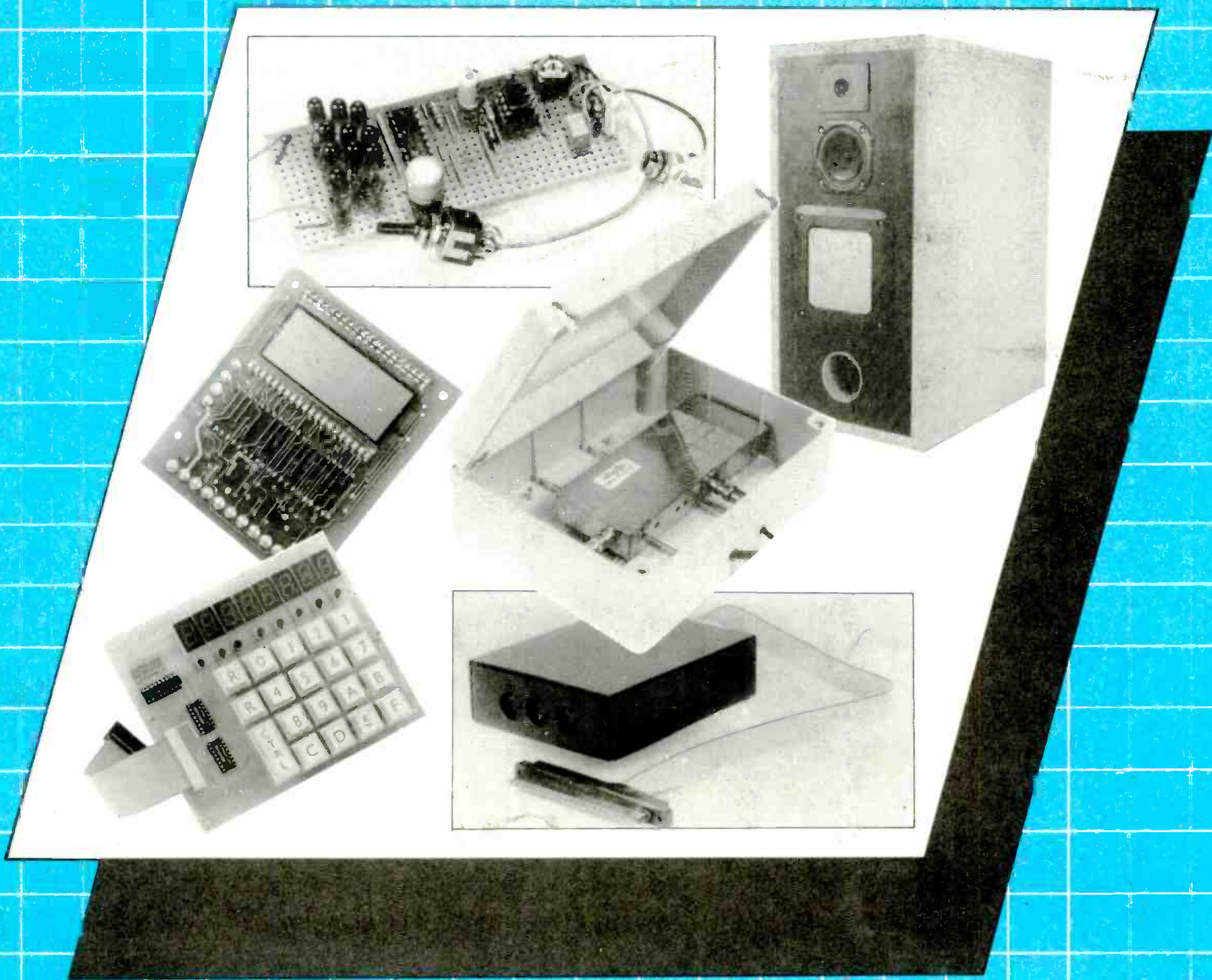
PROJECTS BOOK 22

/// Maplin /// Maplin /// Maplin



23/40 LITRE SPEAKER CABINETS
Z80 CPU HEX KEYPAD
MIDI INTERFACING TECHNIQUES
WEATHER SATELLITE DOWN CONVERTER
4½ DIGIT COUNTER
MINI-CIRCUITS

CONTENTS

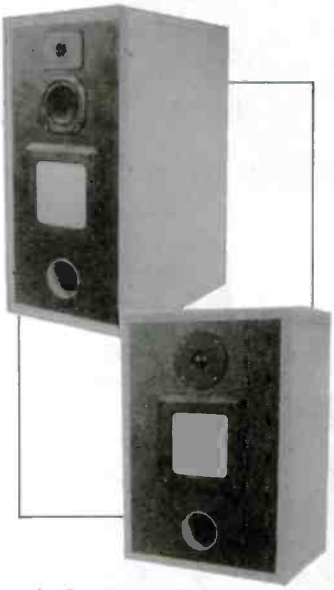
MAPLIN PROJECTS BOOK TWENTY TWO

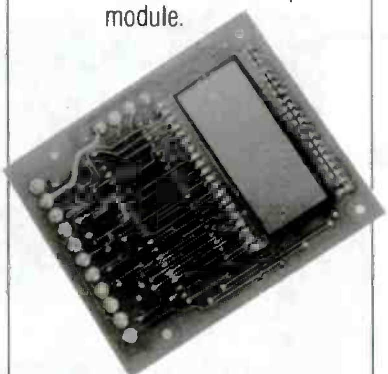
EDITORIAL

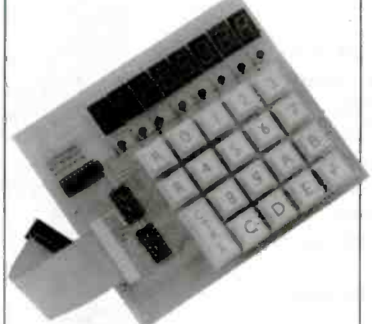
■ 'Maplin Projects Book Twenty Two is a compilation of the projects from 'Electronics – The Maplin Magazine, Issue 22', which is now out of print. Other issues of 'Electronics – The Maplin Magazine' will be replaced by projects books as they go out of print. For kit prices, please consult the latest Maplin catalogue and free price change leaflet, order as CA99H.

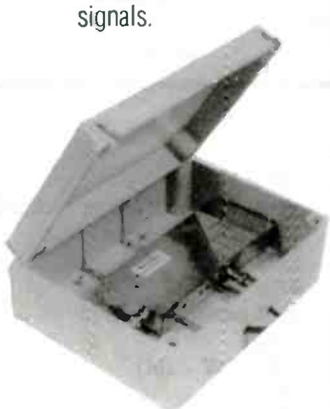
■ **Editor** Robert Ball ■ **Compiled by** Mike Holmes.
■ **Technical Editors** Dave Goodman, Chris Barlow, Gavin Cheeseman, Tony Bricknell, Alan Williamson.
■ **Technical Author** Martin Pipe.
■ **Technical Artists** John Dudley, Lesley Foster, Paul Evans, Ross Nisbet, Nicola Hull.
■ **Art Designers** Peter Blackmore, Jim Bowler.
■ **Published by** Maplin Electronics plc.
■ **Printed by:** Mayhew McCrimmon Printers Ltd.
■ **Mail Order** P.O. Box 3, Rayleigh, Essex SS6 8LR.
■ **Telephone Retail Sales:** (0702) 554161.
■ **Retail Enquiries:** (0702) 552911.
■ **Trade Sales:** (0702) 554171.
■ **Cashtel:** (0702) 552941.
■ **General:** (0702) 554155.
■ **Shops:** See below.
■ **Fax:** (0702) 553935. ■ **Telex:** 995695.

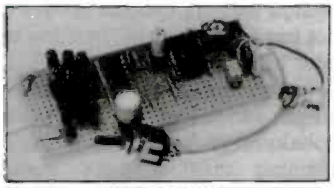
PROJECTS

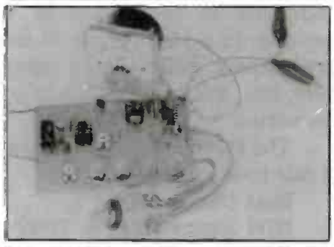


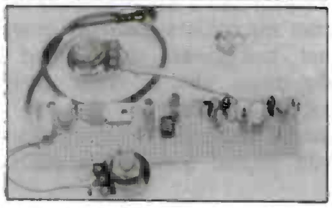


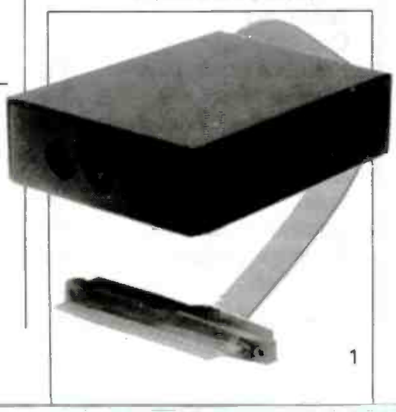












HI-FI LOUDSPEAKERS AND ENCLOSURES

by Dave Goodman

Loudspeakers generally appear to be simple devices, where electrical signals are applied to one end with sound eminating from the other! In fact their design and application is very complex - not just a question of 'bashing up a quick box' hoping it will sound O.K.

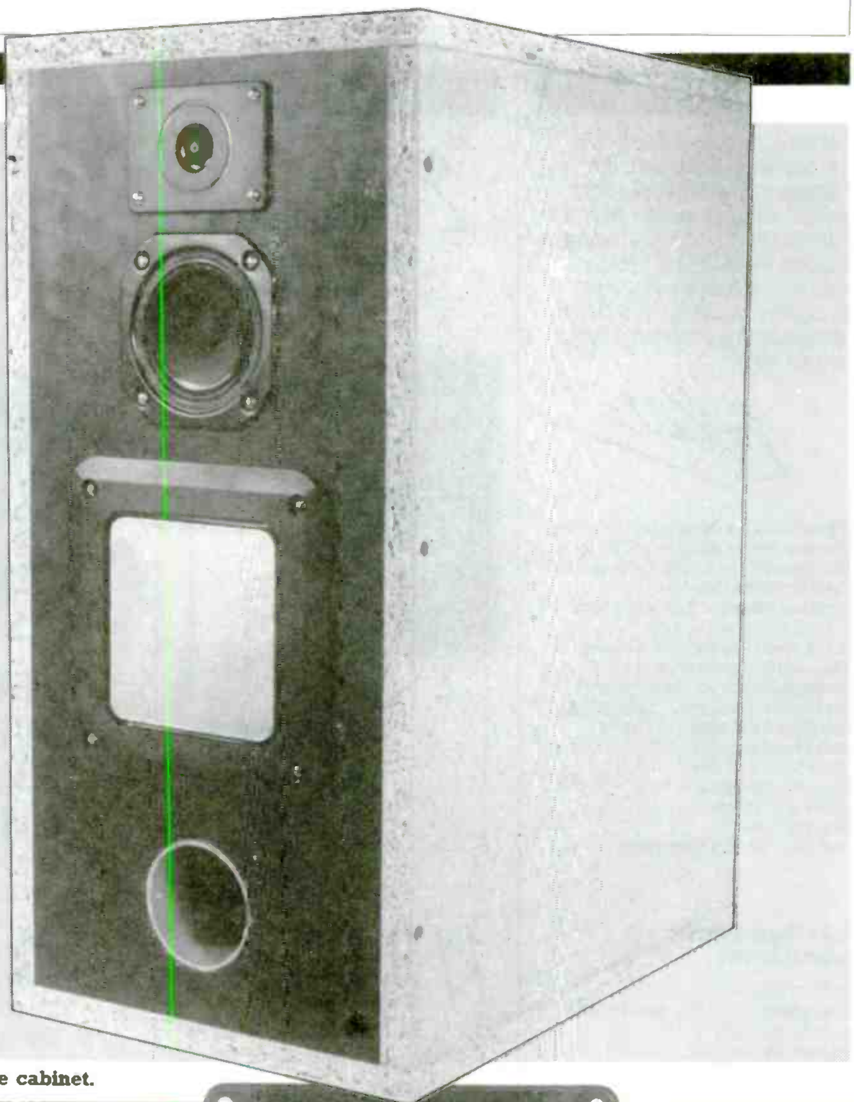
This series will be featuring the very latest range of High Fidelity Loudspeakers in the Maplin catalogue, explaining manufacturers' specifications, and how to use them in your own designs. In addition, kits in the form of pre-shaped baffles with speakers, X-over, etc., will be available to accompany the series.

The three main drive units available at this time are:

Bass Driver YN24B
50W Extended Bass YN47B
Fibreglass Driver YN25C

Each unit has a high temperature aluminium voice coil former allowing good heat dissipation with very low distortion characteristics.

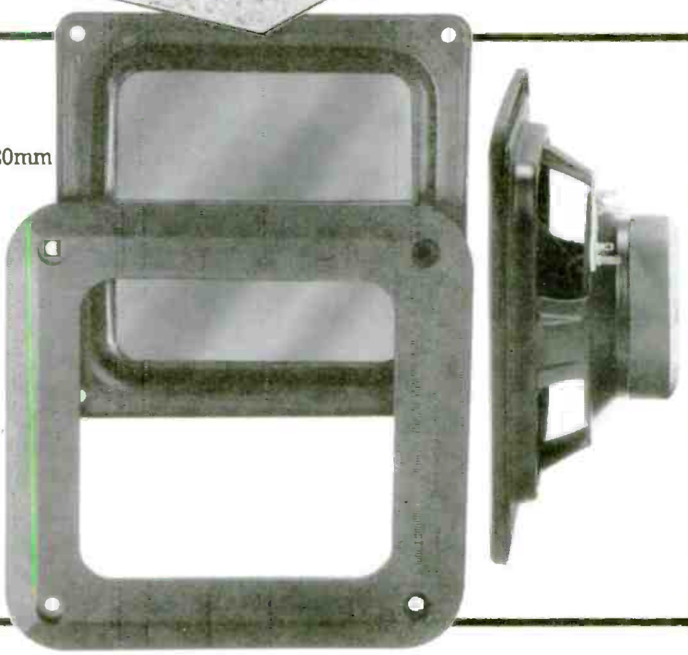
The six inch, square bass driver (YN24B) has a flat diaphram constructed from microfoam damped polymer foam that has been sandwiched between aluminium sheets. This design offers good vibration control with excellent spatial dispersion making it ideal for use with digital recordings (CD). Cabinet



40 litre cabinet.

BASE DRIVER - YN24B

Chassis		150 x 150mm
Baffle		133 x 133mm. Corner radius 20mm
Fixing		134 x 134mm. 5mm clear
Flux Density	(B)	7800 Gauss
Frequency Response		65Hz - 4kHz
Power Handling		60W peak, 30W RMS
NOM Impedance		8 ohm
Sensitivity		87dB (1W - 1M)
Coil Diameter		25mm
Free Air Resonance	(Fs)	50Hz ±8Hz
Moving Mass	(Mmd)	0.0107Kg
Suspension Compliance	(Cms)	0.0009
Mechanical Q	(Qms)	2.26
Electrical Q	(Qes)	0.61
Total Q	(Qts)	0.48
Equivalent Air Load	(Vas)	22 Litres
Effective Surface	(Sd)	$0.0131m^2$



mounting is from the outside using the 170mm square plastic trim supplied.

The eight inch extended bass driver (YN47B) is a more traditional looking unit constructed with a paper cone and dust cover, and a foam rolled surround. The chassis finish is cosmetic, for outside cabinet mounting and the quality of reproduction is excellent.

The six and a half inch bass/mid range driver (YN25C) has a yellow, varnished fibre glass cone with velbex dust cover and rolled surround. This unit has excellent bass response and superbly defined mid range reproduction, producing a clear, airy overall sound with a high degree of realism.

Designing a System

Loud speaker performance characteristics are of great importance when looking for optimum performance in a system. Cabinet dimensions can be calculated quite accurately for a particular drive unit, although the final performance may not always be what the constructor expects. For instance, a relatively small (8in.) woofer may require a 250 litre cabinet for optimum performance. 250 litres of volume is approximately 8.83 cubic feet, which if imagined as a cube would measure over 2 feet high by 2 feet wide by 2 feet deep. Today's 'midi system' popularity in hi-fi systems favours the small book case size cabinet, and it would take some book case to hold a cabinet of these proportions! Therefore, one must decide on a cabinet size suitable for the room available and also the cost and complexity of construction.

For this series, we will be considering the *closed box* and *ported box* (reflex) type of design only, using the mathematical models introduced by an Australian Engineer, A.N. Thiele. Speaker manufacturers supply data for use in these calculations, which makes life a lot easier for the constructor.

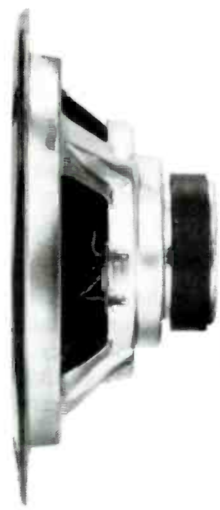
A closed box means just that! A completely sealed and air tight box, whereas the ported box has a tuned port which may be a simple hole or slot cut in the front panel. Closed boxes are usually smaller in size than reflex types and much favoured by the system manufacturer. They exhibit a gradual roll off or slope at low frequencies, and particularly small cabinets can sound lacking in bass. The ported box is larger in size with much greater bass response and a more pronounced cut-off below the tuning point or resonance. However, such a cabinet still requires to be designed and made carefully or it too will have only a poor performance!

Speaker Parameters

The first step in optimising a cabinet design requires information based on the driver specification. Figures 1 and 2 show graphical responses of the square bass driver calculated from the given data. On the vertical X axis, Vb is the box volume shown as both cubic feet and litres, the horizonal Y axis shows the box resonance (Fb or Fc), and the −3dB cut-off frequency (F3). Two plots on each graph are shown for the free air resonance frequency (Fs) of 51.5Hz ±8Hz; one curve for 44Hz and the other for 59Hz. These are however extreme variations, and random tests on various samples have produced Fs figures of 44Hz quite consistantly. If an average

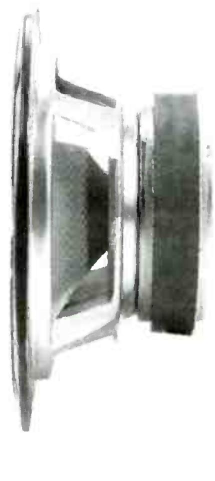
50W EXTENDED BASS - YN47B

Chassis		212 x 212mm
Baffle		186mm
Fixing		154 x 154mm. 5mm clear
Flux Density	(B)	10000 Gauss
Frequency Response		40Hz - 5kHz
Power Handling		50W peak, 25W RMS
NOM Impedance		8 ohm
Sensitivity		91dB (1W - 1M)
Coil Diameter		25mm
Free Air Resonance	(Fs)	42Hz ±7Hz
Moving Mass	(Mmd)	0.0122Kg
Suspension Compliance	(Cms)	0.0007
Mechanical Q	(Qms)	2.57
Electrical Q	(Qes)	0.7
Total Q	(Qts)	0.55
Equivalent Air Load	(Vas)	42 Litres
Effective Surface	(Sd)	$0.0196m^2$



FIBRE GLASS DRIVER - YN25C

Chassis		173 x 173mm
Baffle		145mm
Fixing		124 x 124mm. 5mm clear
Flux Density	(B)	13000 Gauss
Frequency Response		35Hz - 5kHz
Power Handling		45W peak, 22W RMS
NOM Impedance		8 ohm
Sensitivity		91dB (1W - 1M)
Coil Diameter		25mm
Free Air Resonance	(Fs)	44Hz ±7Hz
Moving Mass	(Mmd)	0.0114Kg
Suspension Compliance	(Cms)	0.00017
Mechanical Q	(Qms)	1.01
Electrical Q	(Qes)	0.22
Total Q	(Qts)	0.18
Equivalent Air Load	(Vas)	30 Litres
Effective Surface	(Sd)	$0.0113m^2$



point between both curves is taken, then from Figure 1:

(1) Box volume Vb = 57 litres
Resonance Fc = 60Hz
Cut off F3 = 80Hz

(2) Box volume Vb = 28 litres
Resonance Fc = 69Hz
Cut off F3 = 76Hz

(3) Box volume Vb = 18 litres
Resonance Fc = 75Hz
Cut off F3 = 75Hz

The smallest practical size of a closed box for this speaker is just over 0.5 cubic feet internal volume, but as can be seen, the low frequency response is not very good. From Figure 2:

(1) Box volume Vb = 57 litres
Resonance Fb = 38Hz
Cut off F3 = 32Hz

(2) Box volume Vb = 28 litres
Resonance Fb = 48Hz
Cut off F3 = 45Hz

(3) Box volume Vb = 14 litres
Resonance Fb = 58Hz
Cut off F3 = 63Hz

Using a tuned port in a 0.5 cubic feet box extends the low frequency response below 63Hz, which is a significant improvement over the closed box. To realise the full potential of this speaker, but keeping the cabinet size down, a ported box of 24 to 40 litres internal volume is required.

Predetermined equations are used to calculate box volumes. For a ported box: $Vb = Vas \times Qts^2 \times S$

Where:

Vas is the equivalent air load.
Qts is the total speaker Q.
S is the peak or dip in response at resonance (sensitivity).
Vb is the box internal volume.

The S factor is related to active port surface area in a reflex cabinet, which determines the amount of increase or decrease in output at resonance. Values of S range from 4 to 16 where, S = 4 is −3dB down and S = 16 is +3dB up at the cabinet/driver resonant point (Fb). As an example of determining the effect of S in designs, a value of 8 offers a relatively flat response at Fb with smooth roll off to F3. As a matter of interest classical music tends to favour a slight drop in output at Fb, S = 5.7 (approx. 1dB down), and pop music = 11 (approx. 1dB up).

Using data from the Bass Driver specification and S = 5.7 in the equation as follows:

Vas = 22 litres
Qts = 0.48
S = 5.7

$Vb = 22 \times 0.48^2 \times 5.7$ i.e. Vb = 28 litres.

With S = 8:

$Vb = 22 \times 0.48^2 \times 8$ i.e. Vb = 40 litres.

With S = 11:

$Vb = 22 \times 0.48^2 \times 11$ i.e. Vb = 55 litres.

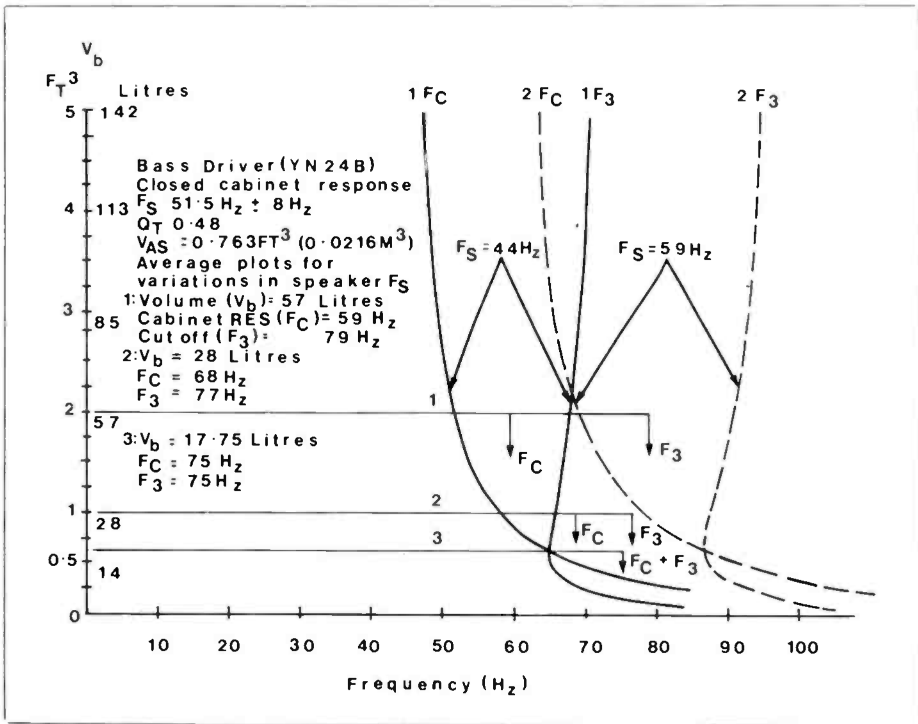


Figure 1. Bass Driver closed cabinet response graph.

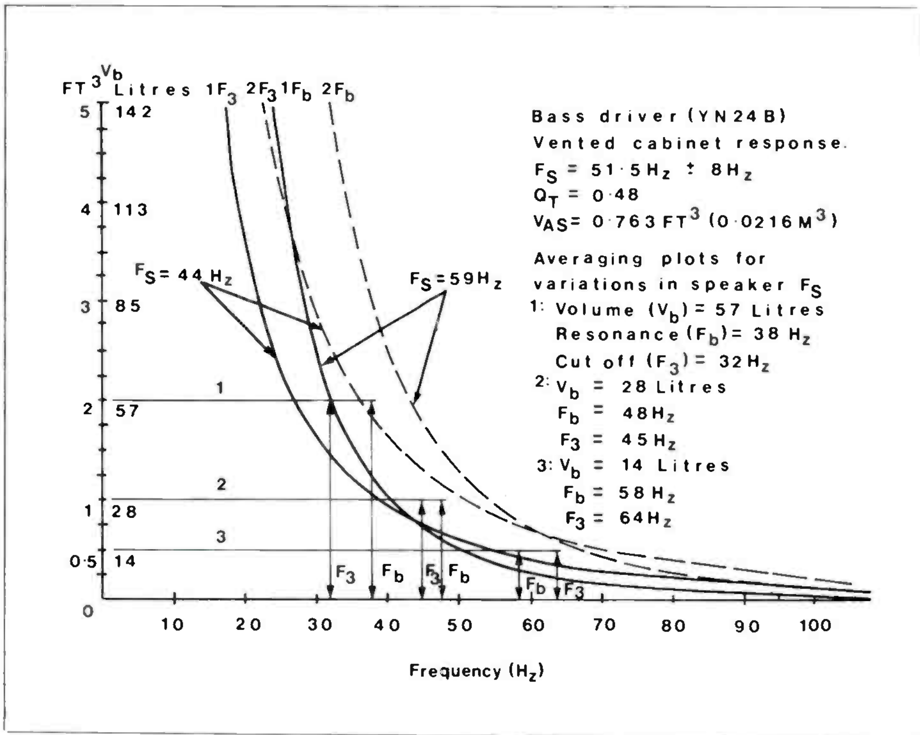


Figure 2. Bass Driver vented cabinet response graph.

A larger box could be designed in this way, but then the driver will be under-damped, thus reducing its power handling capabilities. Similarly, smaller boxes will over-damp the driver, thus lifting the Fb with subsequent loss of bass output.

With values of S equal to eight, Vb is the optimum value. Cabinet resonance can be calculated from Fb = (0.39 x Fs)/Qt and the −3dB cut-off point by squaring Fs, multiplying this by Fs squared, dividing the result by Vb and finding the square root of the total, i.e.

$$F3 = \sqrt{\frac{Vas \times Fs^2}{Vb}}$$

Therefore, for Qts = 0.48, Fs = 44Hz and Vb = 40 litres:

$Fb = (0.39 \times 44)/0.48$ i.e. Fb = 36Hz

$$F3 = \sqrt{\frac{22 \times 44^2}{40}}$$ i.e. F3 = 32Hz

The next step is to find port dimensions to suit Vb and Fb by referring to Figure 3 and 4 for tube length and diameters.

