No 40

OCT-NOV '90 · £1.45

ELECTRONICS

The Maplin Magazine

STOP THIEF!

compuguard

CAR SECURITY SYSTEM

• Take Control with a 1·2kW POWER CONTROLLER • What is SYSTEM X?

• Build a DNR System • PRIZES to WIN in THE WALL Rock Concert Contest

Data File: TDA1514A 50W Power Amplifier • NOISE GATE for musicians

• Read all about TAPE RECORDING and DOLBY SURROUND SOUND

CONTENTS

OCTOBER TO NOVEMBER 1990 VOL.9 No.40

EDITORIAL

Hello! Here we are again with another issue of your favourite electronics magazine. There's a bumper crop of projects and features this time; here's just a taster to whet your appetite: top of the projects list is a highly sophisticated car security system, it's more than just a car alarm; Compuguard actively tries to defeat a theft attempt! For those of you who enjoy watching movie films at home the guide to using the Maplin Surround Sound Decoder will be right up your armchair. The 1·2kW power controller allows lighting, heaters and power drills to be easily controlled. Have you ever wondered how high quality music can be recorded onto magnetic tape? Well, Mike Holmes' article reveals the secrets. If your tape recorder or FM tuner suffers from background hiss, then why not build the Dynamic Noise Reduction System. A new series introduces us to the telephone world of System X. Data File this time is a compact, 50W power amplifier. We look behind the checkpoints of Berlin's recent rock music event of the year and give you the chance to win a few goodies. For the musicians amongst you, there's the Noise Gate project and for the newcomer to electronics, Square One continues by looking at diodes. Of course there's all the usual regulars to complete the line-up. Until next time why not read on and enjoy!

R.T. Smith

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Published by Maplin Electronics plc.
Typesetting by Inline Design Systems Ltd., 258a London Road, Hadleigh, Benfleet, Essex SS7 2DE.
Colour Separations by Stirling Graphics Ltd., 16-22 West St., Southend, Essex SS2 6HJ.
Printed by SVP, Caerphilly, Mid Glam. CF8 3SU.
Distributed by United Magazine Distribution Ltd., 1-11 Benwell Rd, London N7 7AX.
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.
Fax: (0702) 553935. **Telex:** 995695.

PROJECTS

FEATURES

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NEWS REPORT

Intelligent Fish

It is good to know that the European Commission are awarding ten million Ecus to a project designed to develop a machine translation system for the nine official Community languages. Perhaps the EC officials could save time – and Ecus – by contacting Douglas Adams. You will recall that his book "Hitch Hikers Guide to the Galaxy" introduced the Babel Fish. When stuck into your ear, the Babel Fish enables you to instantly understand any language being spoken. However there did seem to be a drawback. The Babel Fish was responsible for producing more problems than it solved, as a result, the world – sorry The Universe – suffered even bloodier wars.

ICL Turns Japanese

Babel Fish apart, the headline news must be the sale by STC of the UK's last remaining major computer manufacturer, ICL to Fujitsu for a somewhat low £743m. for 80% of the company. This follows the recent movement of Hoskyns to the French CGS; Inmos to Thomson; Apricot to Mitsubishi; and Plessey to Siemens. With this take-over, Fujitsu will become the world's second largest computer manufacturer – though whether the company will hold on to ICL's captive government markets remains to be seen. The move has been generally welcomed by industry and government alike. Conservative chairman of the Trade and Industry select committee comments that it is a purely business decision. "I don't think we should be waving national flags about this business". The PA Consulting Group likewise thinks that the take-over is a good thing. "The acquisition of ICL makes sound technological, marketing, and strategic sense for Fujitsu".

Meanwhile industry observer Michael Naughton of comms consultancy Applied Network Research believes that the move can be compared to the Japanese motor industry setting up manufacturing shop in the UK as a method of gaining an entry into the lucrative markets of the EEC.

High Fliers

No doubt those high flying PA consultants will be interested in a new in-line service offered by American Airlines. The airline is introducing an office in the air facility covering not just air-phones but air-fax. But whether providing telecommuting services some eight miles high, will catch on with other airlines remains to be seen. Meanwhile, the stewardesses offer "coffee, tea or me" may henceforth be a much more innocuous "coffee, tea or fax".

Wide Angled Air Safety

What would seem to be a belated development is the introduction by Panasonic Communication Systems of a wide angle micro colour camera which allows pilots to see the external surfaces, wing controls and engines of their aircraft during flight. Mounted on the tail plane, the wide-angle lens gives a 120-degree vision of the whole aircraft without any perceptible effect on the aerodynamics of the aircraft. With the flight stewards busy handling facsimile messages, the pilot obviously needs all the visual help he can get.

Chatting-up Discouraged

The DTI quango body OFTEL seems to be fighting a loosing battle with the spread and popularity of telephone chat-lines. No sooner does the Authority terminate one chat-line service than several more appear. Back in 1989 OFTEL issued a "Code of Practice" covering one-to-one live conversation services available through BT's public telephone network and in particular on the effect of these services on customers' bills. Just who OFTEL is protecting has not been made clear. Possibly the family on the receiving end of a high telephone bill run up by their children or au pair.

Junk the Mail

Another DTI quango The Data Protection Board is also fighting a rear-guard battle with direct mail operators. It seems that the operators are making use of publicly available address lists to send out advertising material. Why recipients should be bothered – unless they don't happen to possess a large wastepaper bin – has never been made clear. The campaign against unsolicited mail is probably deflecting the Board's basic handling of public complaints on the misuse of personal data, in particular the matching of personal information across databases. Over the past several years, the Board has only brought cases against 30 data users, mainly for failing (of forgetting) to reregister. With the rapid development of cross-border data flows taking place in an increasingly unified Europe, the need for a supra-national data protection body possibly organised by the EEC is now seen as being a logical step by many in the communications industry.

Closed Circuit Telephone Calls

One new product which should definitely appeal to fellow American Airline passengers is the latest Panasonic headset for business telephone systems. Approved by the BABT the system gives you that extra pair of hands – essential if you are trying to down the airline free drink quota or leaf through the duty free allowance data sheets.

Misusing your Computer

As if the computer enthusiast does not have enough to contend with, the government has now introduced The Computer Misuse Act. This does not apparently cover the spilling of coffee over your Amstrad, allowing the printer to catch fire or generally shouting abuse when the system is being more than averagely thick. No the somewhat strangely named Act covers the unlawful intrusion into computer systems by hackers or those with a more criminal intent. The Act however still allows BT to provide phone tapping facilities to the police on request, a factor which continues to alarm various civil liberty organisations.

Get Back On Your Knees

Following our comment in the last issue on the development of a range of ecclesiastical software, comes the news that the new Archbishop of Canterbury is a dedicated computer buff. Apparently while in the back of his limousine, he inputs into his laptop and then downloads it into his Amstrad computer. It won't be long before Lambeth Palace ordains on-line collection terminals and random input hymn selection technology.

BT Cares

Communication technology is coming to the aid of the deaf and hard of hearing. In association with the Royal National Institute of the Deaf, BT is providing a £650,000 investment in a new company Sound Advantage, aimed at providing a 'one-stop shop' for hard of hearing people. High tech products range from telephone

Make IT with Lasers

It can be only a matter of time before that office in the air is kitted out with a laser printer. Ready for that happening, Apple Computer have launched a new line of LaserWriter printers. The compact system can handle not just text and graphics, but letterheads, and envelopes. The maintenance time and costs are guaranteed by the design ability to replace sensitive wearing parts alongside the new toner cartridge. Details: Apple Computer. 071-402-3355.

American Airlines may well be interested in what Canon (UK) describe as the first desktop laser filing unit. This is an optical disk-based image storage unit capable of storing 12,000 pages of A4 per disk, yet only the size of a standard-issue PC. The 'Canofile' unit optical disk gives 256 Mbytes of storage available on each side of a single 12in. disk. However at £13,500, the price is not quite in the economic flight class.

Fax news final is the DTI's support of £1m over a four year period to assist with the launching of a National Laser Technology Transfer Programme, 'Make IT with Lasers'. Presumably British Airways are keenly interested.

attachments to flashing doorbells. Text telephones – these allow deaf users to communicate over the public telephone network by means of an incorporated strip screen – and phones which incorporate extra levels of amplification plus a vibrating message pager and alarm clock. For once everyone scores. BT from increasing the usage of its services, the RNID who receive all the profits and not least the deaf themselves. Sound Advantage can be contacted on: 0733-361199.

BT is also providing £4 million in a major initiative to provide a national relay service for deaf and hard of hearing which allows people to make and receive telephone calls in text format on a computer screen through a central exchange. It does seem that the telephone has turned a full cycle. Alexander Graham Bell invented the telephone in an effort to develop a hearing aid for his deaf wife.

BT R & D

Last year BT spent nearly £230 million on research and development. Also a large amount no doubt went on the completion of the BT digital network. The transfer of all trunk traffic – 40 million calls a day – from analogue to digital started in 1985. Now all BT customers have access to the most advanced long distance switched network in the world.

Some of that investment money is now about to be recovered from new tariff and rental increases. BT defends the increases by pointing out that customers who make a lot of telephone calls subsidise those who make less frequent use of the phone. "For historical reasons, what we charge for renting a line falls well short of the cost of providing and maintaining that line. And this shortfall has to be recovered from the relatively high margin we make on calls, particularly long-distance and international calls" says BT Chairman Iain Vallance. Well at least we have been warned . . .

Moving to Mercury

Disgruntled BT users are probably already contacting Mercury Communications who are now competing for private customers. In fact Mercury has launched a £3 million advertising campaign aimed at home telephone users who regularly use their telephones for long distance calls. These are classified by Mercury as being anything over 35 miles, where Mercury savings could be as much as 35% over BT tariffs. What you need, apart from the Mercury authorisation code is a special telephone which incorporates a blue button which gives access into the Mercury network. Mercury are looking to sign 100,000 home users by next April.

Meanwhile, The Post Office have shunned their former partner and set about installing Mercury card phones at the rate of 200 a month in its main offices and sub post offices. Local call costs will be a basic 5p unit. That apart, Mercury is also planning to quadruple the size of its pay-phone network this year. As the industry daily information newspaper "Computergram" warns, watch out for pavement work ahead.

Mickey Mouse Software

Walt Disney, reports "Computergram", is going into software with the first product being a program designed to teach reading and counting skills to children aged two to five. The system includes a speech and music generator which reproduces the voices of Mickey Mouse and other Disney characters. No prizes for forecasting that the first feature will be Snow White and the Seven Dwarfs.

SKY – Clouds Lifting

With something approaching relief, SKY Television has announced that its Movie channel has now attracted well over 750,000 paying subscribers to its premium film service. As from September this year, SKY MOVIES will be on the air 24-hours a day showing, for the cost of £9.92 monthly, some 312 movies a month. Also in the SKY good news department, their Eurosport Channel can now be seen in some 20 million homes in 22 countries throughout the UK and Europe.

IBM Has Home Users In Its PC Sights

Close on the heels of an industry report which suggests that the European PC market is set to continue its fast rate of growth, with IBM and Compaq leading the sales way, IBM has introduced its PS/1 family of microcomputers. The IBM Personal Computer is a new, low priced (at least for IBM products) system designed for "the enterprising individuals and their families. Based on up-to-date technology, with an Intel 80286 processor, VGA screen, 1/44MB diskette, optional 30MB fixed disk drive and DOS and Microsoft Works as part of the package, the system is expected to be fixed at a sub £1,000 price tag. In fact the price tag will be the all important marketing factor in the UK where it will find itself competing with the well established Amstrad PC range.

FAX IT

BT are anticipating that another 250,000 facsimile machines will be sold this year, with small firms being major buyers. Next year, BT are predicting that the home fax market will take off. No wonder therefore that Canon (UK) are concentrating as much on the paper costs as on supplying equipment. There are currently 34 facsimile suppliers operating in the UK, producing 194 different Group 3 fax machines. Of these only 15 machines use plain paper – about 8%. "Plain paper is one area that many users are looking at as an upgrade path" says Chris Jobling, Canon's facsimile product manager. "Thermal paper fades over time and is unsuitable for filing and awkward to write on" Canon see the plain paper fax market as being

at the forefront of user demand in the years ahead. In fact the new Canon Fax-I770 is a compact laser fax machine which gives users plain paper messages of the highest quality. Costing £3,395, the system unit is small enough to sit on a desktop but includes revolutionary new image-enhancing technology. Details: Canon (UK) 081-773-3173.

IBM Develops Do-it-yourself Defect Repairs

An IBM researcher has invented a process in which a defect in wiring between integrated-circuit chips can induce its own repair. The technique covers a situation where the copper wiring on a board has a defect which serves to create a near open circuit. When a sufficiently high current is passed through a wire with a constriction, more heat is generated at that location than elsewhere on the wire, because the thinner the wire, the greater its electrical resistance and the slower the heat dissipation. The heat generated transfers copper from the cooler parts of the wire through the plating solution or electrolyte. The process IBM says could have widescale implications for interconnection wiring defects in integrated circuit chips.

Building Industry Told To Get With IT

At the recent "Infrastructure 90" conference organised by Applied Network Research and CommEd Conferences, John O'Donaghue of engineering consultancy FC Foremans, told the delegates that information technology can now account for 35% of the cost of new building development – 45% in the case of redevelopment. "Office buildings will have to accommodate ever increased layers of comms channels – voice, data, image – with cabling taking up an ever greater proportion of available space".

BT took the opportunity of the industry conference to announce OSCA Voice, a structured voice cabling system which incorporates patching panels and is seen as being a step along the road to ISDN and complete speech-data integration. Given that every two seconds of every working day, at least one BT extension socket is installed, OSCA voice looks like having its work cut out satisfy demand. Details: CommEd: 071-733-3456.

Picture Caption Challenge

"Look – No Hands!"

Is it:

- ★ The Red Arrows display team awaiting further defence cuts.
- ★ BT suspending chatline operators.
- ★ A demonstration of high definition TV.
- ★ BT personnel jumping for joy at the news that their chairman has awarded himself a big rise.
- ★ "I bet they drink Carling Black Label!".

Perhaps not.

These intrepid BT trainee engineers are not just hanging around, but swinging out in space to test their fail-safe harnesses. British Telecom have started a new project to train young riggers and instruct electrical engineers who have to climb to maintain systems on satellite dish aerials and radio towers.

If you can think of any better captions for this picture, then drop the Editor a line at the usual address, no prizes, but we may print some of the funny ones!

1·2kW POWER CONTROLLER

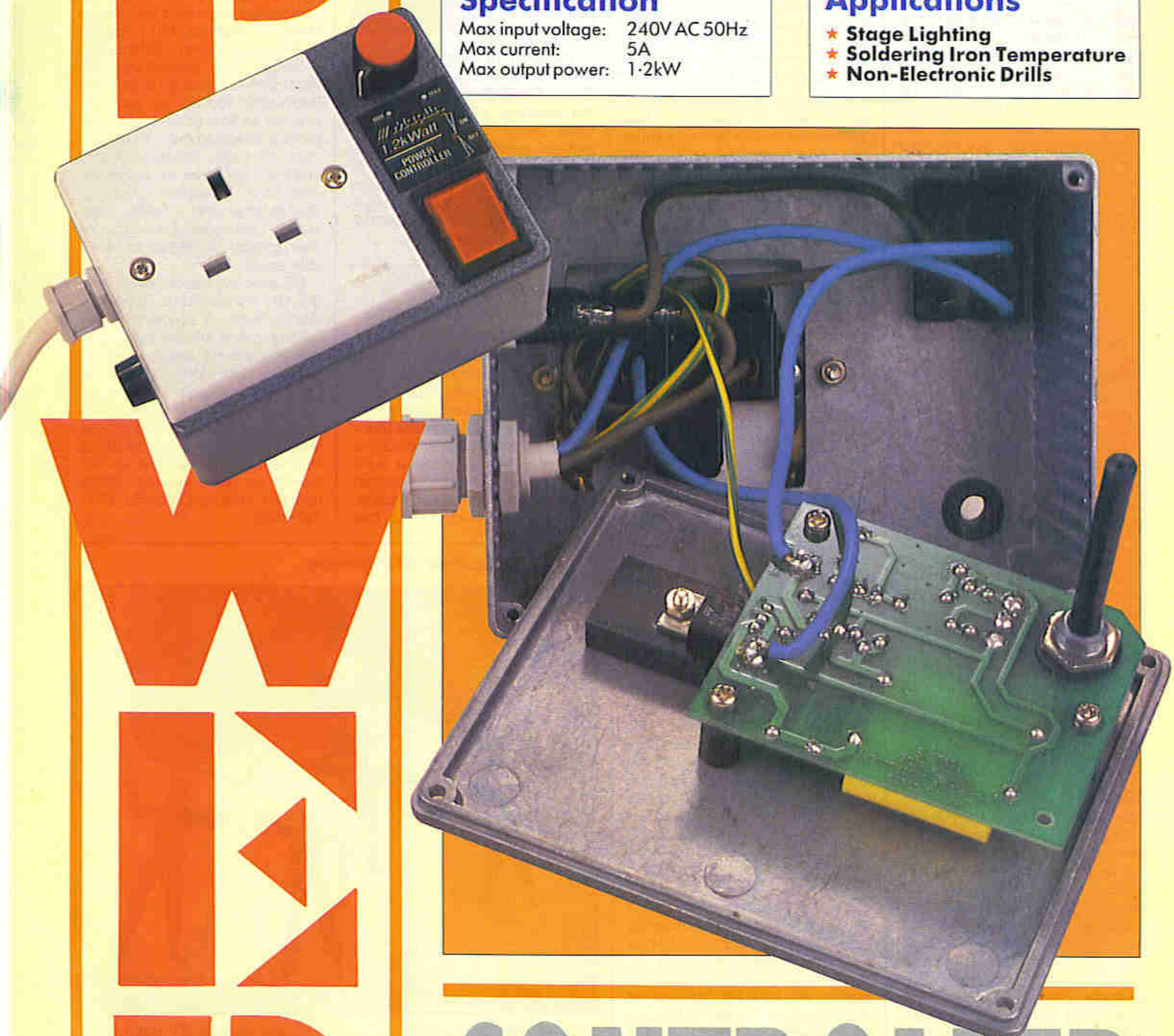
Revised by Alan Williamson
Based on an idea by Robert Penfold

Specification

Max input voltage:	240V AC 50Hz
Max current:	5A
Max output power:	1·2kW

Applications

- Stage Lighting
- Soldering Iron Temperature
- Non-Electronic Drills



Introduction

It is often desirable to control the amount of power applied to a 240V AC mains appliance, such as a power drill, heater, table lamp, etc. There are several ways of providing the necessary control, for example; by using a rheostat, which is a high power variable resistor, connected in series with the supply. This is a wasteful method of control, generating a lot of unwanted heat. Another way is to use a 'variac', a sort of variable transformer which has only one winding, and an adjustable tap. The tap is continuously variable by sliding a copper-carbon brush over the length of the exposed windings, tapping off the required voltage. This is a much better method of AC control, but a variac is a very expensive item. So far, both these forms of control deal with the full sinusoidal waveform of the mains power. Another means of control would be to switch the supply on part way through each half cycle, thus reducing the average power available per cycle. This is illustarted in Figure 1a, b and c. Figure 1a shows near maximum power applied to the load, Figure 1b shows half power applied to the load and Figure 1c shows near minimum power applied to the load.

