charge up with a large current. This causes a voltage drop in the wiring. If any voltage drop gets on to the earth rail, it will be amplified and interfere with the system operation. For this reason separate wiring for the two battery supplies must be used and nothing but the battery may be connected to the left of C1.

## The Coils

Fig. 7 gives the details of the coil assembly. Mount the coils on a plywood frame and cut away as much wood as possible to reduce the weight. A few feet of 3 -core mains cable is suitable for connecting the coil assembly to the 4 mm sockets on the plastic control box. Everything must be plastic or wood. Finally keep in mind that the current in each coil is flowing in the same direction ie they are driven in phase.

## Batteries

Eight 1.2V AA size rechargeable cells provide the -10 V supply. The machine consumes around 80 mA of current so the batteries will give about five hours of continuous running. When the batteries are discharged, the click generator will go out of control. A 9 P PP3 or MNI604 battery provides the positive supply for the op-amps. A voltage converter is not used to obtain this supply as these devices require an oscillator, the output of which might get into the
receiver and cause interference. All the batteries are mounted inside the lid of the plastlc control box and secured with strong rubber bands.

Then encapsulate the coils with Araldite and put the assembly into a warming compartment so that the Araldite melts and permeates into the windings before setting. Use plastic angle material to attach the assembly to a plastic or wooden stem. No metal should be used in the construction of the coil assembly. Any metal nuts, screws, washers or solder tags will upset the system.

An 80 cm length of 20 mm plastic tubing may be used to make the handle for the control box and can be bent into the traditional 'shepherd's crook' shape by means of a bending spring and hot water. A bicycle handlebar grip slipped on to the top end makes an ideal handle hold.

A 50 cm straight length of 16 mm plastic tubing can be used for the stem. One end was dipped in hot water and flattened with pliers and then attached to the coil assembly by means of a plastic nut and bolt. The stem is then slid up into the handle until the total length suits the operator and then bolted into position.



Fig. 6 Component overlay for the detector and off-board wiring


Alternatively one could use a wooden walking stick or adapt whatever non metallic material one has to hand. The only metal materials permitted are a few screws in the control box and the two screws securing the control box to the handle. Finally, insert a rubber washer between stem and coil assembly. This gives a non slip attachment to stop the search head angle being moved by rough grass.

## Testing

The initial testing should be done in a metal free environment. Most work benches and tables contain large numbers of nails, screws and brackets so the reader is advised to suspend the coil assembly from the ceiling on a length of string to ensure that it is well clear of metal. With the click generator set to one click per second the operator will notice a significant increase in the click rate if a two pence coin is taken to a distance of 180 mm from the search coil.

Once small pieces of metal have been located with the general purpose search coil, the final pinpointing can be carried out with a snout probe shown in Fig 7b and in the above photograph. This probe was constructed in a similar manner to the general purpose coil expect that the coils do not overlap. Each coil is made from 48 turns of 30 swg enamelled copper wire making the loops 50 mm in diameter and 70 mm between centres.

## HOW IT WORKS

The operation is asfollows. The two switches in the transmitterclose simultaneously for $165 \mu$ s and allow a current of one amp to flow through each coil. This operation is repeated every 10 ms la frequency of 100 Hzl . The coin signals picked up by the coils along with the interference and ground effect are then routed to the op-amp $A$ in the receiver (Fig. 1). Here the interference and ground effect cancelout and the amplified coin signals are passed on to the detector $D$. Detector $D$ is switched on by the timing circuit $36 \mu s$ after the end of the current pulse and for a duration of $50 \mu \mathrm{~s}$. The $\mu \mathrm{s}$ delay is to allow the coils to settle down because the sudden loss of the currentcauses a verylarge voltage spike to appear across each coil. The DC output of the detector now goes to the click generator which starts to click rapidly as the search coil approaches a coin.

PARTS LIST

| RESISTORS (all $1 / \mathrm{W}$ 5\%) |  | SEMICONDUCTORS |  |
| :---: | :---: | :---: | :---: |
| R1,2,18,22,24 | 47k | IC1,3,4,9 | ICM7555IPA |
| R3,12 | 4k7 | IC2 | NE555 |
| $R 4$ | 15k | IC5 | $\mu \mathrm{A} 709 \mathrm{CP}$ |
| R5,8 | 680R | IC6 | TL081 |
| R6,7 | 150R | 107 | 78L05 |
| R9,11 | 68k | 108 | 79105 |
| R10 | 3 k 3 | 01,2 | TIP31A |
| R13,14 | 470R | 03 | 2N3819 |
| R15 | 470k | 04 | BC178 |
| R16 | 390k | D1.5 | IN4148 |
| R17 | 100k |  |  |
| $R 19$ | 180k | MISCELLANEOUS |  |
| R20 | 220R | BATT1 | $8 \times 1.2 \mathrm{~V}$ AA rechargeable batteries |
| R21 | 1k0 | BATT2 | 1x9V PP9 battery |
| R23 | 1M5 | PLI-3 | 4 mm plugs: 2 red, 1 black |
| R25 | 18k | PL4 | 2.5 mm mono jack plug |
| R26,27 | 2k2 | Sk1-3 | 4 mm sockets: 2 red, 1 black |
| R28 | 180R | SK4 | mono 2.5 mm chassis jack socket |
| RV1 | 100k horiz preset | SW1 | double pole, double throw switch |
| RV2 | 47 k lin | Case Enamelled copper wire, 28swg and 30 swg. Plastictubing, 16 mm |  |
| RV3 | 4 k 7 lin | and 20 mm | ywood. Plastic angle. Cablegrips. Glue (Araidite). |

## CAPACITORS

$\mathrm{C} 1 \quad 2200 \mu$ axial electrolytic
C2,15,17 $\quad 100$ n polyester 7 mm
C3 $\quad 1 \mathrm{nO}$ polyester 7 mm
C4,7,9 $\quad$ 10n polyester 7 mm
C5,10,14 $\quad 22 \mu 16 \mathrm{~V}$ tant bead
C6,8 220 p 63 V ceramic
C11 30363 V ceramic
C12 10p 63 V ceramic
C13 $\quad 470 \mathrm{n}$ polyester 7 mm
C16