Each of the curves represents a different cabinet volume, Vb, and cabinet resonance, Fb, is on the vertical X axis. If a line is drawn horizontally from Fb = 40Hz to a point between both 28 litre and 43 litre curves, corresponding to Vb = 40 litres, then tube lengths can be read along the horizontal scale. Either 50mm internal diameter or 75mm internal diameter tubes are suitable for use as a port on low volume systems and for this design a 50mm(d) x 63mm(l) or 75mm(d) x 167mm(l) tube is called for. Larger diameter tubes are always longer in length and may be too long to fit inside the cabinet. When this happens with a 75mm tube, the 50mm tube used instead will be shorter for the required resonance. At extremely low frequencies, or when speaker/cabinet design offers a high SPL (output sensitivity), air movement through the tube can produce a chuffing noise which can be very disturbing. Always try to use the largest port possible ensuring a clearance between inside back panel to tube of at least 75 to 100mm.

Cabinet Dimensions

Producing an enclosure that adds to the sound from a loud-speaker is to be avoided, or kept to an absolute minimum. Any added resonances will 'colour' and distort the original sound. A cabinet made from thin materials will audibly vibrate and absorb low frequency energy, to the detriment of bass response. The enclosure shape can also affect performance by lumping air resonances at different frequencies and from sound energy reflecting off internal wall panels through the speaker cone. Designing an enclosure with non-parallel walls minimises internal reflection problems, but is extremely difficult to build. The finished triangular cabinet shape may also look like anything but a speaker system!

To keep construction simple with the minimum of panel resonances and reflections, heavy duty or industrial grade chipboard of 19mm (3/4 inch) thickness is best employed. Flakey chipboard that looks like a slab of compacted straw should be avoided. Each of the three cabinet dimensions have a relationship in the ratio of 1 : 1.6 : 2.3, thus ensuring that no side is an exact multiple of any other and thereby spreading resonances instead of lumping them. Figure 5 shows Vb (the internal volume) three dimensionally as (w)idth 1, (d)epth 1.6, and (h)eight 2.3. With volume Vb = 40 litres (40,000cc) the equation used for calculating each dimension is the third root (cube) of Vb divided by the ratio product, and multiplied by each side ratio in turn.

Thus from $\sqrt[3]{Vb/(\text{ratio product})}$ we get:

$$w = \sqrt[3]{40000/(1 \times 1.6 \times 2.3)} \times 1$$

$$d = \sqrt[3]{40000/(1 \times 1.6 \times 2.3)} \times 1.6$$

$$h = \sqrt[3]{40000/(1 \times 1.6 \times 2.3)} \times 2.3$$

Therefore: w = 22.15cm, d = 35.44cm, h = 50.95cm.

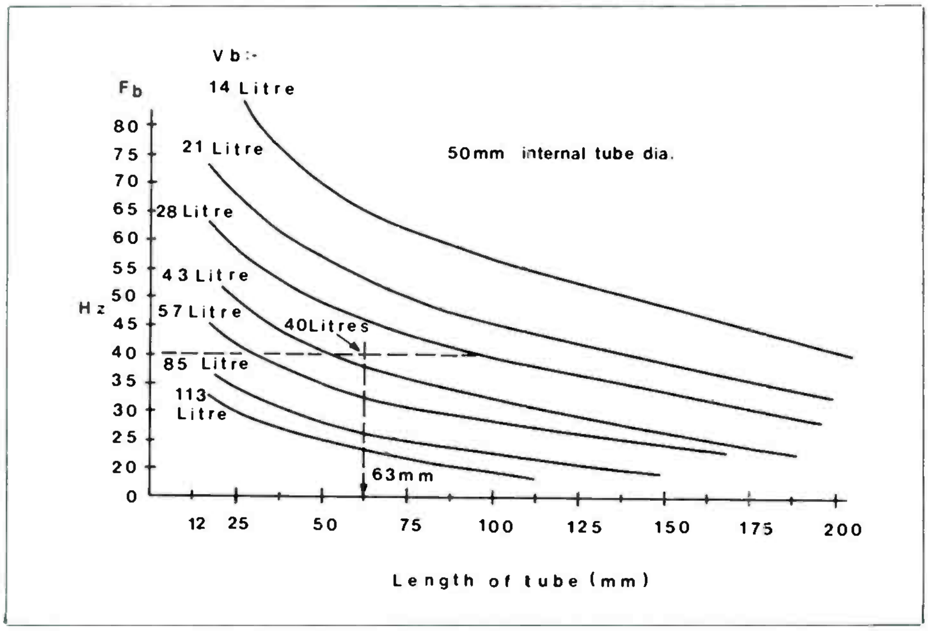
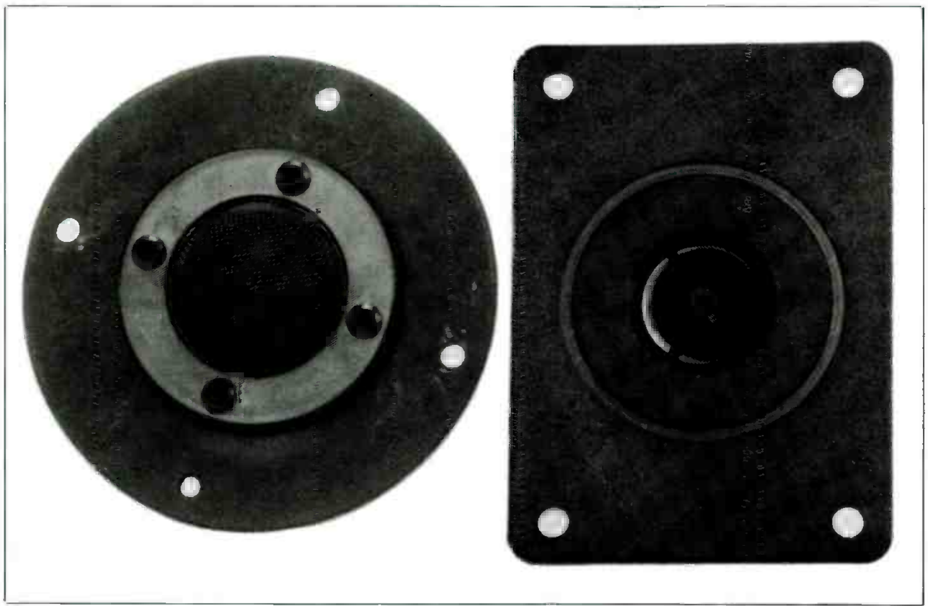


Figure 3. 50mm I/D tube length graph.



Tweeters.

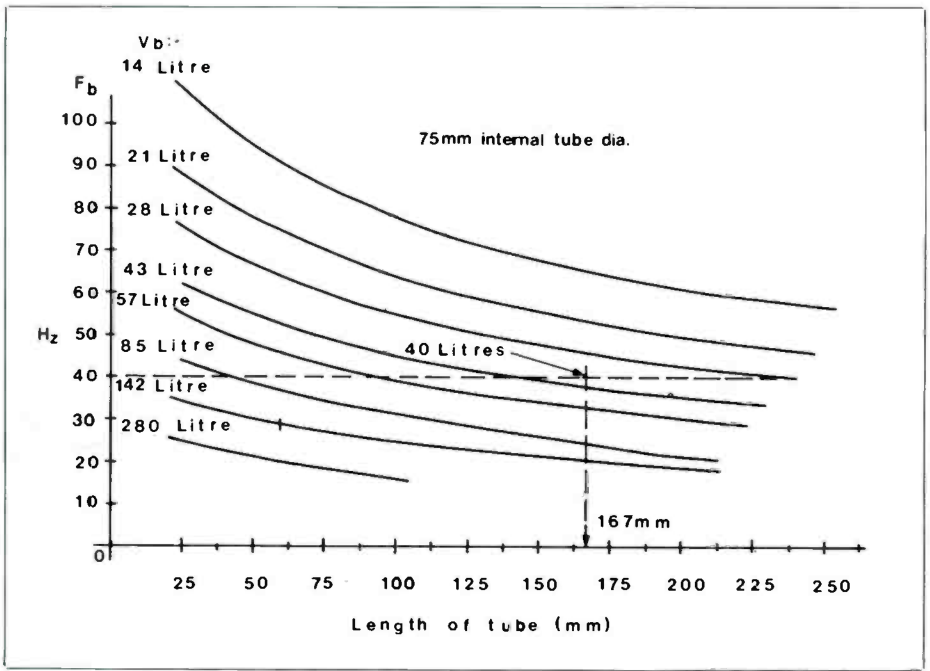


Figure 4. 75mm I/D tube length graph.

If these dimensions are simplified to integers, then cabinet *internal* sizes are w = 23cm, d = 35cm, h = 50cm. The volume Vb is now 40,250cc or 40.25 litres, a small increase of 0.6%.

For Vb = 23 litres (23,000cc),

w = 19cm, d = 29cm, h = 42cm.

Construction

It must be stressed that calculations used in this article are based on theoretical principles and do not necessarily reflect the perfect design. In practice, parameters like volume are reduced by speaker metalwork, cross-over modules, bracing, port tube volumes and stuffing material. Jointing methods affect volume and all of these factors should be considered before cutting any panels. In this particular design, the 40 litre Vb figure allows for approximately 5% over volume to accomodate the extras. Butt joints should be used for assembling panels, fixing with white wood glue and 1.5 inch chipboard screws.

Only a front baffle is supplied in these kits so the constructor must arrange to have five side/back panels made to the cutting lists shown. Most timber suppliers have facilities for cutting chipboard quite accurately to customer requirements and the material costs should not be very high.

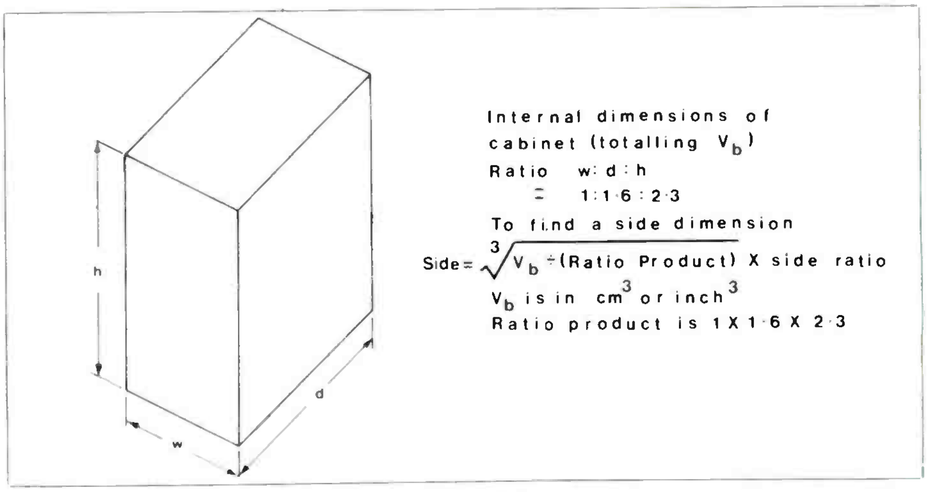


Figure 5. Relationship between width, depth and height.

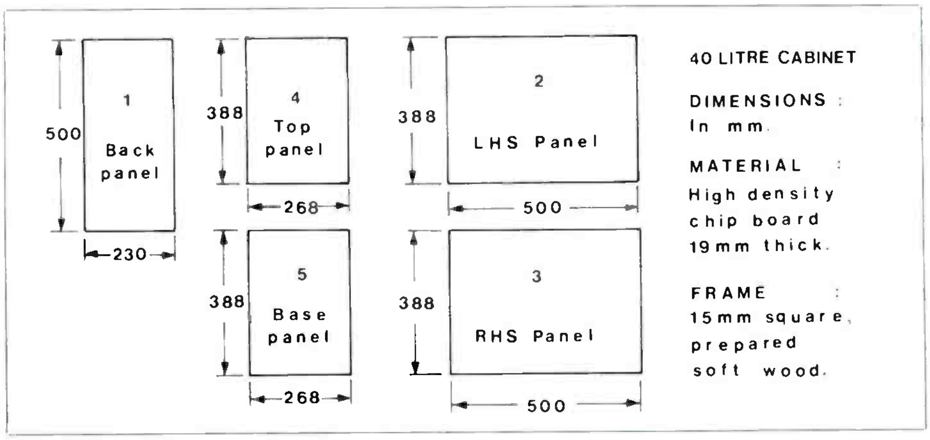
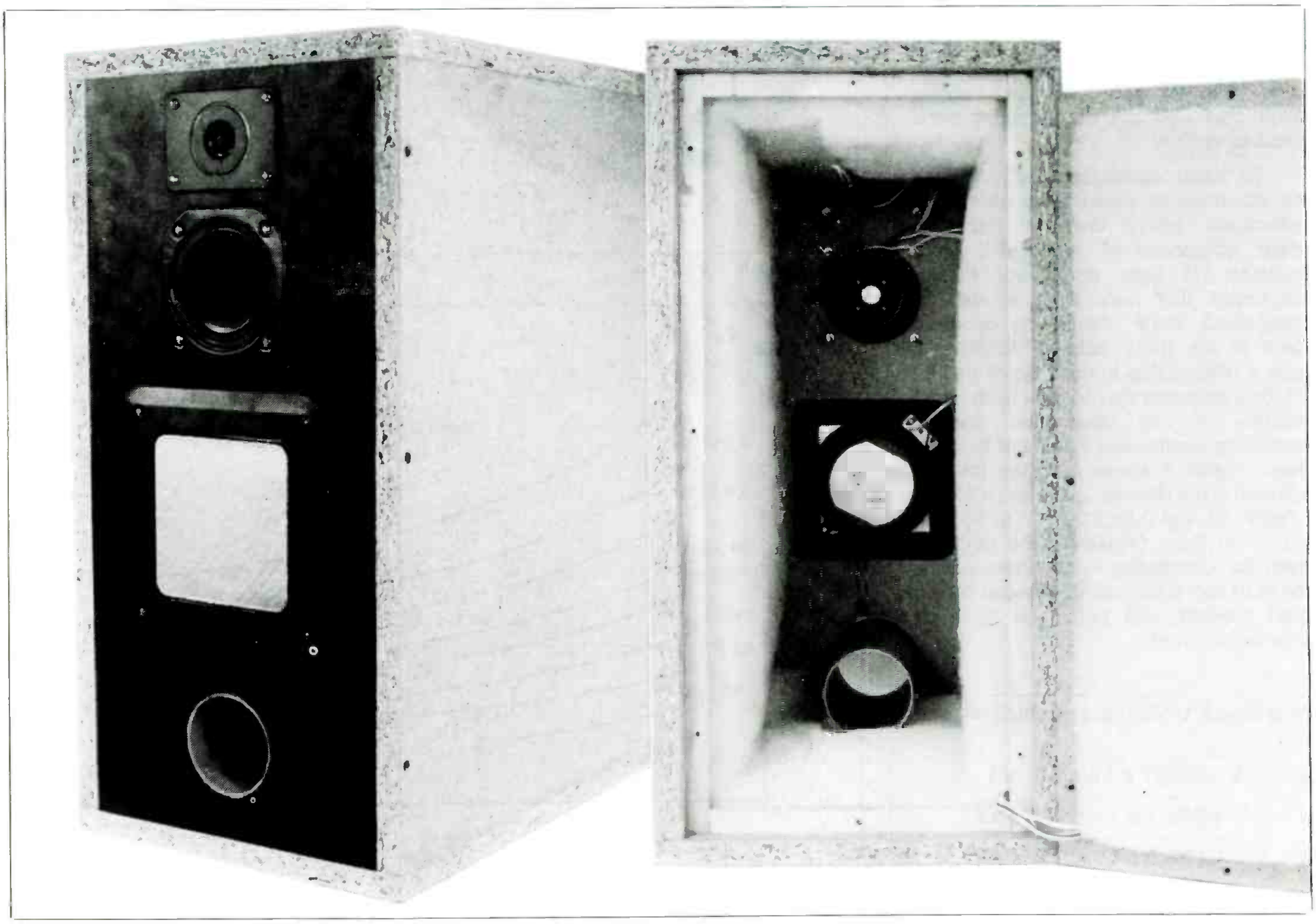


Figure 6. 40 Litre cabinet panels.



Inside the 40 litre cabinet and front baffle layout.

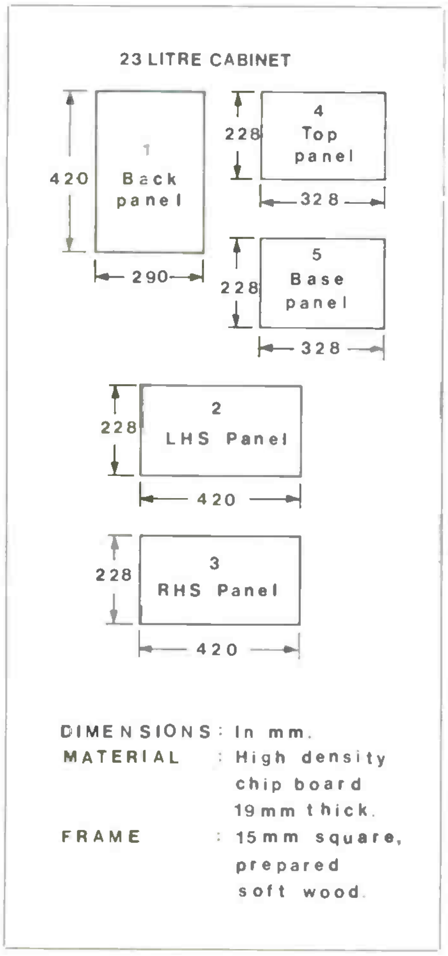
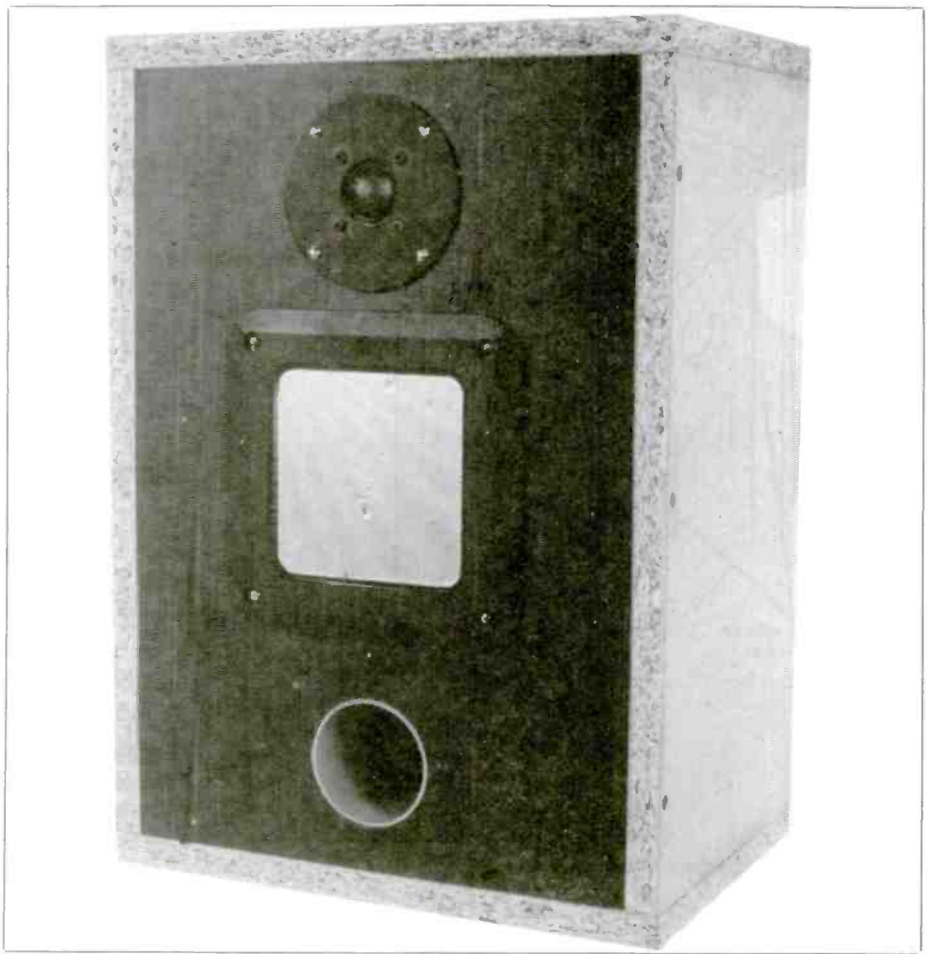
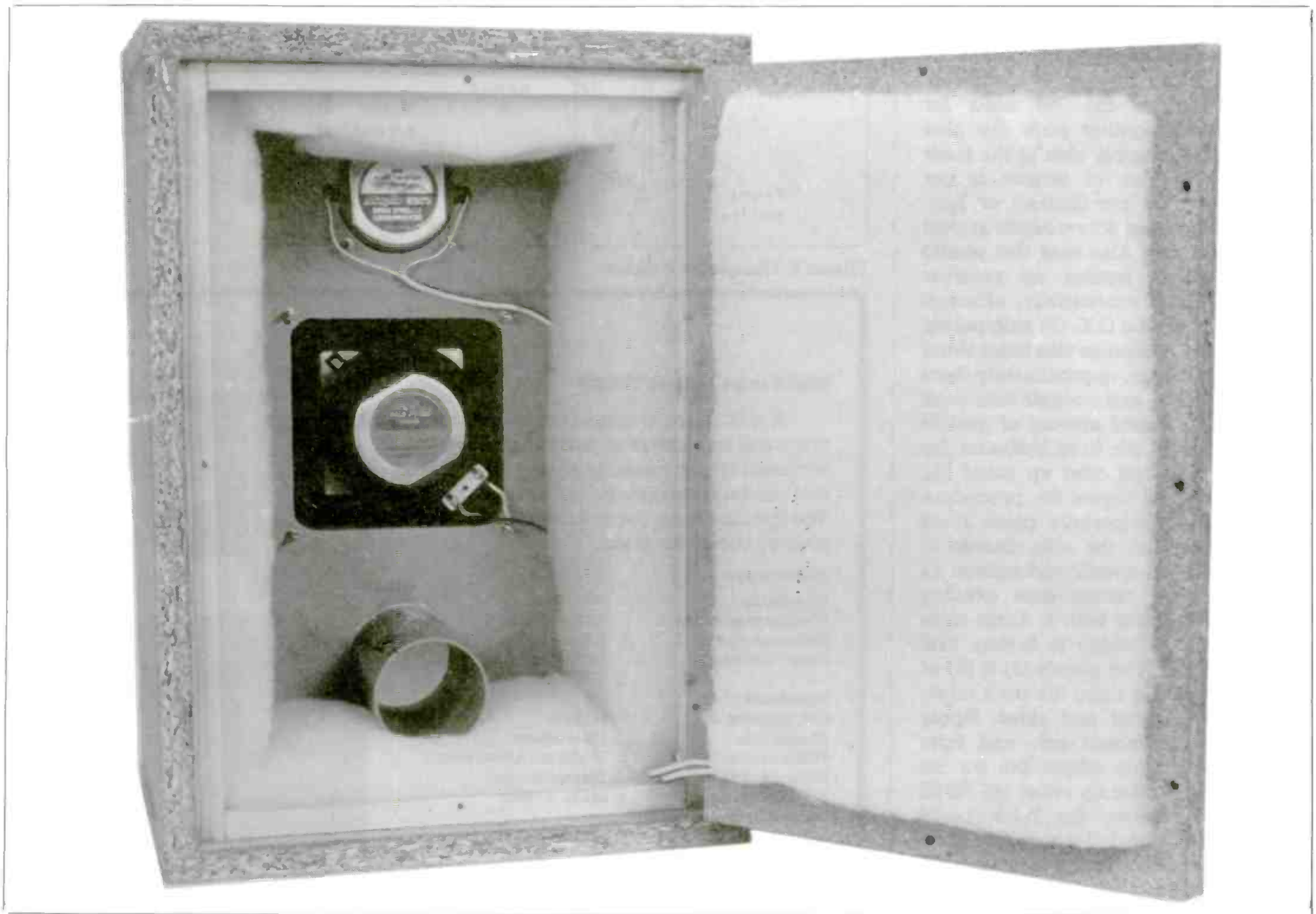


Figure 7. 23 Litre cabinet panels.



23 litre cabinet front baffle.



Inside the 23 litre cabinet.

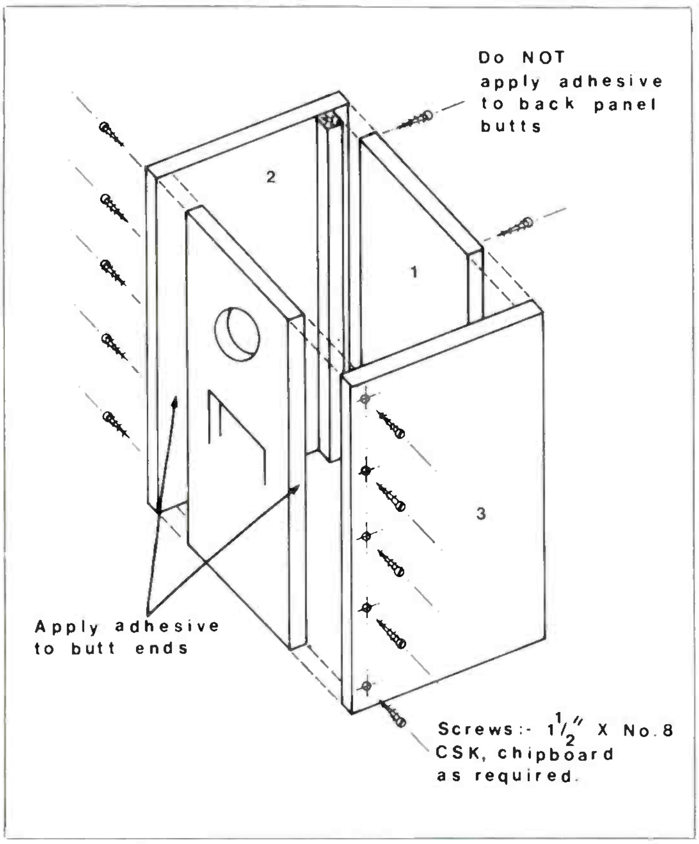


Figure 8. Assembling the panels.

CUTTING LIST (40 LITRE CABINET)

	Material	:	High Density Chipboard 19mm thick
(1)	Back Panel	:	1 off - 500mm x 230mm
(2 & 3)	Side Panels	:	2 off - 500mm x 388mm
(4 & 5)	End Panels	:	2 off - 268mm x 388mm
	Material	:	Soft wood 15mm square prepared.
(6 & 7)	Hor. Frame	:	2 off - 200mm long
(8 & 9)	Vert. Frame	:	2 off - 500mm long

RESIN 'W' wood adhesive.
38mm (1.5") No.8 chipboard screws.
25mm wire nails.

CUTTING LIST (23 LITRE CABINET)

	Material	:	High Density Chipboard 19mm thick
(1)	Back Panel	:	1 off - 420mm x 290mm
(2 & 3)	Side Panels	:	2 off - 420mm x 228mm
(4 & 5)	End Panels	:	2 off - 228mm x 328mm
	Material	:	Soft wood 15mm square prepared.
(6 & 7)	Hor. Frame	:	2 off - 260mm long
(8 & 9)	Vert. Frame	:	2 off - 420mm long

RESIN 'W' wood adhesive.
38mm (1.5") No.8 chipboard screws.
25mm wire nails.