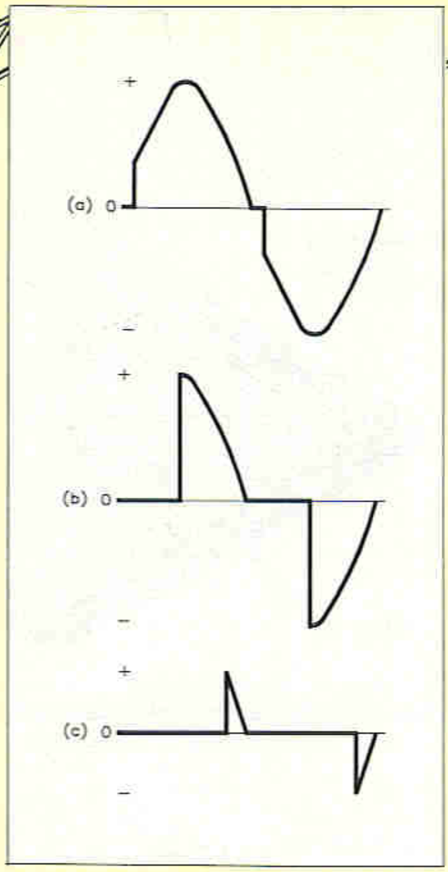


Figure 1. Output waveforms.

The 1·2kW Power Controller described here uses the latter method of control and replaces the original 720W version found in the now discontinued 'Best of E&MM.'

This circuit is particularly useful for controlling fixed speed drills to allow a slow start for accurate drilling, it may be found that using the Controller with an electric drill, there is a small dead spot at the minimum end of the control knob, this is intentional, as it is intended to be used as a pre-heat for light bulbs to minimise 'thermal shock' to the filament. The circuit is also useful for controlling the temperature of a soldering iron, enabling a wide variety of uses from sealing plastic bags to using the soldering iron as a pyrographic pen, (an ancient art form, drawing on the inside of dried wood bark with a hot poker).

The Controller is not suitable for use with fluorescent lighting or any form of electronic equipment. It is strongly advised that beginners do not undertake construction of this project unless supervised by an experienced person.

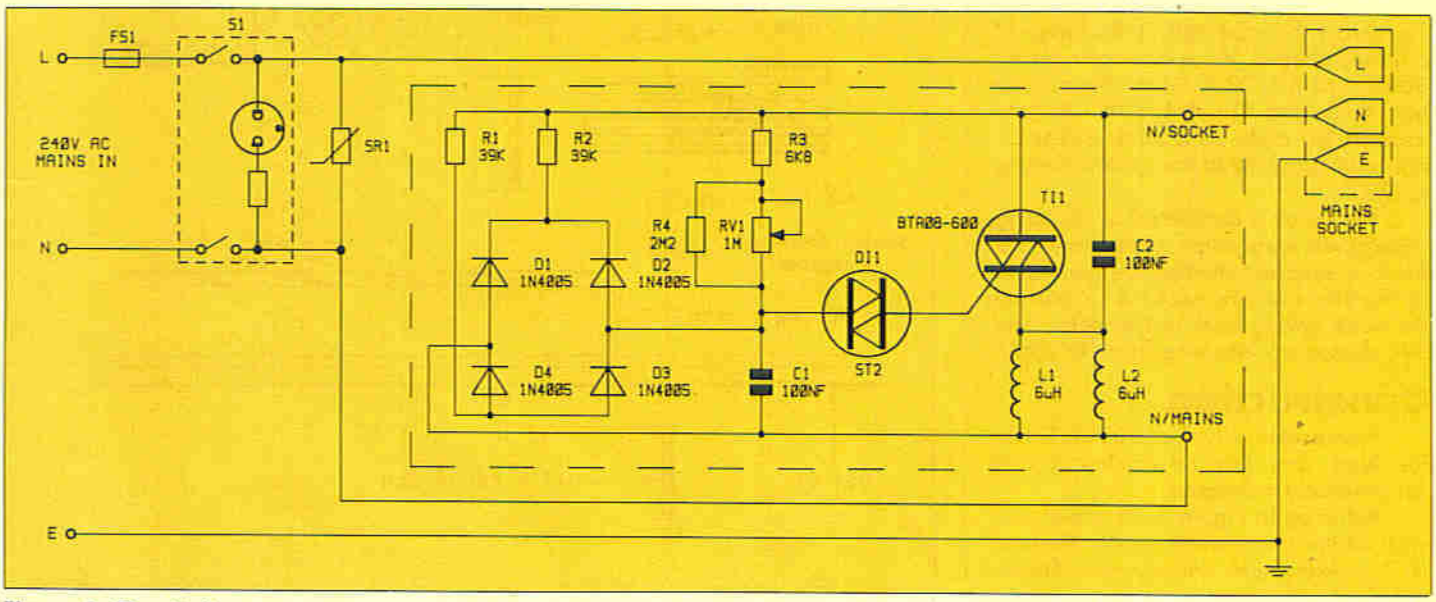


Figure 2. Circuit diagram.

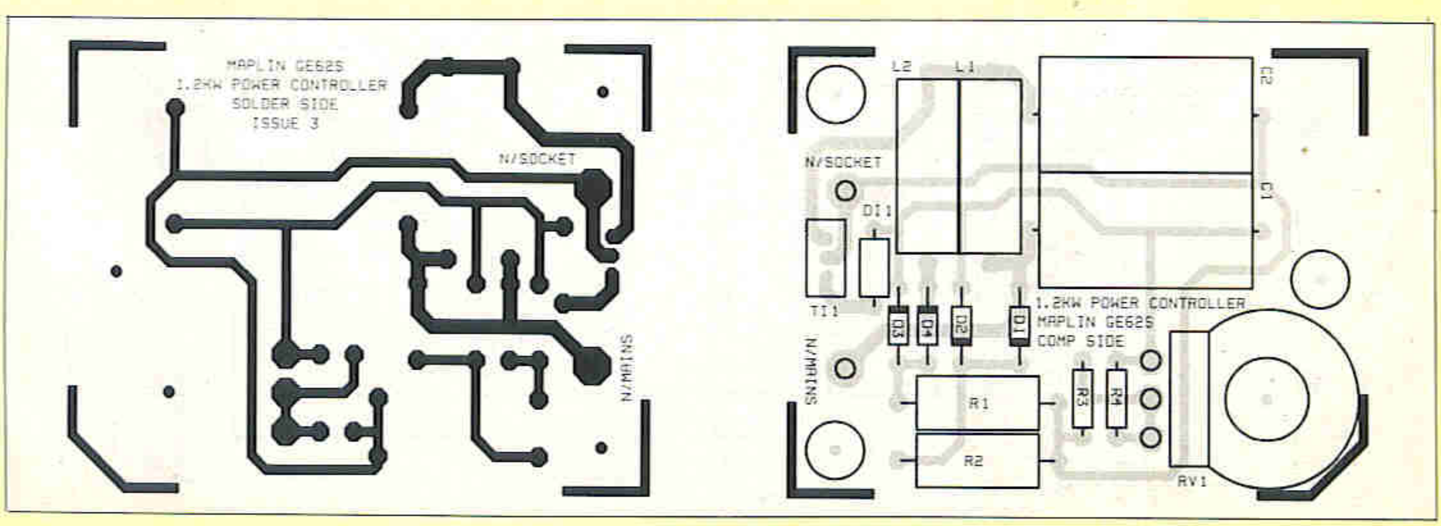


Figure 3. Component legend.

Circuit Description

Referring to the circuit diagram, Figure 2, you will see that the circuit is of the usual diac and triac type.

The circuit operates as follows; R3, RV1 & C1 form a variable time constant, R3 setting the minimum time constant when RV1 is at minimum resistance. R4 is included to reduce the overall value of RV1 giving the required range. This network charges up C1 from the current flowing through R3 & RV1, when a charge of approximately 34V across C1 is reached, diac (DI1) will fire, triggering the triac (TI1) into conduction allowing current to flow through the load.

Just using these components you can make a Controller, but the circuit has a hysteresis problem when RV1 is set to its maximum value. This hysteresis causes a backlash in the system which can be seen when controlling lighting, i.e. the control is advanced and nothing happens, the control is advanced a little more, then suddenly the lights come on brightly. At this point the control can be backed off to dim the lights, as it stands this circuit would be no good to control stage lighting. This hysteresis is caused by a charge left on C1 from the previous half cycle which will counteract the next half cycle, being of opposite polarity. This is overcome by the addition of D1, D2 & R2 for the positive half cycle and D3, D4 & R1 for the negative half cycle. D2 & D3 discharge C1 only during the fall of the cycle (returning to 0V).

The circuit is completed by including some noise suppression components. C2 helps to suppress interference generated by the triac switching on, L1 & L2 prevent the noise getting back to the mains, and SR1 clamps any incoming spikes to 250V.

Construction

Please refer to the Constructors Guide for hints and tips on soldering and constructional techniques.

Referring to Figure 3, Fit solder and crop all the components except the triac (TI1), potentiometer and veropins. Ensure

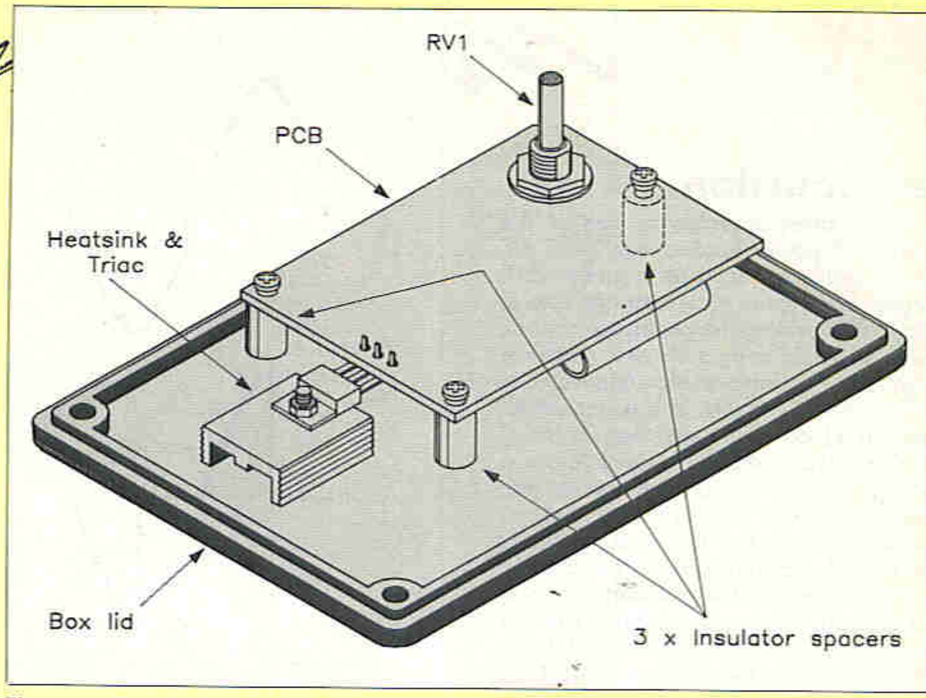


Figure 4. Triac leg/heatsink assembly.

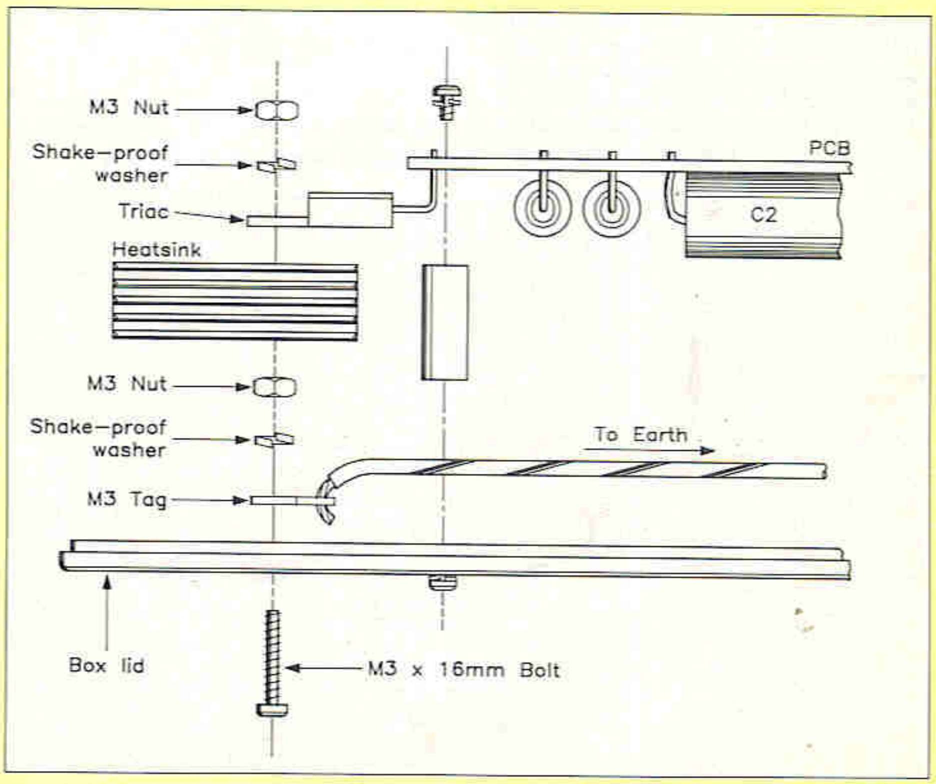


Figure 6. M3 tag/heatsink/triac assembly.

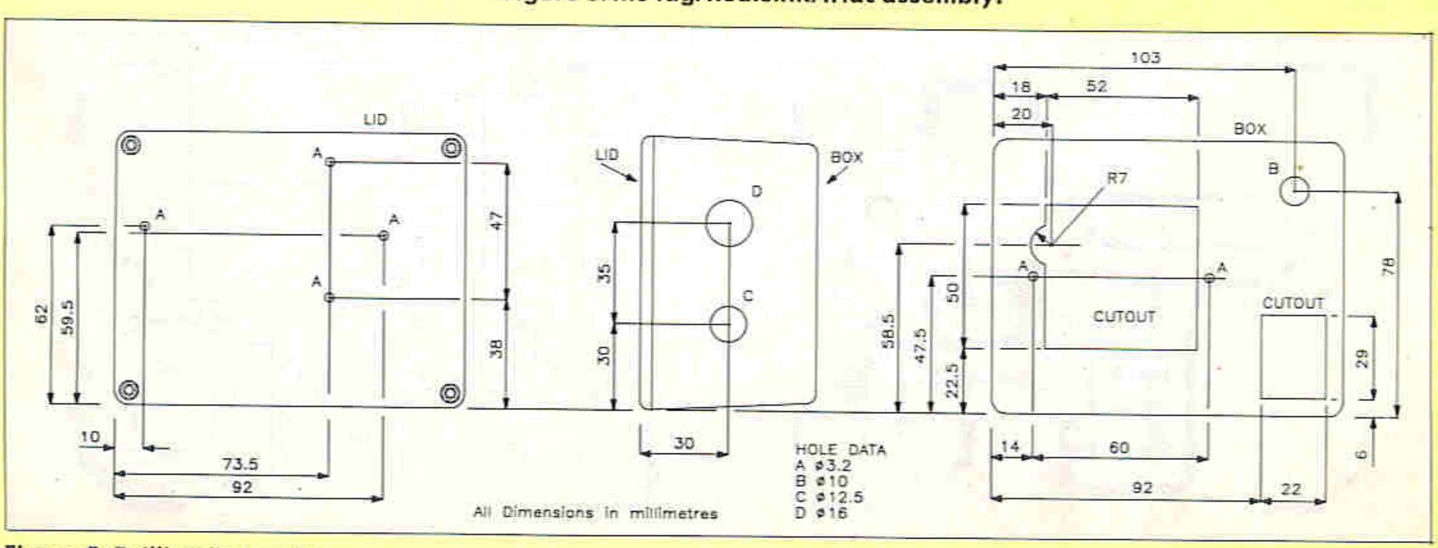


Figure 5. Drilling instructions.

correct orientation of the diodes when fitting them in place. Next insert the wiring pins; noting that the three pins for connection to the potentiometer are fitted from the track side of the PCB, and the other two pins are fitted from the component side. Trim 10mm off the end of the potentiometer shaft, then bolt the potentiometer to the PCB from the component side (see photograph of PCB) and solder the legs to the three veropins. Next, remove all the screws from the three insulating spacers and fit to the PCB so the track side is now face up, then temporarily bolt the triac to the heatsink and bend the legs 5mm from the body as shown in Figure 4. Insert the legs of the heatsink/triac assembly into the pcb and lay on a flat surface with the heatsink and all three insulating spacers touching the surface, solder the triac legs and remove the heatsink. The PCB is now complete, finish by cleaning off any flux residue using alcohol or PCB cleaner YJ45Y.

The next job is to drill the box. Cover the lid, the top and one of the short sides of the box with masking tape; this will enable you to mark out the required holes and the masking tape also provides a non-slip surface for drill. Probably the easiest way of cutting the hole for the 13A socket is to drill a series of closely spaced holes around the perimeter of the cutout-line, the holes can then be joined up using a small round tapered 'rat's tail' file. Drilling information is shown in Figure 5.

After having drilled and de-burred all the holes, final assembly can begin. Cut off approximately 150mm of 13A mains cable, strip off the outer insulation and solder the earth wire to the M3 tag. Bolt the M3 tag to the lid of the box and assemble the heatsink/triac and PCB as shown in Figure 6. Note that the triac, a BTA08-600B is an insulated tab device and therefore does not require the use of an insulating bush and washer. Do not substitute a non-insulated tab device in place of the BTA08-600B specified. Next stick the front panel onto the box and insert the rubber grommet into the hole for the potentiometer. The switch is fitted with the 'mains in' terminals closest to the potentiometer. The fuse holder and cable clamp are installed next, insert the cable through the clamp and tighten. Wire up the pcb, switch and fuse exactly as shown in Figure 7. The four wires (live, neutral and two earths) are then fitted into the socket which is then bolted to the box using M3 nuts bolts and shake-proof washers. Check to see if the wiring is correct and that short circuits cannot occur before screwing the lid onto the box. All that is remaining to do is to fit the knob onto the potentiometer shaft, insert the 5A fuse into the fuse holder and attach a 13A plug (fitted with a 5A fuse) to the mains lead (13A plug and 5A fuse are not supplied in the kit).

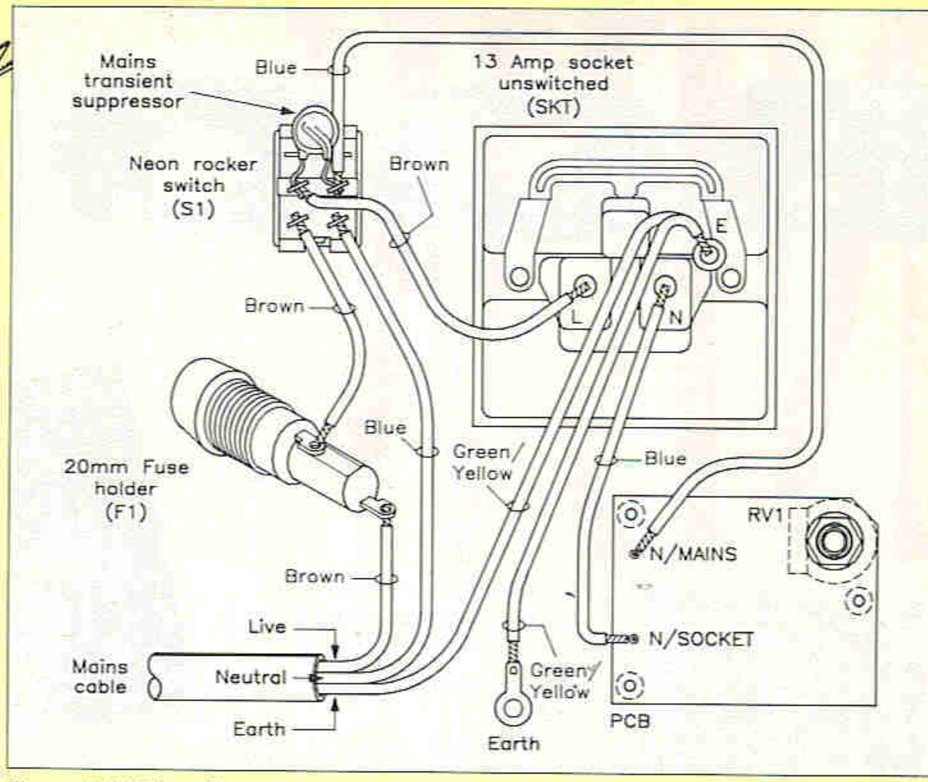


Figure 7. Wiring diagram.