## BUYLINES

$$
\begin{aligned}
& \text { You should have no problem in obtaining components as they are all } \\
& \text { readily available. The enamelled copper wire is available from Maplin } \\
& \text { who can also supply the } 709 \text { op-amp. } \\
& \text { The piastic control box in the prototype was a Vero-box type } \\
& 202-21031 \text { but any box of equivaient size will do. A variety of plastic } \\
& \text { tubingis available in mostDIV stores but must be rigid to make a good } \\
& \text { handle. }
\end{aligned}
$$

## The Archer Z80 8BC

The SDS ARCHER - The $Z 80$ based single board computer chosen by professionals and OEM users $\star$ Top quality board with 4 parallel and 2 serial ports, counter-timers, power-fail interrupt, watchdog timer, EPROM \& battery backed RAM.
$\star$ OPTIONS: on board power supply, smart case, ROMable BASIC, Debug Monitor, wide range of I/O \& memory extension cards.
CIRCLE NO. 122 ON REPLY CARD

## The Bowman 68000 sBC

The SDS BOWMAN - The 68000 based single board computer for advanced high speed applications.
$\star$ Extended double Eurocard with 2 parallel \& 2 serial ports, battery backed CMOS RAM, EPROM, 2 coưntertimers, watchdog timer, powerfail interrupt, \& an optional zero wait state half megabyte D-RAM.
$\star$ Extended width versions with on board power supply and case.
CIRCLIENO. 147 ON REPLY CARD


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# CAMERA CONTROLLER 

Keith Brindley builds a snappy project for beginners to turn your camera into a sophisticated remotetriggered device


To start we must make it clear that this project is not in itself a particularly useful addition to your camera gadget bag. It can be used to control the camera's shutter release but its real advantage will be found later. You see, the ETI Camera Controller acts as a multi-function, electronic shutter release interface which, in the future, can be used with other projects which allow the camera to be remotely operated in a wide variety of ways.


Fig. 1 Block diagram of the camera controller
The later projects (all 1st Class) will include:

- a light-beam trigger - when someone or something interrupts an infra-red beam of light, the camera will be triggered. Ideal for candid shots of, say, birds entering or leaving a bird-house or surprise shots of intruders breaking into guarded premises
- a sound-operated trigger - to trigger the camera on the detection of a sound
- a remote trigger - to operate the camera from a distance for, say, self-portraits
- a fixed-interval timer trigger - allowing photographs to be taken at a number of fixed times, to show, say, a flower head opening up in stages through the day
- a variable-exposure time trigger - allowing the camera to be triggered for longer periods than are available on the camera's built-in exposure time mechanism
The Camera Controller links to your camera with a standard cable release, operating the cable release (with the help of a low-voltage solenoid) to trigger the camera on reception of an electronic pulse. So, quite simply, the project interfaces the mechanical shutter
release of the camera to the electronics world. The camera then, effectively, has an electronicallyoperated shutter release. With the addition of one or more of the later projects listed here it becomes quite a sophisticated photographic tool.

The design is for battery-powered operation - if you're going to use such a device to trigger your camera outside it's hardly likely you'll have a mains power point at the camera's position. So the solenoid used is of lowvoltage ( 12 V ) operation. However, the power required to operate a cable release and a camera's shutter is quite large so the solenoid is a beefy one, which requires a fair current (about 1A). Your common-or-garden dry cell isn'tgoing to be able to provide the current needed. Instead, you should use two, PP3-sized NiCd batteries. These have two advantages. They are rechargeable and their operating voltage $(8.4 \mathrm{~V}, 16.8 \mathrm{~V}$ in series) is within the maximum 18 V limit for the integrated circuit used - a CMOS device. Alkaline batteries could provide a voltage of exactly 18 V - too close for comfort.

Observant readers will note the use of a cheap Darlington pair transistor to ensure a low source current requirement from the CMOS chip, while providing adequate drive current for the solenoid.

## Construction

Circuit of the ETI Camera Controller is shown in Fig. 2. As usual in 1st Class projects, a choice of

## HOW IT WORKS

Figure 1 shows a block diagram of the ETI Camera Controller. A bistable multivibrator (that is, a flip-flop) is used as an interface to incoming electronic shoot pulses. These pulses set (or flip) the bistable into the triggered state. Once set, the bistable cannot be retriggered until it has been reset (flopped!).

Once set, the bistable output triggers a monostable multivibrator, with an on period of about one second. The monostable output controls a power switch which in turn controls alow-voltage solenoid. In addition to this automatic mode, the power switch can be controlled manually.

The circuit is shown in Fig. 2. Gates ICla,b form the bistable multivibrator which is set by incoming shoot pulses and reset by incoming reset pulses. A LED is used to show when the bistable is set and to warn the user it must be reset before it will trigger the solenoid.

Gates IC1c,d form the monostable multivibrator which is triggered by the output of the bistable. Its on state is determined by the values of capacitor C 5 and resistor R8, following the approximate relationship:

Time $=0.8 \times C 5 \times R 8$
which, for the given component values, is as near as dammit 0.8 seconds.

Output of the monostable drives the power switch, formed by power Darlington transistor Q1. This directly drives the solenoid. Diode D 3 is incorporated in parallel with the solenoid, to prevent the back EMF generated when the solenoid is de-energised from damaging the transistor. This is effectively shorted to the positive power supply rail through the diode.

Manual override of the power switch is provided by resistor R10, which when connected to the positive supply rail, creates a secondary base current to transistor 01 cancelling all effects of the preceding circuit.


Fig. 2 Circuit diagram of the camera controller
construction techniques is offered: printed circuit board or stripboard. Both methods are straightforward and apart from a few points are more-or-less self-explanatory.

On PCB, construction doesn't need to follow any particular order, although it's probably best to leave the integrated circuit till last. PCB layout, component overlay and wiring details are shown in Fig. 3. If you aren't too sure, insert components following an imaginarily logical order of complexity. That is, insert and solder passive components first (resistors then capacitors), simple semiconductors second (diodes then the transistor) and the complex semiconductor (the integrated circuit) last.