19mm or 3/4in. chipboard is a common variety and dimensions given are based on this. If thinner or thicker board is used then change the end panel (268mm - 40 litre, 328mm - 23 litre) size accordingly. See Figure 6 and Figure 7.

Figure 8 shows assembly details for the panels. Either large 'sash' clamps or chipboard screws can be used for holding panels together while the glue dries, but do not apply glue to the back panel (1). The use of screws is not recommended on pre-finished or laminated panels unless screw heads appeal to the constructor! Also note that plastic laminated edges butting up together cannot be glued successfully, although wood veneer will be O.K. On side panels (2) & (3), drill 5 clearance size holes along the front edge only, approximately 9mm in from the edge, and counter sink each hole. Apply a liberal amount of glue to the long edge of the front baffle on the right hand side and offer up panel (3). Insert screws and repeat the procedure for panel (2). Temporarily place back panel (1) between the side panels to space them apart evenly and tighten all screws. Remove excess glue exuding from the front joints with a damp cloth and while the assembly is drying, drill and counter sink end panels (4) & (5) as before. Do not drill along the back edge, only along the front and sides. Apply glue to the assembled side and front panels along the top edges, but not the back panel and offer up panel (4). Fit all screws and remove the back panel which may now need cleaning, and wipe away excess glue from all but joints. Turn the assembly over and fit panel (5) as before. Figure 9 shows a completed cabinet.

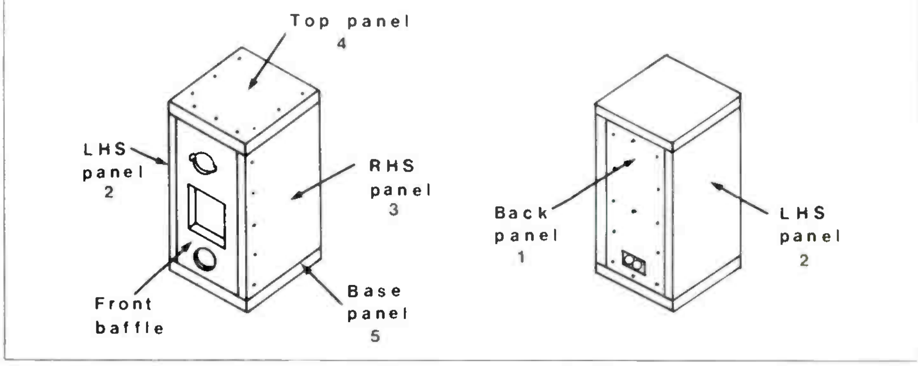


Figure 9. Completed cabinet.

Mid-range Driver YP13P

A mid-range loudspeaker with ferro-fluid cooled voice coil for increased power handling, and an enclosed steel chassis, requiring no special enclosure and can be fitted directly in bass speaker's cabinet. The speaker has a paper cone and dust cover with a pleated paper surround.

Specification

Flux density	: 9500 Gauss
Frequency response	: 500Hz - 12kHz
Enclosure type	: Infinate baffle
Power handling	: 70W @ 1kHz (DIN 45573)
	: 35W RMS
Impedance	: 8Ω
Coil diameter	: 16.5mm
Chassis size	: 100 x 100mm
Fixing centres	: 70 x70mm, 4.5mm clear
Baffle cut-out	: 75mm diameter
Free-air resonance *(infinate baffle enclosure)*	: 850Hz ± 128Hz
Acoustic response	: 91dB (1W @ 1m)

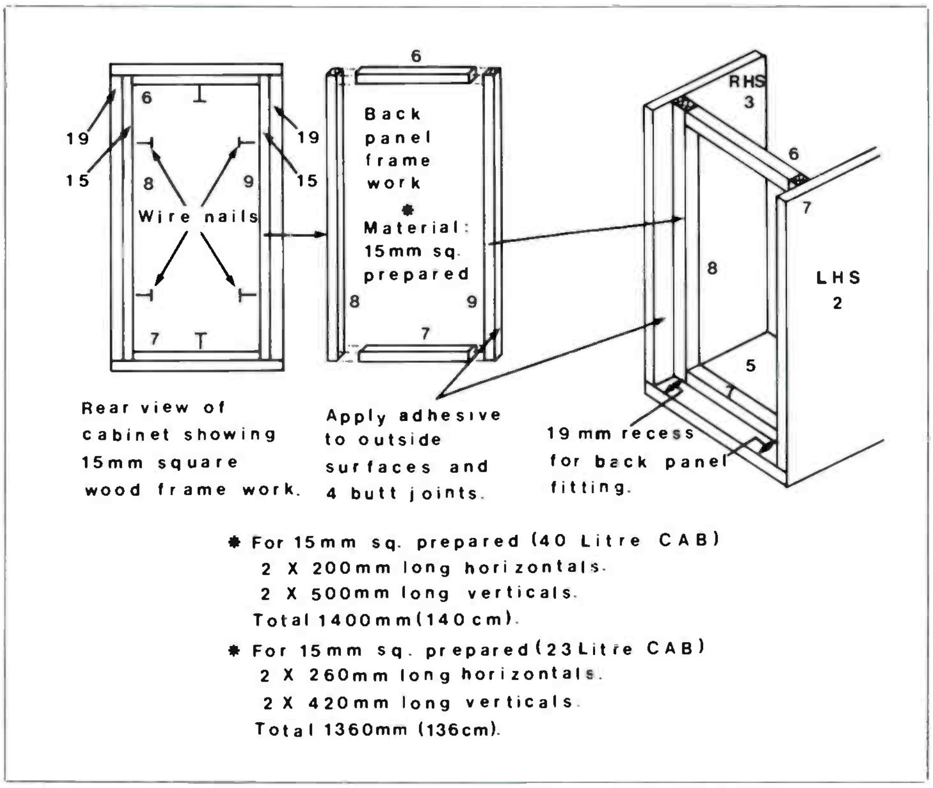


Figure 10. Back panel framework.

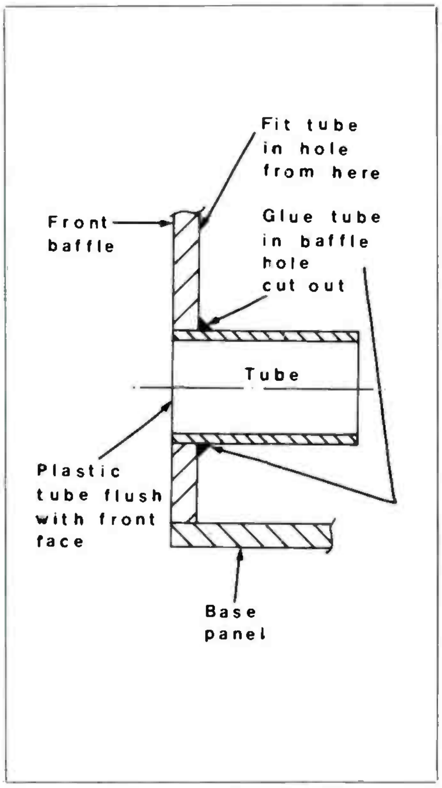


Figure 11. Fitting the port tube.

Figure 10 shows a simple wood framework which is glued and pinned inside the cabinet, and holds the back panel (1) in place. Both longer (8 & 9) verticals are the same length as panels (2 & 3) but horizontals (6 & 7) must be cut to lengths determined by their thickness. Dimensions given are for 15mm square prepared and should be the thinnest material used; thicker material will mean shorter lengths for (6 & 7) accordingly.

Draw a reference line inside the cabinet either 19mm in from the back edges or to suit the back panel thickness. Glue both verticals (8 & 9) to this reference line and use wire nails to hold in position. Do the same for (6 & 7) spreading glue over each cut end. Wipe away any excess glue and re-measure the recess between frame and panel edges, make any adjustments to ensure the frame is square.

Baffle Mountings

Figure 11 shows the port tube fitted in the baffle hole cut-out below the square woofer. Spread a thin layer of glue around the inside of the baffle port hole and insert the tube from behind. Run a thin filler of glue around the tube on the inside face of the baffle keeping the tube flush with the front face and leave to dry.

Before mounting loudspeakers, solder connecting wires to each +V and −V terminal on all units; this can be an awkward job to do with the speakers in place! Insert speakers into the baffle from the front and secure with 4BA x 1.5" bolts, shakeproof washers and 4BA nuts.

40 Litre Baffle - 3-Way System

Refer to Figure 12. Fit the rectangular tweeter FD95D into the topmost position and the mid-range speaker YP13P into the centre hole. Fit the bass driver YN24B into the square cut-out position. and mount the plastic trim cover over the top. Be careful not to force any cones during installation and take precautions while tightening up with a screwdriver. It may be found advisable to cover the speakers with thick card whilst doing this, just in case the screwdriver should slip!

23 Litre Baffle - 2-Way System

Refer to Figure 12. Fit the dome tweeter YN43W into the topmost position and insert the bass driver YN24B into the square cut-out. Mount the plastic trim cover over the top and follow the same precautions during installation as before. If required, a protective grille can be fitted over the dome tweeter to prevent dust and damage from spoiling its performance. The dome is very soft and sticky by design and should not be handled. To fit the grille, very carefully remove all 4 star-head screws, place the grille over the hole positions and replace the screws.

Cross Over and Wiring

When wiring speakers to the crossover module, use a separate cable pair to each unit, connecting terminal 'W' to bass speaker positive, 'T' to tweeter positive, and if used,'M' to mid-range positive. The positive terminal on each speaker will be marked either by a red dot or a + symbol stamped into the plastic housing. Speaker negative or return cables should be terminated to the crossover module terminals marked 'C'.

On the 2-way module only, it is necessary to remove the bass speaker by-pass capacitor. This component is not marked, but can be identified by placing the module with input terminals 'IN' and 'C' facing to the left and output terminals 'N', 'C' and 'T' facing to the right. The capacitor to be removed is then the bottom one which is soldered to the track areas marked with a 'W' and 'C' on the copper side. De-solder both capacitor leads and remove carefully. If required, the capacitor can be left in position, but this has the effect of emphasising the mid-range response on the bass speaker which sounds quite raucous in this particular set-up.

Cabinet Wadding

Cut five pieces of fibre wadding to fit inside the cabinet and fix to the walls with adhesive. Both sides, top and bottom panels and the back panel should be covered, but not the baffle panel. Keep the port tube clear of any obstructions and do not cover the crossover module. Depending on where the module is fitted in the cabinet, cut out a section of wadding so that it fits around the module and not over it. Also, allow clearance for the wood frame when cutting the back panel piece; the wadding should not be sandwiched in the recess when the panel is screwed in place.

Finally drill a small hole in the back panel for the connecting cable to your amplifier. If the speaker is to be sited some distance from the amplifier, then use a fairly thick cable, such as 2-wire mains cable, and not the thin bell-flex variety. On the input side of the crossover module, 'IN' is the positive terminal and 'C' the return terminal. Ensure correct polarity connections to the amplifier on stereo speakers, to keep speakers in phase with each other. If you are not sure about polarity, connect a

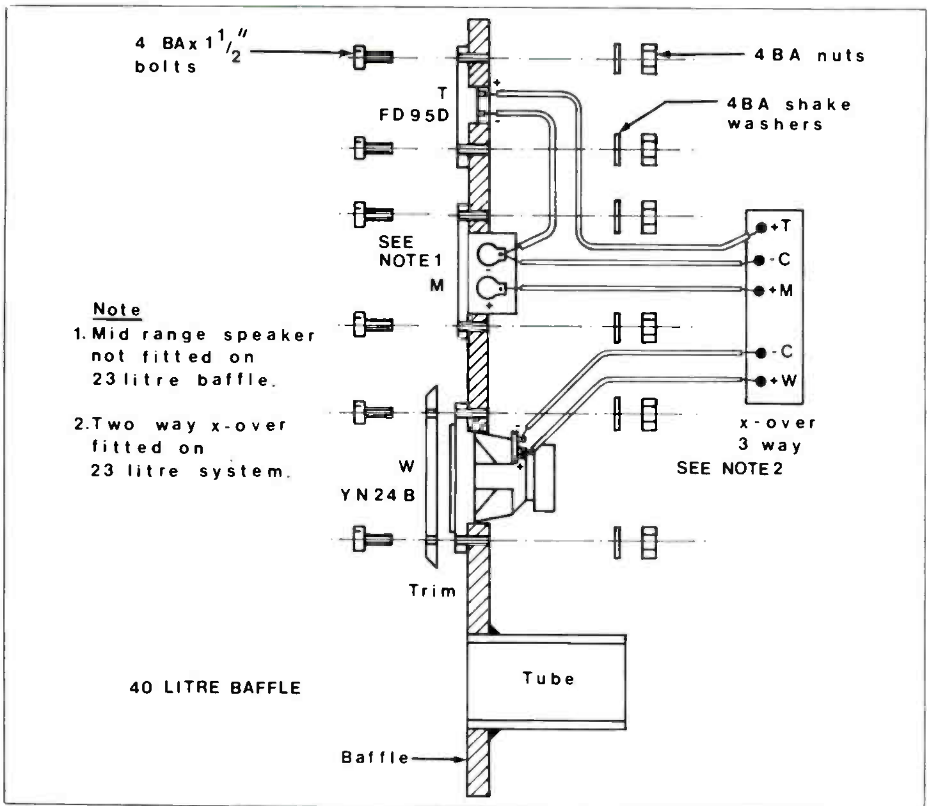


Figure 12. Baffle mounting.

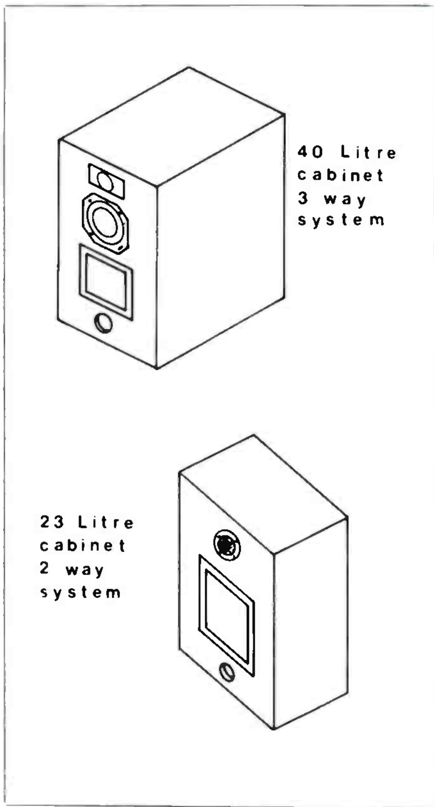


Figure 13. 40 and 23 Litre cabinets.

1.5V battery with positive to 'IN' and negative to 'C' on the module. The woofer cone-panel should pop outward, if not then the polarity is incorrect and the woofer (or crossover) wiring should be reversed. Figure 13 shows the two types of cabinet discussed in this article.

Amplifiers vary in their power output specifications and often 'MIDI' systems rated at 40 Watts per channel refer to peak output power. Both of these designs have a 50W peak continuous sine wave rating which should be adequate for most domestic listening environments.

FINAL SPECIFICATIONS

23 Litre Cab, 2-Way System, Reflex Port

Cross Over	:	2.75kHz (Modified)
Power Rating	:	50W peak, 25W RMS Continuous Sine Wave
Maximum Signal	:	40V peak, 14.2V RMS Sine Wave
Frequency Response	:	50Hz to 16kHz, 4dB peak at 60Hz
Impedance	:	8Ω

40 Litre Cab, 3-Way System, Reflex Port

Cross Over	:	1kHz and 6kHz, −6dB
Power Rating	:	50W peak, 25W RMS Continuous Sine Wave
Maximum Signal	:	40V peak, 14.2V RMS Sine Wave
Frequency Response	:	45Hz to 23kHz
Impedance	:	8Ω

23 LITRE CABINET KIT PARTS LIST

MISCELLANEOUS

Hi-Fi Bass Driver	1	(YN24B)
Hi-Fi Dome Tweeter	1	(YN43W)
Grill for Dome Twtr	1	(FD93B)
Plastic Pipe 130	1	(YP15R)
Crossover 2-Way	1	(WF02C)
Baffle 2W23L	1	(XJ09K)
4BA × 1½in. Bolt	1 Pkt	(LR52G)
4BA Nut	1 Pkt	(BF17T)
4BA Shake	1 Pkt	(BF25C)
No. 4 × ½in. Self Tap	1 Pkt	(BF66W)
Acoustic Wadding	2m	(RY06G)
Hi-Fi Loudspeaker Cable	1m	(XR72P)
Instruction Leaflet	1	(XT62S)
Constructors' Guide	1	(XH79L)

The above items are available as a kit:
Order As LM21X (23 Litre Cabinet Kit)

40 LITRE CABINET KIT PARTS LIST

MISCELLANEOUS

Hi-Fi Bass Driver	1	(YN24B)
70W 3·25in. Speaker	1	(YP13P)
Mini Dome Twtr Rect	1	(FD95D)
Plastic Pipe 180	1	(YP14Q)
Crossover 3-Way	1	(WF03D)
Baffle 3W40L	1	(XJ08J)
4BA × 1½in. Bolt	2 Pkts	(LR52G)
Nut 4BA	2 Pkts	(BF17T)
4BA Shake Washer	2 Pkts	(BF25C)
No. 4 Self Tap × ½in.	1 Pkt	(BF66W)
Acoustic Wadding	3m	(RY06G)
Hi-Fi Loudspeaker Cable	2m	(XR72P)
Instruction Leaflet	1	(XT62S)
Constructors' Guide	1	(XH79L)

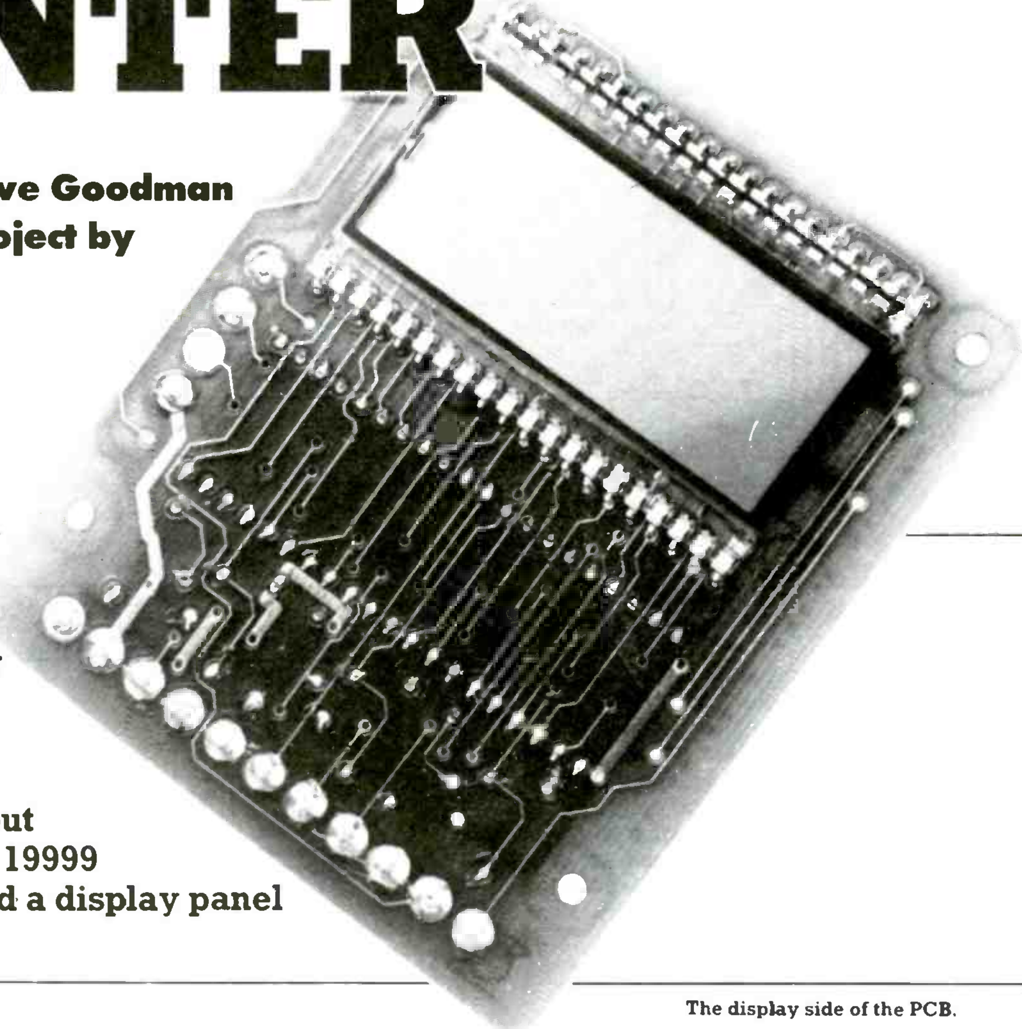
The above items are available as a kit:
Order As LM20W (40 Litre Cabinet Kit)

2ND Time Around

Particular projects from the Maplin range have proved themselves to be very popular, but technology and component specifications have a habit of changing, with the result that some of these projects are in danger of becoming obsolete. Even if this were not the case, it may be equally worthwhile improving the project in question in order to increase its versatility and usefulness, and bring it 'up-to-date.' The '2nd Time Around' series is dedicated to reviewing and improving original, popular Maplin projects by republishing them with the necessary updates and improvements, ensuring their continued availability. This time it is the turn of the 4½ Digit Counter.

4½ DIGIT LCD COUNTER

by Mike Holmes and Dave Goodman based on an original project by Mark Brighton



FEATURES

★ General purpose counter
★ Low power consumption
★ LCD display
★ Buffered inputs
★ Schmitt trigger count input
★ Will display from 0 up to 19999
★ Can be flush fitted behind a display panel

The display side of the PCB.

This is a simple 4½ digit counter intended for general purpose applications, which was first published in the March 1987 issue of 'Electronics'. It can be used either as it stands or in the form of a 'building block' module where a counter function is required as part of a larger system. Please note however, that it is not a complete frequency counter or a timer in its own right, but it can be used in these applications if provided with the necessary external gating signals as supplied by a separate timebase of some description.

The counter is built around the ICM7224 IC, which actually contains all the essential circuitry to operate as an incremental counter and simultaneously drive a 4½ digit LCD. The chip includes a 19kHz oscillator and a divide by 128 divider producing a 150Hz signal for the display's AC backplane, making it very easy to use, and requiring only one +5V DC regulated supply and the display to function.

Circuit Description

The 7224 is shown as IC1 in Figure 1. This is a minimum circuit configuration for this device, and makes use of just three main inputs to the chip for the functions COUNT, COUNT/INHIBIT and RESET. Inverter stages TR1 to TR3 have been added to the minimum configuration to provide a measure of immunity to over-voltage signals, by buffering the chip from the board inputs, and enabling the counter to operate over a wide range of input voltage peaks from approximately +2 to +20V. This is a desirable precaution, given that 'general purpose' covers uses that may require signal voltages to be anything but standard logic levels.

To reset the counter to zero, the RESET input at TB9 is taken to >+0·7V, which also blanks the display if TB14 is connected to +5V or left floating. In this configuration IC1 is normally in leading zero blanking mode. For all other functions the RESET input TB9 should be grounded or left open circuit to ensure normal counting operations. The COUNT input, TB8, is negative going, that is it requires a positive voltage falling to zero for a count of one to be incremented in the counter. This input need not neces-

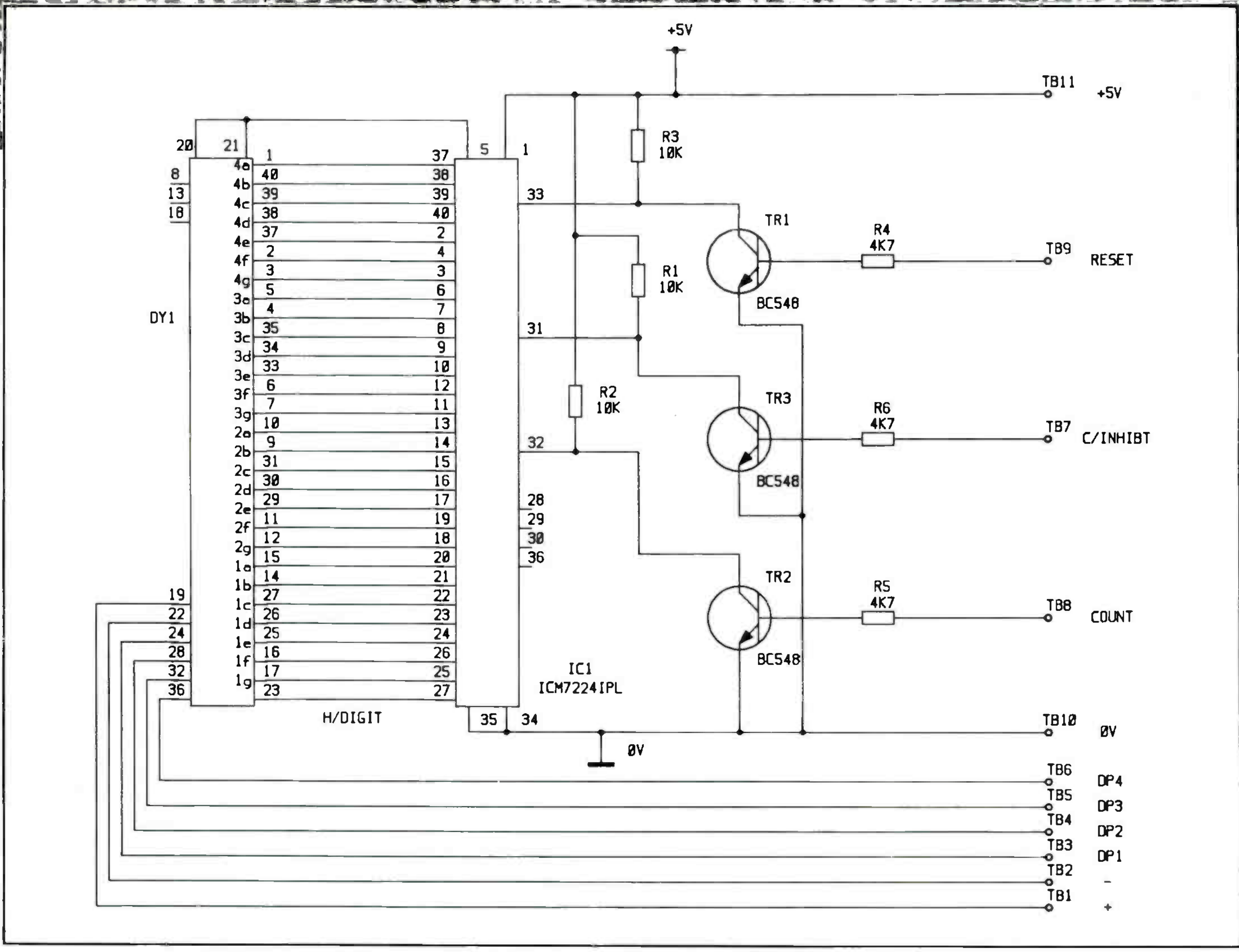


Figure 1. Circuit diagram.

sarily be a hard edged pulse or square waveform, since the COUNT input of IC1 incorporates a Schmitt trigger input stage, which, together with the gain of TR2, will allow slower voltage variations. For example, a low level sine wave signal (less than 5V peak) alternating about 0V can be used to trip the counter on each falling positive half-cycle. Moreover, the Schmitt trigger operation offers some immunity to interference and noise injection, thus preventing erroneous and erratic behaviour of the counter. This flexibility allows a wide variety of COUNT input sources, including various types of sensors.