Testing

On no account must power be applied to the circuit with the cover removed from the box. 240V AC mains is potentially lethal, it must therefore be treated with the greatest respect at all times.

A table lamp is required to test the Power Controller, plug the lamp into the Controller. Check to see if the Controller switch is in the 'OFF' position, and the control knob is set to the minimum (anti-clockwise). Plug the Controller into a 13A power socket, switch the Controller to 'ON', the switch should illuminate and the table lamp should be off. Turn the control clockwise and the lamp should start to glow and increases in brightness as the control is advanced further.

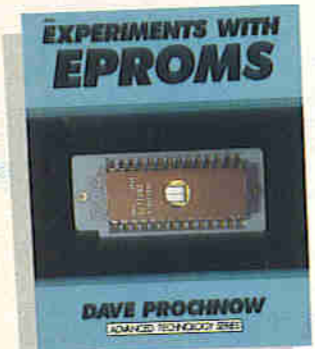
1·2kW POWER CONTROLLER PARTS LIST

RESISTORS: All 0·6W 1% Metal Film (unless specified)			
R1,2	39k 1W Carbon Film	2	(C39K)
R3	6k8	1	(M6K8)
R4	2M2	1	(M2M2)
RV1	1M Pot Lin	1	(FW08J)
CAPACITORS			
C1,2	100nF HV Cap	2	(FA21X)
SEMICONDUCTORS			
DI1	ST2	1	(QL08J)
TI1	BTA08-600B	1	(UK54J)
D1-4	1N4005	4	(QL77J)
MISCELLANEOUS			
L1,2	RF Supp Choke 3A	2	(HW06G)
SR1	Mains Trans Supp.	1	(HW13P)
S1	Dual Rocker Neon Red	1	(YR70M)
	Safuseholder 20	1	(RX96E)
FS1	5A A/S	1	(RA12N)
	PC Board	1	(GE62S)
	Box DCM5007	1	(LH72P)
	Heatsink TO126	1	(JX21X)
	3 Core 13A Mains Black	2Mtr	(XR09K)
	Grommet Small	1	(FW59P)
	Stick on Feet Square	1 Pkt	(FD75S)
	M3 x 15mm Insulated Spacer	1 Pkt	(FS37S)
	Pin 2141	1 Pkt	(FL21X)
	Knob RN18 Red	1	(FD67X)
	Single Socket Unswitched	1	(HL68Y)
	Cable Gland	1	(JR76H)
	M3 x 16mm Isobolt	1 Pkt	(JD16S)
	M3 Steel Nut	1 Pkt	(JD61R)
	M3 Isotag	1 Pkt	(LR64U)
	M3 Isoshake	1 Pkt	(BF44X)
	Stick on Front Panel	1	(JU35Q)
	Constuctors Guide	1	(XH79L)
OPTIONAL (not in kit)			
	Rubber Plug 13A	1	(HL58N)
	Plug Fuse 5A	1	(HQ33L)
	Insulator TO218	1	(UL74R)

The above items, excluding Optional, are available as a kit:
Order As LP41U (1·2kW Power Cntrlr) Price £20.95

The following items are also available separately:
1·2kW Pwr Cntrlr PCB **Order As GE62S Price £2.45**
1·2kW Cntrl Panel **Order As JU35Q Price £1.20**

NEW BOOKS

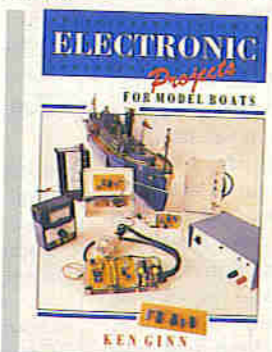


Experiments with EPROM's

by Dave Prochnov

One of the most advanced features in advanced circuit design is EPROM (Erasable Programmable Read-Only Memory) programming. Such memory chips are capable of holding any user specified program in a non-volatile form, and yet which can be quickly and easily erased at the user's discretion and reprogrammed to perform a different function. This book provides an in-depth look at these special integrated circuits to offer you no less than fifteen fascinating hands-on 'EPROM experiments' using the chips on their own in a logic circuit or in computer based applications. Each experiment comprises building different devices to use the EPROM's. These include a 'Boole's box', used for determining the logic levels present on digital IC pins; a keyboard encoder which generates a digit and displays the character of the key pressed; 'Bit Smasher I' and 'Bit Smasher II', extremely simple and more versatile EPROM programmers; 'Eprogrammer II', a versatile, portable, computer-based EPROM programmer; EPT-EPROM program tester, used to test and debug an EPROM's data *before* writing the data to the device; 'ROM Drive', a device that enables the test microcomputer to access the EPROM's code; a serial interface text-to-speech ASCII translation Speech Synthesiser; Music Synthesiser in the form of a programmable sound generator; 'Message Centre', a ROM based character generator; low-cost 4 and 8 watt EPROM erasers; and 'Three Line Burner', the Epromgrammer project converted to 2-way RS232 serial port communication with the host computer. Also contains much information about alternative ROM technologies, boolean logic and a comparison of the various logic gate technologies. American book.
1988. 233 x 186mm. 240 pages, illustrated.

Order As WT13P (Exp with EPROMs) Price £16.30 NV

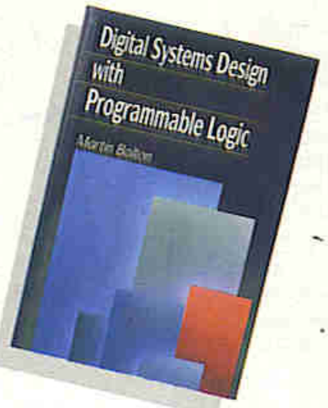


Electronic Projects for Model Boats

by Ken Ginn

Although radio control equipment is reliable and readily available nowadays, there are still many items which are not available commercially or which can be made more cheaply at home. This book describes how to make some of these items, which include servo checkers, a battery tester, glow plug supply and a multimeter, with no prior knowledge of electronics required. The book begins with an overview of the necessary tools, board construction techniques and basic R/C layout, moving on to discuss specific projects, each featuring a circuit description with full construction and testing procedures. The book is aimed at the boat modeller who likes to get involved in more than just the building of the model and who will, with the help of its contents, be able to lift the electronic equipment to a higher standard also. No previous experience as a circuit builder is necessary, and detailed explanations for soldering techniques etc. are provided for the absolute beginner, and appendices cover such aspects as resistor colour codes, component recognition, how to mount power semiconductor devices to heatsinks and component suppliers.
1990. 210 x 145mm. 144 pages, illustrated.

Order As WT12N (Projects for Mdl Bts) Price £6.95 NV

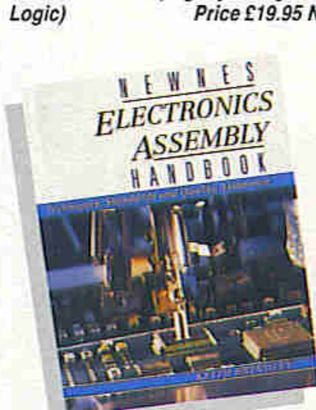


Digital Systems Design

with Programmable Logic by Martin Bolton

Programmable logic devices (PLD's) bring indispensible benefits in terms of speed, flexibility and reliability to digital systems design, and so the modern engineer needs to be familiar with the design principles of programmable logic. This book provides a clear and careful introduction to modern, structured digital systems design from a programmable logic perspective. Extensively covering synchronous and asynchronous techniques, a 'top-down', implementation-independent approach to design is taken throughout. Topics covered include extensive coverage of state machine design, an appendix listing all known PLD's, test objectives, summaries and problems with each chapter, and a very comprehensive bibliography. An excellent introduction to programming logic for the practising professional and dabbling amateur alike.
1990. 240 x 160mm. 384 pages, illustrated, hard cover.

Order As WT10L (Digi Sys Prog Logic) Price £19.95 NV



Newnes Electronic Assembly Handbook

by Keith Brindley

In its broadest sense electronics assembly is all about how to bring together a multitude of electronic components and component types to form a working, reliable piece of equipment. Nowadays many of the basic decisions and problems have been overcome by the integrated circuit manufacturers, who have combined hundreds, thousands or even hundreds of thousands of individual components onto single devices effectively easing the job of the electronics assembler, allowing him to construct more complex appliances for the same amount of work.
Nevertheless many problems still remain and often the very use of large scale integrated circuits has generated more problems. This book is essentially about printed circuit boards, their design, manufacture and assembly, production, component parts, packaging, testing, standardisation, and quality. For PCB's of one sort or another are used in all but the simplest of electronic assemblies. In many respects, the whole topic of electronics assembly may be viewed as just a collection of the multitude of considerations that apply to printed circuit boards.
This is a highly illustrated book, initially aimed at managers, engineers and technicians at a professional level involved at any stage of electronic product manufacture. It is, however, a fascinating insight into commercial electronics assembly techniques and considerations, and will be a source of inspiration and a guide to the amateur projects builder. Did you know for instance that, as far as the industry is concerned at any rate, it is *not* taboo to mount components on *both* sides of a single sided PCB? Is treating the 'component side' and the 'track side' as two entirely separate domains a 'rule' you have always followed implicitly, and the twain shall never meet? Did you know that the solder pads of either single or double sided PCB's need not be drilled at all, but the components soldered directly to the track surface, and that this is how modern micro mounting technology achieves high density PCB's? Ever wondered how flow soldering works?
The book covers electronics assembly to a depth which, the publishers believe, has not been seen in print before. Contents include an introduction to electronics assembly, PCB and surface mount assembly, electromechanical assembly, packaging, soldering and quality testing, standardisation of manufacture and the worldwide standards, quality assurance, and an interesting chapter on the performance involved for a manufacturer seeking to secure the MoD as a customer.
1990. 250 x 195mm. 350 pages, illustrated, hard cover.

Order As WT15R (Electronics Assembly) Price £40.00 NV

compuguard

VEHICLE ALARM

Part 1 by Tony Bricknell

FEATURES

- ★ Fully programmable
- ★ Infra-red arm/disarm remote control
- ★ Low power consumption
- ★ Optional battery backup
- ★ Suitable for all negative earth vehicles
- ★ Microprocessor controlled

Introduction

In Britain a car is stolen every 30 seconds costing £722 million in 1988. In Essex alone, the past twelve months have seen over 24,000 thefts of, or from, vehicles. Approximately 30% of nation-wide crime is against vehicles. To add to this, 70% of all cheque book and credit card thefts are from cars. This all adds up to a £750 million bill for insurance and police time.

With over 20 million car owners in the U.K., the chances of *your* vehicle being broken into is frighteningly high. A high percentage of vehicles are not alarmed, surprising when you consider that owners will part with over £6,000 for a car, but do not install an alarm system costing only a minute proportion of the total cost of the vehicle.

Many vehicle alarms presently on the market work by monitoring battery voltage, movement, normally open and normally closed switches/security loops, but few (you can count the number on one hand) allow the user to program which sensors are active at any one time. For instance, while parked outside your house which is situated in a quiet street, all sensors can be activated. But if parked in your local High Street, vibrations from large goods vehicles may trigger the movement sensor. Compuguard allows the user to 'deactivate' this one sensor, whilst leaving the rest of the alarm system fully functioning. Being microprocessor controlled, Compuguard is semi-intelligent and will actually 'mould' itself to the surrounding environment, constantly monitoring and modifying its outlook on the security of the vehicle.

Compuguard comes in three parts:

1. Main Processing Unit. This houses a microcontroller, optional battery backup, shock and voltage drop sensors, control relays and service keyswitch. The unit is impregnable to water and is intended to be mounted in the engine compartment.

2. Switch/Infra-red Receiver Unit. This houses the infra-red receiver/decoder, and the control switches to select the activated sensors. This unit is mounted inside the passenger compartment.

3. Infra-red Transmitter Unit. This hand-held transmitter that can easily be attached to a key-ring. Depressing the recessed tactile switch button toggles the alarm between activated and deactivated states.

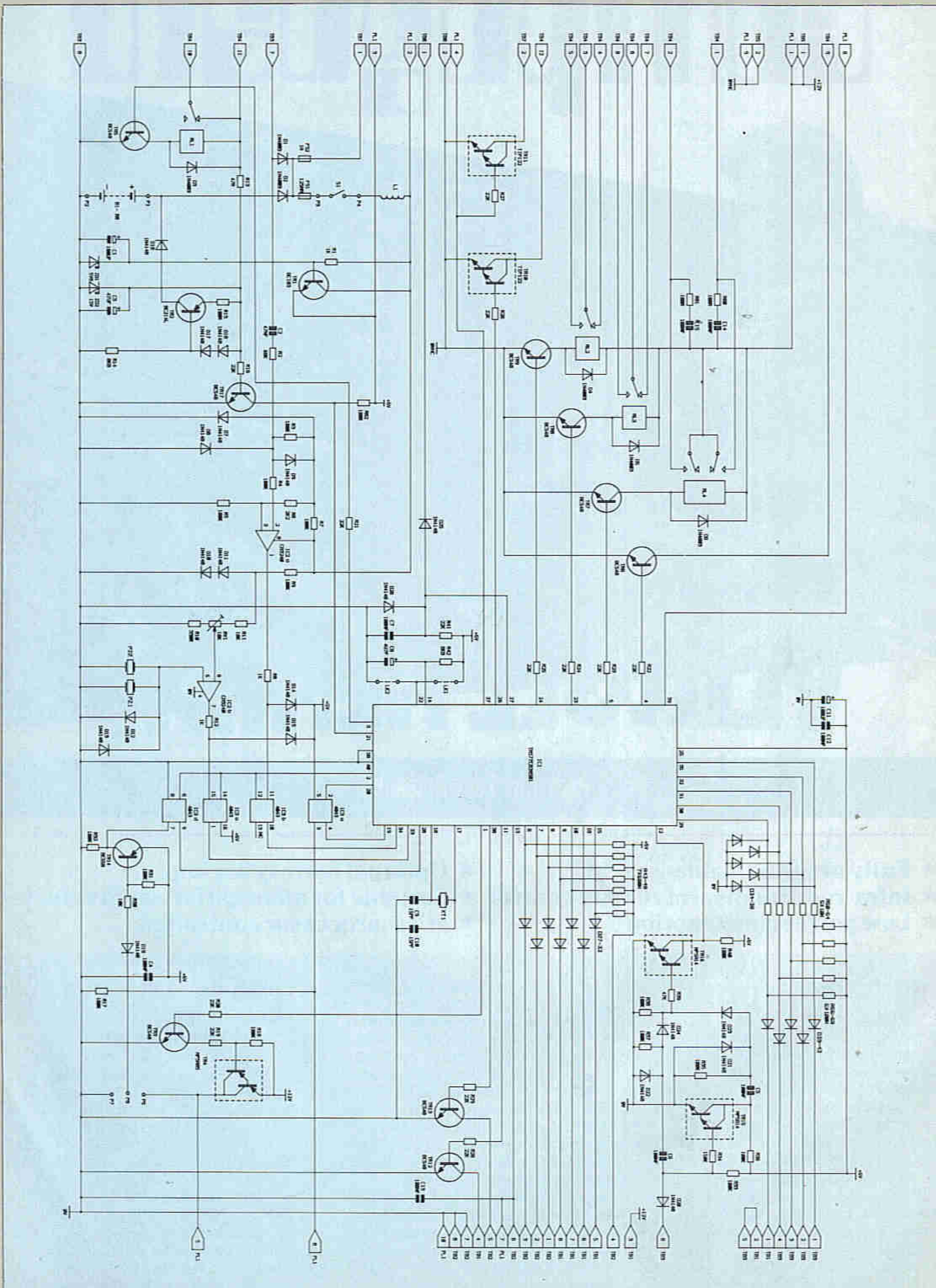


Figure 1. Circuit diagram.

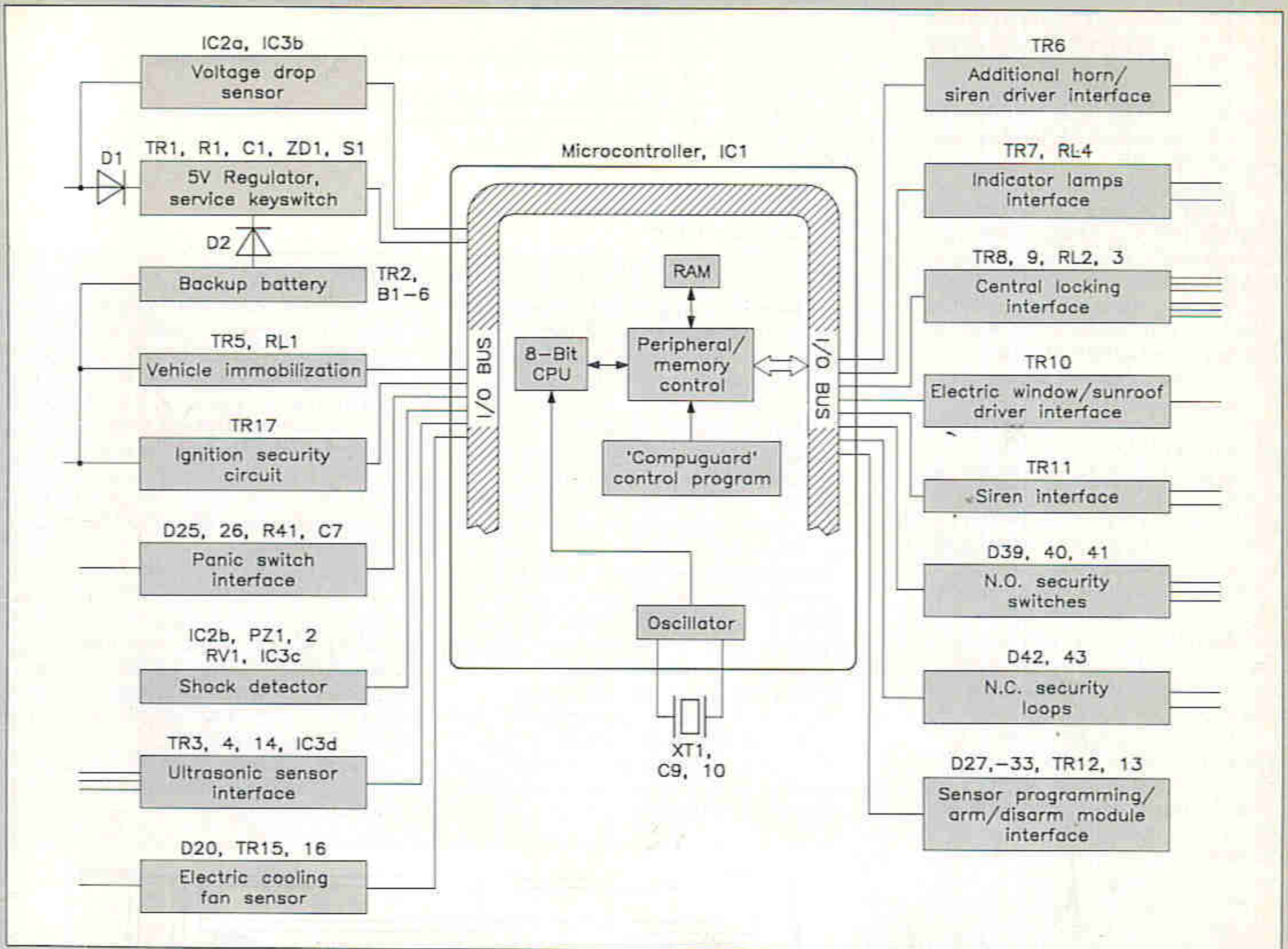


Figure 2. Block diagram of Compuguard.

The alarm system protects the vehicle by disabling the ignition circuit (or fuel pump for diesel engines), and triggering an audio-visual alarm lasting for 60 seconds, immediately upon any one or more of the following occurrences:

- Opening any door/boot which activates a light of at least 3 watts;
- Opening any door/boot/bonnet protected by a normally open pushbutton;
- Physical shock to the vehicle;
- Tampering with connecting cables, internal and external accessories (radiocassette, fog lamps etc).