Although a socket was used in the prototype to mount the IC, it's by no means essential, and 4011s aren't too pricey anyway, so it can be soldered directly into the PCB without worry. Nevertheless, if you're not using a socket, go easy on the heat. Solder one pin then leave the IC to cool before moving on to solder the next pin.

On stripboard, it's probably best to follow this order of inserting and soldering components pretty rigidly. The stripboard layout, component overlay and wiring details are shown in Fig. 4. Before you start, however, make all copper track cuts then insert and solder all wire links. It's far easier doing all these fiddly

PARTS LIST



Fig. 3 The PCB component overlay


Fig. 4 The stripboard track cuts and component overlay

things before components are mounted.
On either PCB or stripboard, it's a good idea (though, again, by no means essential) to use circuit board pins where all off-board wire connections are to be made. Their use means it is simple to make a connection, particularly so after the board has been fastened down.

Whichever construction method you choose, check that no solder links or bridges are present between components or IC pins.

Although we've offered no suggestions for housing your completed project, any suitably-sized box can be used. It's here, however, that the interface between the project and the cable release takes place, using a solenoid. So this is where readers may have to use a little ingenuity to ensure the solenoid operates the release satisfactorily.

A possible method of linking the solenoid to the cable release is shown in Fig. 5. As the solenoid is energised, its plunger must push in the cable release's operating pin. The method shown couples the cable


Fig. 5 Attaching the solenoid to the cable release
release's operating pin directly to the solenoid's plunger giving a pretty 'solid' and robust interface.

## Setting Up

There's not a lot to do here. Test the project initially unconnected to the camera. When power is first applied, the LED may light. If so, connect between the two reset terminals (using, say, a screwdriver) and the LED will go out. Now, connect between the shoot terminals - the solenoid should energise for a short while (about one second) then stop. The LED should now be lit. Further attempts to connect the shoot terminals should not operate the solenoid. Resetting the project, however, by connecting the reset terminals will allow the shoot terminals' connection to energise the solenoid again.

It should be noted, here, that although we've been talking about connecting between terminals to provide shoot and reset triggers - by touching the terminals with a screwdriver - there's nothing to stop switches being used. Also, for future use, these triggers will be generated elecronically by other circuits - negativegoing pulses will trigger both shoot and reset.

The shoot-then-reset operation can be effectively over-ridden by connecting the manual terminals the solenoid should energise for as long as the manual terminals are connected.

## BUYLINES

The solenoid is the only difficult bit and it's available from Electromail as part 349-709. For a suitable cable release for your camera, call in at your local camera shop!

## C.A.D. SOFTWARE MADE EASY

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# A PRESENT FOR GRANNY 

0ne of the most common problems that elderly people have to put up with is a gradual loss of hearing sensitivity. Unfortunately, this problem is made more irritating by the way that the rest of us tend to assume that they are daft as well - when they simply didn't hear what we said.

The NHS does its best to help but they have only a limited range of appliances which they can dispense. These are mainly small units which sit on top of the ear, with a microphone which points backwards fine for hearing what people are saying behind one's back but not as useful for hearing what the person standing in front of one is saying.

The NHS units, being small, are also liable to get trodden on, mislaid or dropped into unlikely places, and are dear (hundreds of pounds-ish) to replace when lost.

However, from the point of view of the DIY electronics fraternity, the task of making a gadget which will amplify the output from a small microphone and will operate an earphone is neither difficult nor costly, provided that one doesn't aim to make it too miniaturised. This, in a sense, can be an advantage since if it isn't too small, it won't be so easily lost.

A word of warning is necessary at this point. The manufacture and supply of 'hearing aids' is a very profitable business - especially for the private enterprise side of the market. It is therefore, quite predictably, hedged around with official restrictions to prevent others from homing in on this market. So, no-one other than an officially approved manufacturer can market such devices as 'hearing aids', and no-one but a registered 'hearing aid' dispenser can sell them.

This does not restrict what one can do oneself if one has a mind to but it must be called something other than a 'hearing aid"! The few firms which exist in this country outside the patronage of the NHS refer to their products as 'binaural amplifiers' or 'microphone amplifiers'. Again, I suppose one can call it what one likes so long as it isn't sold.


## Design Considerations

There is quite a range of small 'electret' condenser-type microphones available, nearly all of which contain an integral FET buffer amplifier, of the general form sketched in Fig. 1. These require a small DC voltage in the range $2-15 \mathrm{~V}$ to make them work.


Fig. 1 Internal connections to an electret microphone

Some of these are ' 3 -terminal' devices, with the FET load resistance inside the unit but most require the load resistor to be added externally, as shown in the diagram. Either of these types are usable.

There is also quite a range of small headphones on sale, for use with 'Walkman' type personal cassette players or personal radios. These are typically 30R impedance and require a few milliamps of drive. Check these before you buy, since some of the very cheap ones are too insensitive to be of much use.


Fig. 2 The 2-transistor gain block with gain control



Fig. 3 The headphone driver amplifier


Fig. 4 Automatic volume control system
Between the microphone and the earpiece one must insert an amplifier giving an adjustable gain somewhere in the range $30-500$ which will operate from 3V DC (the electret microphone requirements rule out 1.5 V operation, as does the need to produce enough volume from a 30R earphone) and which won't take too much current.

Finally, it has to fit inside some fairly strong metal

container to give electrostatic screening and protection, such as the little hinged lid tins in which, in more expensive days, tobacco and cough sweets were sold and which are now largely to be found in the workshop or mending box, full of screws or safety pins.

## Circuit Design

The basic 2 -transistor gain block of Fig. 2 will operate quite happily down to about 1.5 V and will provide a peak-to-peak output voltage swing which is close to the limits set by the supply lines. It also provides a convenient means of adjusting the gain, by varying the amount of negative feedback. R1 and R2 are chosen to set the DC output level at about half the supply line voltage.