The COUNT/INHIBIT input at TB7 is used to defeat normal counting operation so that the display remains static without the need to remove the signal at the COUNT input. To do this, COUNT/INHIBIT is taken to >+0·7V. In this condition the last count will be retained and displayed indefinitely, or can be cleared to zero with RESET. COUNT/INHIBIT must be grounded or open circuit at TB7 to enable, or resume, normal counting. In this way the counter can be 'gated' by a means based on a time period or a specific condition (e.g., counting allowed provided condition is true, etc.).

Some additional terminals are provided on the board, which are direct

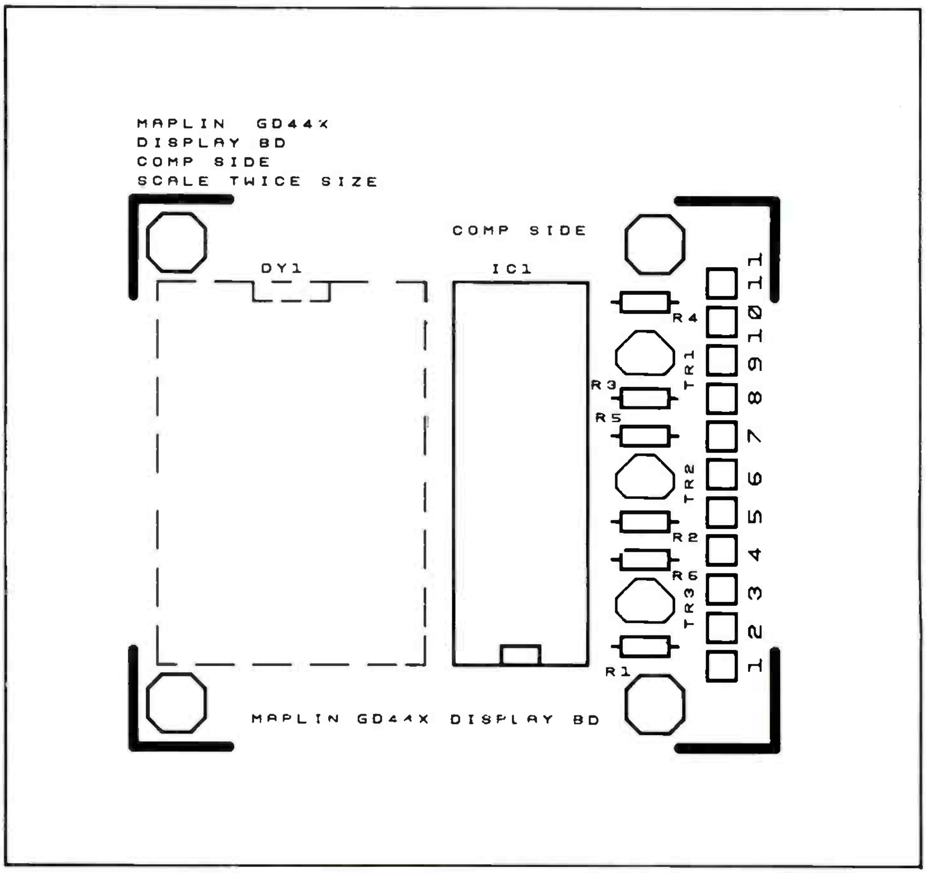
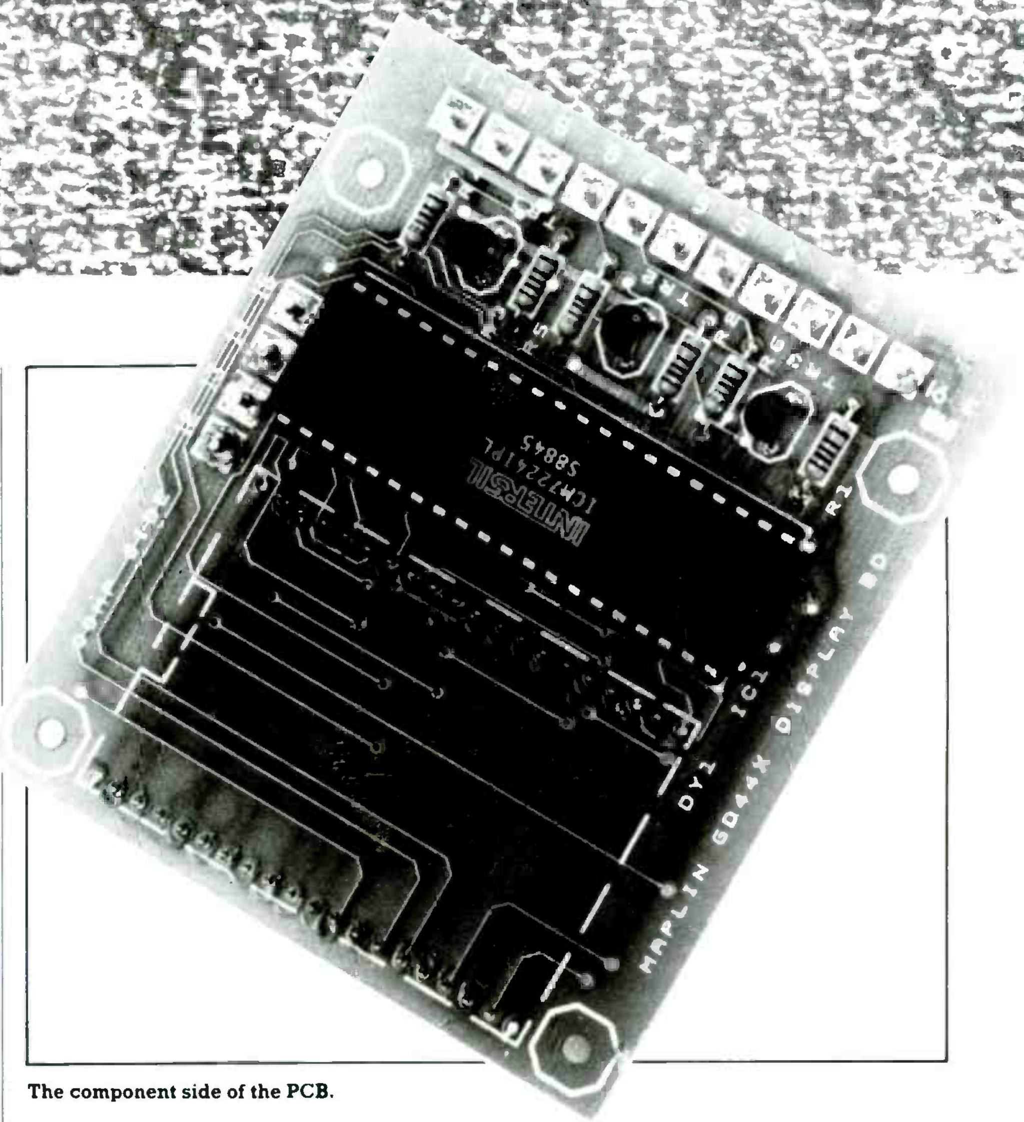


Figure 2. PCB layout.

connections to both the LCD array and IC1, and brought out and made available to the user. These include the four decimal point positions TB3 to TB6, the plus sign TB1, and the minus sign TB2. If the counter module is to be incorporated into a system for a particular task, then these can be hard-wired to 0V as required, or temporarily connected for some general purpose use to clarify the display. TB15 is provided as a second convenient earth terminal for this purpose. The decimal points may be switched if the module is to form the basis of some sort of frequency or period counter, over several switched ranges. If these terminals are not to be used, they should be left floating.



The component side of the PCB.

Construction

With reference to the Parts List and the board layout and legend shown in Figure 2, insert and solder all six resistors, then the 15 veropins from the bottom (non-legended) side of the board at the positions TB1 to TB15 using a hot iron to push them home, then solder in place. Insert and solder TR1 to TR3, making sure orientation is correct by aligning their package style with the 'D' shaped legends. Carefully fit the LCD array DY1 to the *underside* of the board, ensuring that the glass pip or similar marker at one end aligns with the rectangular marker on the legend shown in dotted lines on the *top side*, and that all the pins line up with their respective holes. Be *very careful* when soldering that the display does not get hot; if it does, wait between soldering operations until it is cool. It may be helpful to solder each corner pin first to ensure the display is seated properly before soldering all the pins. In fact the pins may be quite long, and if so then it is possible to gain some extra height above the board by slipping a small, rectangular piece of corrugated cardboard underneath the display, which can then be removed after soldering. This will make the top surface of the display approximately 5·5mm above the PCB, and the longer lead length will add extra resistance to heating during soldering.

The same soldering precautions can be applied to the 40-pin DIL socket, which is fitted at IC1 position with its end notch adjacent to the rectangular marker on the legend. Upon completion, check your work for possible short circuits and incomplete solder joints. In particular closely examine the areas between the pins of the DIL socket and the display. When you are satisfied that all is well, carefully insert IC1 into its socket, ensuring that its end notch aligns with that of the socket, and commence testing the module.

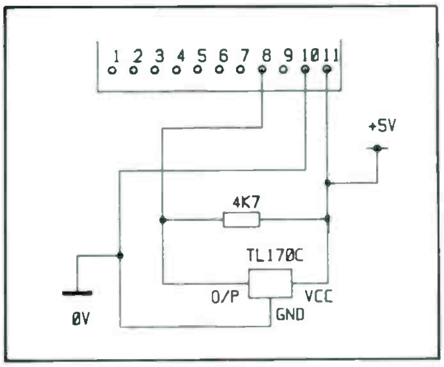


Figure 3. Hall effect device input.

Testing

Temporarily connect TB14 to TB10. The module requires a regulated +5V DC supply to operate – *do not* attempt to connect anything other than +5V to the supply pin or damage may occur. Upon switching on a random number may appear on the display. Operate the RESET input by connecting TB9 to TB11 with a test lead. The display should go blank.

Similarly, pulsing the $\overline{\text{COUNT}}$ input in the same way should cause the module to count up each time TB8 is released. Link the COUNT/INHIBIT pin TB7 to TB11, and the module should not then respond to any further changes at TB8. The display should be 'frozen' showing the last value incremented before COUNT/INHIBIT was taken high.

Pins TB1 to TB6 on the board access the 'plus' and 'minus' signs and the decimal point symbols of the display. Earthing each in turn to TB10 should cause these to appear.

Using The Module

How the module is used is entirely up to the constructor, one of the most obvious applications being of the event counting variety. Means of electrical input to the $\overline{\text{COUNT}}$ pin have already been described, and a mechanical switch can also be used between TB10 and TB8 with a pull-up resistor between TR8 and TB11, to cause an increment of one whenever the switch is closed. The switch may be a thumb operated push-button for counting items by hand, or a micro-switch on a machine. Note however, that some form of 'debouncing' is desirable and a minimum requirement would be a 100nF capacitor connected across the switch terminals. Figures 3 and 4 show alternative arrangements utilising a magnetic hall effect device (with a magnet) and an opto-coupler respectively as sensors. Either of these latter two methods can produce a counter for a hand operated wire winding machine, or for use with a wheel to make an odometer etc.

Some more sophisticated functions are available if desired. TB12 provides access to the CARRY output of IC1 (active low), which can be used to drive an overflow indicator or even a second, cascaded counter module.

Connecting TB13 to ground via a wire link to TB10 defeats the leading zero blanking mode, causing the display to show zero on RESET.

TB14 provides access to the chip's STORE input (active low). The STORE function controls internal latches which transfer decoded display data from the count decoders to the display drivers; whilst TB14 is low the latches are 'transparent' and the display reflects real time counting. On TB14 being high, the last count value is latched, causing the display to hold this value, even though counting is continuing. Pulsing TB14 low will show updates of the count value, remaining static between times. RESET does not effect the display. For continuous real time counting TB14 should be wired to 0V at TB15.

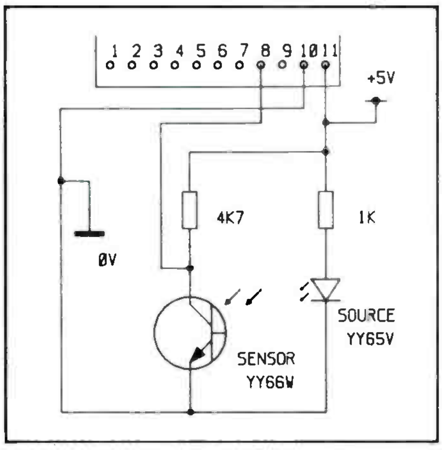


Figure 4. Opto-coupled input.

A simple frequency counter can be made using a suitable external clock and logic timebase producing, in sequence, COUNT/INHIBIT (TB7) low for the gating (sampling) period, followed by STORE (TB14) low to update the display, then RESET (TB9) to clear the counters for the next gating period. Signal input is to COUNT continuously. If a gating period of 1 second is used, you will have an AF frequency counter with a resolution of 1Hz and an upper display limit of 19,999Hz, covering the entire AF band with no range switching necessary.

Other applications of this sort include machine rpm indicators, engine rpm and maybe even mph counters using suitable sensors. The LCD is deliberately placed on the underside of the PCB so that it can be incorporated into a front panel display design with the actual LCD mounted flush behind its display window, but leaving all connections accessible.

4½ DIGIT COUNTER PARTS LIST

RESISTORS: All 0·6W 1% Metal Film			
R1-3	10k	3	(M10K)
R3-6	4k7	3	(M4K7)
SEMICONDUCTORS			
IC1	ICM7224IPL	1	(FP62S)
TR1-3	BC548	3	(QB73Q)
MISCELLANEOUS			
DY1	4½ Digit LCD	1	(FP61R)
	PCB	1	(GD44X)
	DIL Socket 40-pin	1	(HQ38R)
	Pin 2141	1 Pkt	(FL21X)
	Instruction Leaflet	1	(XT55K)
	Constructors' Guide	1	(XH79L)

The above items are available as a kit:
Order As LM19V (4½ Digit Countr Kit)

The following item (which is included in the kit) is also available separately.
4½ Dig Countr PCB **Order As GD44X**

Keypad for the Z80

★ Type in your own code easily
★ Visual display aids checking
★ Simple construction

by Graham Dixey C. Eng., M.I.E.R.E.

PLEASE NOTE THAT THIS PROJECT WILL SHORTLY BE REVISED AND RE-DESIGNED

Introduction

The Maplin Z80 CPU card which was published in Issue 15 of 'Electronics' offers an inexpensive way to get to grips with computerised control systems for those who can write their own control programs and put them into an EPROM. The provision for 8K of on-board memory is generous for such applications, and input/output decoding for peripheral chips is also provided. Unfortunately the module is totally devoid of any kind of resident software and is completely inaccessible to the 'user' in its basic form. Provision is made for a keyboard to be added however, the suggested device is the 8279 programmable keyboard/ display interface IC, which can look after a variety of input sources (keypads, full keyboards, sensor arrays, etc.) and can control up to sixteen 7-segment displays if required. Thus, it is obviously possible to produce a small computer which can be programmed directly from a HEX keypad, with a 7-segment display to monitor addresses, status and data, both in and out.

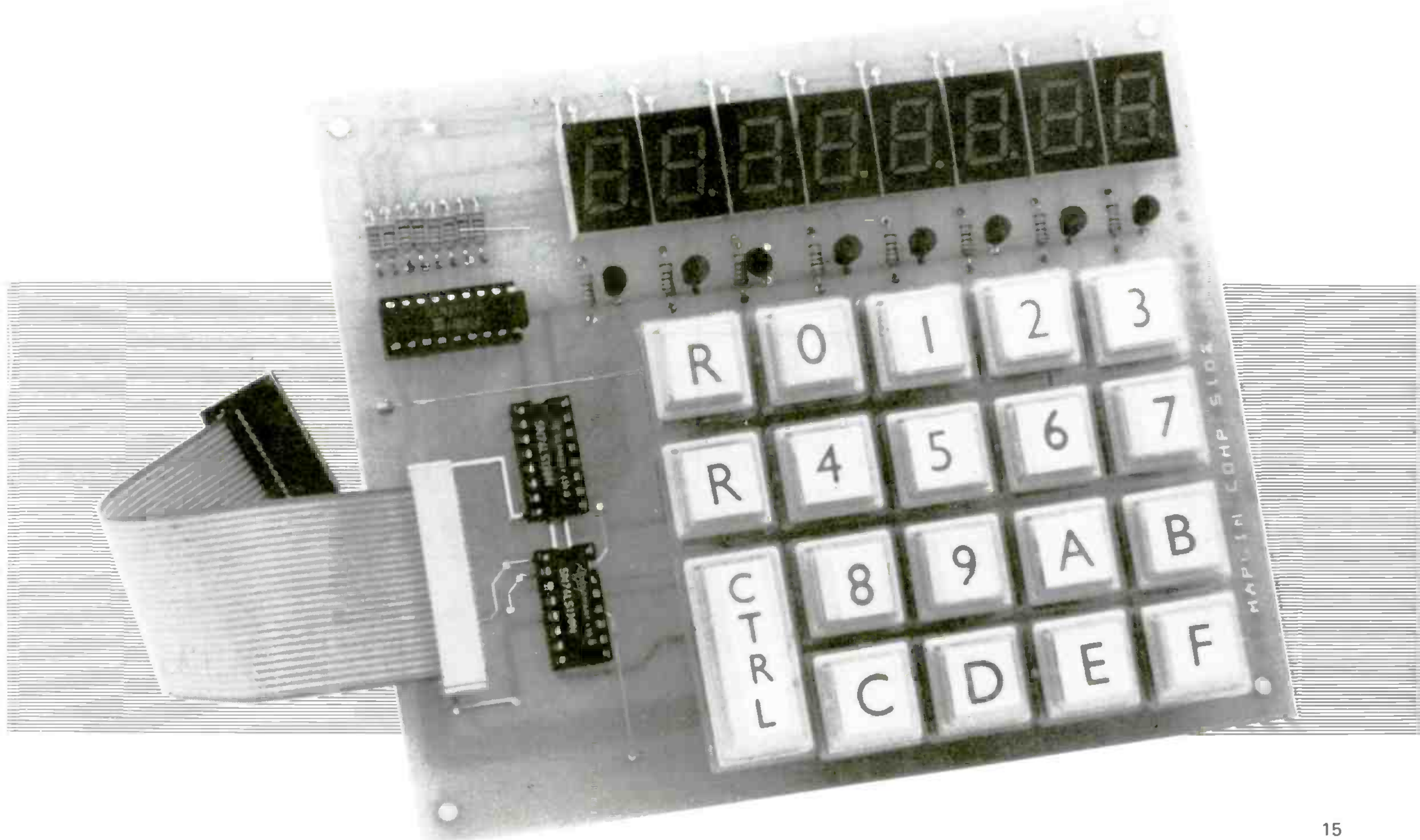
In the design presented here, there are eight 7-segment displays, from left to right, the first four form the 'address field', the next two the 'status field' and the two on the right, the 'data field'. What is needed to achieve this simple objective is the module described here, a monitor resident in EPROM (which has been developed and is available - see Parts List), and some knowledge of Z80 machine code or the desire to acquire it, which can be assisted by my current series on the subject.

Note that it is advisable to obtain a copy of Issue 15 of 'Electronics' to be sure exactly what is provided in the Z80 CPU kit and what is otherwise 'optional'.

The Keyboard Display Circuit

The circuit is shown in Figure 1, and is quite straightforward, largely due to the built-in sophistication of the 8279 IC.

The display is multiplexed at a rate determined by the system clock and a control word sent by the monitor. As a result, a binary counter output appears



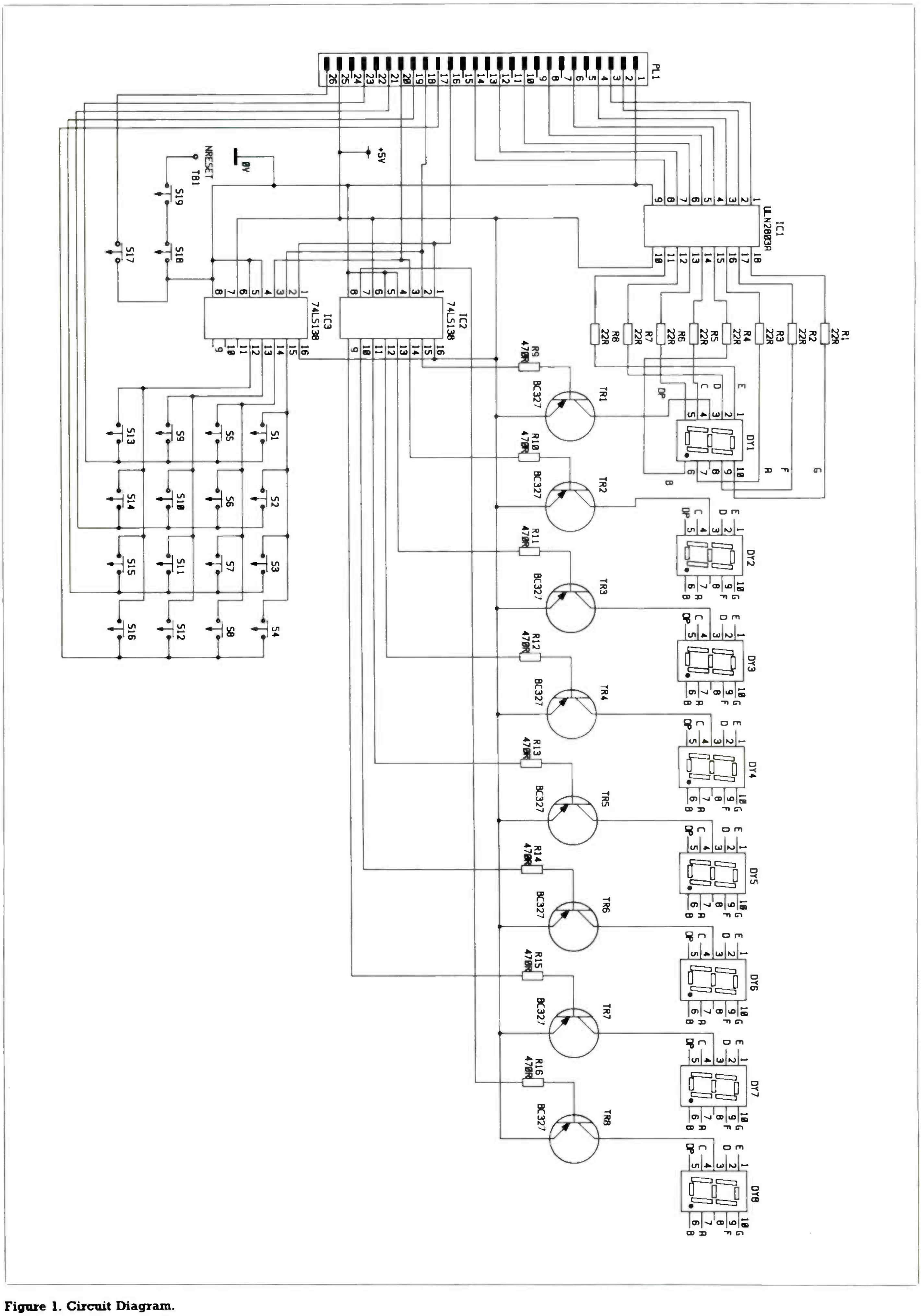


Figure 1. Circuit Diagram.

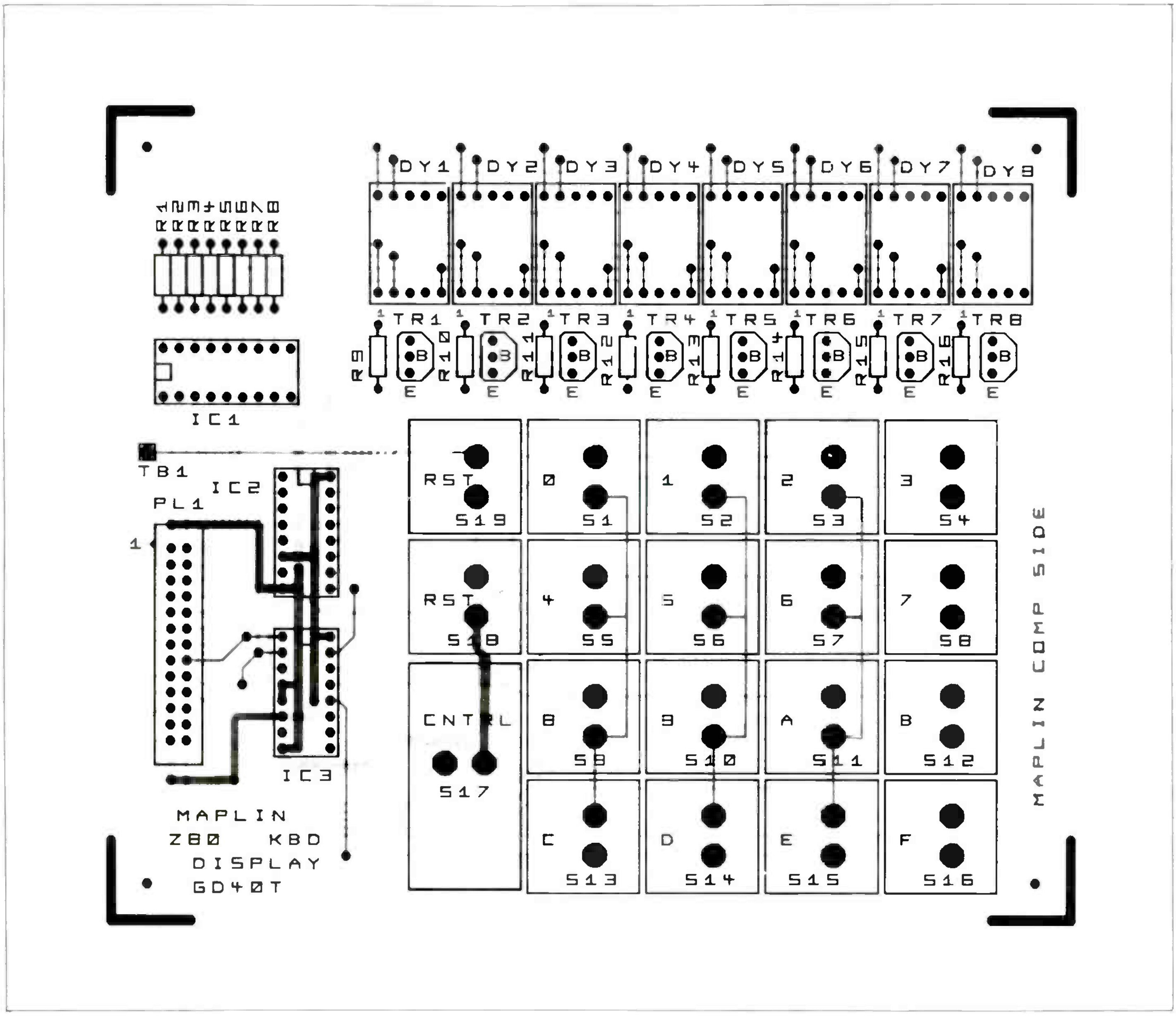


Figure 2. PCB Layout.

on the scan lines SL0-SL2, which is applied to both 74LS138 3-8 line decoders, IC2 and IC3. The eight output lines of IC2 drive the bases of driver transistors TR1-8, which are PNP types since common-anode displays are used. The ULN2803A, IC1, is an octal inverting buffer, that can sink more than enough current for the 7-segment displays. Its input is a set of eight data lines A0-A3, B0-B3 from the 8279 internal display RAM. The 22Ω resistors, R1-R8, limit the segment current to a value that will ensure reliability and long life for the displays.