In addition to these conditions, the alarm may be triggered manually into a 'Panic' state by:

a) Depressing an internal switch for more that 2 seconds,

b) Depressing the IR/TX button twice within 2 seconds (this will also disable the ignition).

To reset a 'Panic' condition, the IR/Tx button needs to be pressed once.

An ignition security circuit is incorporated to prevent the alarm system accidentally arming itself or being armed whilst the vehicle is in use. This means that the IR remote control is inactive whilst you are driving.

The triggered, 60 seconds alarm cycle duration is repeated every time a new violation to the vehicle occurs. Delay times are provided when the alarm arms, and between alarm cycles to allow all sensors to settle.

The intelligent processor monitors all sensors and, in the event of a sensor becoming faulty, will 'learn' to ignore it, thereby stopping the vehicle's battery being drained due to the alarm being sounded repeatedly or continuously.

In addition Compuguard has outputs for automatic operation of power door locks, and will also close electric windows/sunroofs when the alarm is armed.

Two security loops are provided, allowing protection of internal and external accessories. Removing an accessory breaks the loop, triggering the alarm.

All sensors (except bonnet) are programmable so that, for instance, the boot sensor only can be deactivated, very usefully allowing you to be able to load/unload the boot whilst the rest of the vehicle is protected.

Other alarm systems currently on the market which offer you features approaching those of Compuguard can cost in the region of £400. This makes Compuguard the ideal solution to your vehicles security needs at a *very* competitive price!

Circuit Description

In addition to the circuit shown in Figure 1, a block diagram is detailed in Figure 2. This should assist you when following the circuit description or fault finding in the completed unit.

The heart of Compuguard is the TMS77C82 Microcontroller IC1. This device, a complete micro-computer system on a chip, requires a stable supply of +5V which is provided by R1, C1, ZD1 and TR1. This is also used to supply IC3. During a power failure, Compuguard runs from the optional Ni-Cad batteries via D2, which are charged when the ignition is on via TR2 and associated circuitry.

IC2a provides a highly effective voltage drop detector. Under normal conditions pin 2 is held at 6V, and Pin 3 at 0.59V, maintaining a low output. A voltage drop on the +12V rail generates a negative going pulse which is coupled via C2 and R2 to pin 2 and causes pin 1 to pulse high. This output, clamped by R8 and D14 is latched by IC3b. It is the sensitivity of this part of the circuit that enables the switching on of bootlid or door courtesy lamps down to 3W to be noticed.

The second half of IC2 forms a movement detector. Under normal conditions (i.e., no movement), pin 6 is held at a higher potential than pin 5, keeping its output pin 7 low. When the unit is subjected to mechanical shock, two piezo transducers develop a voltage which is clamped by D12 and D13. As soon as this voltage exceeds that set by RV1 on pin 6, the output pulses high. This is clamped by R12 and D15, and latched by IC3c.

Provision for an external ultrasonic sensor has been allowed via P5, P6 and P7. P5 provides a switched +12V output when Compuguard is active and P6, when taken low, triggers the alarm.

TB3-6 (Cathode of D20) is connected to the thermal switch of vehicles which are fitted with electric cooling fans. TR15 and TR16 and associated circuitry generate an interrupt to IC1 whenever the fan switches on or off. This provides immunity to false triggering caused by the cooling fan switching while the alarm is active.

External switches/security loops are connected to IC1 via R50 – R53 and D39 – D43. The infra-red receiver/sensor programming PCB (to be described in part two of this series) connects to IC1 via D27 – D33 and TR12 and TR13. Diodes D27 – D43 are needed to protect IC1 from the harsh environment of a vehicle.

Vehicle immobilisation is provided by RL1 cutting the supply to the ignition coil, electric fuel pump, etc. To reduce quiescent current consumption, the relay is only energised when an attempt is made to start the vehicle.

RL2 and RL3 provide change over contacts to control central locking (RL2 being active to lock all doors, RL3 active to unlock). To cater for the fact that vehicles with vacuum operated locks require an extended lock/unlock pulse, the active duration of RL2, RL3 is set by LK1, LK2.

RL4 provides +12V outputs to allow the vehicle indicator lamps to flash and acknowledge arming/disarming of the unit. It will also provide a visible indication of an alarm condition.

TR10 provides a firm ground for 15 seconds after the alarm arms to activate an external relay used to close electric windows and/or sunroof.

Table 1 shows the signals brought out to PL1. This connector will be used for future developments.

Pin 1	+12V High Current
Pin 2	+12V Constant Live (drops to approx 7·2V in battery backup mode)
Pin 3	+5V Regulated
Pin 4	+5V When alarm sounding (low current)
Pin 5	+12V When alarm armed (300mA max.)
Pin 6	Reserved for future expansion
Pin 7	Arm/Disarm Control
Pin 8	Trigger Input
Pin 9	0V High Current
Pin 10	0V Logic

Table 1. Compuguard PL1 expansion bus pin definitions.

Construction

The glassfibre PCB is of the double-sided, plated-through hole type, chosen for maximum reliability and stability. However, because the holes are plated through, removal of a misplaced component will be quite difficult, so *please* double-check each component type, value and its polarity where appropriate, before soldering! For further information on component identification and soldering technique please refer to the Constructors' Guide included with the kit.

Figure 3 shows the PCB, with printed legend, to help you correctly locate each item. The sequence in which the components are fitted is not critical; however, the following instructions will be of use in making these tasks as straightforward as possible.

Start by inserting the PCB pins P1 – P7, and press them into position using a hot soldering iron. When the pins are heated in this way, very little pressure is required to push them in place. Once the pins are in position they can be soldered. Generally it will now be easier to start with the smaller components, such as the resistors, and work upwards in size until the relays are fitted last.

Semiconductors (diodes and transistors) need special precautions when fitting. In particular, take care not to overheat them during soldering. All the silicon diodes have a band identifying one end, be sure to position these adjacent to the white blocks marked on the legend. Take care with the polarity of the electrolytic and tantalum capacitors. The polarity for both types of capacitor is shown by a plus sign (+) matching that on the PCB legend. However, on the actual body of most electrolytic capacitors, the polarity is designated by a full length stripe with negative symbol (−), in which case the lead nearest *this* symbol goes in the hole *opposite* to that adjacent to the positive sign on the legend.

Install all the transistors, matching the shape of each case to its outline on the legend. Beware of over-heating transistors, allow several seconds between soldering each lead-out for re-cooling. When fitting the IC sockets, ensure that you install the appropriate one at each position, matching the notch in the end of the socket with the white block on the legend. *Do not* install the IC's until they are called for during the test procedure!

If you are installing the alarm into a vehicle with central locking, then it is necessary to ascertain whether the vehicle employs vacuum operated locks (the most common cars to use these are Mercedes and Audi). If so, install link LK2, otherwise fit link LK1 (even if you are not using the central locking outputs).

Insert and solder the seven terminal blocks, taking note of the orientation as shown in Figure 4.

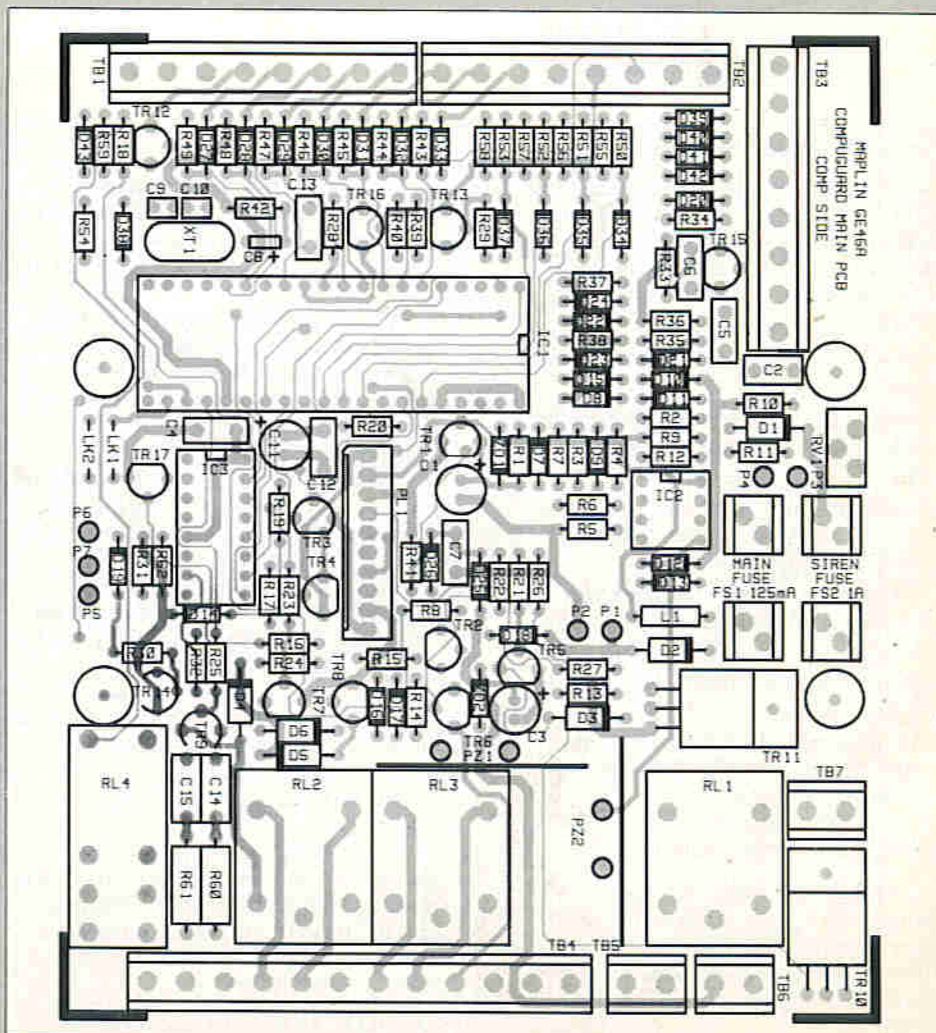


Figure 3. PCB layout and legend.

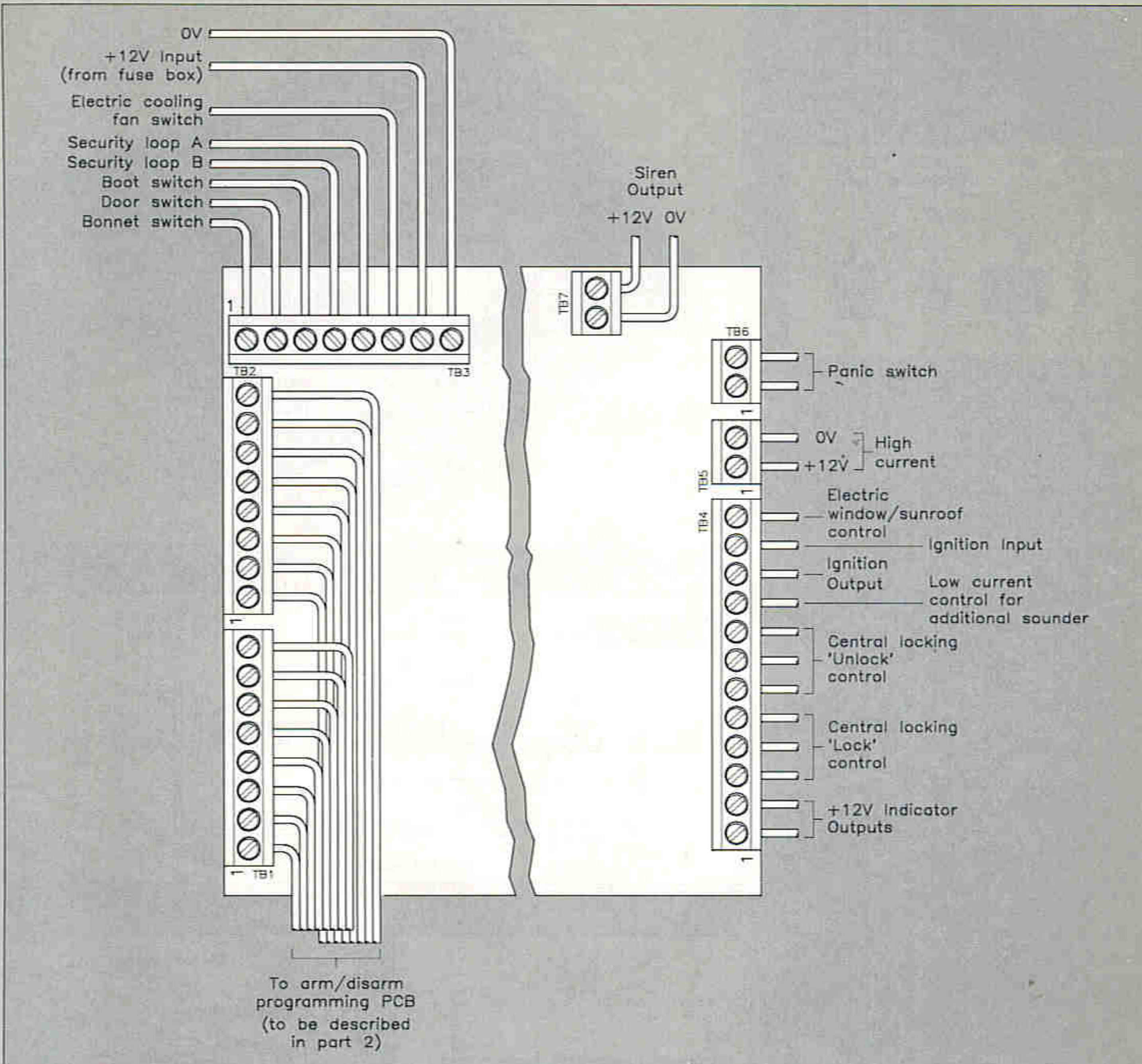


Figure 4. Orientation and connection of terminal blocks.

Remove the red and black wires from both piezo transducers PZ1 and PZ2. This must be carried out extremely carefully and with the minimum of heat for a fraction of a second, or destruction of the silvered electrode begins to occur. The brass rim of the transducer acts as a heatsink and will require more heat, but the shorter the contact time with the soldering iron the better.

You now need to attach to the small solder pads on both PZ1 and PZ2 two short, solid core wire off-cuts left over from the components previously installed (I hope you didn't throw them away!), or else use short lengths of 22 s.w.g. tinned copper wire. Carefully insert and solder the ends of these new wires for PZ1 and PZ2 into their adjacent holes on the PCB, ensuring that the transducers are not touching each other.

- If you are using the optional battery backup, solder the PP3 battery clip to P1 (+ or red) and P2 (– or black), otherwise join P1 and P2 with a shorting link.

Final Assembly

The PCB is designed to fit into a waterproof box type YM91Y. Drill the sides of the box as in Figure 5. Fit the water-tight cable glands as shown in Figure 6. The PCB

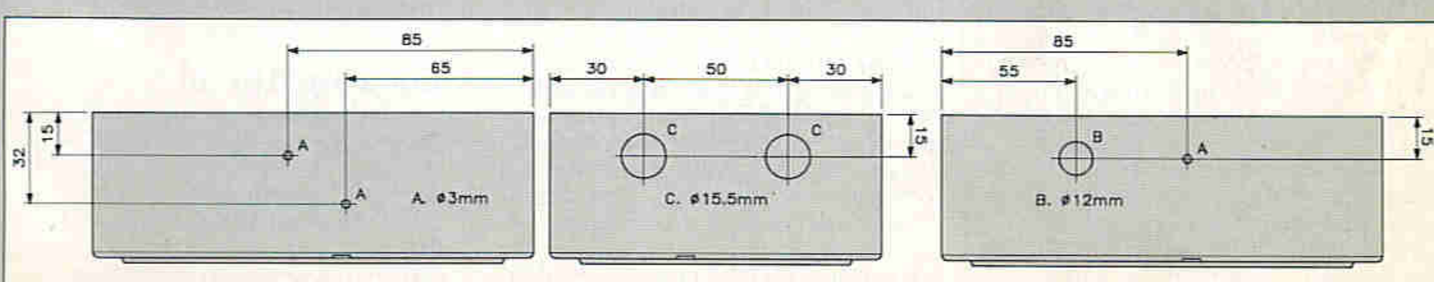


Figure 5. Drilling dimensions.

is fitted into the box by four 3/8in. No.4 self-tappers and four 1/8in. M3 spacers, shown in Figure 7.

Fit the key-switch in the side of the box and connect it to P3 and P4 via short lengths of cable.

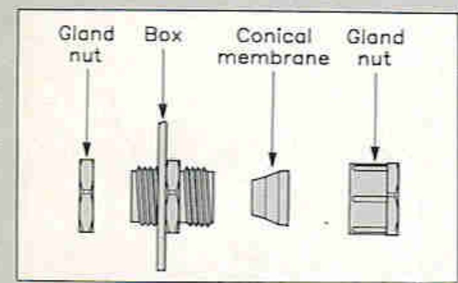


Figure 6. Fitting the water-proof cable glands.

If the option of an external ultrasonic sensor is required, then an additional hole has to be drilled for the 3·5mm Stereo Jack Socket as shown in Figure 8a. Figure 8b shows the wiring between the PCB P5, P6, P7 and the socket. Most ultrasonic sensors fitted with a 3·5mm stereo jack plug will work from this socket.

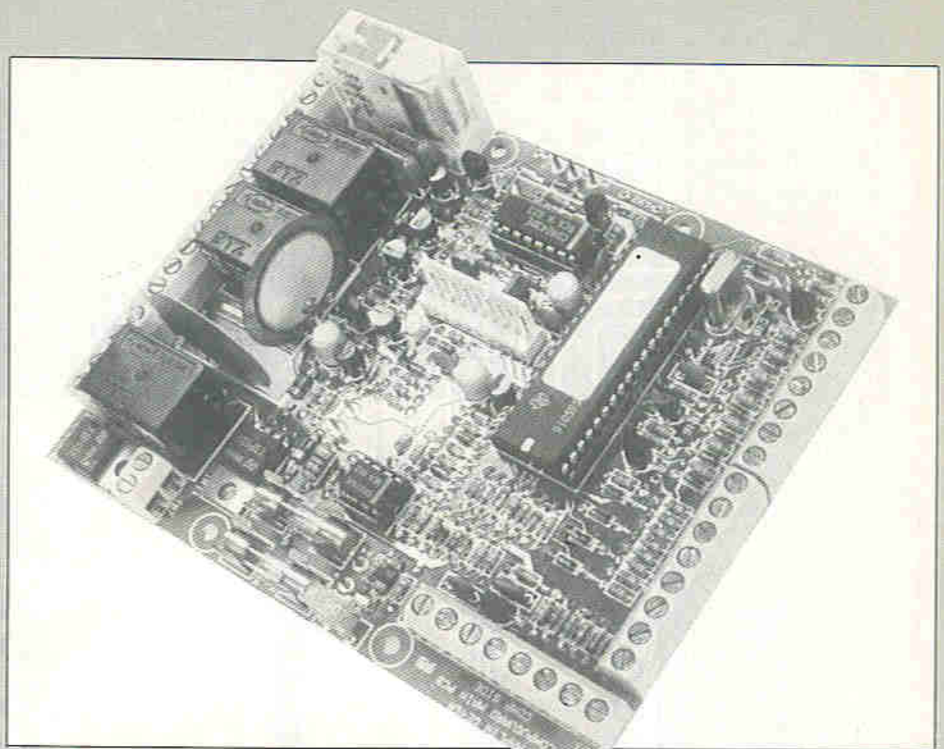
Holes should be made in the top of the box for the mounting of a siren. The choice of siren (and therefore arrangement of the mounting holes) has been left to the user. Maplin supply a wide range of sirens at various output sound pressure levels. Table 2 gives a comparison of all the sirens suitable at the time of this issue of *'Electronics'* going to press. The best way to locate the hole positions in the top of the box is to remove the siren's mounting bracket and use this as a template. Note that one additional hole will be required for the cable.