The following stages Q3-Q11, shown in Fig. 3, comprise a miniaturised hi-fi amplifier with a pushpull output operating on a 1 mA quiescent current setting, giving a total quiescent battery demand of 2 mA .

The 'bootstrapped' driver load (C9/R16), though now not used much in true hi-fi designs where a constant current source load for the driver stage is preferred, makes the best use of the limited supply voltage. Prior to clipping, with a 30 R load and 3 V supply, the overall THD is of the order of $0.02 \%$.

## Automatic Gain

This is a facility which lifts this circuit out of the ordinary 'run of the mill' microphone amplifiers and greatly adds to its usefulness. Unfortunately, with the very limited available supply voltages and output swings, most of the normal AGC systems are inapplicable.

The circuit of this is shown in Fig. 4, and it operates by using the dynamic impedance of a small signal MOSFET device as part of the feedback loop of the amplifier to Fig. 3. When the MOSFET device (in this case a VN1210 but any similar enhancementmode MOSFET will work as well) is conducting, it has a dynamic drain-source impedance of only a few ohms and the gain of the amplifier of Fig. 3 is $22 \mathrm{k} / 470$ $=47$.

In the normal quiescent state, the MOSFET is biased into conduction by the voltage developed across RV3, at a forward gate-source potential of about $1-1.5 \mathrm{~V}$. The AGV action arises because the output signal voltage from the amplifier is then rectified by the diode-pump circuit of D1, D2, C11 and C 13 , to provide a negative-going output voltage which progressively biases off the MOSFET and causes its dynamic impedance to rise.

As the MOSFET becomes an open-circuit, the gain of the output amplifier drops towards unity since the load resistor for the MOSFET is also bootstrapped by C15 and R21

Sadly, this kind of AGC wouldn't be useful in true hi-fi applications since, when it is operating with the MOSFET somewhere between fully on and cut-off, it worsens the THD from about $0.02 \%$ to about $0.5 \%$, mostly second harmonic. At small signal levels or large, ones, when the MOSFET is either fully on or cut-off, the THD reverts to the lower figure.

This circuit will hold the output constant over at least a 100:1 input signal range and will do a lot to compensate for the wide range of ambient sound levels which exist in practice.

## Construction

The complete circuit of the unit is shown in Fig. 5. A suitable PCB layout is shown in Fig. 6. The mic should be mounted at one end of the case in a small rubber grommet to give some measure of sound isolation

PARTS LIST


## 



Fig. 5 The complete circuit diagram
from the case. The 3 mm phono jack for the headphone is mounted at the other end. A small preset pot is mounted on the hinged lid of the tin to allow an easily accessible adjustment to the sensitivity of the unit.

Power is provided by a pair of AAA sized batteries in small battery holders and should give a life of some hundreds of hours before battery replacement is needed. Ideally, the works would have been held in place by screws and stand-off spacers but on the prototypes the relative shallowness of the tin did not give adequate height, so the PCB was held in place by a couple of small strips of double-sided adhesive foam, after the copper side of the PCB was liberally covered with insulating tape to prevent any shortcircuits from the protruding solder blobs to the tin can.

Setting up is simple. Set RV1 and RV3 to maximum and RV2 to minimum resistances and then adjust $R V 2$ to give a total supply current of 2 mA . Set the gain (RV1) to a level at which there is an adequate level of loudness on some sound (such as a radio) and gradually reduce RV3 from its maximum resistance value until the output sound level just begins to fall.

With devices of this type, it is undesirable that the LF response should be too good since most LF sounds are just noise or room resonances and add nothing to the information content. The LF gain can be reduced further, if needed, by reducing the values of C4, C7 or C9. In the prototypes, I mainly used
tantalum bead types for all the larger values in the interest of compactness but this is at the constructor's choice, depending on how much space there is available.

## $E 1$



Fig. 6 The component overlay

John Brockhurst explains the key to converted keyboards

# HOW TO MIDI A PIANO 



0ver recent years there have been considerable advances in the level of technology available to the working musician. There are however still occasions when that favourite old keyboard or piano sound would still be best suited to the music but the absence of MIDI means either having no synthesiser part or working as a one handed pianist. Lack of adequate funds is a saddeningly common problem that precludes the purchase of one of the latest fully-equipped keyboards.

The project presented here is the solution to the problem. A stand-alone MIDI processor tailored to the needs of the pianist that is both inexpensive and relatively undemanding from a constructional point of view. It handles up to 88 notes - that's over seven octaves, certainly enough for any keyboards and about right for most pianos.

The unit consists of three modules: the CPU/PSU section, the control/display panel and the keyboard contact assembly. The first two will be common to all constructors. The keyboard contact arrangement however will need a certain amount of customising to whatever instrument it is fitting obviously no single design could be presented to fit all pianos, synths and stylophones. The photographs and illustrations shown here use the example of a Fender Rhodes piano and will be helpful as a guide.

Please note that tt is much easier to produce MIDI note information without the velocity information. This system does handle this velocity information but it should be noted that such a system does require a reasonable level of mechanical competance.

Much has been written in these and other pages over recent years regarding the MIDI specification and those who are unfamiliar with the techniques involved would probably benefit by reading through those ETI back issues (you do keep them don't you?). Also recommended in this respect is The ETI Guide to

Making Music from last autumn, which also looks at the subject in some detail.

## Processor Prelude

Okay so where do we start? Clearly we will need some means of recognising which keys are being played and how hard. We also need to know if the sustain pedal is being used. Realising that not all synthesisers are equipped with a sustain function, we must find a means of providing this. It would also be nice if we can select the sound that the synth produces, and provide visual indication of the selected patch and MIDI channel in use.

We then have to set up a serial link conforming to the MIDI protocol. Clearly this is a task well suited to a microprocessor and this design uses the Z80A.