The keypad consists of a 16-key HEX matrix and, to obtain additional functions, a further key, CTRL, is provided. This, used together with any other key, allows up to 16 *control* functions. Some of these are used in the system monitor, described later. The keypad returns four lines RL0-RL3, plus CTRL, to the 8279, which scans them to detect a key press and to identify its position in the matrix. It then sends an 'interrupt' signal to the Z80, to initiate a 'read keyboard input' routine.

Construction and Testing

No problems should be encountered provided that you negotiate the usual hazards of dry joints, bridged tracks, wrong polarity for IC's and transistors, etc. Refer to Figure 2 for assembly and afterwards carry out a visual examination for such defects and check with a meter for shorts across the supply. Some resistance checks with the power off will give an indication whether all is well or not. Where there are semiconductors in circuit, the reading will be different depending upon which way round the meter leads are applied to the circuit.

For example, a measurement between the 0V line and any of the lines A0-A3, B0-B3 should show high resistance one way and approximately 22Ω the other way. Similar results should be obtained between 0V and the lines SL0-SL2, but RL0-RL3 to 0V should show open circuit until a key is pressed, when the results are much the same as for the others.

These are all static tests, carried out with the chips in place but no supply connected.

When the CPU and keyboard/display modules are connected together (a suitable cableform is available and its pin-outs are shown in Figure 3), and power is applied, the Z80 sends clock pulses to the 8279 IC which then generates the scan voltages on SL0-SL2. These can be detected by using a logic probe in the 'pulse' mode. Naturally, on power-up the display could show almost anything, unless the monitor ROM is fitted, in which case, if all is well, you'll see eight dots to tell you that the monitor's running. Pulse trains should also be found on pins 7, 9-15 inclusive of IC2 (and hence on the bases of TR1-8), and on pins 12-15 of IC3.

Pressing a key in any of the four vertical rows causes pulses to appear on RL0-RL3 respectively. Both CTRL and RESET should send logic low to the CPU card when pressed. To avoid the embarrassment of accidentally resetting the computer, two Reset Keys are provided, which have to be pressed at the same time to force a reset.

The Monitor EPROM

A simple monitor occupies rather less than half of a 2716 EPROM and provides the following facilities:

(a) Access to any memory location to examine and/or modify data,

(b) Step backwards or forwards through memory to examine, edit or enter program data,

(c) Run a program from a given start location.

No sophistication is claimed for the monitor, but it is easy to use with a little practice. It operates as follows.

On power up, eight dots appear on the display to indicate that the monitor is running. Now press CTRL-A. CTRL-A means 'CTRL key plus A key, press down together'. OA now appears in the 'status field' to indicate 'address mode'. Type in an address, which you'll notice goes into the display in 'typewriter' mode, i.e. left entry. If you now enter CTRL-D, the status field changes to Od (data mode), though the dots remain in the data field. However, anything typed on the keypad now will appear in the data field and is entered into memory at the location stated in the address field. Entering CTRL-F takes you forward to the next location, while CTRL-B takes you back to the previous location. CTRL-F and CTRL-B only operate in data mode and always refresh the display, in other words, an address must be entered first before you can move forward or backward from that location. You can use this facility to only examine or edit memory rather than entering a program. Instead of typing in the actual start address, type in the one 'before it'; enter data mode and then use CTRL-F. This takes you to the location you want and brings up the data actually at that location on the display. Now as you step forward or back through memory, you will have a simultaneous display of memory and data. You can look anywhere in ROM or RAM in this way. CTRL-A and CTRL-D allow you to 'toggle' back and forth between the two modes, so you can nip about in memory quite niftily.

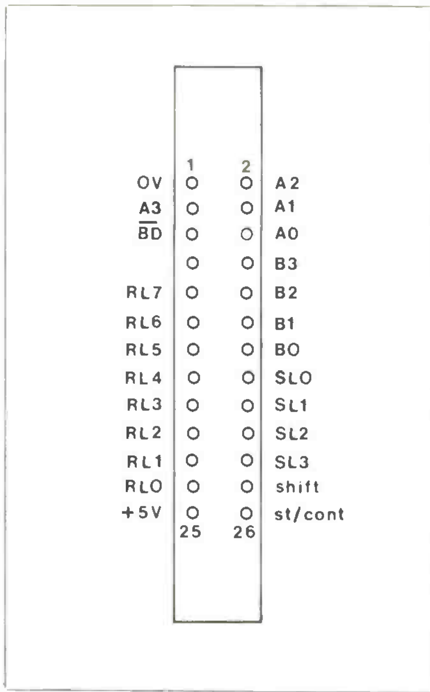


Figure 3. Connections to Z80 CPU.

Note that the data display is only refreshed by CTRL-F or CTRL-B, which means that whenever you toggle back to 'data' mode from 'address' mode the data shown is arbitrary. To see the correct data at the new location, type CTRL-F followed by CTRL-B (or vice-versa).

Thereafter, any memory locations examined by use of either of the latter control keys is correct.

To run a program that you have entered, type CTRL-E and OE appears in the status field. Enter the start address of the program, and operate CTRL-C. The program will now run and eight dashes on the display is the sign that it is doing so. If you put a HALT instruction at the end of your program, it will wait and you can then leave the program by using the 'reset' keys, which will take you back into the monitor. This, of course, allows you to examine any memory locations that might have been modified by the program simply by re-entering the sequence via CTRL-A.

Scratchpad RAM

As with all monitors, this one requires a small amount of RAM for its own use. This it uses for the storage of variables and also as a 'stack' during the running of certain monitor routines. The monitor ROM occupies the addresses &0000 to &07FF; the following area of the memory map is normally occupied by RAM, starting at address &0800. The monitor claims the bytes from &0800 to &085F for its scratchpad. This means that the lowest address at which user programs should be stored is &0860.

Summary of Control Functions

Command	Status Code	Mode & Action
Power-on	8 dots	Monitor ready
Reset	8 dots	Monitor ready
CTRL A	0A	Address mode. Enter address.
CTRL D	0d	Data entry mode, following Address mode only. Enter data.
CTRL F	0d	Step forward one location and refresh display. Data mode only.
CTRL B	0b	Step backward one location and refresh display. Data mode only.
CTRL E	0E	Enter start address of program to execute.
CTRL C	8 dashes	Execute program.

Z80 CPU MODULE HEX KEYPAD KIT PARTS LIST

RESISTORS: All 0·6W 1% Metal Film

R1-8	22Ω	8	(M22R)
R9-16	470Ω	8	(M470R)

SEMICONDUCTORS

TR1-8	BC327	8	(QB66W)
IC1	ULN2803A	1	(QY79L)
IC2,3	74LS138	2	(YF53H)

MISCELLANEOUS

DY1-8	½in. Display Type 1	8	(FR39N)
	PCB	1	(GD40T)
S1-19	Keyboard Switch	19	(FF61R)
	Pin 2145	1 Pkt	(FL24B)
	Keytop 1 Position	18	(FF62S)
	Keytop 2 Position	1	(FF63T)
	DIL Socket 16-pin	2	(BL19V)
	DIL Socket 18-pin	1	(HQ76H)
	Constructors' Guide	1	(XH79L)

OPTIONAL

PL1	Keypad Cableform	1	(FP63T)
	EPROM 2716/M12	1	(UH87U)

The above items (excluding Optional) are available as a kit.
Order As LM18U (Z80 Hex Keypad Kit)

The following items (which are included in the kit) are also available separately.
Keypad PCB **Order As GD40T**
Keypad Cableform **Order As FP63T**
EPROM 2716/M12 **Order As UH87U**

Weather Satellite Down Converter

by Robert Kirsch Part 1

PLEASE NOTE THAT THIS PROJECT AND THE WEATHER SATELLITE SERIES WILL BE UPDATED IN THE NEAR FUTURE!

A system for receiving and decoding data from polar orbiting weather satellites in the 137MHz band has been described in previous issues of 'Electronics'. This article describes a Down Converter that may be connected directly to this receiving system to enable it to receive S band signals from Meteosat 2, a geostationary weather satellite.

The Meteosat System

Meteosat forms one of a chain of five geostationary weather satellites located above the equator that cover the entire world, see Figure 1. Meteosat 1 was launched during November 1977 by the European Space Agency (E.S.A.) and continued to work until November 1979, being replaced in June 1981 by Meteosat 2, which is still in service.

This satellite is located on the Greenwich meridian, nearly 36,000km above the equator (0 degrees N, 0 degrees E) and orbits on the same axis as the Earth at a speed that maintains it in a fixed position above the Earth (geostationary orbit).

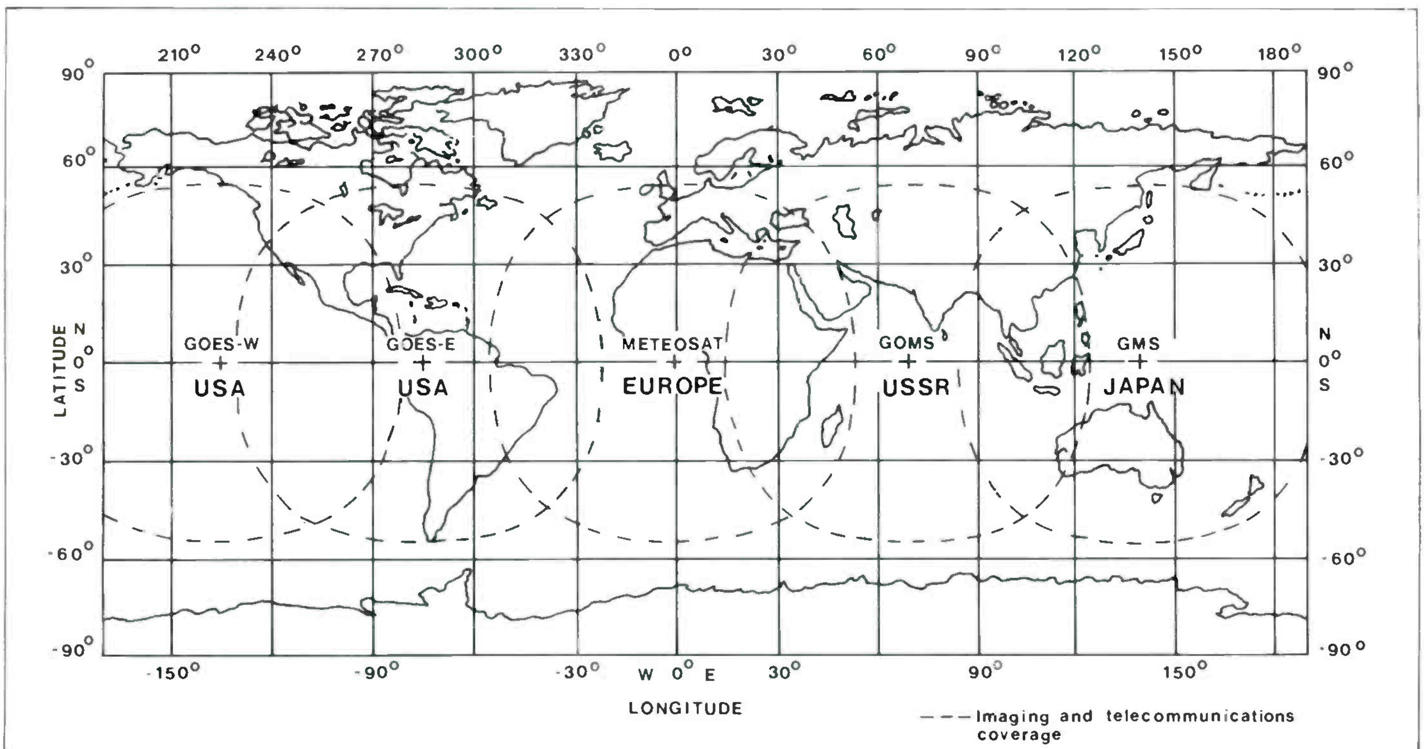
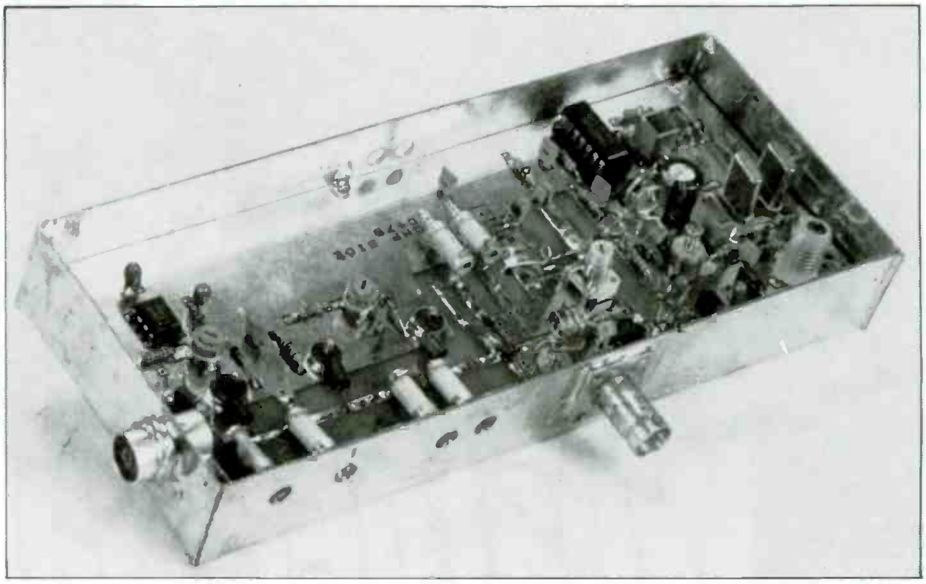


Figure 1. Weather Satellite system.

The Meteosat satellite system differs from the polar orbiting satellites in that the higher orbit of Meteosat enables it to see the whole disk of the Earth at once, thus it can make constant observations of the area covered. The satellite looks at the Earth through a device called a multispectral radiometer which provides four separate images, two visible and two in different infra-red spectral bands. One infra-red channel gives a thermal image of the Earth, the other responds to the water vapour absorption band giving an indication of the levels of atmospheric humidity.

The raw data from the satellite's radiometer is transmitted to the Earth for processing in the S frequency band (1670-2110MHz). The receiving station for these signals, called the DAATS (Data Acquisition, Telemetry and Tracking Station), is located in the Federal Republic of Germany, about 50km from Darmstadt. Signals received are fed to the Meteosat Ground Computer System at the European Space Operations Centre in Darmstadt, see Figure 2.

The data from the satellite is processed by a pair of large mainframe Siemens computers from which information is fed by land-line to world wide users and also back to the DAATS for re-transmission to the satellite in WEFAX format with coastline added. The satellite re-radiates this information on two S band frequencies, channel A at 1694.5MHz and channel B at 1691MHz, these are the signals received by the Down Converter described in this article. These signals are then translated to 137.5MHz by the Down Converter ready for detection and decoding by the Mapsat receiving system.



The Down Converter.

The Complete System

Figure 3 shows a block schematic of the complete receiving system, details of the Pre-amplifier, Down Converter and Channel Switching Unit are shown. Information on the Mapsat receiver and decoder can be found in issues 18 and 20 of 'Electronics' (XA18U and XA20W), which are available as back issues, see inside back cover.

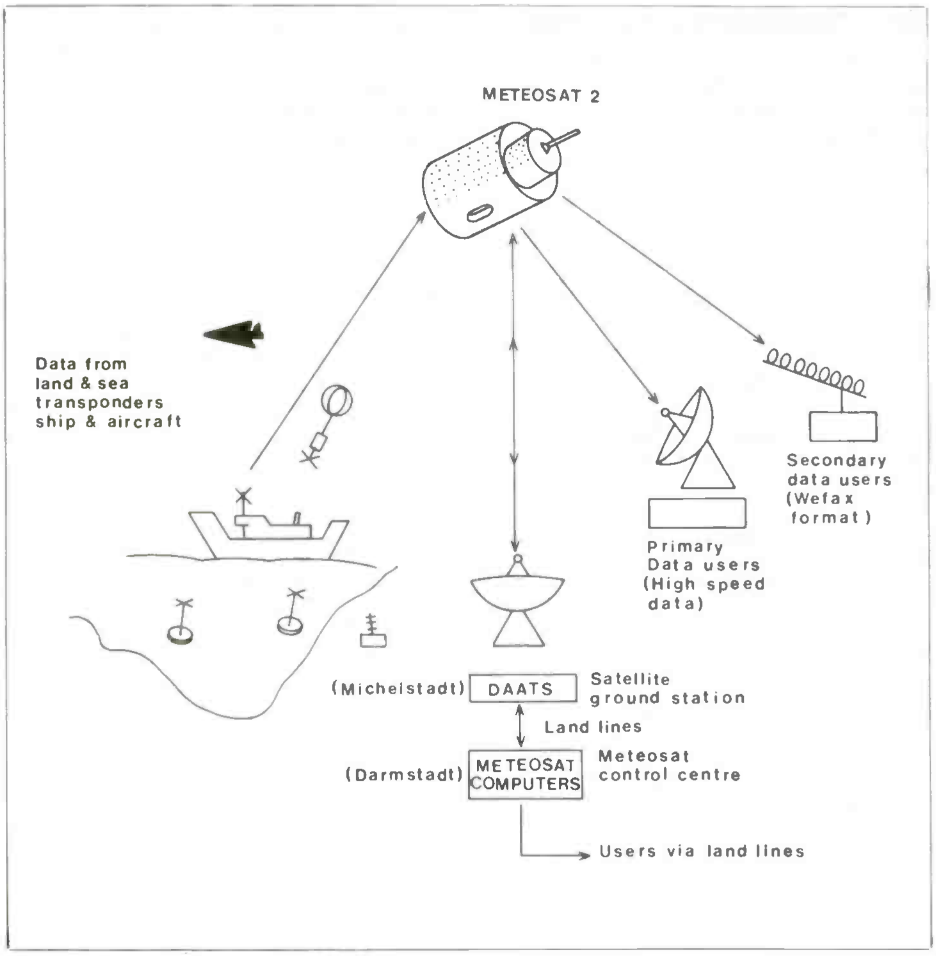


Figure 2. Meteosat 2 receiving station.

Aerial

The aerial used is of the loop-yagi type which is fairly unusual for this kind of application, but it has several advantages over more conventional satellite receiving aerials. The relatively small size of this aerial makes mounting it a simple exercise as it can be attached to most types of standard television aerial masts, its small surface area producing a low wind resistance. The beam-width of this system is fairly broad and aligning is quite easy. The complete receiving system may be operated from batteries in the vicinity of the aerial during adjustment and this enables the aerial to be aimed for the best signal from the satellite. This type of aerial is equivalent to a small dish and has a sharp horizontal polarisation. Due to the low level and high frequency of the signals received it is important to keep feeder losses as small as possible and for this reason a high gain low noise pre-amplifier is mounted directly below the receiving element of the aerial.

Pre-amplifier

Figure 4 shows the essential circuit diagram of the aerial pre-amplifier, which, due to the nature of its application, is only available as a complete ready-built module. The circuit is a straightforward three stage design, having for the first two stages low noise GASFET transistors, TR1 and TR2, which combine both high gain with a low noise level of less than 1dB. The third stage is a moderately low noise MMIC (Monolithic Microwave Integrated Circuit), IC1, to provide sufficient overall gain in the region of 35dB. This combination ensures that the gain of the twin GASFET front-end will be enough to completely overcome the noise contribution of the MMIC and deliver something very close to the specified noise figure of the input GASFET, TR1, for overall noise, namely 0.5dB.

TR1 and TR2 are source biased for

simplicity and device protection, and are powered from a 5V regulator IC. The MMIC is powered from the 'raw', 12V supply via its biasing resistor R4. Series inductors are used in the matching network at each gate and simple moulded chokes are used for gate DC return. Interstage impedance matching is provided by the stripline 'L' networks which combine both L3 and C4, and L6 and C8 respectively.

As with any microwave circuit, this schematic is only part of the design since the physical dimensions of the construction are significant fractions of the electrical wavelengths handled. For this reason it is not possible to copy the circuit from scratch, nor should the assembled and aligned module be 'fiddled with'!

Down Converter

The incoming signals from the pre-amp are coupled via C1 (see Figure 5) to the input stripline of TR1; this forms an amplifier virtually identical to that of the pre-amp. The output from this stage is coupled to L7, the input stripline of the mixer GaAs FET TR2. This line is tuned by VC4 and VC5. The dual gates of this FET are also fed via C5 with signals from the oscillator/multiplier stages. The resultant output from the mixer at 137.5MHz is tuned by L6 and VC6 and fed to the output socket via C9.

Two crystals are provided to enable both channels of the satellite to be received without re-tuning the 137.5MHz receiver. The switching of these crystals is accomplished by interrupting the power to the Down Converter for set periods. The logic that detects these pulses and switches the crystals is formed by IC2 and its surrounding components. TR5 is the crystal oscillator whose collector circuit, VL1 and C23, is tuned to the overtone frequency of the selected crystal. The output from this stage is fed to the multipliers formed by TR4 and TR5 producing an output of 18 times the crystal frequency. This signal is filtered by L9 and L10 before being fed to the gate of the mixer via C5.

A 9 volt supply is fed from the Mapsat receiver via the channel switching unit and downlead coax and is isolated from the 137.5MHz signal by L5. This supply feeds the oscillator stage direct and the rest of the Down Converter via the 5 volt regulator RG1. The negative supplies for the two GaAs FETs is generated by IC1. The 5 volts required by the pre-amp is fed to SK1 via LK1 and L14 which isolates the signal path from the power supply.

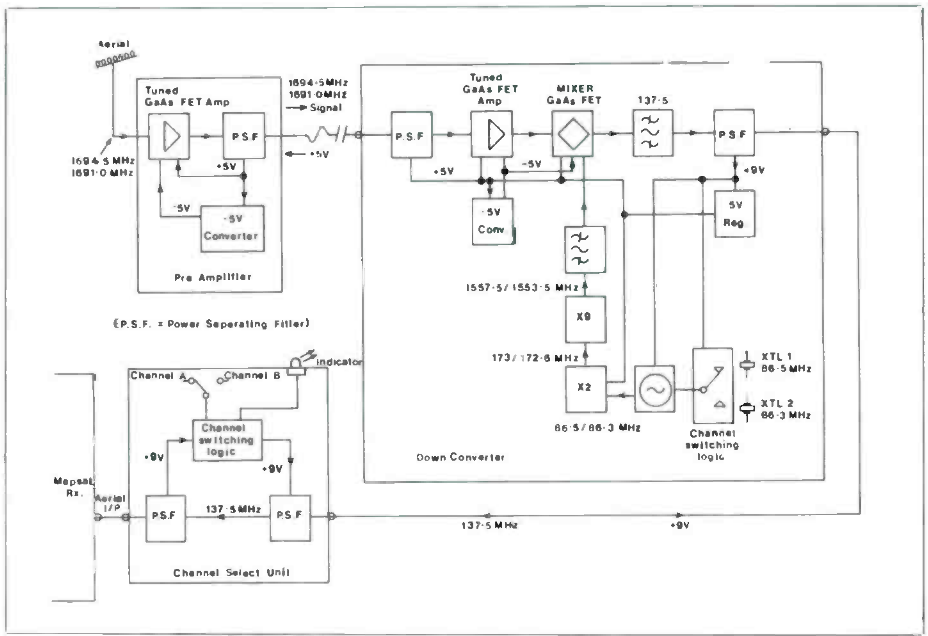


Figure 3. Block schematic of the system.

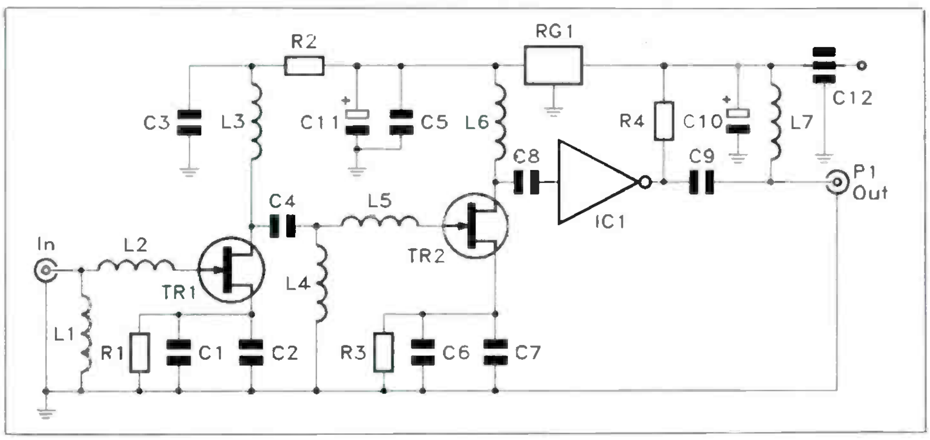


Figure 4. Pre-amplifier circuit.

Power Supplies

The power for both the Down Converter and the Aerial Pre-amplifier comes from the power unit in the Mapsat receiver via the same coaxial cable that carries signals to the receiver from the Down Converter. This simplifies the installation of the system as only one cable connects all the units together. The Channel Switching Unit (to be described in the next issue) is connected in series

The aerial showing the attached pre-amplifier.

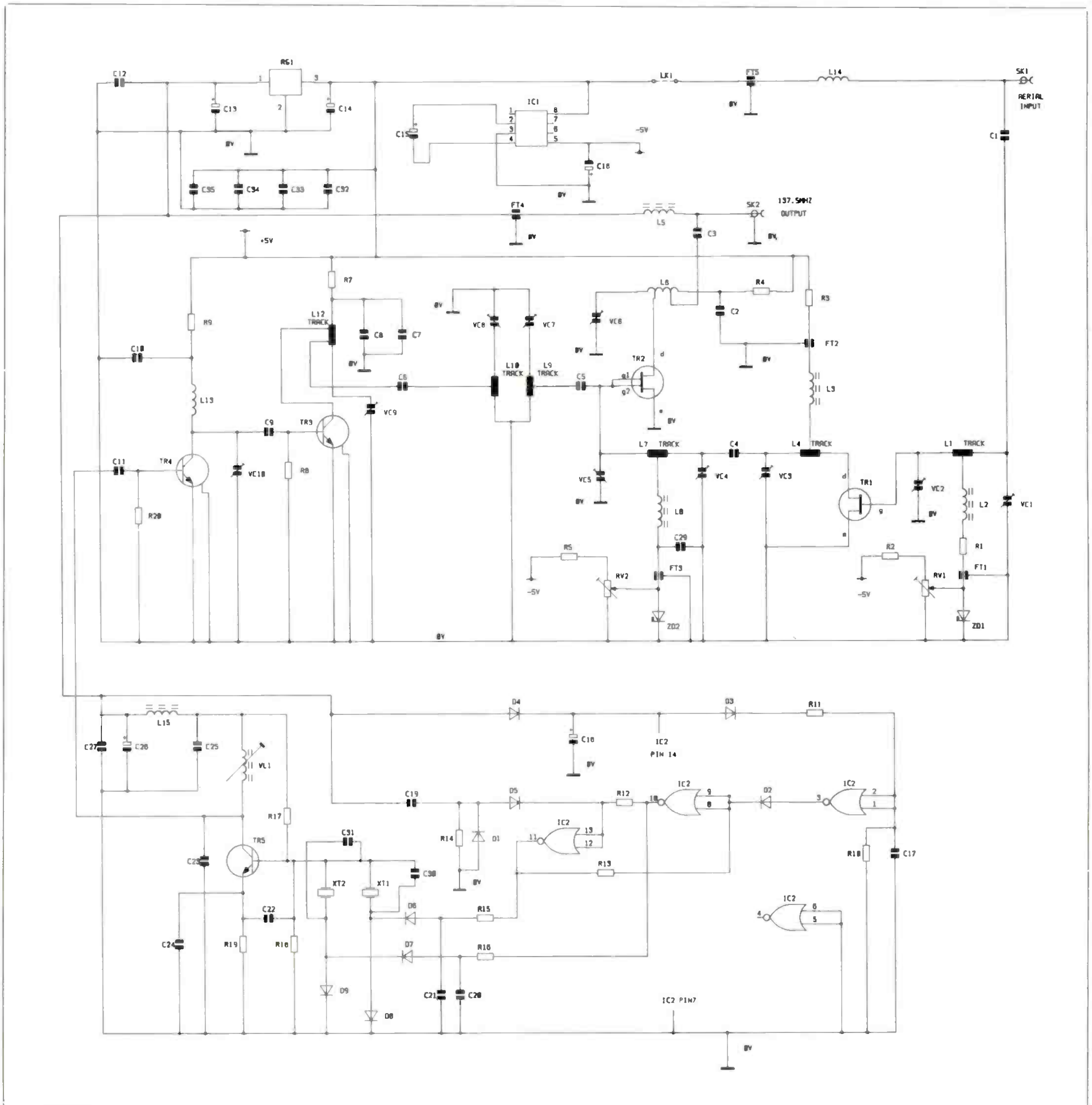


Figure 5. Down Converter circuit.