Testing

Table 3 gives the connections of TB1 to TB7. Note that not all wires need to be connected for a fully working system! Testing the unit can be made with the minimum of equipment. You will need a multimeter and a +12V supply (a car battery will be suitable). The readings were taken from the prototype using a digital multimeter, some of the readings you obtain may vary slightly depending upon the type of meter employed.

Double check your work to make sure that there are no dry joints or short circuits, that none of the IC's have been fitted into the sockets on the board and that FS1 and FS2 are installed. Insert and turn the service-key clockwise. The first test is to ensure that there are no short circuits. Set your multimeter to read OHMS (Ω) on its resistance range, connect the test probes to TB3-7 and TB3-8. With the probes either way round a reading greater than 10kΩ should be obtained. Now connect the probes to TB5-1 and TB5-2. Again, a reading of higher than 100kΩ should be obtained with the probes either way round.

Set your meter to read DC mA. If installing the optional backup-batteries then place the probes between P1(+) and P2(−). Apply +12V to TB4-11 and 0V to TB3-8. A current reading of approximately 6mA should be obtained.



The completed PCB.

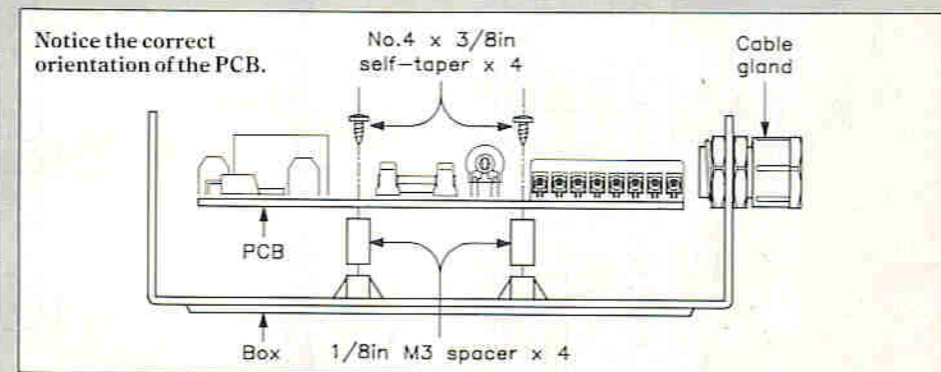


Figure 7. Fitting the PCB in the box.

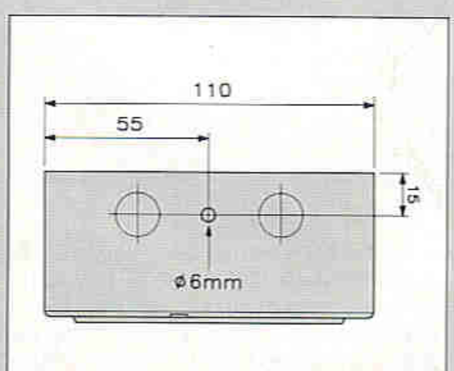


Figure 8a. Drilling for the 3·5mm socket.

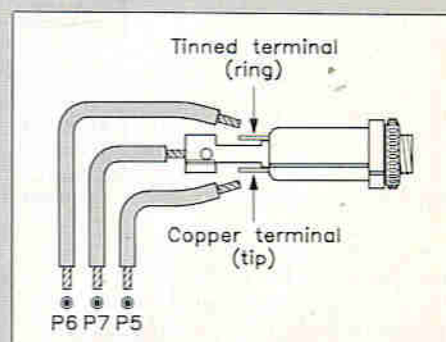


Figure 8b. Wiring the 3·5mm socket to the PCB.

Turn the service key-switch counter-clockwise and monitor the supply current. With your meter reading DC mA place it in the positive line of the power supply. Connect TB3-7 to +12V (through the multimeter) and TB3-8 to 0V. A reading of approximately 0mA should be registered. When the key-switch is turned clockwise this should increase to approximately 6mA. Remove the supply and turn the service key-switch counter-clockwise. Install the IC's making sure that all the pins go into their sockets properly and that the pin one marker on the IC package is at the notched end of the socket. Power up the unit by turning the key-switch clockwise and observe the current reading which should now be approximately 11mA. Remove the power supply and the test meter, and turn the key-switch counter-clockwise.

Looking Ahead

All you have to do now is sit and wait for the next edition of *'Electronics – The Maplin Magazine'*, December 1990 – January 1991, in which we will give full constructional details of the Sensor Programming PCB, Infra-red remote control and details on installing the complete system in your vehicle.

Description	Current Consumption @ 12V	Sound Output	Size	Stock Code
Micro Piezo Siren	150mA	110dB @ 1m	43 x 39 x 59mm long	JK42V
Miniature Piezo Siren	300mA	115dB @ 1m	61 x 56 x 86mm long	JK43W
Piezo Electronic Siren	30mA	110dB @ 1m	60mm dia. x 98mm long	YP11M
Staccato Electronic Sounder	300mA	116dBA	90mm dia. x 110mm long	YZ03D

Table 2. Comparison of suitable sirens.

TB1–1 to TB1–8	To Infra-red Receiver/ Sensor Programming PCB (Described in Part Two)
TB2–1 to TB2–8	To Infra-red Receiver/ Sensor Programming PCB (Described in Part Two)
TB3–1	Bonnet Switch Input
TB3–2	Door Switch Input
TB3–3	Boot Switch Input
TB3–4	Security Loop B Input
TB3–5	Security Loop A Input
TB3–6	Electric Fan Switch Input
TB3–7	+12V Low Current Input (Taken from fuse box)
TB3–8	0V Low Current Return
TB4–1	+12V Indicator Output
TB4–2	+12V Indicator Output
TB4–3	Central Locking 'Lock' Signal POLE
TB4–4	Central Locking 'Lock' Signal N.C.
TB4–5	Central Locking 'Lock' Signal N.O.
TB4–6	Central Locking 'Unlock' Signal POLE
TB4–7	Central Locking 'Unlock' Signal N.C.
TB4–8	Central Locking 'Unlock' Signal N.O.
TB4–9	Low Current Pulsed Output For Additional Siren/Horn
TB4-10	Ignition Output
TB4-11	Ignition Input
TB4-12	Electric Window Control Output
TB5–1	+12V High Current Input
TB5-2	0V High Current Return
TB6–1	Panic Switch Input
TB6–2	Panic Switch 0V
TB7–1	+V Output for Siren
TB7–2	0V Return for Siren

Table 3. Connection of TB1 to TB7.

COMPUGUARD MAIN UNIT PARTS LIST

RESISTORS: All 1/8W Carbon Film (Unless specified)			
R1,8,12	1k	3	(U1K)
R2	68k	1	(U68K)
R3-5,7,9,17,18, 31,32,33,35-38, 40,55-59,62	100k	21	(U100K)
R6	2k2	1	(U2K2)
R10	330R	1	(U330R)
R11,30,50-54	10k	7	(U10K)
R13	47R	1	(U47R)
R14,42	6k8	2	(U6K8)
R15	100R	1	(U100R)
R16,19-29,41	22k	13	(U22K)
R34	150k	1	(U150K)
R39	47k	1	(U47K)
R43-49	680k	7	(U680K)
R60,61	100R Metal Film	2	(M100R)
RV1	10k Vert. Encl. Preset	1	(UH16S)
CAPACITORS			
C1,11	100μF 10V Minelect	2	(RK50E)
C2	47nF Monores Cap	1	(RA47B)
C3	47μF 16V Minelect	1	(YY37S)
C4-7,12,13	100nF Minidisc	6	(YR75S)
C8	4μ7F 16V Tantalum	1	(WW64U)
C9,10	15pF Ceramic	2	(WX46A)
C14,15	100nF Polyester	2	(BX76H)
SEMICONDUCTORS			
D1-6	1N4003	6	(QL75S)
D7-43	1N4148	37	(QL80B)
ZD1	BZY88C5V6	1	(QH08J)
ZD2	BZY88C15V	1	(QH18U)
TR1	BC183L	1	(QB56L)
TR2	BC214L	1	(QB62S)
TR3,5-9,12,13,17	BC548	9	(QB73Q)
TR4	MPSA65	1	(QH61R)
TR10,11	TIP122	2	(WQ73Q)
TR14	BC558	1	(QQ17T)
TR15,16	MPSA14	2	(QH60Q)
IC1	TMS77C82 MS01	1	(UL63T)
IC2	CA3240E	1	(WQ21X)
IC3	4043BE	1	(QW29G)

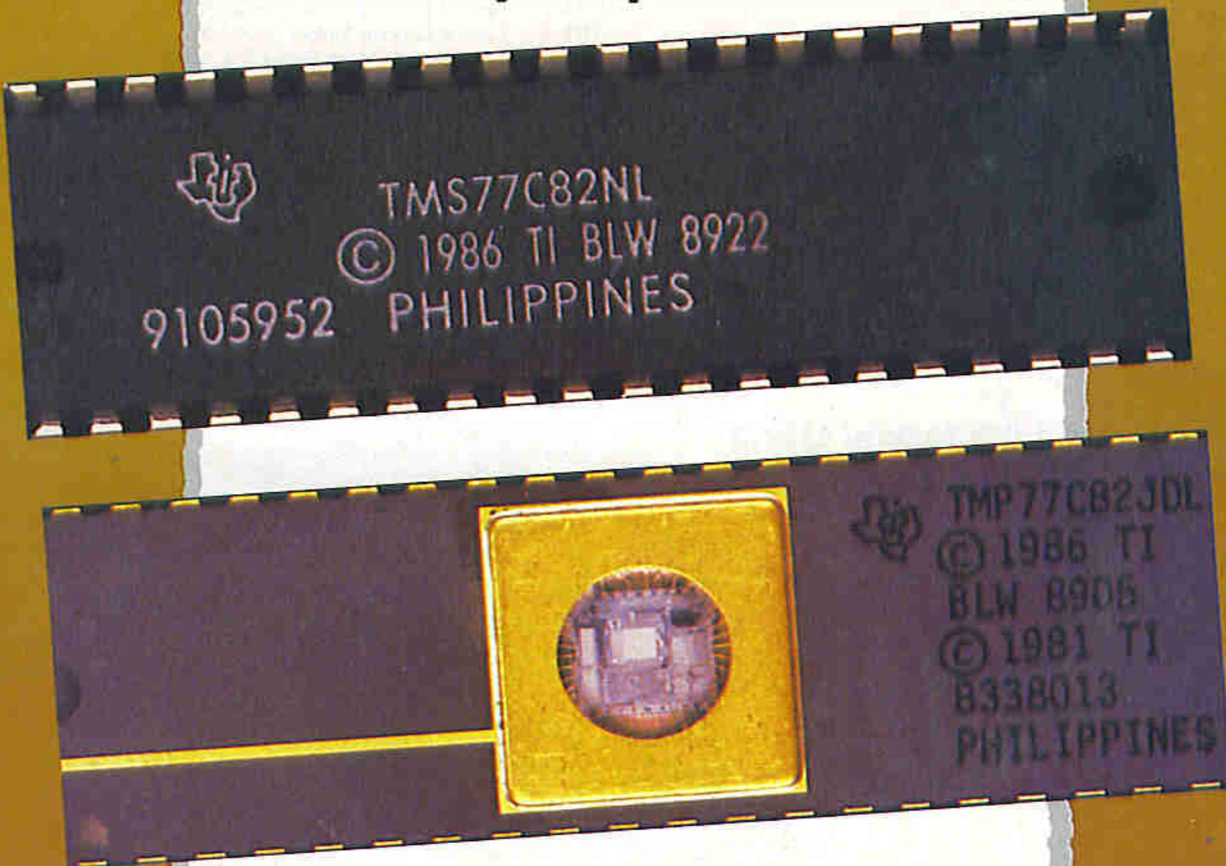
MISCELLANEOUS			
SW1	Min Key Switch	1	(FE44X)
TB1,2,3	8-Way PC Terminal	3	(RK38R)
TB4	12-Way PC Terminal	1	(RK74R)
TB5,6,7	2-Way PC Terminal	3	(FT38R)
PL1	Minicon Latch Plug 10-way	1	(RK66W)
P1-7	Veropin 2145	1 Pkt	(FL24B)
RL1-3	12V/10A Min Relay	3	(JM67X)
RL4	5A Mains Relay	1	(YX98G)
FS1	Fuse 20mm 125mA	1	(UJ75S)
FS2	Fuse 20mm 1A	1	(WR03D)
	Fuse Clip	4	(WH49D)
L1	100μH Choke	1	(WH41U)
PZ1,2	Piezo Transducer 27/4·2	2	(QY13P)
XL1	2MHz MP Crystal	1	(FY80B)
	Medium Waterproof Box	1	(YM91Y)
	Compuguard Mounting Bracket	1	(JR77J)
	Cable Gland 5-8	2	(JR76H)
	Compuguard Main PCB	1	(GE46A)
	Alarm Sticker	2	(JR91Y)
	Quick Snap Connector	6	(JR88V)
	Constructors Guide	1	(XH79L)
	Isobolt M3 12mm	1 Pkt	(BF52G)
	Isoshake M3	1 Pkt	(BF44X)
	Springwash M3	1 Pkt	(JD96E)
	Isonut M3	1 Pkt	(BF58N)
	M3 Spacer ⅛in	1 Pkt	(FG32K)
	Self-Tapper No. 4 x ⅜in	1 Pkt	(BF65V)
	DIL Socket 8-Pin	1	(BL17T)
	DIL Socket 16-Pin	1	(BL19V)
	DIL Socket 40-Pin	1	(HQ38R)
OPTIONAL (not in kit)			
B1-6	Re-Chargeable Type AA Batt	6	(YG00A)
	9V Battery Holder	1	(HQ01B)
	PP3 Battery Clip	1	(HF28F)
	3·5mm Stereo Chassis J/Skt	1	(FK03D)
	Additional Q/Snap Connctrs	As Req	(JR88V)
	12V Siren		'See Text'
	Motor-Start Press	1	(FH91Y)
	Power Connection Wire Black	As Req	(XR32K)
	Power Connection Wire Red	As Req	(XR36P)

The above items, excluding Optional, are available as a kit:
Order As LP22Y (Compuguard Main Unit) Price £64.95

The following items are also available separately:
Compuguard Bracket **Order As JR77J Price £3.95**
Compuguard Main PCB **Order As GE46A Price £11.95**
TMS77C82 MS01 **Order As UL63T Price £16.95**

TMS77C82 MICRO CONTROLLER

Reviewed by Tony Bricknell Part 2.



Features:
- ★ CMOS technology
- ★ Low power 'sleep' modes
- ★ 8K on-chip eraseable EPROM
- ★ 256 byte on-chip RAM
- ★ 32 I/O pins
- ★ On-chip serial port
- ★ Three on-chip timers/event counters

Applications:
- ★ Security systems
- ★ Autodialers
- ★ Robotics
- ★ Bar-code readers
- ★ Trip computer for motor vehicles

ADDRESSING MODE	EXAMPLE		
Single Register	LABEL	DEC INC CLR	B R45 R23
Dual Register	LABEL	MOV ADD CMP	B,A A,R17 R32,R73
Peripheral File	LABEL	XORP MOVP	A,P17 P42,B
Immediate	LABEL	AND ANDP BTJO	%>C5,R55 %VALUE,P32 %>D6,R80,LABEL
Program Counter Relative	LABEL1	JMP DJNZ BTJO BTJOP	LABEL A,LABEL %>16,R12,LABEL B,P7,LABEL
Direct Memory	LABEL	LDA CMPA	@>F3D4 @LABEL
Register File Indirect	LABEL	STA	★R43
Indexed	LABEL2	BR	@LABEL(B)

Table 8. TMS77C82 addressing modes.

SYMBOL	DEFINITION	SYMBOL	DEFINITION
A	Register A or R0 in Register File	B	Register B or R1 in Register File
Rn	Register n of Register File	Pn	Port n of Peripheral File (0 ≤ n ≤ 255)
s	Source operand	d	Destination operand
Rs	Source register in Register File	Ps	Source register in Peripheral File (0 ≤ s ≤ 255)
Rd	Destination register in Register File	Pd	Destination in Peripheral File (0 ≤ d ≤255)
Rp	Register pair	iop	Immediate operand
ST	Status Register	SP	Stack Pointer
PC	Program Counter	pcn	Location of the next instruction
$	Current value of Program Counter	b	Bit number, as in b7 (0 ≤ b ≤ 7)
offset	Relative Address (offset = ta – pcn)	ta	Target Address (ta = offset + pcn)
@	Indicates an address or label	%	Indicates immediate operand
★	Indicates Indirect Register File Addressing mode	<XADDR>	Indicates an extended address operand
?	Binary number	>	Hexadecimal number
MSB	Most significant byte	LSB	Least significant byte
MSb	Most significant bit	LSb	Least significant bit
cnd	Condition	( )	Contents of
→	Is assigned to	←	Becomes equal to
[ ]	Indicates an optional entry. The brackets themselves are not entered.	< >	Indicates something that must by typed in. For example, <offset> indicates that an offset must be entered. The brackets themselves are not entered.

Table 9. Symbols used in the instruction set.

Instruction Set

The TMS77C82 instruction set contains 62 instructions that control input, output, data manipulation, data comparison, and program flow. The instruction set can be divided into eight functional categories:

- Arithmetic instructions
- Branch and Jump instructions
- Compare instructions
- Control instructions
- Load and Move instructions
- Logical instructions
- Shift instructions
- I/O instructions

TMS77C82 Assembly Language supports eight addressing modes, listed in Table 8, to provide the flexibility to optimise programs to the user's applications. Addressing modes that use 16-bit operands are sometimes referred to as extended addressing modes.

Table 9 lists and defines the symbols used in the instruction set. Formats, opcodes, byte lengths, cycles/instruction, operand types, status bits affected, and an operational description for each instruction is given in Table 10.

Programming the TMS77C82

The 77C82 can be programmed like any 27C64 on a wide variety of PROM programmers. Programming the 77C82 requires a 40-to-28 pin adaptor socket with the RESET and XTAL2 pins grounded. Figure 9 shows the necessary connections for the adaptor socket.