Much has been said about choosing microprocessors and it is worth mentioning the reasons for choosing the Z80A for this application. Most of the time, our processor will be scanning the keyboard to see which keys are being used and how hard they are being played (how long the key depression takes). This is a repetitive task with speed being of prime concern. However, fast processors are relatively more expensive and demand the use of fast peripheral


A look at the specification sheet for a typical microprocessor shows that operations completed within the chip are usually much faster than those requiring access to external devices. The 280 has 14 internal general purpose registers available to the programmer, enabling much of the work to be completed within the CPU. It is also one of the cheapest processors currently available and would appear quite well suited to the job.

By opting for the Z80A which runs at the increased speed of 4 MHz we are able to allow ourselves the luxury of adding wait states to memory cycles, relieving the speed constraints on these chips and further reducing cost. '

The $\mathrm{Z80}$ also automatically includes wait states in all input and output instructions, thus placing lower demands on our peripheral chips.

Having decided on our processor we will need some memory, an ACIA and some ports to communicate with the outside world. This design uses the 6850 ACIA and 8255 PPI in these positions.

Finally, we have to decide if we will use interrupt driven timing routines for the keyboard scanning or rely on software techniques. We need to know the time taken to depress a key to determine how loud a note is. To achieve this, we must examine the key at regular intervals keeping these intervals as short as possible. The interrupt technique would certainly keep the timing constant but if we are to avoid the interrupt service routine being called whilst it is still being executed, the period between interrupts must be somewhat longer than the time required by the routine and this results in a reduction in speed.

This design uses a different approach. Since the hardware will handle 88 keys, the time spent examining any key is just over one percent of the total time - if we include the other routines the processor has to cope with, this figure falls quite a lot further. By keeping the time spent processing a key which is in use as short as possible, the difference in time required to process an active key is very nearly the same as for a dormantone and makes no difference to the subjective response of the keyboard.

## Construction

The first decision to be taken is how the modules are going to be fitted to the keyboard, with particular reference to the keyboard contact assembly. The processor will handle keyboards with up to 88 notes and if the recipient keyboard is longer than this it will be necessary to decide which keys are to be excluded.


Fig. 1 Cross section of a piano showing the contact arrangment
Bear in mind that some synths or expanders only work within a limited range and either transpose or ignore notes outside these limits. A further point to consider here is that the software provided expects the lowest keyboard note to be E. Although facility is provided within the software to transpose the keyboard, if the power supply to the keyboard is interrupted the processor will default to the E setting. Any unused contacts need only be ignored, so provided there are sufficient spare positions available it would make sense for the unit to power up in tune!

Next a point must be found where there is sufficient clearance to accommodate the contact assembly. Ideally this would be either under the front of the keys, or above and behind the key pivot as in Fig. 1, at a point where the key travel is about 3 or 4 mm . A layout for a keyboard PCB is provided in Fig. 2 but as the spacing between notes is not always the same from keyboard to keyboard and the errors are cumulative along the length of the instrument, it might be preferable to use stripboard. The PCB allows 164 mm per octave.

The recommended contact construction (Fig. 3) uses bus-bars and contact wires since this is both




Fig. 3 Detail showing contact wire and bus-bar assembly
cheap and reliable and has the added advantage of being easily adjusted. However it might be possible to use single pole changeover switches provided these are not of the kind with tactile feedback (click action).

Whatever form of contact is chosen, it is important to realise that the completed contact assembly must be as rigid as possible. If the key travel is 4 mm and the contact assembly moves by 2 mm relative to the keys as a result of springing or vibration, then clearly most of the dynamics of the keyboard will be lost. A support brace made from extruded aluminium can be recommended here as quite substantial sections are easily obtained from the local hardware store and offer good support for minimum weight.

Assuming that you will be using bus-bars and contact wires, the bus-bars will need to be mounted in parallel with the PCB and cut to length to accommodate eight notes each. This requires that a large number of support blocks be made or purchased with a similar design to the one illustrated in Fig. 4 (three support blocks per key bank is suggested).

If these are to be made, it is strongly recom mended that time is invested in designing a jig to ensure that the blocks are all exactly the same since this will be reflected in the performance of the


Fig. 4 Suggested design for support block
completed unit. Such a jig need only consist of a piece of metal arranged to enable accurately positioned holes to be drilled in the support block.

The keyboard circuitry can now be assembled with the contact wires being brought out between the bus bars as shown, care being taken to check that there is sufficient spring in the wires to ensure reliable contact with the upper bus-bar when the key is at rest. Then check carefully for shorts and track continuity. The diodes can also be checked with a test mefer at this stage, as once assembled within the depths of the piano these can prove time consuming to replace. Finally, the wires can be connected to the bank select pads and the keyboard bus tracks and arranged neatly to form the wiring loom.

The next stage is to fit the contact adjustment screws to the keys. These screws will probably be about 25 mm long and should have a slotted head. 6BA brass cheese-head screws are recommended.

The keys should be drilled with a hole which will allow the screw to be threaded in or out of the key as required. Again, care should be taken to keep misalignment to a minimum.

The screws should be rotated until their head slots are parallel to the contact assembly fitted to the keyboard with the contact wires being laid through the

## HOW IT WORKS

The Processor circuit diagram is shown in Fig. 5.
ICla and IC1b form a crystal 0 oscillator running at 4 MHz and this is buffered by IClc for use as the system clock. This signal is also. buffered byIC1c and thendivided by 8 in $1 C 4$ toprovide a 500 KHz clock for the ACIA iC11, where itis internally divided again by 16 to give the 31.125kHz MIDIClock.

IC3 and IC2C are used to insert wait states whilst the processor is accessing memory to ensure that slow devices will work without difficulty. $1 C 2$ a in conjunction with $C 4, R 5$ and $D 5$ provides a positive goingreset pulse at power-up to treset the outputports 1 C12 This pulse is inverted by $1 C 2 b$ to reset the processor.

With the exception of I 6 c whichserves to buffer the outputfiom IC11, all the remaining gates perform the address decoding for the processor: At firstglanceitmightseemthat|C7ddoesnothingbutthis gate is essential to meet the timing requirements for IC12.

The port outputs of CC 12 are used to connect the keyboard and control panel to the processor, PA and PC being configured as outputs and PB as input. Output ports are only rated to source 1 mA or sink 1.6mA and this means that the circuits connected here should have a high impedance.