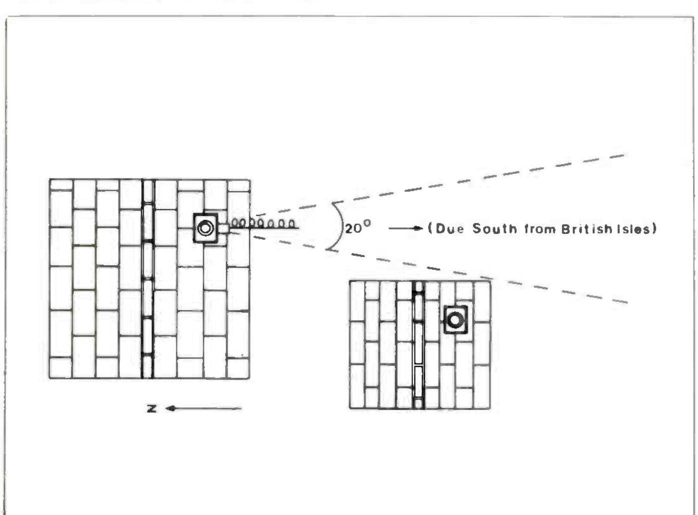


Figure 6. Aerial azimuth.

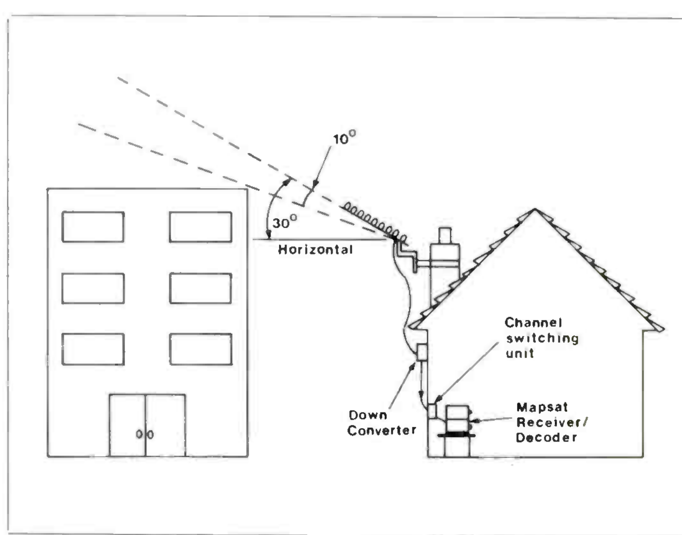


Figure 7. Aerial elevation.

with this cable at any convenient point near the Mapsat receiver.

Sighting the Aerial

The aerial *must* be positioned in such a way that it can point at clear sky in a due south direction (from the British Isles), see Figure 6, at an angle of 30 degrees above the horizon. There should also be no major obstructions below about 10 degrees of the aerial axis, see Figure 7. No mast or other aerial should be within 1 metre of the aerial other than directly behind, where there is no restriction. The angle of elevation is set by the special clamping bracket supplied with the aerial when the support mast is mounted vertically. Small changes in elevation may be affected by setting the mast off vertical in the appropriate direction. The heading and elevation of the aerial when used in other countries may be calculated from the position of the satellite above the Earth and the known height of 36,000km. Note this may require special mounting arrangements to be made. Further details of the aerial assembly and installation will be given in Part 2.

Kits

There are 3 main kits available for this project. The Down Converter kit, an Aerial and Pre-amplifier kit, and the Channnel Switching Unit kit. The Down Converter and Pre-amplifier are supplied ready-built, tested and aligned. The Channel Switching Unit and the Aerial are all that can be constructed by the hobbyist. In addition, other items will be required to complete the installation, these include a mast for the aerial, a water proof box for the Down Converter (if it is to be sited out of doors), various connectors and cable for connecting the Down Converter to the Mapsat receiver via the Channel Switching Unit. The aerial *must* be coated with a good quality polyurethane clear varnish before installation, and this can usually be obtained from your local DIY shop.

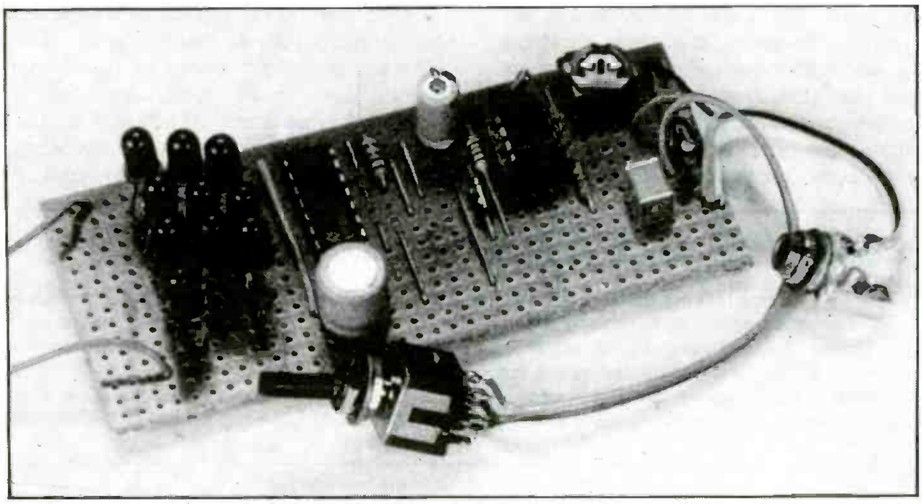
The Down Converter may be used with other aerial systems provided they produce a sufficient signal level, and do not have DC continuity across their output terminals as the Down Converter supplies power to the Pre-amplifier via the coaxial cable inner and outer conductors.

MINI-CIRCUITS

from Robert Penfold

Audio Level Tester

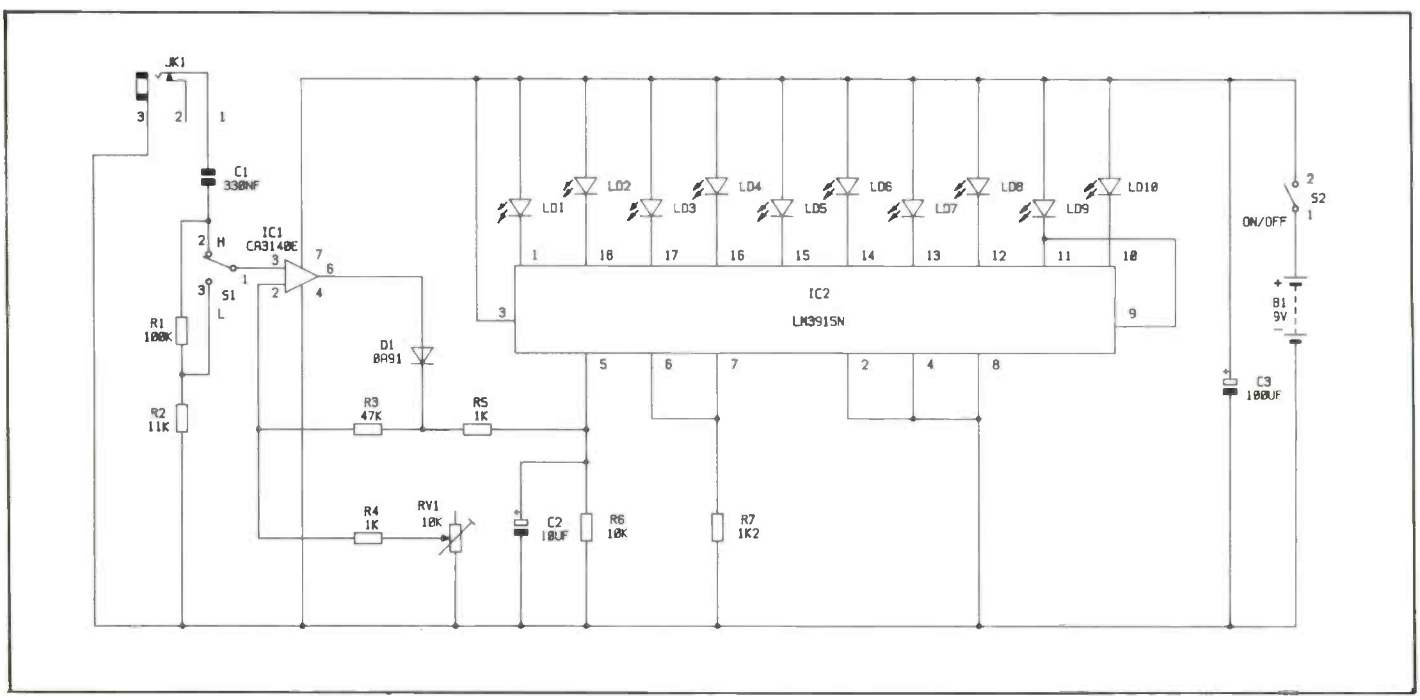
This device is intended as an aid to checking audio equipment, and it is a form of audio signal tracer. However, rather than the usual audio amplifier feeding an earphone or a miniature loudspeaker, this design has a ten LED bargraph display with the LEDs at 3dB intervals on the scale. The unit therefore provides a fairly accurate indication of the audio level present in the circuit, provided it is within the measuring range of the unit of course. When used in conjunction with an audio signal generator or function generator this makes the unit suitable for such things as voltage gain measurement and frequency response testing. In other words, it is in many ways more like a very basic audio millivolt meter than a conventional audio signal tracer, and it can often provide more meaningful results than an ordinary signal tracer. The user should be aware of the devices's weaknesses though, and these are mainly that it will not detect very low level signals, and it does not give any idea of the waveform that it is measuring. It is therefore useless trying to test something like a low impedance dynamic microphone (with its sub 1 millivolt RMS output level) using this tester, or searching for the stage in an amplifier which is producing clipping. Like virtually any simple piece of test gear, it can be very helpful and worthwhile if used thoughtfully and in the right context, and worse than useless if it is not.



Circuit Operation

The circuit is similar to that of a LED audio level indicator, and it could actually be used in this role if desired. Basically the unit comprises a precision half wave rectifier driving a smoothing circuit and a bargraph driver.

Looking at the circuit in more detail, IC1 is the active device in the precision rectifier, which is a simple half wave type. For reasonably accurate results with low level signals, it is essential to use some form of active circuit in order to overcome the inherent non-linearity of semiconductor diodes. This lack of linearity is most severe with silicon devices which require a forward bias of about 0.5 volts or so before they will start



to conduct significantly. In this circuit a germanium type is used, and although these offer much improved performance on low level signals, they still need some external assistance in order to give really good results.

In this circuit, the standard approach of including the diode in the negative feedback look of an amplifier has been adopted. This boosts the output voltage of the amplifier by an amount which is equal to the voltage drop through the diode, and this exactly compensates for the forward voltage drop. The gain of the amplifier has been made variable by means of RV1 to permit calibration of the unit. R1 and R2 form a simple 0dB/−20dB attenuator at the input of the unit, and this enables the sensitivity of the unit to be cut by a factor of ten so that relatively high level signals can be accommodated.

The bargraph driver is an LM3915N logarithmic type which has 3dB increments between the LEDs. Here it is wired in the 'dot' mode and it does not provide a true bargraph display. This gives slightly less clear results, but the current consumption is substantially reduced so that the unit can be powered from a small 9 volt battery if desired. C2 smooths the input signal to the bargraph driver, and this is important as the signal would otherwise be varying at a very fast rate making the display dim and blurred. R7 sets the LED current at approximately 10 milliamps.

Construction

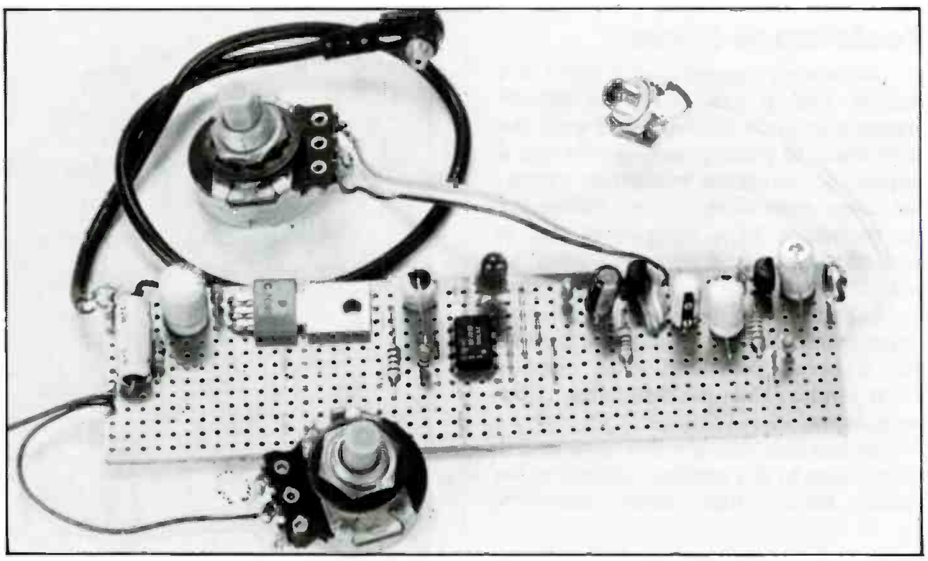
There are two basic constructional forms that can be used with a device of this type. The first method is to build it as a conventional instrument with a set of screened test leads which connect to JK1. The second approach, and perhaps the better one, is to build the unit as a probe, with C1 connecting to the probe tip and an earthing lead fitted with a crocodile clip connecting to the negative supply rail (JK1 then being omitted). The gain of the circuit is not particularly high, and at 111k the input impedance is only moderately high, and the component layout is not all that critical, although reasonable care needs to be taken with the layout of any fairly sensitive audio circuit. On the prototype the LEDs are ten separate 5 millimetre types. A proper bargraph can be used if perferred and will give the unit a neat appearance, albeit at a greater cost.

RV1 is adjusted to give the desired full scale sensitivity, which can be anything from under 100 millivolts peak to peak, to over 800 millivolts peak to peak with S1 in the 'H' position. With it set to the low sensitivity 'L' position the full scale voltage is boosted by a factor of ten. The best setting to use is to some extent a matter of personal preference, and also depends to some degree on the equipment that will be tested, but a full scale value of about 200 millivolts peak to peak with S1 at the 'H' setting probably gives coverage of the most useful dynamic range.

Sound Triggered Flash

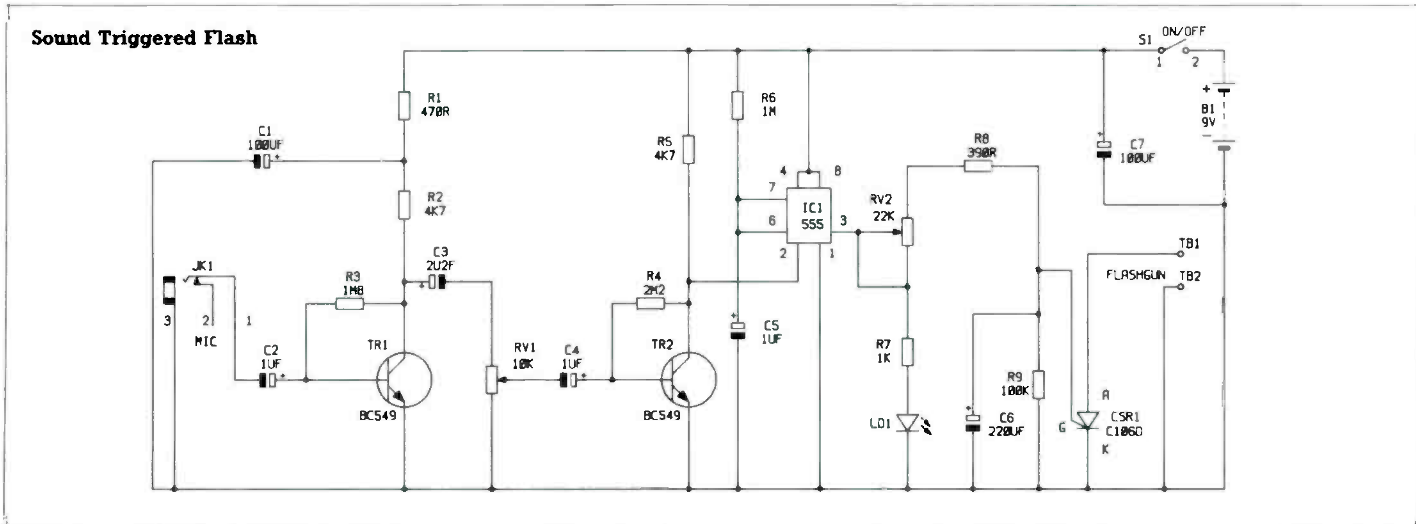
With the aid of an automatic flashgun trigger, it is possible to take a variety of action shots that would be practically impossible by any other means, and which would take innumerable attempts in order to catch just the right moment. The most popular form of automatic trigger is probably the sound triggered type. These are suitable for shots such as balloons in mid-burst, water splashes, champagne corks 'popping', etc., and the very short duration of the output from an electronic flashgun usually 'freezes' the action well enough to give a sharp image. Of course, shots of this type must be taken under quite dark conditions as the camera's shutter must be set to 'Bulb' and left open for a few seconds while the exposure is made, and shots really need to be carefully set up and rehearsed if a lot of wasted film is to be avoided.

Circuit Operation



This unit is designed primarily for use with a low impedance dynamic microphone (the type used with inexpensive cassette recorders), although it will function reasonably well with crystal, high impedance dynamic types, or types which have similar output characteristics to the latter. With any of these types of microphone the output level is only going to be something in the region of 1 millivolt, and a considerable amount of amplification is needed in order to boost the signal to a level that can drive a switching device of some

Sound Triggered Flash



kind. In this case a two stage common emitter amplifier is used, and this provides over 80dB of voltage gain. RV1 is the gain control, and it is generally advisable to have the gain no higher than is really necessary as this would almost certainly result in frequent spurious triggering of flashgun, and possibly a lot of wasted frames.

The output from the amplifier is used to trigger a monostable based on 555 timer device IC1. This gives a nominal output pulse duration of 1.1 seconds, and it activates LED1 to show that the unit has been triggered properly. This is of little value in normal use when the flashgun will obviously fire if the unit is triggered properly, but it is very useful for 'dry' runs when preparing to take a shot. The output pulse from IC1 is also used to trigger a thyristor which in turn activates the flashgun. RV2, R8 and C6 provide a variable delay before triggering occurs, and this is useful for taking a sequence of shots to show (say) a balloon at the instant it starts to burst and the first tears that start to appear in it, through to the point where it has fully collapsed. Note that the distance from the sound source to the microphone also introduces a small delay, and for rapid triggering this distance should be as short as possible. If very fast triggering is required it would also be advantageous to include a switch to permit C6 to be cut out of circuit.

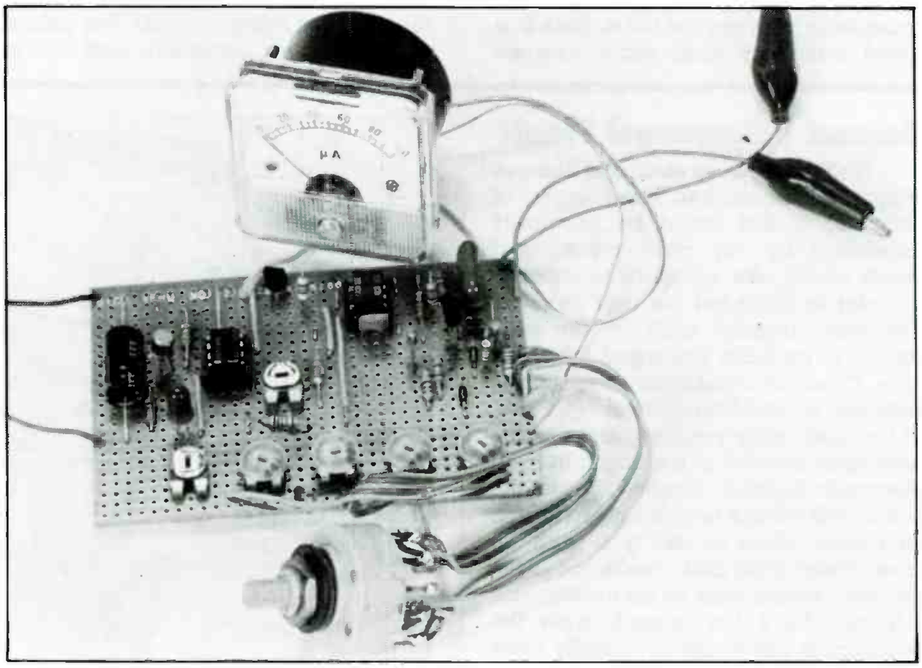
Construction

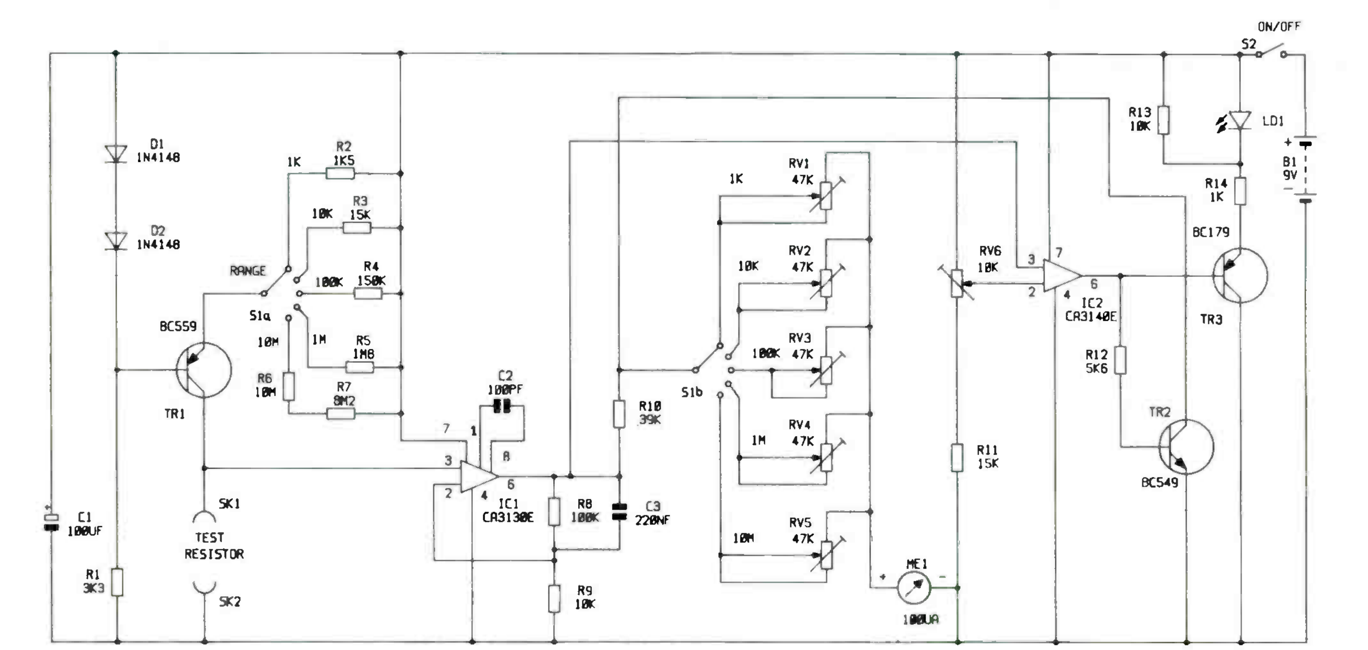
The amplifier section of the unit has quite a high level of voltage gain, and consequently the component layout for this section of the unit needs to be designed with suitable care. An output socket to suit the miniature coaxial plugs fitted to flashgun leads might be obtainable from a large camera accessory stockist, but a simple alternative is to use a socket and short piece of lead cut from a flash extension lead. The lead must be connected through to CSR1 with the correct polarity, and this is something that can be checked using a multimeter set to a high DC voltage range. CSR1 will not be damaged if the lead is connected with the wrong polarity, and it is therefore quite alright to determine the correct method of connection using trial and error. Note that the circuit will only function properly with a sensitive thyristor such as the C106D, which will trigger from a gate current of well under a milliamp, which precludes most other devices from use in the unit.

In-Circuit Resistance Meter

Accurately measuring the value of a resistor that is out of circuit usually presents no great problem, and even the most basic of multimeters usually has a number of resistance measuring ranges that cover most likely values. Measuring the resistance of a component that is in-circuit is a very different proposition, and there is the problem of other components in the circuit placing resistance in parallel with the component that is being measured. This gives a lower reading than the true value of the component.

In modern circuits this problem is largely due to the semiconductors in the circuit rather than other resistors.





In-circuit Resistance Meter

Semiconductor junctions (which can be in diodes, transistors, integrated circuits, or practically any semiconductor component) are often forward biased by the voltage source in the multimeter, and this results in a very low reading. Sometimes simply reversing the test leads will cure the problem, but although this might eliminate the original junction by reverse biasing it, it may well forward bias a second junction and re-introduce the problem.

The most simple solution to the problem is to use a resistance measuring circuit that operates with a maximum voltage across the test resistance that is too low to bring a silicon semiconductor junction beyond the threshold of conduction. This means keeping the voltage down below about 0.4 volts, which is substantially less than that used by most resistance measuring devices. It must be stressed that this system should eliminate the shunting effect of semiconductor junctions, but it will not cut out any pure resistance in parallel with the component under test. Very often there will be no significant resistances of this kind, but if a low reading is obtained one lead of the resistor should be disconnected from the board to isolate the component from any parallel resistance, and the check then repeated. Of course, if an excessive resistance reading is obtained this certainly means that the device under test is a 'dud' (or you have forgotten to switch off the supply before starting work on the equipment!).