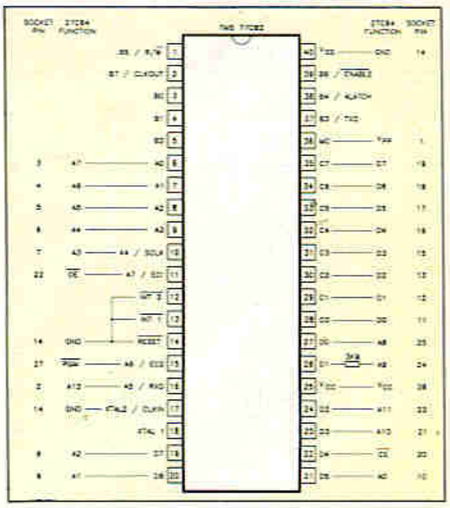


Figure 9. Connections required for 40-to-28 pin adaptor.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
ADC	B,A	69	1	5	R R R x	Add with carry instruction
	Rs,A	19	2	8		(s) + (d) + (C) → (d)
	Rs,B	39	2	8		Add the source, destination
	Rs,Rd	49	3	10		and carry bit together. Store
	%iop,A	29	2	7		at the destination address.
	%iop,B	59	2	7		
	%iop,Rd	79	3	9		
ADD	B,A	68	1	5	R R R x	Add instruction
(Continued	Rs,A	18	2	8		(s) + (d) → (d)
overleaf)	Rs,B	38	2	8		Add the source and destination

Table 10.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
	Rs,Rd	48	3	10		operands at the destination address.
	%iop,A	28	2	7		
	%iop,B	58	2	7		
	%iop,Rd	78	3	9		
AND	B,A	63	1	5	0 R R x	AND instruction.
	Rs,A	13	2	8		(s) .AND. (d) → (d)
	Rs,B	33	2	8		AND the source and destination
	Rs,Rd	43	3	10		operands together and store at
	%iop,A	23	2	7		the destination address.
	%iop,B	53	2	7		
	%iop,Rd	73	3	9		
ANDP	A,Pd	83	2	10	0 R R x	AND Peripheral File Register
	B,Pd	93	2	9		(s) .AND. (Pn) → (Pn)
	%iop,Pd	A3	3	11		AND the source and destination operands together, and store at the destination address.
*See note 1					0 R R x	
BTJO	B,A,Ofst	66	2	7 (9)		Bit Test And Jump If One.
	Rn,A,Ofst	16	3	10 (12)		If (s) .AND. (d) ≠ 0,
	Rn,B,Ofst	36	3	10 (12)		then (PC) + (offset) → (PC)
	Rn,Rd,Ofst	46	4	12 (14)		If the AND of the source and
	%iop,A,Ofst	26	3	9 (11)		destination operands ≠ 0,
	%iop,B,Ofst	56	3	9 (11)		the PC will be modified to
	%iop,Rn,Ofst	76	4	11 (13)		include the offset.
*See note 1					0 R R x	
BTJOP	A,Pn,Ofst	86	3	11 (13)		Bit Test And Jump One-Peripheral
	B,Pn,Ofst	96	3	10 (12)		If (s) .AND. (Pn) ≠ 0, then (PC) + (offset) → (PC)
	%>iop,Pn,Ofst	A6	4	12 (14)		If the AND of the source and destination operands ≠ 0, the PC will be modified to include the offset.
*See note 1					0 R R x	
BTJZ	B,A,Ofst	67	2	7 (9)		Bit Test And Jump If Zero Instruction.
	Rn,A,Ofst	17	3	10 (12)		If (s) .AND. NOT (d) ≠ 0, then (PC) + (offset) → (PC)
	Rn,B,Ofst	37	3	10 (12)		If the AND of the source and NOT (destination) operands ≠ 0,
	Rn,Rf,Ofst	47	4	12 (14)		the PC will be modified to include the offset.
	%>iop,A,Ofst	27	3	9 (11)		
	%>iop,B,Ofst	57	3	9 (11)		
	%>iop,Rn,Ofst	77	4	11 (13)		
*See note 1					0 R R x	
BTJZP	A,Pn,Ofst	87	3	11 (13)		Bit Test And Jump If Zero-Peripheral Instruction.
	B,Pn,Ofst	97	3	10 (12)		If (s) .AND. NOT (Pn) ≠ 0, then (PC) + (offset) → (PC)
	%>iop,Pn,Ofst	A7	4	12 (14)		If the AND of the source and NOT (destination) operands ≠ 0, the PC will be modified to include the offset.
BR	@Label	8C	3	10	x x x x	Branch Instruction.
	@Label(B)	AC	3	12		(d) → (PC)
	*Rn	9C	2	9		The PC will be replaced with the contents of the destination operand.
CALL	@Label	8E	3	14	x x x x	Call instruction
	@Label(B)	AE	3	16		(SP) + 1 → (SP)
	*Rn	9E	2	13		(PC MSB) → (Stack)
						(SP) + 1 → (SP)
						(PC LSB) → (Stack)
						Operand Address → (PC)
CLR	A	B5	1	5	0 0 1 x	Clear Instruction.
	B	C5	1	5		0 → (d)
	Rd	D5	2	7		Clear the destination operand.
CLRC		B0	1	6	0 R R x	Clear Carry Instruction.
						0 → (C)
						Clears the carry bit.
CMP	B,A	6D	1	5	R R R x	Compare Instruction.
	Rn,A	1D	2	8		(d) – (s) computed
	Rn,B	3D	2	8		Set flags on the result of the source
	Rn,Rn	4D	3	10		operand subtracted from the destination
	%iop,A	2D	2	7		operand.
	%iop,B	5D	2	7		
	%iop,Rn	7D	3	9		

(Continued overleaf)

Table 10. Continued.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
CMPA	@Label	8D	3	12	R R R x	Compare With An Extended Instruction.
	@Label(B)	AD	3	14		(A) – (s) computed.
	*Rn	9D	2	11		Set flags on result of the source operand subtracted from A.
DAC	B,A	6E	1	7	R R R x	Decimal Add With Carry Instruction.
	Rs,A	1E	2	10		(s) + (d) + (C) → (d) (BCD)
	Rs,B	3E	2	10		The source, destination, and the carry bit
	Rs,Rd	4E	3	12		are added, and the BCD sum is stored at the
	%>iop,A	2E	2	9		destination address.
	%>iop,B	5E	2	9		
	%>iop,Rd	7E	3	11		
DEC	A	B2	1	5	R R R x	Decrement Instruction.
	B	C2	1	5		(d) – 1 → (d)
	Rn	D2	2	7		Decrement destination operand by 1.
DECD	A	BB	1	9	R R R x	Decrement Double Instruction.
	B	CB	1	9		(rp) – 1 → (rp)
	Rp	DB	2	11		Decrement register pair by 1.
						C = 0 on 0 – FFFF transition.
DINT		06	1	5	0 0 0 0	Disable Interrupts Instruction.
						0 → (global interrupt enable bit)
						Clear the I bit.
*See note 1					x x x x	
DJNZ	A,Ofst	BA	2	7(9)		Decrement Register And Jump If Non-Zero Instruction.
	B,Ofst	CA	2	7(9)		(d) – 1 → (d)
	Rd,Ofst	DA	3	9(11)		If (d) ≠ 0, (PC) + (offset) → (PC)
DSB	B,A	6F	1	7	R R R x	Decimal Subtract With Borrow Instruction.
	Rs,A	1F	2	10		(d) – (s) – 1 + (C) → (d) (BCD)
	Rs,B	3F	2	10		The source operand is subtracted from the destination;
	Rs,Rd	4F	3	12		this sum is then reduced by 1 and the carry bit is then
	%>iop,A	2F	2	9		added to it. The result is stored as a BCD number.
	%>iop,B	5F	2	9		
	%>iop,Rd	7F	3	11		
EINT		05	1	5	1 1 1 1	Enable Interrupts Instruction.
						1 → (global interrupt enable bit)
						Set the I bit.
IDLE		01	1	6	x x x x	Idle Until Interrupt Instruction.
						(PC) → (PC) until interrupt
						(PC) + 1 → (PC) after return from interrupt
						Stops μC execution until an interrupt.
INC	A	B3	1	5	R R R x	Increment Instruction.
	B	C3	1	5		(d) + 1 → (d)
	Rd	D3	2	7		Increase the destination operand by 1.
INV	A	B4	1	5	0 R R x	Invert Instruction.
	B	C4	1	5		NOT (d) → (d)
	Rd	D4	2	7		1's complement the destination operand.
JMP	Ofst	E0	2	7	x x x x	Jump Unconditional Instruction.
						(PC) + (offset) → (PC)
						The PC is modified by an offset to create a new PC value.
*See note 1					x x x x	Jump On Condition Instruction.
JC	Ofst	E3	2	5 (7)		Jump If Carry.
JEQ	Ofst	E2	2	5 (7)		Jump If Equal.
JHS	Ofst	E3	2	5 (7)		Jump If Higher or Same. — If conditions are met,
JL	Ofst	E7	2	5 (7)		Jump If Lower. — then (PC) + offset → (PC)
JNC	Ofst	E7	2	5 (7)		Jump If No Carry. — If the needed conditions
JNE	Ofst	E6	2	5 (7)		Jump If Not Equal. — are met, the PC is modified
JNZ	Ofst	E6	2	5 (7)		Jump If Non-zero. — by the offset to form a
JP	Ofst	E4	2	5 (7)		Jump If Positive. — new PC value.
JPZ	Ofst	E5	2	5 (7)		Jump If Positive or Zero
JZ	Ofst	E2	2	5 (7)		Jump If Zero.
LDA	@Label	8A	3	11	0 R R x	Load A Register Instruction.
	@Label(B)	AA	3	13		(s) → (A)
	*Rn	9A	2	10		Move the source operand to A.

Table 10. Continued.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
LDSP		0D	1	5	x x x x	Load Stack Pointer Instruction. (B) → (SP) Load SP with Register B's contents.
MOV	A,B	C0	1	6	0 R R x	MOVE Instruction. (s) → (d) Replace the destination operand with the source operand.
	A,Rd	D0	2	8		
	B,A	62	1	5		
	B,Rd	D1	2	7		
	Rs,A	12	2	8		
	Rs,B	32	2	8		
	Rs,Rd	42	3	10		
	%>iop,A	22	2	7		
	%>iop,B	52	2	7		
	%>iop,Rd	72	3	9		
MOVD	%>iop,Rp	88	4	15	0 R R x	Move Double Instruction. (rp) → (rp) Copy the source register pair to the destination register pair.
	%>iop(B),Rp	A8	4	17		
	Rp,Rp	98	3	14		
MOVP	A,Pd	82	2	10	0 R R x	Move To/From Peripheral File. (s) → (d) Copy the source operand into the destination operand.
	B,Pd	92	2	9		
	%>iop,Pd	A2	3	11		
	Ps,A	80	2	9		
	Ps,B	91	2	8		
MPY	B,A	6C	1	44	0 R R x	Multiply Instruction. (s) ★ (d) → (A,B) Multiply the source and destination operations, store the result in Registers A (MSB) and B (LSB).
	Rs,A	1C	2	47		
	Rs,B	3C	2	47		
	Rn,Rn	4C	3	49		
	%>iop,A	2C	2	46		
	%>iop,B	5C	2	46		
	%>iop,Rn	7C	3	48		
NOP		00	1	5	x x x x	No Operation Instruction. (PC) + 1 → (PC) Add 1 to the PC.
OR	B,A	64	1	5	0 R R x	OR Instruction. (s) .OR. (d) → (d) Logically OR the source and destination operands, and store the results at the destination address.
	Rs,A	14	2	8		
	Rs,B	34	2	8		
	Rs,Rd	44	3	10		
	%>iop,A	24	2	7		
	%>iop,B	54	2	7		
	%>iop,Rd	74	3	9		
ORP	A,Pd	84	2	10	0 R R x	OR Peripheral File Register Instruction. (s) .OR. (d) → (d) Logically OR the source and destination operands, and store the results at the destination address.
	B,Pd	94	2	9		
	%>iop,Pd	A4	3	11		
POP	A	B9	1	6	0 R R x	POP From Stack Instruction. (Stack Top) → (d) (SP) − 1 → (SP) Copy the last byte on the stack into the destination address.
	B	C9	1	6		
	Rd	D9	2	8		
POP	ST	08	1	6	0 R R x	POP From Stack To Status Register. (Stack Top) → (Status Register) (SP) − 1 → (SP) Replace the Status Register with the last byte of the stack.
PUSH	A	B8	1	6	0 R R x	Push On Stack Instruction. (s) → (Stack) (SP) + 1 → (SP) Copy the Status Register onto the stack.
	B	C8	1	6		
	Rs	D8	2	8		
PUSH	ST	0E	1	6	0 R R x	Push Status Register On Stack. (Status Register) → (Stack) (SP) + 1 → (SP) Copy the Status Register onto the stack.
RETI (Continued overleaf)		0B	1	9	Loaded from the stack.	Return From Interrupt Instruction. (Stack) → (PC) LSB (SP) − 1 → (SP) (Stack) → (PC) MSB

Table 10. Continued.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
						(SP) – 1 → (SP) (Stack) → (Status Register) (SP) – 1 → (SP)
RETS		0A	1	7	x x x x	Return From Subroutine Instruction. (Stack) → (PC) LSB (SP) – 1 → (SP) (Stack) → (PC) MSB (SP) – 1 → (SP)
RL	A B Rd	BE CD DE	1 1 2	5 5 7	b7 R R x	Rotate Left Instruction. Bit (n) → Bit (n+1) Bit (7) → Bit (0) and Carry
RLC	A B Rd	BF CF DF	1 1 2	5 5 7	b7 R R x	Rotate Left Through Carry Instruction. Bit (n) → Bit (n+1) Carry → Bit (0) Bit (7) → Carry
RR	A B Rd	BD CD DC	1 1 2	5 5 7	b0 R R x	Rotate Right Instruction. Bit (n+1) → Bit (n) Bit (0) → Bit (7) and Carry
RRC	A B Rd	BD CD DD	1 1 2	5 5 7	b0 R R x	Rotate Right Through Carry. Bit (n+1) → Bit (n) Carry → Bit (7) Bit (0) → Carry
SBB	B,A Rs,A Rs,B Rs,Rd %>iop,A %>iop,B %>iop,Rd	6B 1B 3B 4B 2B 5B 7B	1 2 2 3 2 2 3	5 8 8 10 7 7 9	R R R x	Subtract With Borrow Instruction. (d) – (s) – 1 + (C) → (d) Destination minus source minus 1 plus carry; stored at the destination address.
SETC		07	1	5	1 0 1 x	Set Carry Instruction. 1 → (C) Set the carry bit.
STA	@Label @Label(B) *Rd	8B AB 9B	3 3 2	11 13 10	0 R R x	Store A Register Instruction. (A) → (d) Store A at the destination.
STSP		09	1	6	x x x x	Store Stack Pointer Instruction. (SP) → (B) Copy the SP into Register B.
SUB	B,A Rs,A Rs,B Rs,Rd %>iop,A %>iop,B %>iop,Rd	6A 1A 3A 4A 2A 5A 7A	1 2 2 3 2 2 3	5 8 8 10 8 8 9	R R R x	Subtract Instruction. (d) – (s) → (d) Store the destination operand minus the source operand into the destination.
SWAP	A B Rn	B7 C7 D7	1 1 2	8 8 10	R R R x	Swap Nibbles Instruction. d(Hn,Ln) → d(Ln,Hn) Swap the operand's hi and lo nibbles.
TRAP	0-23	E8-FF	1	14	x x x x	Trap To Subrouti e Instruction. (SP) + 1 → (SP) (PC SB) → (Stack) (SP) 1 → (SP) (PC LSB) → (Stack) (Entry Vector) → (PC)
TSTA		B0	1	6	0 R R x	Test A Register Instruction. 0 → (C) Set carry bit; set sign and zero flags on the value of Register A.
TSTB		C1	1	6	0 R R x	Test B Register Instruction. 0 → (C) Set carry bit; set sign and zero flags on the value of Register A.

(Continued overleaf)

Table 10. Continued.

Mnemonic		Opcode	Bytes	Cycles	Status C N Z I	Operation Description
XCHB	A Rn	B6 D6	1 2	6 8	0 R R x	Exchange With B Register Instruction. (B) ←→ (d) Swap the contents of Register B with (d).
XOR	B,A Rs,A Rs,B Rs,Rd %>iop,A %>iop,B %>iop,Rd	65 15 35 45 25 55 75	1 2 2 3 2 2 3	5 8 8 10 7 7 9	0 R R x	Exclusive Or Instruction. (s) .XOR. (d) → (d) Logically exclusive OR the source and destination operands, store at the destination address.
XORP	A,Pd B,Pd %>iop,Pd	85 95 A5	2 2 3	10 9 11	0 R R x	Exclusive Or Peripheral File Register Instruction. (s) .XOR. (Pn) → (Pn) Logically exclusive OR the source and destination operands, store at the destination address.

* Note 1 : Add two to cycle count if branch is taken.

Status Legend :
0 Status Bit always set to 0.
1 Status Bit always set to 1.
R Status Bit set to a 1 or a 0 depending on results of operation.
x Status Bit not affected.
b Bit () affected.

Table 10. Complete.

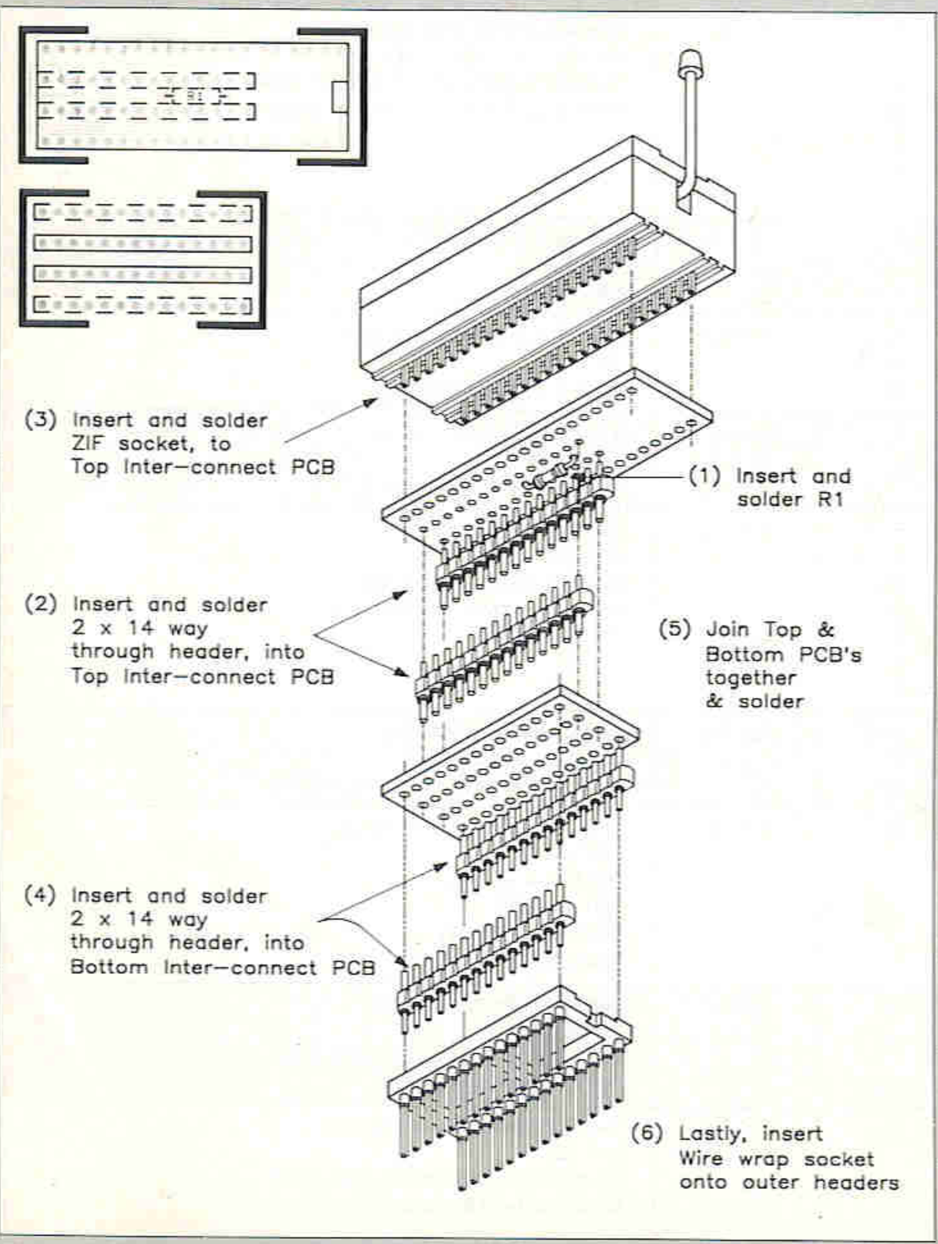


Figure 10. Construction of the adaptor.