Unfortunately, because of the necessarily long leads required to connect thekeyboardcontactassembly, this wouldmean bothreduced noise immunity and slowerrise times for the keyboard scanning signals, particularly truein the case of PC 3 and PC4 which switch betweenthe upper andlower bus-bars and see all 11 contact banks in parallel.. This problemis isemoved by buffering thesedata lines and this is the purpose of01 and Q2. Resistors R9 and R10 in the MIDloutputcircuitserve to protect the unit from failure in the event of a short arising in the MIDI lead.

Withreference to the circuitdiagram for the control panel (Fig. 6), PC1 and PC2 are used asenable signals for latches IC2 and IC4. IC 1 is

## aBCD to decimal decoder which is used todecode the address output

 by the microprocessor on PAO to PA4 and held in IC2.Daringtondriver array IC3 provides the esegmentdrive current sink for the displays LEDI to LED4. 01 to 06 provide the corresponding source currents for the display and the switch array. Diodes D1 to D8 prevent the data outputbus PB being shorted in the event of more than one switch being pressed at any one time.

D6 and R16 (which are mountedon the CPUpanell and the circuit around 07 serve to cancel the effect of capacitance in the lead connecting the sustainfoot switch tothe processor which, if mounted externally, will probably be several feet long. When the processor switches 01 on via $P A$, the capacitance in this lead is seen in parallel with the switch and because of the high frequencies involvedpull $P 87$ high indicating that the switch is closed. However, 01 also turns 07 on via $R 20$ taking $P B 7$ low via $R 23$ and $D 9$ cancellingthis effect. When the switch really is closed $D 6$ allows 01 to pull $P B 7$ high and signal the processor accordingly.

In the circuitdiagram of the keyboard contact assembly (Fig. 7), the key contacts are arranged in groups of eight and read into the processor via PB.

The circuitcomprising D1 to D12 isrepeated everyeightnotes as many times as necessary to cover the length of the keyboard. As we are examining upto 88 notes in groups of eight, we could have 22 busbars. To avoid the expense of anotherport these busses arediode AND gated onto PB. In order for a key to pull the data bus PB high we need simultaneous logic high signals on both on the bank select line PA1-7 or PC5-7 and on U or D which decide whether the key is being tested to bein the upperor lower reststate, allowingthe bus to be pulled high by R1 or R2. Anykey which does not pull the bus high is deemed to be in use, but as unused positions never reach a high level false MIDI triggering can not occur.
slots of the screw heads ensuring precise control of the contact both vertically and laterally. A piece of thin stiff wire with a small hook formed in the end can be invaluable here to lift the contact wires in or out of the slot in the screw head whilst the screw is being adjusted.

A correctly adjusted contact wire should leave the upper bus-bar when the key is about $15 \%$ depressed and should make contact with the lower bus-bar as the key nears the end of its travel. This final adjustment is most easily checked when the unit is fully assembled using a synth sound with a short attack and decay envelope as the contact points can then be accurately assessed.

The effect of any dimensional errors in the busbar support blocks will by now be apparent and any unacceptable errors can be rectified. When you are quite sure that the contacts are correctly adjusted, a small application of contact adhesive to the screw head will ensure that the wire does not come adrift.

The processor and power supply are assembled on a double sided PCB (Fig. 8). There are several links between the upper and lower PCB tracks and these will need to be fitted first. Some of these links are


Fig. 5 Circuit diagram for the processor


Fig. 6 Circuit diagram for the control panel


Fig. 7 Circuit diagram for the keyboard contact assembly
beneath IC sockets and it is well worth a second check to be quite sure that all of these links have been soldered correctly.

Next fit the diodes and capacitors (observing polarity), resistors, transistors, IC sockets, connectors and power supply components. The regulator should be mounted on a small heatsink but it does not lead a hard life and will not need heatsink compound.

The board can now be powered up after checking for shorts to ensure that the power supply is giving +5 V and that this is present at the respective pins of the IC sockets. If all is well, disconnect and discharge the power supply using a suitable resistor and fit the crystal and the TTL ICs. Care should be taken when fitting the crystal as the wires can short to the case if they are bent too close to the body. (This would not be good practice in any case).

## Testing

Next if you have access to an oscilloscope, there is an easy way to check the operation of the microprocessor. Push short lengths of fine tinned copper wire into pins 9 to 17 of the EPROM socket and join them all together, temporarily shorting D0-D7 to ground.

Now fit the processor observing static precautions. When the unit is powered up the processor will sequentially count through its address range and this can be seen on the scope giving an indication that the processor is working. Address lines A 0 to A 6 will have refresh addresses present but $A 7$ to $A 15$ should have clean square wave signals of reducing frequency on them. This works because the $Z 80$ op code $\& 00$ is the NOP instruction and tells the processor to do nothing. As the data bus has been shorted to ground by the wires, the prccessor reads NOPs from what it thinks is the EPROM and interprets these as program data ${ }_{*}$ The processor thus scans its entire 64 K memory map looking for something to do (poor thing) and this activity is easily seen on the scope. Any problems can be diagnosed far more easily at this stage.

Assuming all is well, remove the wires from the EPROM socket (observing static precautions as the EPROM socket is connected to the CPU by the PCB tracks) and fit the remaining ICs. The completed panel can now be mounted in the piano ready for final testing.

## SOFTWARE

A hexadecimal listing of the software for the unitis given in Listing 1 . Operation of the keyboard supported by this code is as follows:

On power up, the unit defaults to operation on MIDIchannel 1 with the selected synthpatch as bank 1 program 1 . This will beindicated on the display. At thistime, the default tuning for the keyboard is that the lowest note will be E (middle C is the 33 rd note on the keyboard).

PressingSW2 orSW3 will advance or lowerthe MIDIchannel and this will be displayed on LED1 and LED2.

Program selection on the active channel is provided by switches SW4 to SW15, giving access to the first 32 patch memories of the synthesiser beingused. These are arranged as four banks of eight, the last selected sound being displayed on LED3 and LED4.