Circuit Operation

The unit operates on the well established principle of supplying a constant current to the test component, and then measuring the voltage developed across it (which is proportional to its resistance value). A point in favour of this system is that it gives a forward reading linear scale rather than the reverse reading logarithmic type of a conventional analogue instrument.

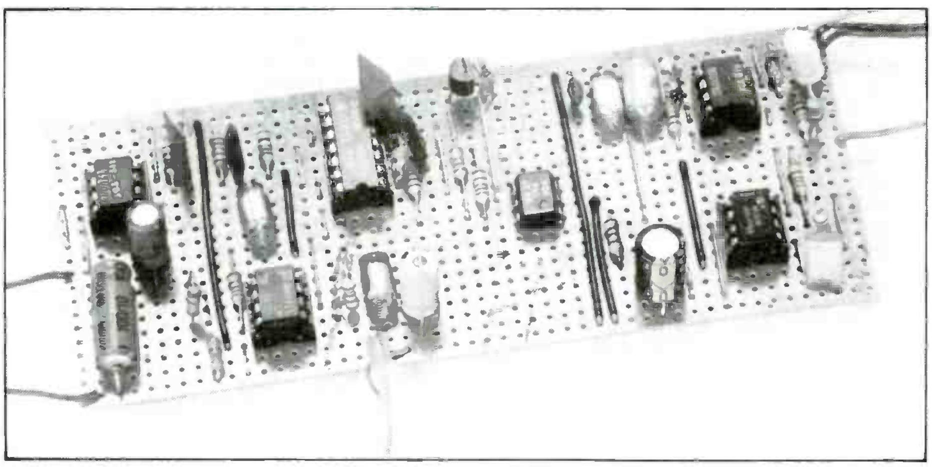
In this circuit TR1 operates as a standard constant current source, and the five switched emitter resistances provide five output currents. These give the unit measuring ranges having full scale values of 1k, 10k, 100k, 1M and 10M. Normally with circuits of this type low resistance ranges cause problems as they require quite high test currents. In this case the currents are relatively low, and it would be acceptable to add a 100Ω range if desired by using the otherwise unused position of S1 to switch a 150Ω resistor into circuit. This still only gives a test current of about 4.3 milliamps. An advantage of the low test currents and voltages is that there is little risk of damaging anything in the circuit under test. On the other hand, it does mean that the voltmeter section of the unit must have an extremely high input impedance in order to give good accuracy, particularly on the higher ranges. This requirement is fulfilled by using an operational amplifier having a MOSFET input stage to provide buffering and a certain amount of voltage amplification. C3 rolls off the frequency response of this stage to avoid problems with stray pick-up of mains 'hum' and other noise. There is a separate preset resistor for each range in the voltmeter circuit (switched by S1b) so that each range can be individually calibrated.

With no test resistance the meter will be driven beyond full scale deflection. In order to avoid this, IC2 is used to detect an excessive output voltage from IC1 and to switch on TR2 which diverts the output current from the meter. This could cause confusion, where it is unclear whether the meter is showing a low and valid reading, or an overload is being suppressed. LED1 avoids this by lighting up when a valid reading is present.

Construction

The unit offers little that is difficult as far as construction is concerned, but bear in mind that both integrated circuits are MOS types and consequently require the usual antistatic handling precautions. For calibration purposes five 1% resistors are required, and these should have values equal to the full scale values (1k, 10k, etc). It is just a matter of connecting the appropriate resistor for the range that is being calibrated, and then adjusting the correct preset for precisely full scale reading on the meter. RV6 is given any setting that permits a full scale reading to be obtained, but which also suppresses the meter with no test resistor connected, and its exact setting will probably not be critical.

Audio Isolator



It is sometimes necessary to couple an audio signal from a piece of equipment without making any direct connection to the equipment. The most common example of this is where a headphone or tape output is required from a television or other item of gear which has a 'live' chassis, and the isolation is required to avoid problems with electric shocks and short circuits to Earth. Another use for an audio isolator is where (say) a number of musical instruments are connected together and there are problems with 'hum' loops. the easiest solution to the problem is to use an isolation transformer, but in practice suitable components seem to be unobtainable. Where an output is required from a circuit that has a 'live' chassis, even if a transformer with the right input/output characteristics could be obtained, that is not to say that it would necessarily guarantee to withstand a few hundred volts without breaking down.

What is generally a more satisfactory solution to the problem is to use an opto-isolator plus some simple electronics to provide the signal transfer, and that is precisely what this circuit does. It has to be emphasised that if the unit is to be used to provide an isolated output on equipment which has a 'live' chassis, it is essential that the person installing it knows exactly what they are doing and have the requisite experience to undertake this task. This is quite definitely not a beginners project, and is potentially very dangerous if used incorrectly with 'live' chassis equipment.

Circuit Operation

The obvious method of using the audio input signal to vary the input current to the LED in the opto-isolator, and then taking the audio output from a load resistor on the transistor side of the device does not generally work well in practice. The two problems that compromise results are rather poor linearity through the system, and a vulnerability to the pick-up of 'hum' and other electrical noise. Better results are usually obtained using an ultrasonic carrier wave having either pulse width or frequency modulation. In this circuit frequency modulation is used.

On the input side of the unit IC1a acts as a buffer amplifier which gives an input impedance of about 50k, and IC1b is the buffer stage in an active third order lowpass filter. This prevents high frequency signals from reacting with the carrier signal to give distortion products

AUDIO LEVEL TESTER PARTS LIST

RESISTORS: All 0·6W 1% Metal Film (Unless specified)			
R1	100k	1	(M100K)
R2	11k	1	(M11K)
R3	47k	1	(M47K)
R4,5	1k	2	(M1K)
R6	10k	1	(M10K)
R7	1k2	1	(M1K2)
RV1	10k Hor Encl Preset	1	(UH03D)
CAPACITORS			
C1	330nF Poly Layer	1	(WW47B)
C2	10μF 50V PC Electrolytic	1	(FF04E)
C3	100μF 10V PC Electrolytic	1	(FF10L)
SEMICONDUCTORS			
IC1	CA3140E	1	(QH29G)
IC2	LM3915N	1	(YY96E)
D1	OA91	1	(QH72P)
D2-11	Red LED	10	(WL27E)
MISCELLANEOUS			
JK1	3·5mm Jack Socket	1	(HF82D)
S1	SPDT Min Toggle	1	(FH98G)
S2	SPST Min Toggle	1	(FH97S)
B1	9V PP3 Battery	1	(FK62S)
	PP3 Battery Clip	1	(HF28F)
	8-pin DIL Socket	1	(BL17T)
	14-Pin DIL Socket	1	(BL18U)

IN-CIRCUIT RESISTANCE METER PARTS LIST

RESISTORS: All 0·6W 1% Metal Film (Unless specified)			
R1	3k3	1	(M3K3)
R2	1k5	1	(M1K5)
R3,11	15k	2	(M15K)
R4	150k	1	(M150K)
R5	1M8	1	(M1M8)
R6	10M	1	(M10M)
R7	8M2	1	(M8M2)
R8	100k	1	(M100K)
R9,13	10k	2	(M10K)
R10	39k	1	(M39K)
R12	5k6	1	(M5K6)
R14	1k	1	(M1K)
RV1-5	47k Hor Encl Preset	5	(UH05F)
RV6	10k Hor Encl Preset	1	(UH03D)
CAPACITORS			
C1	100μF 10V PC Electrolytic	1	(FF10L)
C2	100pF Ceramic	1	(WX56L)
C3	220nF Poly Layer	1	(WW45Y)
SEMICONDUCTORS			
IC1	CA3130E	1	(QH28F)
IC2	CA3140E	1	(QH29G)
TR1	BC559	1	(QQ18U)
TR2	BC549	1	(QQ15R)
TR3	BC179	1	(QB54J)
D1,2	1N4148	2	(QL80B)
LD1	Red LED	1	(WL27E)
MISCELLANEOUS			
S1	6-Way 2-Pole Rotary Switch	1	(FH45W)
S2	SPST Sub-Min Toggle	1	(FH97F)
ME1	100μA Panel Meter	1	(RW92A)
SK1,2	2mm Socket	2	(HF42X)
B1	9V PP3 Battery	1	(FK62S)
	PP3 Battery Clip	1	(HF28F)
	8-Pin DIL Socket	2	(BL17T)

SOUND TRIGGERED FLASH PARTS LIST

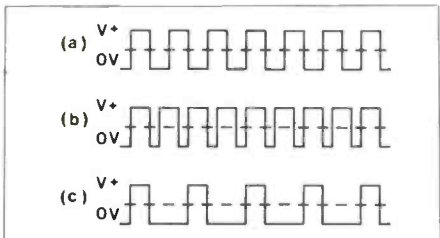
RESISTORS:All 0·6W 1% Metal Film (Unless specified)			
R1	470Ω	1	(M470R)
R2,5	4k7	2	(M4K7)
R3	1M8	1	(M1M8)
R4	2M2	1	(M2M2)
R6	1M	1	(M1M)
R7	1k	1	(M1K)
R8	390Ω	1	(M390R)
R9	100k	1	(M100K)
RV1	10k Log Pot	1	(FW22Y)
RV2	22k Lin Pot	1	(FW03D)
CAPACITORS			
C1,7	100μF 10V PC Electrolytic	2	(FF10L)
C2,4,5	1μF 100V PC Electrolytic	3	(FF01B)
C3	2μ2F 100V PC Electrolytic	1	(FF02C)
C6	220μF 16V PC Electrolytic	1	(FF13P)
SEMICONDUCTORS			
IC1	NE555	1	(QH66W)
TR1,2	BC549	2	(QQ15R)
LD1	LED Red	1	(WL27E)
CS1	C106D	1	(QH30H)
MISCELLANEOUS			
S1	SPST Min Toggle	1	(FH97F)
B1	9V PP3 Battery	1	(FK62S)
SK1	3·5mm Jack Socket	2	(HF82D)
SK2	Min Coax Socket	As req.	*(See text)*
	8-Pin DIL Socket	1	(BL17T)
	Microphone	As req.	

AUDIO ISOLATOR PARTS LIST

RESISTORS: All 0·6W 1% Metal Film			
R1,2	100k	2	(M100K)
R3,4,5,7,11,12,13,14,15	10k	9	(M10K)
R6	4k7	1	(M4K7)
R8	1k2	1	(M1K2)
R9	470Ω	1	(M470R)
R10	2k2	1	(M2K2)
R16,17	47k	2	(M47K)
R18	2k7	1	(M2K7)
R19	5k6	1	(M5K6)
CAPACITORS			
C1,15	100μF 10V PC Electrolytic	2	(FF10L)
C2	1μF 100V PC Electrolytic	1	(FF01B)
C3	2n2F Poly Layer	1	(WW24B)
C4	3n3F Poly Layer	1	(WW25C)
C5	220pF Polystyrene	1	(BX30D)
C6	330pF Polystyrene	1	(BX31J)
C7,9,14	1nF Poly Layer	3	(WW22Y)
C8	680pF Polystyrene	1	(BX34M)
C10	4n7F Poly Layer	1	(WW26D)
C11	180pF Ceramic	1	(WX59P)
C12	4μ7F 63V PC Electrolytic	1	(FF03D)
C13	10μF 50V PC Electrolytic	1	(FF04E)
SEMICONDUCTORS			
IC1	1458C	1	(QH46A)
IC2	NE555	1	(QH66W)
IC3	4001BE	1	(QX01B)
IC4,5	μA741C 8-Pin DIL	2	(QL22Y)
OP1	Opto Isolator	1	(WL35Q)
MISCELLANEOUS			
JK1	3·5mm Jack Socket	1	(HF82D)
	8-Pin DIL Socket	5	(BL17T)
	14-Pin DIL IC Socket	1	(BL18U)

on the output. The carrier oscillator is a 555 astable circuit which is modulated by the normal means of coupling the audio input signal to pin 5. The carrier frequency is around 100kHz incidentally. IC2 drives the LED in the opto-isolator via current limiting resistor R8.

Unfortunately, ordinary opto-isolators are not particularly fast in operation, and the output signal across load resistor R9 is a fairly weak triangular waveform. This is amplified by TR1 though, which gives a virtual square wave output signal having quite fast rise and fall times. The F.M. demodulator is a monostable circuit which is formed by two of the 2 input NOR gates of IC3. A monostable might seem to be an unlikely form of F.M. demodulator, but it can work very well in this roll. Under quiescent conditions the output waveform of the monostable is a squarewave signal, as shown in (a) of the



Waveform

waveform diagram. If the input frequency is increased, the output pulse length remains the same, and the gap between the pulses narrows (as in (b)). With reduced input frequency the duration between output pulses increases, giving a waveform of the type shown in (c). The point here is that the average output voltage varies in sympathy with changes in the input frequency, and there is a linear relationship between the two.

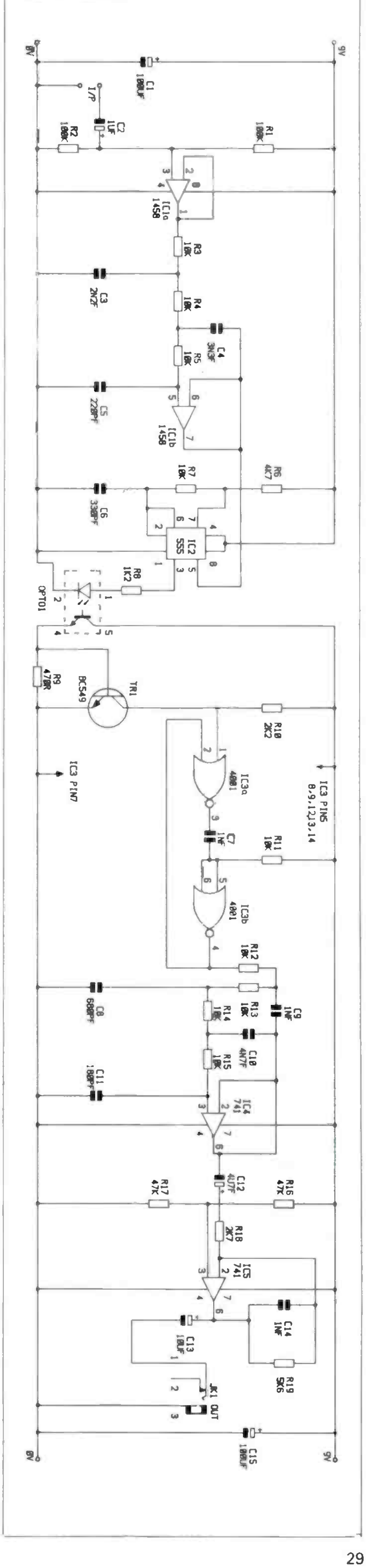
In order to recover the demodulated audio signal a lowpass filter is needed, and it must give a high degree of attenuation in order to give an output signal that contains an insignificant carrier content. In this circuit a 24dB per octave active filter is used. This is followed by a low gain amplifier based on IC5 which gives a further stage of filtering. This also boosts the output signal slightly so that there is approximately unity voltage gain through the unit.

Construction

The main point to check when constructing the unit is that there is no path of conduction through the unit. It is worthwhile checking the finished unit with a multimeter set to a high resistance range to ensure that no connection through the unit has been inadvertently left. It may be possible to tap off the supply for the input side of the circuit from the equipment which supplies the input signal, but the output side of the unit must, of course, have an entirely separate and safe power source.

As far as audio quality is concerned the unit is not up to true hi-fi standards, and it has something less than the full audio bandwidth. It provides a very acceptable level of performance though, and is more than adequate for most purposes.

Audio Isolator



M·I·D·I INTERFACING TECHNIQUES

by R.D. Ball

The System

MIDI is the acronym for Musical Instrument Digital Interface, MIDI is now the universal standard for connecting and controlling electronic musical instruments.

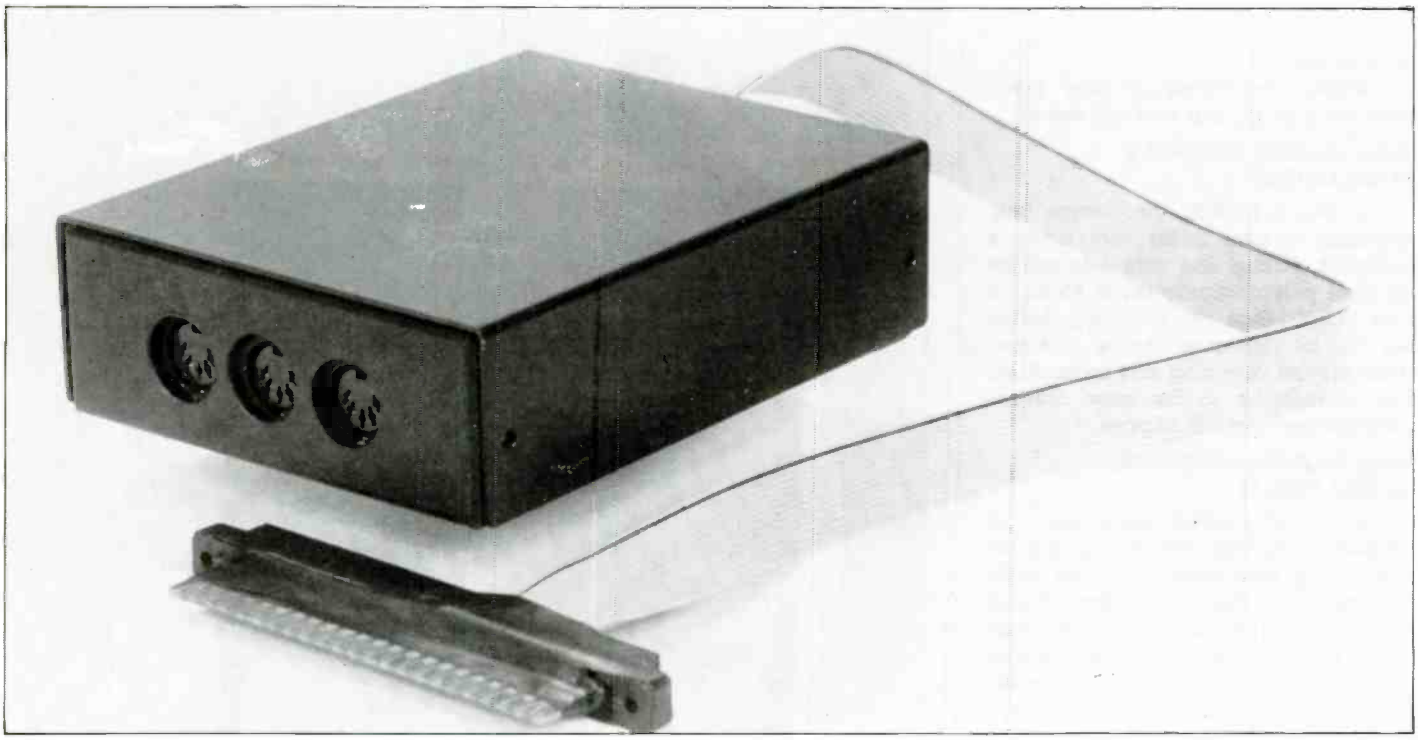
Originally, synthesisers were controlled using two signals; 'gate' and 'CV', the 'CV' signal is simply a DC voltage corresponding to the pitch of the note, a change of 1V would give a change in pitch of one octave. The 'gate' signal is used to control the sample and hold circuitry, i.e. it gates the control voltage. It is also used to trigger the envelope generator. This system suited monophonic analogue synthesisers but is not really practical for modern polyphonic digital synthesisers.

Fortunately, synthesiser manufacturers adopted a single standard, this system was developed by Roland in conjunction with other manufacturers. MIDI was the result.

The outline of MIDI is that all synthesisers and drum machines (or any other MIDI compatible device) use the same set of codes to initiate and stop notes. If certain information does not apply to a particular machine (i.e. velocity information on a non-velocity sensitive synthesiser) it is ignored.

The system outline may seem limited at first, but special functions exclusive to a particular manufacturer or machine are incorporated using special 'system exclusive' codes to indicate a non-standard function.

To allow easy connection between devices, a serial system is used. Operation is similar to RS423, except that MIDI uses a current rather than voltage loop, however the main difference is speed; most serial systems have a maximum speed of 9600 BAUD. MIDI uses an incredible 31.25 Kilo BAUD!! The



A MIDI interface box

system runs at this speed to avoid apparent gaps between simultaneous events. The data word format is 8 bits, 1 stop bit and no parity.

A full implementation of MIDI utilises three sockets on the equipment. These are MIDI IN, MIDI OUT and MIDI THRU'. MIDI IN is the serial data into the equipment. MIDI OUT is the serial data from the equipment. MIDI THRU is a buffered version of the data that appears at MIDI IN. The connection between equipment is similar to the way that Commodore printers and disk drives may be chained together.

To prevent possible hum and earth loop problems all MIDI inputs are Opto Isolated. Connection between equipment is usually via 5 pin 180° DIN leads (with some equipment intended for use 'on the road' XLR connectors may be found), only two cores and screen are used on 5 pin leads though. Ready made MIDI leads can be purchased or leads may be made up using good quality screened cable and connectors. MIDI leads should not be longer than 50 metres.

What can MIDI be used for?

(a) Producing a 'thicker' sound

This means connecting two or more synthesisers together so that notes played on the MASTER synth will also be played on the SLAVE synth(s). This can be very effective, especially if the attack rate of the envelope generator on the SLAVE is considerably slower than that of the MASTER. This effect is generally termed layering or doubling.

(b) Allowing computer control

By using MIDI, electronic musical instruments can be controlled by a home computer. This means that synths can 'talk' to computers and vice-versa. This opens up possibilities of using the computer as a sequencer or to produce effects such as appregiation of chords or even to act as an extra modulation generator. This does however require the computer to be MIDI compatible, this article also includes a circuit for a MIDI interface and it is hoped that this will provide a stepping stone into computer controlled electronic music.

(c) Reducing equipment space and cost

By using a master keyboard a number of slave synths can be driven, since the slave synths do not need keyboards, the keyboards can be omitted!! Keyboardless synths are much cheaper and occupy a lot less space than full synths. The master keyboard may be a special MIDI MASTER KEYBOARD which has no sound generation circuitry; just MIDI sockets. Master keyboards generally feel and respond more like a real piano keyboard and normally include good facilities for program and parameter changes on slaves. Some also allow the keyboard to be 'split' so that different octaves transmit on different MIDI channels. A good master keyboard will transmit velocity and aftertouch information as well as note information.

(d) Reducing obsolescence

All MIDI equipment is compatible, so that old equipment will quite happily work along side the very latest equipment and that applies to all equipment yet to be designed!! Even old analogue synths can join the MIDI revolution as MIDI to CV converter boxes are now available.

(e) The MIDI studio

By using a sequencer, different instruments can be recorded into memory, just like multi-track recording on tape. Each 'track' is assigned to a different MIDI channel allowing simultaneous playback, and because the information is in memory it can easily be modified or edited. It is also possible to change instruments once tracks have been recorded. One great advantage is the total absence of recording medium noise. Once multiple tracks have been laid down, the slave devices can now be mixed onto conventional tape.

MIDI Modes

Mode 1: OMNI ON/POLY (OMNI MODE)

In OMNI MODE, the MIDI device receives data transmitted on all 16 MIDI channels, note data is assigned to the device's voices polyphonically.

This means that whether the transmitting device sends data down channel 1 or 16 (or any channel for that matter), the device in omni mode will receive it. Data corresponding to notes will turn on and off the devices voices polyphonically, i.e. a number of different notes simultaneously sounding; as in a chord. The degree of polophony will depend on the device and is usually around six or eight notes.

Mode 2: OMNI ON/MONO

In this mode, data is received from any channel as above, but note data is assigned to only one voice (monophonically).

If multi note information is sent, the device will either respond to the highest, lowest or last note to be received, this depends on the device and some manufacturers allow the assignment to be changed.

Simply, regardless of how many notes are played, only one will sound.

Mode 3: OMNI OFF/POLY (POLY MODE)

In POLY MODE, the device only responds to data being sent down a particular channel and note data will be assigned polyphonically (as in Mode 1). With POLY Mode the receiving device has to be assigned to a channel, obviously the receiving and transmitting devices must be on the same channel otherwise nothing will happen.

Mode 4: OMNI OFF/MONO (MONO MODE)

In MONO MODE each voice is assigned to its own channel, i.e. if a six voice synth was assigned so the basic channel is 1, then successive voices would be assigned to channels 2 through to 6. Synths with multi timbral facilities allow each voice to have a different sound, for these reasons MONO mode is very powerful but needs careful use to get good results. Facilities in this mode can vary and the owners manual should outline how to use this mode fully.

MIDI Data Messages

NOTE ON

The following sequence of data will initiate a note. Exchange consists of a three byte transfer.

BYTE 1 Channel No. (0 to 15) and NOTE ON command.

BYTE 2 Key No. (0 to 127) 0=lowest, 127=highest. MID 'C' = 60.

BYTE 3 Key Velocity; how hard key is pressed, (0 to 127) 0=no velocity, 127=max velocity. Non-velocity keyboards use 64.

NOTE OFF

The following sequence of data will stop a note. Exchange consists of a three byte transfer.

BYTE 1 Channel No. and NOTE OFF command.

BYTE 2 Key No.

BYTE 3 Key Release Velocity; how quickly key is released, (0 to 127).

POLYPHONIC KEY PRESSURE (AFTERTOUCH)

Some keyboards also respond to the pressure applied to the key after the key has been pressed. Exchange consists of a three byte transfer.

BYTE 1 Channel No. and KEY PRESSURE command.

BYTE 2 Key No.