Adaptor Socket Construction

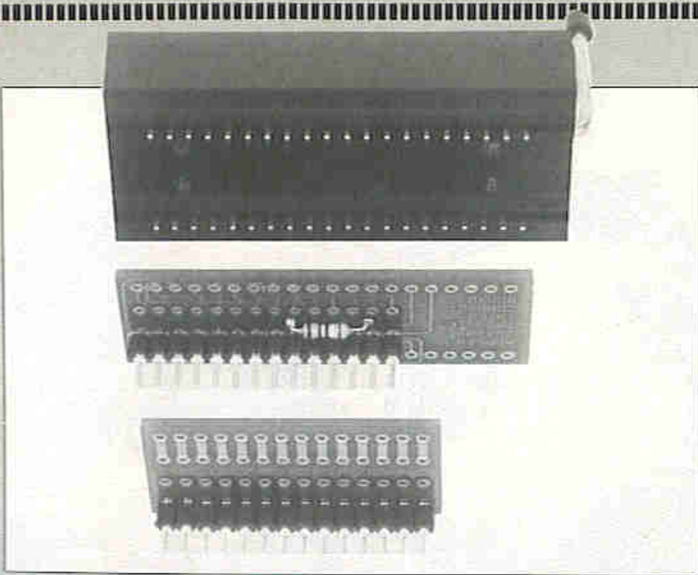
A kit is available (stock code LP36P) allowing the TMS77C82 microcontroller to be programmed by most PROM programmers. The 40 to 28-pin adaptor consists of two PCB's (one upper GE47B, one lower GE48C), one 3k9Ω resistor, one 28-pin wire wrap DIL socket, and two strips of SIL header pins.

The order of construction is *very* critical as components overlap and insertion in the wrong order will result in inaccessible solder joints. Before construction commences, break the two 32-way SIL headers down into four 14-way strips. The small 4-way strips you are left with can be discarded. Identify the top side of both the upper and lower PCB's (printed legend facing upwards). Construction can be broken down into six steps shown in Figure 10.

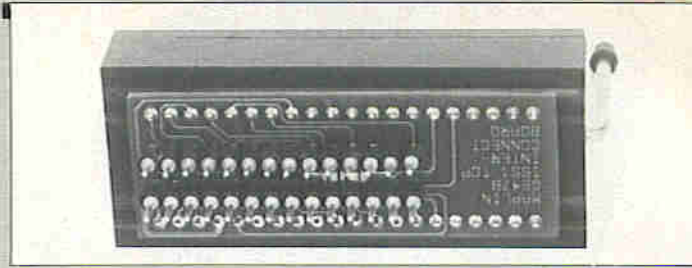
1. Solder R1 into the BOTTOM side of the upper PCB.
2. Solder two of the four 14-way SIL headers into the BOTTOM of the upper PCB (same side as R1).
3. Solder the ZIF socket into the TOP of the upper board.
4. Solder the remaining two 14-way SIL headers into the BOTTOM of the lower PCB.
5. Connect the upper and lower PCB's together by offering up the lower PCB to the SIL headers on the upper PCB. Take care in soldering the headers into the bottom PCB as the outer headers can be easily damaged.
6. Plug the 28-pin wire-wrap socket into the bottom SIL headers.

EPROM Integrity Protection using the R bit

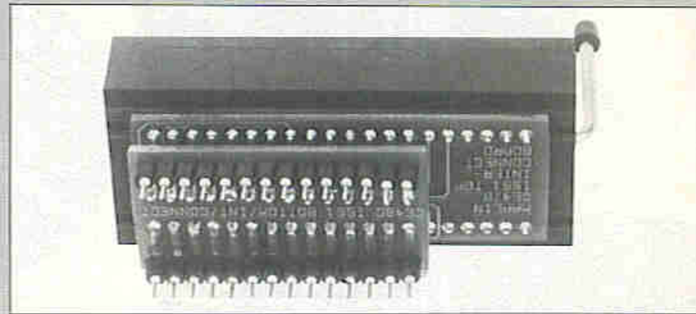
Once the TMS77C82 has been programmed with the desired code, the contents of the EPROM may be protected with the use of the R bit integrity feature.



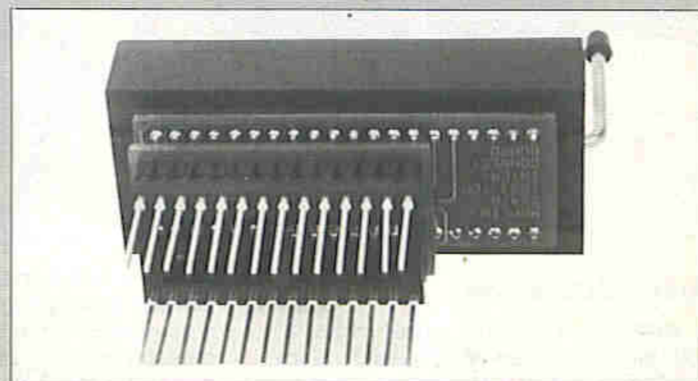
The relevant parts are ready to assemble.



Top pcb fitted to ZIF socket.



Bottom pcb fitted to top pcb.



IC socket fitted to bottom pcb.

TMS77C82		R Bit	
Function	**Pin #**	**Program**	**Verify**
XTAL/CLKIN	17	V_{ss}	V_{ss}
RESET	14	V_{ss}	V_{ss}
MC	36	12.5V	V_{cc}
INT3	12	12.5V	12.5V
D4	22	V_{cc}	V_{ss}
A7/EC1	11	V_{cc}	V_{ss}
A6/EC2	15	V_{cc}	V_{cc}
A3	9	V_{cc}	X
A1	7	X	V_{ss}
C7	35	$V_{IH}/V_{IL}/V_{IH}$	Refer to Step 3 in R bit verify procedure

Note: X=Don't care.

Table 11. Modifying the 'R' bit.

The function of the R bit is to disable all external access to the on-chip EPROM while in the EPROM mode, which will prevent a protected code from being modified or read externally. The only way to 'unprotect' the TMS77C82 after the R bit has been programmed is by erasing the EPROM, thereby destroying the protected code. Table 11 shows the required connections needed to program the R bit.

R bit Programming Procedure:

1. Configure all referenced pins for the R bit program mode.
2. Power up the device.
3. Toggle C7 from a logical high (1), to a logical low (0), and back to a logical high (1).
4. Power down the device.

R bit Verify Procedure:

1. Configure all referenced pins for the R bit verify mode.
2. Power up the device.
3. Read C7. Zero (0) is programmed, one (1) is not programmed.
4. Power down the device.

Erasure

The TMS77C82 (EPROM version *only*) is erased by exposing the chip to shortwave ultraviolet light that has a wavelength of 253.7 nanometers. It should be noted that normal ambient light contains the correct wavelength for erasure. Therefore, when using the 77C82 the window should be covered with an opaque label.

The TMS77C82 is available in two versions:

TMP77C82JD, supplied in a ceramic package with a 7.5mm window (stock code UL66W, price £39.95) or TMS77C82NL (One Time Programmable), supplied in a plastic package with no window (stock code UL62S, price £13.95).

For those wishing to investigate deeper into the processing power of this device a TMS7000 Assembly Language Programmer's Guide (stock code WS91Y, Price £3.95) is available. A fully comprehensive data sheet giving full electrical specifications, memory allocation, and programming procedure is also available through our normal data sheet service.

TMS77C82 Adaptor Parts List

R1	Min Res 3k9	1	(M3K9)
	ZIF Socket 40-way	1	(FT15R)
	32 Way SIL Through Header	2	(JR74R)
	28 Way Turned Pin Wire Wrap IC Socket	1	(JR75S)
	TMS77C82 Adaptor Top PCB	1	(GE47B)
	TMS77C82 Adaptor Bottom PCB	1	(GE48C)
OPTIONAL			
	TMS77C82NL	1	(UL62S)
	or TMP77C82JD	1	(UL66W)
	TMS7000 Assm Lang Book	1	(WS91Y)

A complete kit of parts, excluding Optional, is available: **Order As LP36P (TMS77C82 Adaptor Kit) Price £19.95**

The following parts are also available separately but are not shown in our 1990 catalogue:

Top PCB **Order As GE47B Price £2.95**

Bottom PCB **Order As GE48C Price £2.95**

32W SIL Header **Order As JR74R Price £1.98**

28W Wire Wrap Skt **Order As JR75S Price £2.25**

TMS77C82NL **Order As UL62S Price £13.95**

TMP77C82JD **Order As UL66W Price £39.95**

TMS7000 Assembly Language Book **Order As WS91Y Price £3.95** *NV*

TAPED

by
Mike
Holmes

Part 1

Introduction

In 1943 the U.S. Army signal corps, at that time stationed in England, were puzzled by sometimes hearing German radio broadcasts of operas and music during the middle of the night, and noticed that these were free of record noise (clicks and scratches) that usually accompany recordings, but at the same time were highly unlikely to be 'live' broadcasts either, given the time of day. Even more peculiar was the voice of Hitler who could be heard giving radio speeches from entirely different parts of Germany, within the same hour.

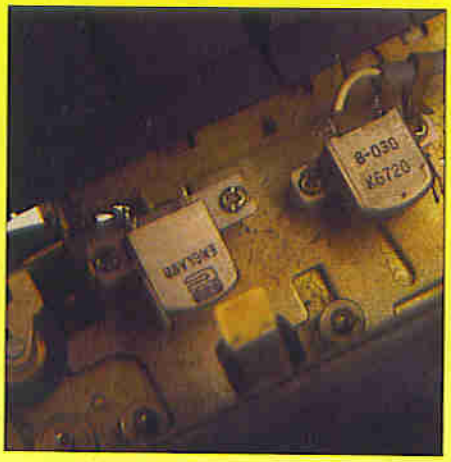
The answer was found in 1945 in Frankfurt, upon the Allied occupation. Now magnetic recording machines have been around for a time; the 'Telegraphone', which was patented in 1905, was a typical example of the device of this time which recorded onto steel wire. In 1928 the Germans patented an alternative medium comprising iron particles bonded to a strip of paper, and the machine which used this was called the 'Magnetophone'. But what made the machines in Frankfurt different was that they had plastic backed tape and used AC bias.

'Liberated' by the allies, the tape recorder, as it is now known, has never looked back since. To begin with it was immediately sought after by anybody and everybody in the business of recording sound with any resemblance to decent quality. Up until this time there had been only two practical methods for doing this – either by cutting a 'master' gramophone record, or the photo-electric method used for film sound tracks. The quality of the latter left much to be desired, and so the master recording was always a record, made by a machine built to a precision of engineering equal to that of a machine-shop lathe, and which was probably more expensive and nearly as massive. Although, believe it or not, portable battery powered versions have existed.

It is difficult for us to imagine now the kind of difficulties involved in making a sound recording in those days, not just in the music studio, but also on the film set – the camera could not record the sound simultaneously. A large cast, comprising a full orchestra and a chorus of singers and dancers could be a producer's nightmare – whenever there was a mistake or similar problem the half finished master record had to be discarded, a new 'blank' put in its place and the whole thing restarted from the beginning. What's more, very little if any editing of the final recording could be done to correct tone or remove extraneous noise, so it had to be perfect at the 'first cut'.

The tape recorder changed all that. For the first time here was a medium which could be used over and over if necessary, where recording could be stopped and restarted anywhere, where mistakes could be re-recorded over the top for just that particular piece, and bits could be cut out and thrown away, or moved to another part of the tape. Also for the first time – if the final product was a record as it invariably was – then the recording could be corrected for tone and amplitude electronically before being sent to the master cutting machine. This principle is still used to this day, where a final cut is made from the mixed combination of several edited and corrected tape recordings.

In this modern day and age, the tape recorder, in its cassette form, is such a common sight everywhere that it is easy to take for granted the amount of development that went into it to make it perform as well as it does. There still are a few idiosyncrasies in the most modern recorder of today which provide an



Cassette head in close-up – erase head at left is upstream of record/play head near capstan pinch roller.

insight into the sort of problems those early German innovators were trying to grapple with while developing their revolutionary machine.

The First Major Stumbling Block

The primary problem with magnetic tape is that it is a magnetic medium. What is required is to produce from an alternating electric current a pattern of alternating magnetic fields along a strip of tape, that is, areas of ferrous material alternately magnetised in opposite polarity and with varying strengths. The trouble is that the iron particles don't want to co-operate just like that.

When a ferrous material comes into the presence of an external magnetic field, if the field is fairly weak the material is not very interested in becoming polarised with it. More magnetic energy is required to get the material magnetised. If the field strength is enough to saturate the material, and then the field turned off, the material would retain a certain amount of magnetism. A field of opposite polarity is required to reduce this magnetism to zero.

However, with magnetic tape it is usually required to magnetise the tape particles in one or other direction only, and not reduce or reverse this magnetism. A hysteresis curve, which illustrates the manner in which the field strength of

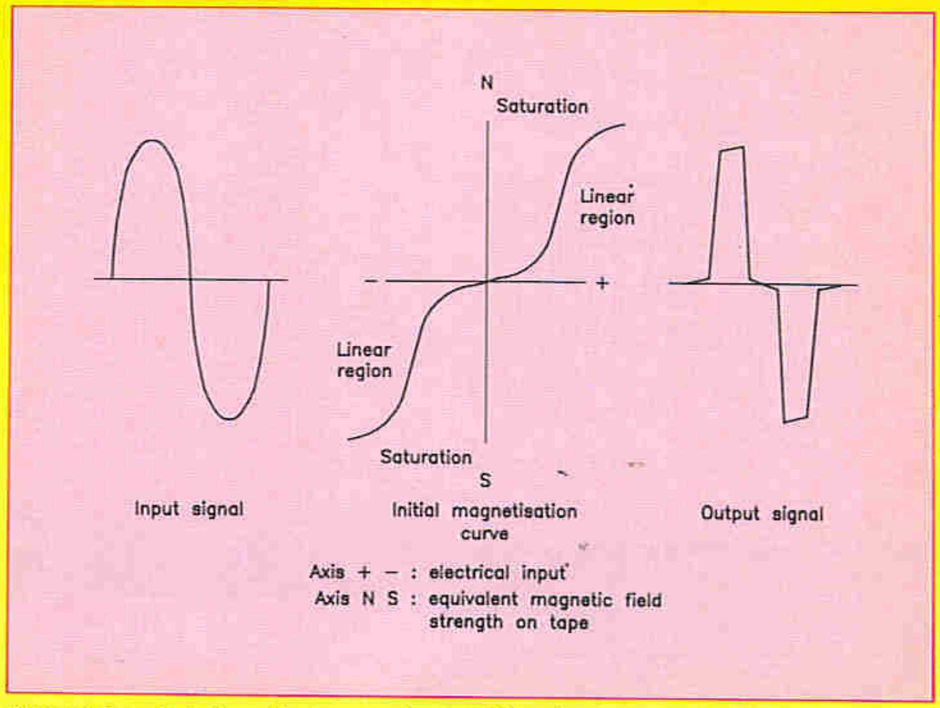


Figure 1. Result of attempting to record onto unbiased tape.

a ferrous material behaves under the influence of altering applied fields, is not really applicable in this case. Instead what is of more interest to us is the 'initial magnetisation curve', i.e., how the medium behaves upon being first influenced by an applied field when it had no magnetisation itself to begin with.

Figure 1 shows a typical output from magnetic tape recorded with one complete cycle of an alternating magnetic field. The original source waveform was sinusoidal, the recorded copy is a mess. The non-linear nature of the initial magnetisation curve prevents the medium correctly registering the relative lower parts of the waveform at its weakest levels close to zero, resulting in gross distortion. At the other extreme, it so happened that in this example the recording field was much too strong at the peaks of the waveform, saturating the tape. However it will be noticed that between the two extremes there is a linear region in the magnetisation curve, and it operates for both halves of the cycle.

Biasing

So if we could get the complete waveform to fit in just one of these linear regions we solve the problem at a stroke. This can be done with DC biasing, as it's called, where the recording electromagnet, or record head, has a constant current flowing through its windings in one direction. The resultant field must be sufficient to push the tape's response up out of the non-linear part of the initial magnetisation curve into the middle of the linear region. Now all we need do is add the AC waveform to the existing direct current and we get the result shown in Figure 2 – Perfect.

Well no, not quite. The principle works very well, but has two flaws. One is that the iron particles are constantly energised, whether a signal is present or not, and are all polarised in one direction. During playback, they all contrive to impress on the playback head their own magnetic energy, and as each particle passes the magnetic pole gap of the head it produces a pulse in the playback winding. The result can be called 'shot

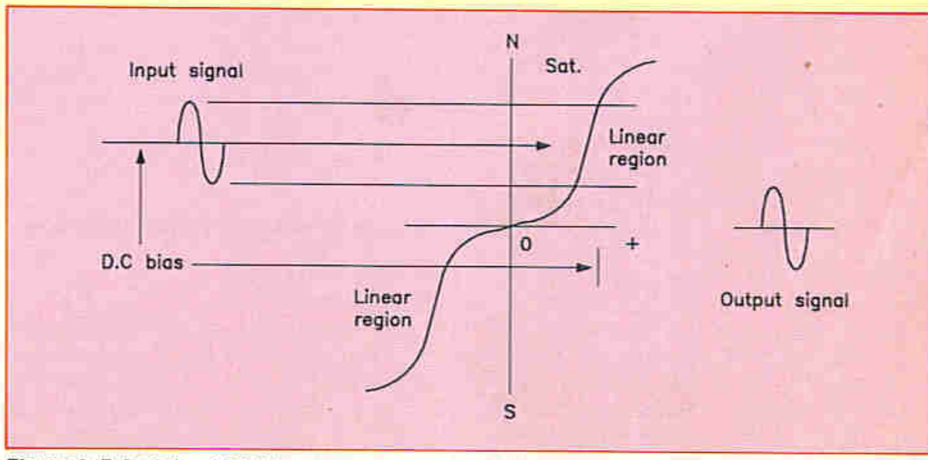


Figure 2. Principle of DC bias.

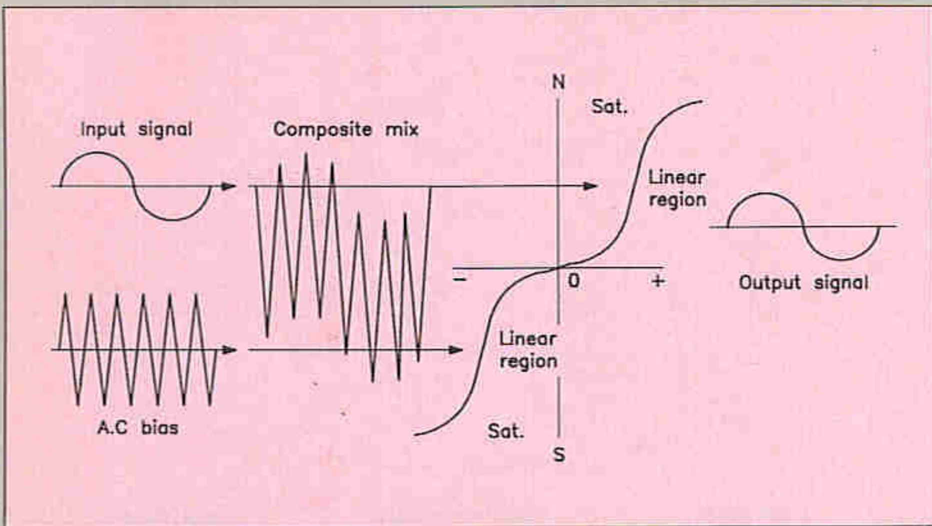


Figure 3. AC biasing with record signal superimposed.

noise', white noise or just plain hiss, and you've not heard tape hiss like that produced by a DC biased recorder.

The other problem is more serious. If the recording head has a DC current flowing through it the whole time it is recording, then the pole pieces of the head themselves will become permanently magnetised. After the current is removed, the head remains a permanent magnet and attempts to wipe the tape 'clean' during playback, as any other magnet would. Not all at once certainly, but over the course of several playings the recording may become quite weakened.

The Only Solution

There is only one thing for it, we must use an alternating bias current. An alternating current prevents the record head becoming permanently magnetised in one direction or the other, whilst simultaneously 'scrambling' the iron particles, so that they are not all polarised in the same direction and so cannot produce tape hiss – well actually they do, but because of their differing polarities they tend to cancel each other out, thus greatly reducing noise. The final requirement is that this AC bias signal does not also record itself onto the tape.

This last is taken care of by making the frequency of the waveform way above that which the combination of record head magnetic gap width and tape speed are capable of recording. This will become clearer later while dealing with frequency response.