If the MIDIchannel is changed whilst the keyboard is beingplayed, any notes which are in use will be turned off to prevent these notes latching on. Notes are not turned off when the program is changed.

When the channelischanged, the programselectionisstored and recalled when the channel is next selected. Pressing SW1 toggles between MIDImodes 1 and 3 (OMNIon POLY and OMNI off POLY) and corresponding MIDImessages are sentonall 16 channels. Thisenables all instruments to be played either individually or coupled together. In mode 1 , selection of a program will be accepted by all active instruments capable of receiving program change data although the
keyboard will still be transmitting on the displayed channel. The unit does notupdate itsmemory for otherchannels as this greatly speeds access to sounds when returning to mode 3.

SW1 has a second function. When pressed simultaneously with SW4 the processor willscan the keyboard and assign the firstkey press it encounters to be middle C. Whilst it is primarily intended that this transpose function should enable the synth to be in the same key as the piano, thickening effects can be obtained when transposing by an octaveorafitth. Since the software suppresses 'illegal' notes, therange of tuning is limited only by the receiving synthesiser.

To enable maximum compatability with other equipment, the software does not use 'running status', and sends genuine note off messages. Active sensing is not used.

Finally, a word about sustain. Many expanders do not have a sustain function, whilst some manufacturers use a dedicated MIDI code to access sustain routies provided within their equipment. To gain the best of both worlds, the software emulates the sustainfunction by not turning notes off until either the pedal is released or the key is restruck. Sustain will therefore be provided up to the polyphonic limit of the synth beingused, and thereafter will be dependant on the synth's own operating system.

Assembly of the control panel (Fig. 9) should present few problems. Care should be taken to avoid overheating the push switches when soldering them in as the plastic supporting the leads melts easily.

Otherwise, apart from observing correct orientation of components and careful checking for shorts, this should be a routine operation.


Fig. 8 Component overlay for the processor and power supply



PARTS LIST


## INTERCONNECTIONS

Since the positioning of the modules for this project will largely be determined by the keyboard to which they are to be fitted, it is not possible to give concise details of the interconnecting loom. Instead, here are the wiring details in tabular form.
If the connecting wires between the CPU panel and the control panel are more than about six inches long, it is recommended that $\mathrm{C} 31220_{\mu}$ 16 V ) is fitted to (or close tol the rear of the control panel with the case connected to pins 1-3 and the positiveterminal connected to pins 15-16.

Note: This table shows the data and address wiring for the control

| CPU Connector <br> Pin Number. | Control Panel <br> Connector Pin <br> Number. | ContactAssembly <br> Connection Point. |
| :---: | :---: | :---: |
| 1 Wired to | 17 |  |
| 2 Not used |  |  |
| 3 Not used |  |  |
| 4 Wired to |  |  |
| 5 Wired to |  | Bank select pad 1 |
| 6 Wired to |  | Bank select pad 2 |
| 7 Wired to |  | Bank select pad 3 |
| 8 Wired to |  | Bank select pad 4 5 |
| 9 Wired to |  | Bank select pad 6 |
| 10 Wired to |  | Bank select pad 7 |
| 11 Wired to |  | Bank select pad 8 |
| 12 Wired to |  | Bank select pad 9 |
| 13 Wired to |  | Bank select pad 10 |
| 14 Wired to |  | Bank select pad 11 |
| 15 Wired to |  | Keyboard U bus |
| 16 Wired to |  | Keyboard D Bus |
| 17 Wired to | 11 |  |
| 18 Wired to | 12 |  |
| 19 Not used |  |  |
| 20 Wired to | 15 |  |
| 21 Wired to |  |  |
| 22 Wired to |  | Keyboard D7 line |
| 23 Wired to |  | Keyboard D6 line |
| 24 Wired to |  | Keyboard D5 line |
| 25 Wired to |  | Keyboard D4 line |
| 26 Wired to |  | Keyboard D3line |
|  |  | Keyboard D2 line |

panet coming viathe keyboard contact assembly. In some cases itmay be preferable tore-route these wires directly to the CPU pane if this will give shorter or tidier wiring.

The connections to the 5-pin connector PL. 2 are as follows:

| 1 | Footswitch screen |
| :--- | :--- |
| 2 | MIDI +ve to pin 4 or MIDI socket |
| 3 | MIDI ground to pin 2 of MDI socket |
| 4 | MIDI loutput to pin 5 of MIDI socket |
| 5 | Footswitch inner. |


| CPU Connector <br> Pin Number. | Control Panel <br> Connector Pin <br> Number. | ContactAssembly <br> Connection Point. |
| :---: | :---: | :---: |
| 27 Wired to |  | Keyboard D1 line |
| 28 Wired to |  |  |
| 29 Wired to |  |  |
| 30 Wired to | 1 | Keyboard DO line |
|  | 1,2 |  |
|  | 3 Not used |  |
|  | 4 Wired to | Bank select pad 7 |
|  | 5 Wired to | Bank select pad 6 |
|  | 6 Wired to | Bank select pad 5 |
|  | 7 Wired to | Bank select pad 4 |
|  | 8 Wired to | Bank select pad 3 |
|  | 9 Wired to | Bank select pad 2 |
|  | 10 Wired to | Bank select pad 1 |
|  | 11,12 | (already connected) |
|  | 13,14 | Not used |
|  | 15 | (already connected) |
|  | 16 Wired to | Keyboard +5V line |
|  | 17 | (already connected) |
|  | 18 Wired to | Keyboard D7line |
|  | 19 Wired to | Keyboard D6 line |
|  | 20 Wired to | Keyboard D5 line |
|  | 21 Wired to | Keyboard D4 line |
|  | 22 Wired to | Keyboard D3 line |
|  | 23 Wired to | Keyboard D2 line |
|  | 24 Wired to | Keyboard D1 line |
|  | 25 Wired to | Keyboard DO line |



## BUYLINES

Most of the components are available from Maplin and other major suppliers. The ACIA 6850 is availablefrom Fannell (tel: (10532) 636311 or Trilogic (fel: (0274) 684289). The control panel IC3 (ULN2003) is available from Electromail (tel: ( 0536 ) 204555).