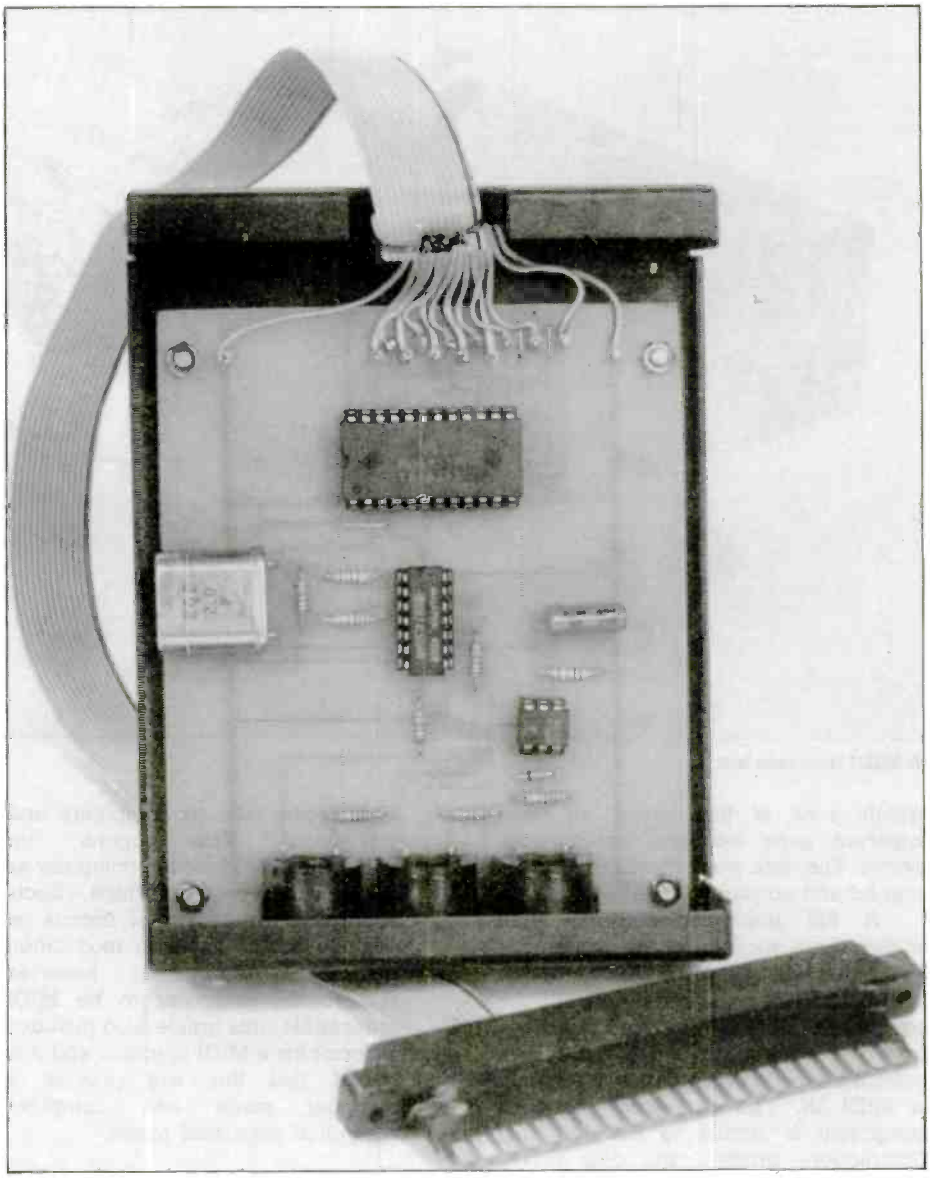
BYTE 3 Key Pressure (0 to 127) 0=no pressure, 127= max pressure.

OVERALL PRESSURE (AFTERTOUCH)

Similar to above, but is the overall pressure of all the notes down. Exchange consists of a two byte transfer.

BYTE 1 Channel No. and OVERALL PRESSURE command.

BYTE 2 Overall Pressure (0 to 127) 0=no pressure, 127 = max pressure.



Inside the box

CONTROLLERS

These are used to transmit information corresponding to operation of such things as portamento, damper and operation of modulation wheels or joysticks. This means, for example, that when the damper is operated on the master, it is also operated on the slave.

There are two main types of controller command:

CONTINUOUS - these correspond to turning a pot on the control panel or operating a modulation wheel, i.e. anything that needs to have a continuous value corresponding to position, to be transmitted.

ON/OFF - these correspond to switch operations such as the damper pedal being pressed.

For the different types of controllers a set of sub-channels or controller numbers are used:

0 to 31 for continuous controllers (low and high resolution).

32 to 63 for continuous controllers (high resolution).

64 to 95 for on/off controllers.

96 to 121 undefined.

122 to 127 for channel mode messages.

Controller messages consist of a three byte transfer.

BYTE 1 Channel No. and CONTROLLER message.

BYTE 2 Controller No. (0 to 127).

BYTE 3 Controller Value (0 to 127) 0=full off, 127=full on.

PITCH BEND

Pitch bend is considered such an important controller it is assigned as a separate message. Pitch bend is a three byte transfer.

BYTE 1 Channel No. and PITCH BEND command.

BYTE 2 Pitch Bend value (0 to 127) LSB.

BYTE 3 Pitch Bend value (0 to 127) MSB.

PROGRAM CHANGE

This enables program or patches corresponding to different sounds to be selected, this is a two byte transfer.

BYTE 1 Channel No. and PROGRAM CHANGE command.

BYTE 2 Program No. (0 to 127).

MODE MESSAGES

This group of messages affect the modes of operation and are under the class of controllers.

LOCAL/REMOTE

This is used for selecting whether note data for voices within the synth is obtained from the keyboard and MIDI or just MIDI. In the second option, operation of the keyboard just sends out MIDI data. This option is useful for adding such features as appregiation, auto chording or other MIDI effects and is achieved by using a MIDI effects box or micro computer.

ALL NOTES OFF

Causes all notes on that channel to cease sounding. This command should be used with some care as not all devices respond to it!

OMNI/POLY/MONO SELECT

This allows selection of the channel and note assignment mode as previously discussed.

SYSTEM EXCLUSIVE (★)

This allows totally non-standard information to be exchanged between compatible equipment. Such as data for a particular patch or even data for a sampled sound.

The exclusive message was provided for manufacturers different needs and is one facility that means MIDI cannot become obsolete.

The exclusive message is a multi byte transfer and the length depends on the application.

BYTE 1 SYSTEM EXCLUSIVE Command.

BYTE 2 MANUFACTURERS ID No.

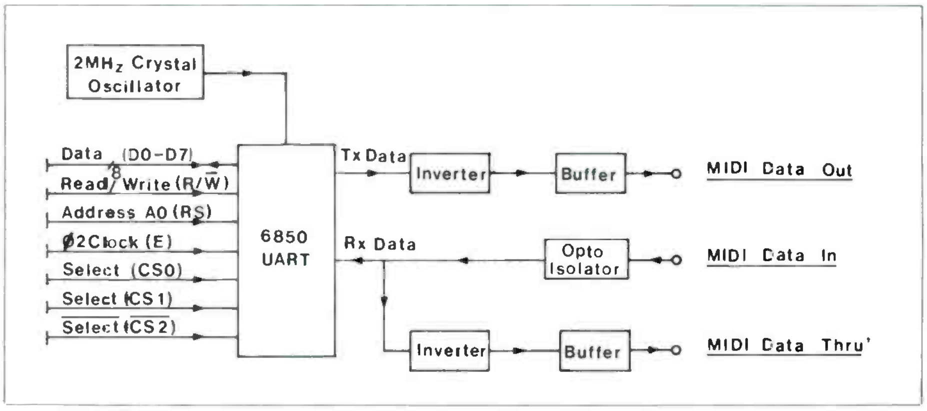


Figure 1. Block diagram.

BYTE 3 }
'DATA'
BYTE n }

BYTE n+1 END OF EXCHANGE Command.

SYSTEM RESET (★)

Puts all the equipment into the reset state, as if it had just been powered up. There are many other messages but only the main ones have been covered here.

N.B. (★) = Any 'SYSTEM' message that is transmitted, is received on all channels.

MIDI Requirements

INTERFACE HARDWARE

Data rate 31.25 K BAUD ± 1%, Asynchronous.

1 start bit, 8 data bits, 1 stop bit.

Total 10 bits, one complete word takes 320μs to transmit.

Circuit operates using 5mA current loop. The receiver must be opto-isolated. Opto isolator must require less than 5mA to turn on.

MESSAGES

i) Channel messages

The status byte contains a four bit number corresponding to the MIDI channel required (1 to 16); the number transmitted is 0 to 15 (0000_2 to 1111_2).

ii) System messages

These are not given a channel

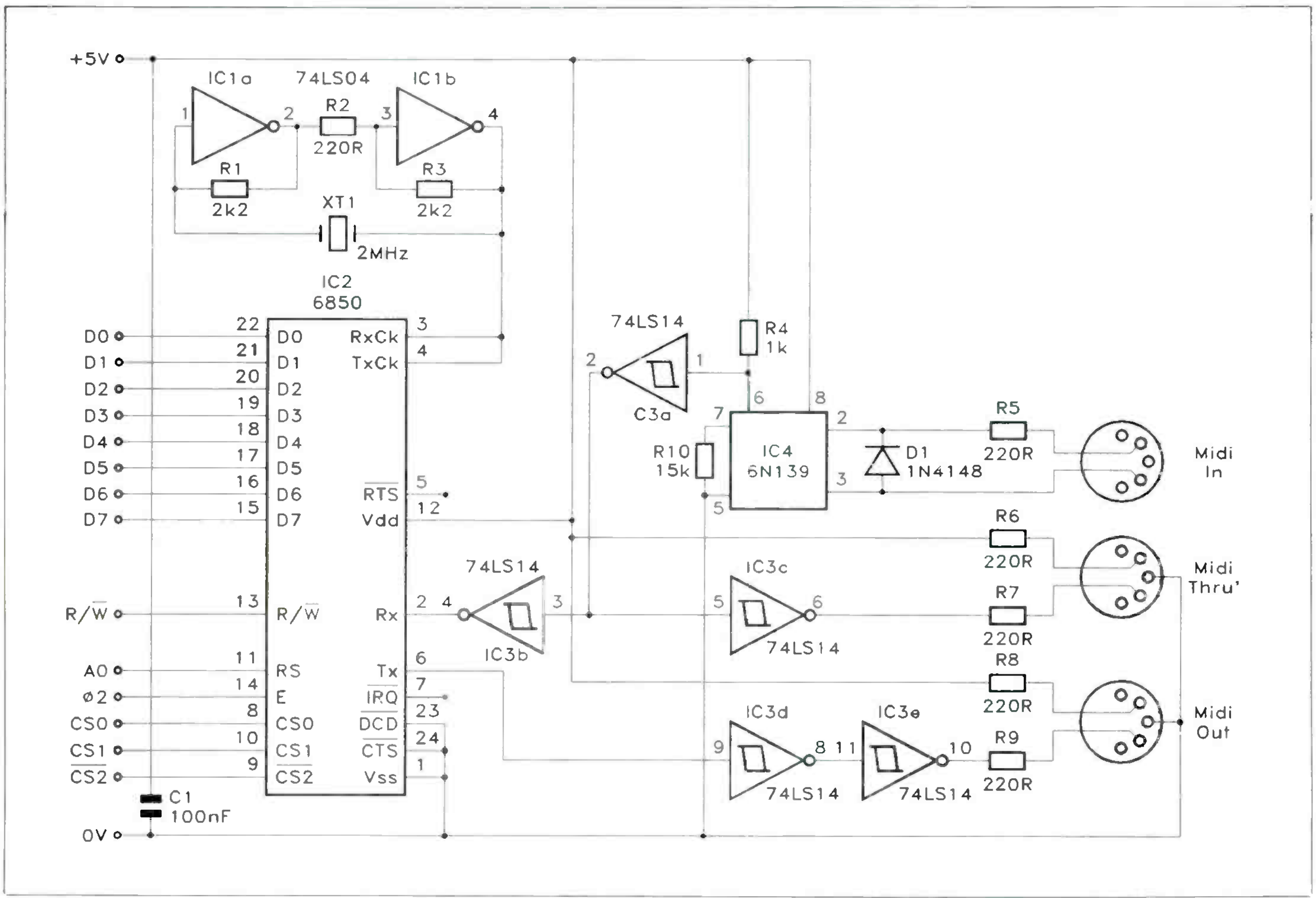


Figure 2. Circuit.

number and are received on all channels.

COMMON - for all devices connected.

REAL TIME - for timing and synchronisation purposes.

EXCLUSIVE - manufacturers exclusive data exchange.

Block Diagram

Figure 1 centres around the 6850 Universal Asynchronous Receiver Transmitter; this device is used to convert parallel data from the computer to serial MIDI data and also to convert serial MIDI data to parallel data for the computer.

The data from the 6850 UART is not suitable to directly drive a MIDI device. The data has to be buffered to drive the MIDI bus. The buffer has quite a low output impedance so as to prevent problems caused by induced noise or a low cable capacitance. The output is current limited to protect the equipment.

As previously mentioned, MIDI inputs need to be Opto Isolated, this gives added safety and prevents troublesome (and annoying!) hum loops from being formed through cables and equipment. After data is passed through the Opto Isolator it follows two routes; into the 6850 UART and via an inverter into another buffer (duplicate of MIDI OUT buffer) to provide MIDI THRU' data.

The data rate is derived from a 2MHz crystal oscillator, this is subsequently divided by 64 in the 6850 to give the 31.25 K BAUD data rate used in the MIDI system.

Connection to the computer data bus is via the tri-state buffers in the 6850, data transfer is controlled by read/write, enable and select signals.

Register selection is achieved by using read/write and register select lines.

Circuit Description

In Figure 2, IC1a, IC1b, R1, R2, R3 and X1 form a conventional two inverter crystal oscillator which runs at 2MHz. The output of which is fed to IC2's transmit and receive clock inputs, the 31.25KHz data clock is obtained by an internal device by 64 circuit in IC2. In the interface, IC2 does most of the work; parallel/serial and serial/parallel conversion, as well as interfacing to the computer expansion bus. Bus control is achieved using 'E' (enable) and 'R/W' (read/write). The E line is normally connected to the processors '∅2' (phase 2) system clock. Data transfer takes place when the ∅2 line is high. The R/W line controls the data direction. With R/W high data is read from the 6850's registers and with R/W low data is written to the 6850's registers. Since the 6850 has two pairs of registers, a selection signal is required, this is the 'RS' (register select) line, and is connected to the least significant bit of the address bus. Data is exchanged via 8 bi-directional lines (D0-D7), when IC2 is de-selected D0-D7 are in the tri-state mode. To insert the 6850 into the computer's memory map, 3 select lines are provided; two active high (CS0 and CS1) and one active low (CS2), these lines may have to be fed from a separate address decoder if suitable memory map decode lines are not available on the computer.

Data Bus	Addr+1 (TX) Transmit	Addr+1 (RX) Receive	Addr+0 (WR) Control	Addr+0(RD) Status
D0	Data bit 0	Data bit 0	CK divide	RX reg full
D1	Data bit 1	Data bit 1	CK divide	TX reg empt
D2	Data bit 2	Data bit 2	Word form	★not★ ★used★
D3	Data bit 3	Data bit 3	Word form	★not★ ★used★
D4	Data bit 4	Data bit 4	Word form	Framing err
D5	Data bit 5	Data bit 5	(TX ctrl)	RX overun
D6	Data bit 6	Data bit 6	(TX ctrl)	Parity err
D7	Data bit 7	Data bit 7	(INT en)	(INT requ)

Table 1. Register contents.

Function	D0	D1	D2	D3	D4	D5	D6	D7	Decimal Value
Reset	1	1	★	★	★	★	★	★	3
Divide by 64	0	1	★	★	★	★	★	★	2
Divide by 16	1	0	★	★	★	★	★	★	1
Divide by 1	0	0	★	★	★	★	★	★	0

Table 2. Control register (Reset and Divide).

Serial data transmitted from the 6850 is not suitable to directly drive the MIDI bus, so it has to be buffered. This is achieved using IC3e, since this is an inverting buffer, the data has to be first inverted using IC3d. By taking the MIDI output from between the +5V line and IC3e, IC3e is sinking rather than sourcing current. R8 and R9 limit the maximum current that can be drawn under possible fault conditions, protecting the computer, interface and MIDI device. IC3e forms the MIDI OUT buffer. Data received drives the LED half of IC4. R5 serves to limit current and D1 affords reverse bias protection for the LED. When the LED turns on, the transistor half of IC4 is biased on, and pulls the input to IC3a and the receive data input of IC2, low. When the LED is off, the transistor is also off, the input to IC3a and the receive data input of IC2 are pulled high via R4. IC3a, IC3c, R6 and R7 form the MIDI THRU' buffer, operation is identical to the MIDI OUT buffer except that data is obtained from the opto isolator instead of the transmit data output of IC2.

6850 UART Registers

Table 1 shows the registers contents on all 8 data lines. Table 2 shows the reset and divide settings in the Control register and Table 3 gives the various settings required for various word formats. Bit D7 of the Control register is the Receive Interrupt Enable bit and if bit D7 is set, an interrupt will be generated when RX Register Full bit goes high. If bit D7 is not set, receive interrupts are disabled. To RESET 6850 UART; set bits D0 and D1 to 1 (i.e. decimal 3). To ENABLE 6850 UART; set bit D0 to 0, bit D1 to 1, bit D2 to 1, bit D3 to 0, bit D4 to 1 (i.e. decimal 22). (This corresponds to Divide by 64 and 8 bits, No Parity, 1 Stop Bit). To TRANSMIT data; check that the TX Register Empty bit, (D1; Control Register) is high, if not wait until bit D1 is high, then place transmit data into bits D0-D7 of TX Register. To RECEIVE data; check that

Word format	D0	D1	D2	D3	D4	D5	D6	D7	(Decimal
7 bits, even parity 2 stop bits	★	★	0	0	0	★	★	★	0
7 bits, odd parity 2 stop bits	★	★	1	0	0	★	★	★	4
7 bits, even parity 1 stop bit	★	★	0	1	0	★	★	★	8
7 bits, odd parity 1 stop bit	★	★	1	1	0	★	★	★	12
8 bits, no parity 2 stop bits	★	★	0	0	1	★	★	★	16
8 bits, no parity 1 stop bit	★	★	1	0	1	★	★	★	20
8 bits, even parity 1 stop bit	★	★	0	1	1	★	★	★	24
8 bits, odd parity 1 stop bit	★	★	1	1	1	★	★	★	28

Table 3. Control register (word format)

the RX Register Full bit, (D0; Control Register) is high, if not wait until D0 is high, then retrieve data from bits D0-D7 of RX Register.

MIDI Interface Software Writing

Sequencing can be accomplished from BASIC quite easily. All that is required is that the appropriate data be output to the 6850 in the correct order. This data may be stored in an array or in data statements.

For recording or processing of MIDI data received, it is really necessary to use machine language, either entirely or using subroutines which can be called from BASIC; this should not be too much of a problem to anyone who has an understanding of assembly language, however guidelines will be given.

Before the 6850 is used it has to be RESET and the CLOCK DIVISION rate set. This is achieved by writing decimal 3 then decimal 22 to the control register of the 6850.

Data being exchanged between devices is passed through the TX and RX data registers. To achieve correct data transfer, the STATUS register must be checked; i.e. that the TRANSMIT DATA REGISTER is empty before sending more data; TDRE bit of STATUS REGISTER will be high when it is OK to send more data, and similarly for receiving data; data must be present in the RECEIVE DATA REGISTER before it can be read; RDRF bit of STATUS REGISTER will be high when it is OK to read the data.

The Registers will be configured in the computers memory map as shown in Table 4; the actual locations will depend on the machine and the address decoding used, but addresses for the VIC 20 and the Commodore 64 are shown. Figures 3 and 4 show the pin functions of VIC 20 and Commodore 64 edge connectors respectively, and Table 5 shows connections from the two edge connectors to the circuit.

To illustrate how simple the software can be, a few examples are shown in Listings 1 to 5, these show how to RESET the 6850 and READ and WRITE data, examples are given in both BASIC and 6502 assembly code. In the examples:

'base' means base address,

'base+1' means base address+1 location,

'data' means either variable or memory location containing data.

Please note that BASIC should not be used to directly get data from the 6850 since BASIC is slow. However BASIC may be used to send data without problem. If data needs to be fetched in BASIC, a machine language subroutine must be used.

Table 6 is intended as a guide to what data should be sent and in what order. Table 7 gives Controller assignments and Table 8 describes Mode Messages.

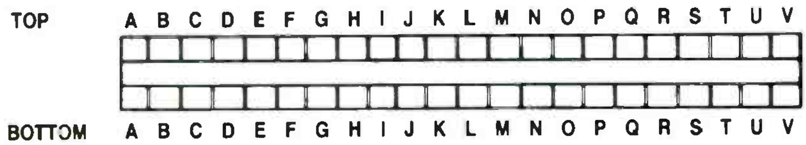
Base address	(read)	Status Register
Base address	(write)	Control Register
Base address +1	(read)	Receive Data Register
Base address +1	(write)	Transmit Data register

		VIC-20 (dec)	(hex)	CBM-64 (dec)	(hex)
Base address	=	39936	$9C00	57088	$DF00
Base address +1	=	39937	$9C01	57089	$DF01

The base address is the position in memory where the 6850 has been placed.

Table 4. Register Memory Map.

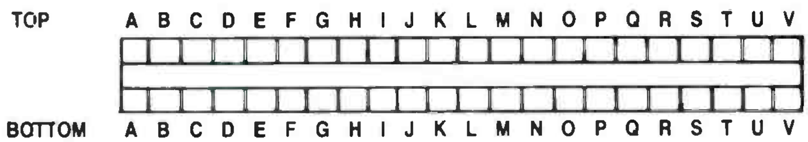
(Viewed looking into expansion slot)



Top		Bottom	
A GND	L BLK 3	A GND	L CA10
B CD0	M BLK 5	B CA0	M CA11
C CD1	N RAM 1	C CA1	N CA12
D CD2	O RAM 2	D CA2	O CA13
E CD3	P RAM 3	E CA3	P I/O 2
F CD4	Q V R/W	F CA4	Q I/O 3
G CD5	R C R/W	G CA5	R SO2
H CD6	S IRQ	H CA6	S NMI
I CD7	T nc	I CA7	T RESET
J BLK 1	U +5V	J CA8	U nc
K BLK 2	V GND	K CA9	V GND

Figure 3. VIC 20 Edge Connector.

(viewed looking into expansion slot)



Top		Bottom	
A GND	L ROML	A GND	L A10
B D0	M I/O 2	B A0	M A11
C D1	N EXROM	C A1	N A12
D D2	O GAME	D A2	O A13
E D3	P I/O 1	E A3	P A14
F D4	Q Dot clock	F A4	Q A15
G D5	R R/W	G A5	R SO2
H D6	S IRQ	H A6	S NMI
I D7	T +5V	I A7	T RESET
J DMA	U +5V	J A8	U ROMH
K BA	V GND	K A9	V GND

Figure 4. Commodore 64 Edge Connector.

MIDI Circuit	VIC 20	CBM 64
0V	GND	GND
D0	CD0	D0
D1	CD1	D1
D2	CD2	D2
D3	CD3	D3
D4	CD4	D4
D5	CD5	D5
D6	CD6	D6
D7	CD7	D7
R/W	C R/W	R/W
RS	CA0	A0
E	SO2	SO2
CS2	I/O 3	I/O 2
+5V	+5V	+5V

Note the CS0 and CS1 lines should be tied to +5V, as only CS2 is used.

Table 5.

Example 1

```
10 REM **RESET 6850**
20 POKE 'base',3
30 REM **SET DIVIDE RATE & WORD FORMAT**
40 POKE 'base',22
```

Listing 1

Example 2

```
@ reset   LDA#$03       load acc with hex 03
          STA$'base'    store acc at 'base'
          LDA#$16       load acc with hex 16
          STA$'base'    store acc at 'base'
          RTS           return from subroutine
```

Listing 2

Example 3

```
@ get data LDA $'base'      load acc with STATUS
           AND#$01          AND with bit 0
           CMP#$01          test
           BNE$@ get data   branch if no data
           LDA$'base+1'     get data
           STA$'data'       store acc at 'data'
           RTS              return from subroutine
```

Listing 3

Example 4

```
10 REM **IS TX REG EMPTY?**
20  IF (PEEK('base') AND 2)=0 THEN 20
30 REM **SEND DATA to 6850**
40 POKE 'base+1','data'
```

Listing 4

Example 5

```
@ send data LDA$'base'      load acc with STATUS
            AND#$02         AND with bit 1
            CMP#$02         test
            BNE$@ send      branch if not ready
            data
            LDA$'data'      get data to send
            STA$'base+1'    send data
            RTS             return from subroutine
```

Listing 5

MIDI INTERFACE PARTS LIST

RESISTORS: All 0·6W 1% Metal Film

R1,3	2k2	2	(M2K2)
R4	1k	1	(M1K)
R2,5,6,7,8,9	220Ω	6	(M220R)
R10	15k	1	(M15K)

CAPACITORS

C1	100nF Polyester	1	(BX76H)

SEMICONDUCTORS

IC1	74LS04	1	(YF04E)
IC2	MC6850P	1	(WQ48C)
IC3	74LS14	1	(YF12N)
D1	1N4148	1	(QL80B)
OP1	Opto Isolator 6N139	1	(RA59P)

MISCELLANEOUS

XT1	2MHz Crystal	1	(FY80B)
SK1-3	PC DIN Skt 5-pin A	3	(YX91Y)
	24-pin DIL Skt	1	(BL20W)
	14-pin DIL Skt	1	(BL18U)
	Box AB10	1	(LF11M)

First byte	Second byte	Third byte	Description
1001 mmmm	0nnn nnnn	0vvv vvvv	NOTE ON (Velocity 0 = note off) [1]
1000 mmmm	0nnn nnnn	0vvv vvvv	NOTE OFF [1]
1010 mmmm	0nnn nnnn	0ppp pppp	POLYPHONIC KEY PRESSURE (Aftertouch) [2]
1101 mmmm	0ppp pppp	———	OVERALL KEY PRESSURE (Aftertouch) [2]
1011 mmmm	0ccc cccc	0ddd dddd	CONTROLLER CHANGE [3]
1110 mmmm	0ddd dddd	0ddd dddd	PITCH BEND [4]
1100 mmmm	0PPP PPPP	———	PROGRAM CHANGE [5]
1111 0000	0 iiiiiii	0*** ****	SYSTEM EXCLUSIVE [6]
1111 0111	———	———	EOX (End of exchange) [7]
1111 1111	———	———	SYSTEM RESET

Key

mmmm	=	MIDI channel No. (0 to 15)
0nnn nnnn	=	NOTE on keyboard (0 to 127)
0vvv vvvv	=	VELOCITY (0 to 127)
0ppp pppp	=	PRESSURE (0 to 127)
0ccc cccc	=	CONTROLLER No. (0 to 127)
0PPP PPPP	=	PROGRAM No.
0ddd dddd	=	DATA (0 to 127)
0 iii iiii	=	I D Code (0 to 127)
0*** ****	=	Undefined number of data bytes (as 0ddd dddd)

Notes

[1]	0nnn nnnn	= 60 is middle 'C'
	0vvv vvvv	0 = off, 1 = ppp, 64 = between mp and mf, 127 = fff
	0vvv vvvv	= 64 in non velocity sensitive devices
[2]	0ppp pppp	0 = no pressure, 127 = max pressure
[3]	0ccc cccc	= controller number; see list
	0ddd dddd	= data for controller
[4]	0ddd dddd	2nd byte = LSB, 3rd byte = MSB
		2nd byte = 0, 3rd byte = 64, gives no bend
[5]	0PPP PPPP	= program or patch number
[6]	0 iii iiii	= manufacturers ID code
	0*** ****	= undefined number of data bytes, terminate by sending [7]
[7]		Use to return to normal use after EXCLUSIVE data exchange

Table 6.

Controller assignment

Controller No.	Description
0	Continuous controller 0 MSB
1	Modulation wheel MSB
2 to 31	Continuous controllers 2 to 31 MSB
32	Continuous controller 0 LSB
33	Modulation wheel LSB
34 to 63	Continuous controllers 2 to 31 LSB
64 to 95	Switch controllers
96 to 121	**undefined**
122 to 127	MODE MESSAGES (see Table 8)

Controller data

For continuous controllers 0 to 127 (min to max).
Fore switch controllers 0 = off, 127 = on.
For continuous controllers; if only 7 bits of resolution required, send only MSB. If full resolution is required send MSB first, then LSB. If only LSB has changed in value, LSB can be sent without MSB.

Table 7.

Mode Messages

Controller No.	Description	Data
122	Local control	0 = off, 127 = on
123	All notes off	0
124	Omni off (all notes off)	0
125	Omni on (all notes off)	0
126	Mono on [poly off] (all notes off)	number of channels
127	Poly on [mono off] (all notes off)	0

Note

All messages which can be sent successively (e.g. NOTE ON) under the same STATUS byte, can be sent without STATUS byte until a different STATUS is required.

Table 8.