The author car supply pre-programmed EPROMs at $£ 15.00$. Contact J. Brockhurst, 57 Berwick Road, Rainham, EssexRM133XO.

Two alternative versions of the sot tware are available. The first is a cut-downversion allowing operation of the unit without the control panel, supplied with details of the small hardware modification
necessary. The second is anenhanced version allowing simultaneous operation on two MIDI channels with selectable cutoff points and keyboard response. This permits lavered sounds and velocity crosslading, plus control viaMDID of volume and channel selective MIDI mode.

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The moulded plastic cover for the control panel (see photographs) is available from the author for $£ 17.50$ including the printed label.


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Capacitance; 2nF-20uF Frequency: $2 \mathrm{kHz}-10 \mathrm{MHz}$ Continuity, diode. HFE. logic \& LED test.
dc volts: 200 mV - 1 kV ac volts: $200 \mathrm{mV}-750 \mathrm{~V}$ dc current: 200uA-10A ac current: 200uA-10A

Resistance: $200 \Omega-2000 \mathrm{M} \Omega$ Temperature: $200^{\circ}-750^{\circ} \mathrm{C}$ Capacitance: $2 \mathrm{nF}-20 \mathrm{uF}$ Diode. HFE \& continuity test

Resistance: $200 \Omega$-2000M $\Omega$ Continuity. diode \& HFE test Basic dc accuracy $\pm 0.5$

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MEA

[^0]
## PEDAL POWER

The background to this project is quite simple. A friend plays guitar semi-pro and uses several effects pedals. He had a problem with battery eliminators and cables cluttering his stage area and so he asked for help. The solution was equally simple. A small box fitted to the rear of the amplifier providing a 8 V feed for the effects pedals. This power feed and signal return became combined into a single multi-way cable and the power supply box evolved into the form presented here.

A basic design consideration is thaf it should fit unobtrusively into the rear of an amplifier. The unit must be compact, yet robust and so the enclosure chosen is a very sturdy aluminium extrusion that neatly houses a single $100 \times 160 \mathrm{~mm}$ Eurocard-size PCB. All the components mount onto the card and this simply slots into the housing

To ensure simple operation there are only three connections to the unit. Firstly, mains power is tapped from the amplifier, preferably after the on/off switch. A second lead carries the output to the main amplifier signal input. This is soldered to the circuit board inside the main amplifier but could be provided on a flying lead/jack plug. Finally, a multiway DIN socket provides a connection to the effects pedals.

As this is an 'add-on' in a critical position, it is vital that no compromises are made in component selection. Consequently high quality components are used throughout to ensure that the unit never becomes the 'weak link' in the chain.

The input stage uses a basic differentiation amplifier to accept the incoming signal and a voltage follower to buffer the output to the main amplifier.

The power supply is configured to supply approximately 9 V for the effects pedals. With reference to the circuit diagram in Figure 1, the power supply follows traditional linear supply practice of


Gordon Tomlinson shows how to cut the cabling on stage


## HOW IT WORKS

The signal handling circuitry is built around two OP27 operational amplifiers. These are quite expensive but you only need the lowest grade version.

The first stage is a basic differential amplifier. Marry different formats of differential amplifier exist, and the most simple single chip version was chosen. The differential amplifier works by effectively looking at the signals presented to its inputs, and it acts in one of two ways. If the signals are different it amplifies the difference by a factor determined by R4/R3 (when R4 = R6 and R3 $=$ R5). If the signals are the same then they are attenuated by the common mode rejection ratio (CMRR) of the circuit. The amount of CMRR is determined by the choice of op-amp, the auxiliary components used and circuit topology. If we had expected high levels of interierence, we could have
exploited the $90+d B$ CMRR capability of the OP27. But we would have been forcod to use closely matched (0.01\%) resistors and a trimmer for C 12 . However in our application we do not expect high levels of interference. We can happily use $1 \%$ standard resistors. With the values shown we have an overall gain of one.

A disadvantage of a single chip stage is that the inputs are presented with unequal impedances. However this is not important here.

The network R5, C13 serves as a passive low pass fitter, progressively attenaating unwanted high frequency signais. Finally the second $O P 27$ forms a simple voltage follower fits output follows its input) and this provides a low output impedance to drive into the standard amplifier.


Fig. 3 Component overlay for input amplifier and power supply

## Construction

The end cover of the housing must be drilled to accept the sockets and glands. This is the only metalwork involved unless you decide to make a small divider to keep the mains cable away from the other circuitry.

Basic board construction follows standard practice. Start with the small components and work your way through to the larger items, finishing with the transformer. When finished have a break and then recheck everything, especially components that are polarity conscious.

Wiring the unit to the outside world is quite straight forward and is just common sense as shown in Figure 2. The 9 V power supply and signal returns are taken by a single multiway cable to the five pin din connector. You may prefer something more beefy and expensive such as an XLR connector

How you interface the signals at the pedal end is up to you. Remember to keep cable lengths to a bare minimum. Two important points to note: capacitors C11 and C12 are soldered piggy back to resistors R4 and R6 and capacitors C15 and C16 are used at the pedal end of the line.

## Getting Going

If you own a scope or signal generator then testing will probably be second nature. If you do not and run into a problem, the first thing to check is the power supply. Do you have mains? Are the regulators correctly oriented? Do you have voltage at the IC supply pins? If you are using IC sockets, have you plugged them in the wrong way? It is basically a simple circuit and should not be hard to fault find.

Where and how you fix it in the amplifier is up to you. Make sure that it is secure and not too close to the 'hot bits. Do not be tempted to compromise on component quality. If the main amplifier has a capacitor at its input it will be possible to remove capacitor C14. Over to you . . .

PARTS LIST


## BUYLINES

The parts are not cheap but they are easily available. The transformer is RS/Electromail catalogue number 208-333. Tel: (0536) 204555. The alloy case is available from West Hyde Developments Ltd. Tel: 102961 20441.



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