All this immediately adds complexity to our tape recorder. We now need a special sine wave oscillator operating at ultrasonic frequency to apply AC bias in just the right controlled quantity to our record head. The AC bias signal lifts the tape particles out of the bottom of their magnetisation curve into *both* the linear regions of Figure 1, possible because it 'switches' the magnetic strength and polarity of each particle between 'north' and 'south' at a point which is never allowed to fall below the equivalent DC bias level, or in other words halfway along both linear regions.

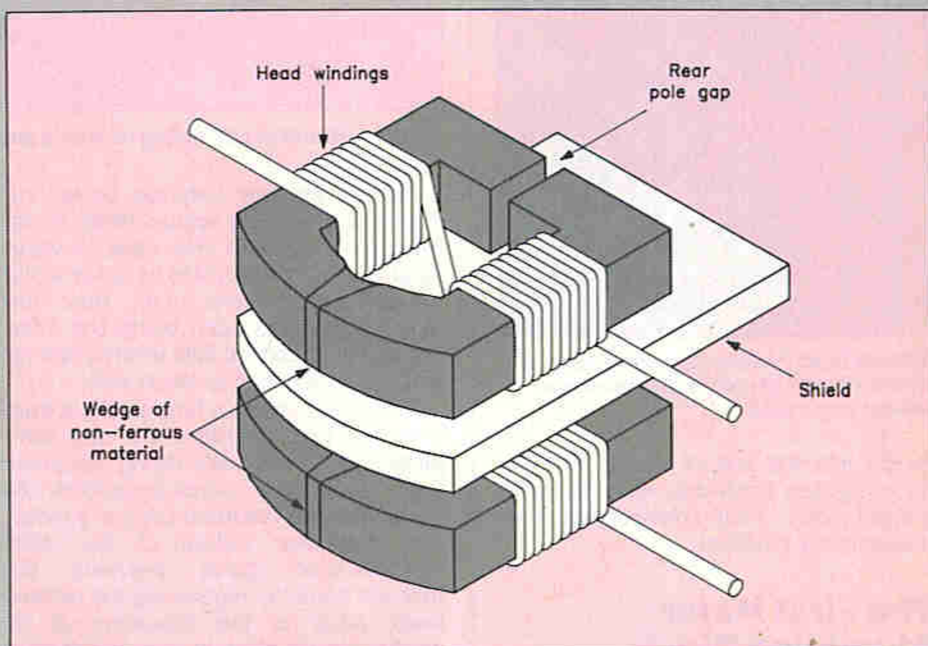


Figure 4a. Construction of a magnetic tape head (stereo).

The actual signal required to be recorded is simply mixed with or superimposed on top of the bias signal, and these waveforms, and the result, are shown in Figure 3.

We'll agree that making this AC bias principle work properly is a bit complicated and somewhat critical in practice, but the results are certainly worth the effort. There can't be much else wrong with it now, surely?

Sorry, but there is. Several things, actually.

Problems arising from AC Biasing:

1. Distortion

The 3rd harmonic distortion content of a recorded signal on tape is inversely proportional to the AC bias level. The higher the bias, the less the distortion. So to keep distortion to a minimum, the bias level must be high.

2. Sensitivity

The sensitivity of the tape is directly proportional to the AC bias level. Literally the strength of the recorded signal on playback is directly dependent on the bias level at the recording stage. Generally then, in order to record a nice, strong signal onto the tape, which itself will both improve the dynamic range and signal to noise ratio, the bias level must be high.

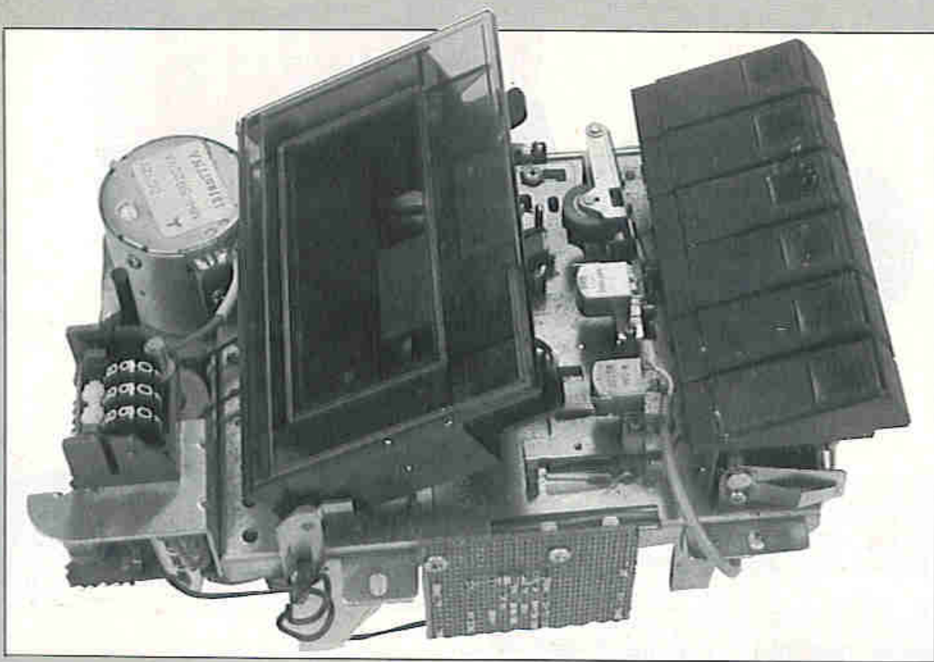
3. LF Response

Usually the magnetic tape recording principle is not very efficient at low or bass frequencies, but fortunately the low frequency response is proportional to the bias level. To achieve a good low frequency response, which is in keeping with say that of the mid-range, and to prevent the bass frequencies becoming weak and ragged, the bias level should be high.

4. HF Response

This is the crunch. High or treble frequency response is inversely proportional to the AC bias level. Nothing impairs the top end performance of a tape recorder more surely than having the AC bias level just a gnat's whisker too high. To ensure that the recorder can record the highest frequency signal that it is mechanically capable of so doing, the bias level should be low.

So in practice the AC bias level is set up as follows. To begin with it is set to a high value, and then test recordings are made. The test consists of recording a high frequency sine wave signal, making sure that it is the highest frequency which the recorder is known to be able to put onto the tape at maximum strength. The signal is recorded at maximum level, i.e. just below saturation. Successive tests are made, and each time the bias level is reduced in small steps until the recorder produces the test signal at a maximum amplitude on playback. It should then be



Top loading cassette transport.

noted that any further reductions in the bias level fail to achieve any further increase in playback amplitude, and in fact if the bias is taken too low then the signal amplitude will begin to reduce again as the tape becomes less sensitive, for reasons explained in paragraph 2 above. So the bias setting is then left at this point, because this level satisfies the fourth requirement above, but without compromising the first, second and third requirements.

Tape Erasing Method

While we are on the subject of AC bias it might be a good opportunity to mention the erase system at this point. Again it's probably taken for granted that it is the simplest matter to record something else over a previously recorded tape, and that the previous recording is entirely removed and replaced with the new. However the means for recording as just explained cannot do this alone, if it did then some of the original recording will still be left and can be heard plainly in the background. The tape must be 'wiped clean' first, and in all modern recorders this is done automatically by an erasing head 'upstream' of the recording head, while in record mode. Again the design of the erasing system should not be taken lightly. Designers of cheap recorders go to all the trouble of including a decent enough AC biasing system, and spoil it all through the necessity to cut costs by using a DC 'driven' erase head. The principle is exactly the same as previously described for DC bias, except that the problems are much worse; the erase head is much more likely to remain quite strongly magnetised with the consequent risk to all recordings on tape every time they are played. The reason for this is that the erasing system has to erase tapes, i.e. thoroughly 'soak' the tape with a magnetic field many times stronger than any produced by a record head. This ensures that anything previously recorded on it is utterly obliterated, and it requires a comparatively higher electric current.

The proper method is to use AC erasing, because again this prevents the erase head being permanently magnetised in one direction, and secondly because this 'scrambles' the iron particles thus ensuring that noise is kept to a minimum (during quiet or 'silent' passages). As with AC bias, the frequency is ultrasonic and the driving signal is a reasonably pure sine waveform, so that it doesn't leave any traces of itself on the tape. Almost universally the same oscillator that produces the AC bias signal also generates the erase. The added complexity here is that now it is a power oscillator and may be required to drive the erase head at several hundred milliwatts; the thing becomes dedicated to this function and produces the AC bias signal almost as a by-product.

Erase Head Degaussing

But what if the record switch were turned off just as the erase output was at the peak of a half cycle? There is a slim possibility that it will leave the erase head magnetised in one direction – the peak power may exceed half a watt just before the oscillator was 'killed'. Superior oscillator designs don't cease immediately, but 'die down' to zero, neatly demagnetising, or 'degaussing', the erase head. In a few rare instances where a recorder has a 'pause' function, the 'pause' period may be 'timed out'. What's happening here is that the tape has been stopped but both record and erase heads are still operating, and the small portion of tape against the erase head is getting rather warm. There may be a risk of heat distortion to the plastic backing, so if the 'pause' button is not released after say five minutes everything is turned off. It also goes without saying that given that the biasing level is critical, then the output stability of this oscillator must be above reproach, and so perhaps some form of automatic gain control may be called for against the possibility of supply voltage and temperature fluctuations. A few years ago many hi-fi tape decks had a front panel bias control and a level monitoring meter, so that the right bias level could be assured before any recording was made, and allowed an optimum setting to be chosen for each different batch of tape. Understandably this facility was a bit complicated for mere mortals to use and so you don't see it any more.

Where it Matters – The Tape to Head Interface

We shall take it for granted that the actual mechanical parts of a tape recorder, the 'tape transport', must be up to a standard of engineering to move tape from one reel to another at constant speed past the tape heads and keep the tape always in contact with the critical surfaces of the heads as required. Aside from this, the performance of a recorder is utterly dependent on two vital components – the tape, and the heads themselves.

We all know about the differences between various brands of tape and their equally differing quality, and so it goes without saying that any recorder has a good head start if it's using a decent tape. But the quality of the heads also matters and the way in which the tape 'interfaces' with them, and this is purely physical.

Figure 4a shows the internal construction of a tape head. It consists of an electro-magnet ring comprising a pair of cores mated together to form a magnetic circuit. There are two 'pole gaps', one at the back, and the other important one at the front. It is the one at the front which produces the alternating north/south magnetic patterns which 'encode' the tape during recording. The core material is typically laminated electrical steel, albeit on a tiny scale; advanced designs may use a ferrite material. Although the two ends at the back butt together face to face, and only exist to make assembly easier, the front are 'interrupted' with a gap: this could be an air gap, but in practice it is a wedge of non-ferrous material, which might be brass as an example. An air gap would quickly clog with scraped-off tape particles, which would rather defeat the object of the exercise! In order for the magnetic lines of force to complete their circuit through the cores, they must 'jump around' this gap, and very conveniently find that the ferrous particles of the tape make a handy bridge, see Figure 4b. This time honoured technique ensures a deep penetration of the tape.

Before the age of cassette – early 1960's portable reel-to-reel recorder by Teleton.

When the assembly is installed in its case the whole front face, including the case, is polished as one complete surface, and at this stage the width of the front gap can be finally set. The steel pole pieces may actually meet to begin with but removing material at the polishing stage exposes the non-ferrous wedge as a fine strip. Further polishing will make this strip effectively wider.

This 'strip' is the front magnetic pole gap of the tape head, and its width determines the upper frequency limit of the recorder for a given tape speed. It's all to do with wavelengths, and in this case the waves are 'laid down' along the length of the tape. For a tape speed of 9.5cm/sec. there will be 10.52 cycles for a signal of 100Hz on one centimetre length of tape. At 10kHz, there are 1,052.6 of them. At 15kHz the recorder has reached its cut-off frequency, because there would have been 1,578.9 cycles in one centimetre, which means that each cycle should have been 6.2μm long if it could have got onto the tape. But guess what? Divide that by two and that's how wide the pole gap is, 3.1μm. The record head defeats itself by undoing what it's just done because the tape won't move out of the way quickly enough. Double the tape speed, and the cut-off frequency immediately leaps to 30kHz.

The above quoted example is for a typical 'consumer quality' (as opposed to a studio machine) reel to reel recorder. The now universal cassette tape standard starts out with the disadvantage that the speed is only 4.75cm/sec., because this is the only way a reasonable playing time can be got out of what is really a small amount of tape in a cassette. This means that to equal the above example, a cassette tape head has to have a pole gap of an incredibly narrow 1.5μm, an engineering feat not to be sneezed at, and it is only by sheer volume of production that keeps cassette tape heads at a reasonable price.

The construction of an erase head is identical to that of Figure 4a, with just two important differences. Firstly the windings are of a very low impedance so that they will accept a higher current for a given frequency and voltage input, and secondly there are often two front pole gaps side by side, so that the erase head can have 'two shots' as it were at erasing the tape.

Head Adjustment

This is something you may already be familiar with, namely twiddling the little screws which make a tape head 'lean' one way or the other. Actually these adjusters exist only to set the head's pole gap exactly at right angles to the direction of tape travel, once this has been achieved, no further improvements are possible. As we have seen, given the extremely short wavelengths that a recorded signal on tape can be reduced to, an off-vertical tape head is effectively one that has a widened pole gap, see Figure 5, and consequently impaired treble response. (The adjustment is sometimes known as 'azimuth' adjustment because it alters the effective pole gap width.) Not only that but the full track width of tape particles pass the gap at different times, causing all sorts of weird phase errors. Unless of course the same head has also done the recording, as is most often the case, and if so the phase might be alright but the HF response will still not be at its best. These adjustments are extremely sensitive and it is particularly noticeable for example on playback what a world of difference that 1/16 of a turn on the screw makes!

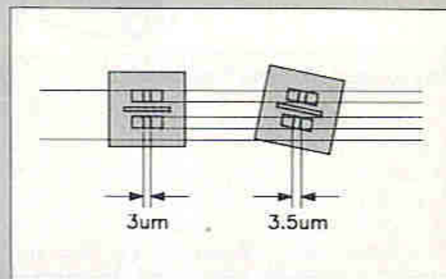


Figure 5. Head alignment error increases effective pole gap width, angle is exaggerated.

Tape Heads are Inductors Too

We have been talking about erase, bias and recording signals as though these were things that will remain constant and which we need no longer worry about once we've got them. These signals will be applied to, and in the case of playback, originate from magnetic tape heads. But as we have seen, a tape head is essentially an electro-magnet, and that makes it very definitely an inductor. And the behaviour of inductors is very frequency dependent. And in this respect, as far as the electronics of a tape recorder is concerned, tape heads make real nuisances of themselves.

Take the case of the erase oscillator. Say that the erase/bias signal frequency is 50kHz, a typical value, and that the inductive reactance of the erase head is 300Ω at this frequency, again typical. Erase current is usually in the order of tens of mA, so for say 50mA erase current, again typical, the erase power oscillator output voltage will have to be 15V rms, and straight away RF injection into every other amplifier circuit block throughout the internals of the recorder becomes an issue.

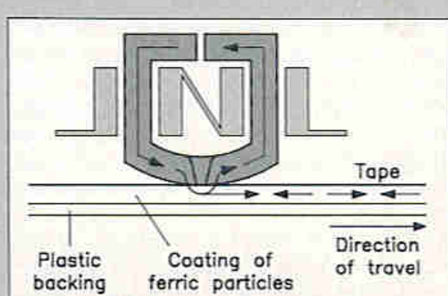


Figure 4b. How magnetic lines of force are made to flow through tape particles by wedge in pole gap.

Similarly bias current, curious as it may seem, needs to be several times that of the maximum signal record current, typically five or six times more. At 50kHz the record head has a reactive impedance of several tens of kΩ, requiring the bias voltage signal to be as an example 15V to 25V rms; the actual level depends on the bias frequency and the inductance of, and value of bias current required by the head. If the impedance of a recording head is 30kΩ at 50kHz, then the bias voltage has to be 18.6V rms as measured across the winding if a bias current of 620μA is required, to use a real example. The record signal current however is 'only' a maximum of 92μA before saturation occurs. This gives an indication of the energy required on the part of the bias current, to lift the tape particles up into the linear regions of their initial magnetisation curve.

Next Time

In the next issue we shall look at how the actual amplifier circuits of a tape recorder cope with these problems.

COMPUTERS IN THE REAL WORLD

Part 8 by Graham Dixey C.Eng., M.I.E.E.

Introduction

Many applications of microcomputers result in an output to drive a motor. Without stopping to think about this fact, it may not seem immediately obvious. After all, wordprocessors and databases seem totally divorced from the world of industrial machine control. Yet motors of one sort or another are frequently essential in both.

To take the use of a wordprocessor, as a familiar example. No less than *four* separate motors are involved in the process! The software to run the program is invariably contained on disk (don't write in if yours is in ROM!), whether it is a floppy or a hard disk. To access the application program itself, as well as the files for the work being done, requires two motors, as follows:

(a) One to spin the disk itself at 360 rev/min.

(b) One to drive the read/write head radially across the disk surface to find the required tracks and sectors where the data is stored.

As a result of the wordprocessor being used to produce a document, whether a short letter or a block-busting novel, a printer will be used to turn the 'silicon copy' into something rather more permanent and substantial. This also requires two motors:

(c) One to rotate the platen, to feed the paper.

(d) One to drive the print head across the paper as the characters are struck onto it. This assumes a dot-matrix printer, which doesn't need a motor to rotate its print head.

In fact applications of this nature may actually make more use of motors than rather more 'physical' tasks. In some of the latter cases no motors are needed at all. A temperature control system, for example, unless it was using an output device such as a printer for a hard copy record of temperature/time variations, would merely be converting between analogue and digital quantities (and vice-versa), purely electronically.

Motors are, traditionally, analogue devices, or rather they were until relatively recently. Because the computer can only input and output digital data, there would seem to be a need for conversion when a conventional motor is to be driven by a computer. As it happens, there are several choices, as follows:

(a) Use a conventional motor and a digital-analogue converter.

(b) Use a conventional motor and drive it 'digitally'.

(c) Use a motor that takes a digital input directly.

Figure 1 shows the first case. A digital quantity, representing motor speed, is delivered to a digital-to-analogue converter (DAC). The analogue output drives current through the field winding and, hence, determines the motor speed. Naturally, the binary output will be generated by a software program. This, in its turn, may have received input data – perhaps from a set of switches at an input port – that demands a certain motor speed.

Figure 2 shows the same motor being used, but instead of the supply being an analogue quantity, it is a square-wave of variable mark/space ratio. The speed depends upon the ratio of on-time/off-time. It is easiest if the off-time is a constant value and the on-time only is varied.

Figure 3 shows two possible ways of using inputs to control motor speed. In the first case, (i), a potentiometer is used as the speed controller; its wiper voltage is converted to a digital value which the program reads at the input port. This is used to set the length of the time waste loop and, hence, mark/space ratio and motor speed.

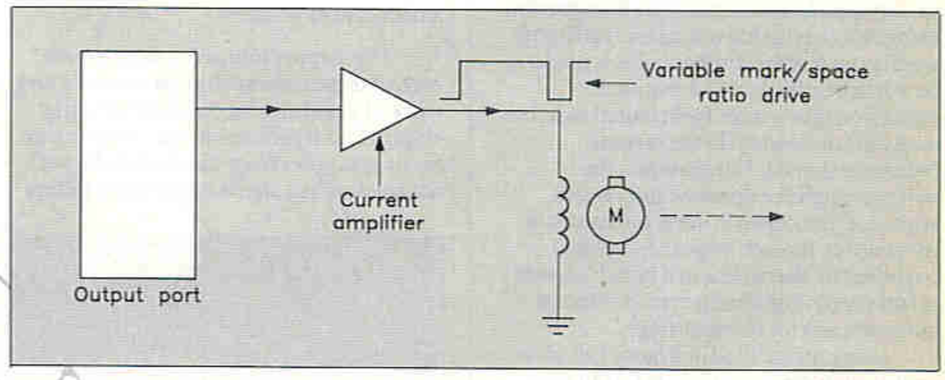


Figure 1. Using a DAC-derived analogue voltage to drive an analogue motor.

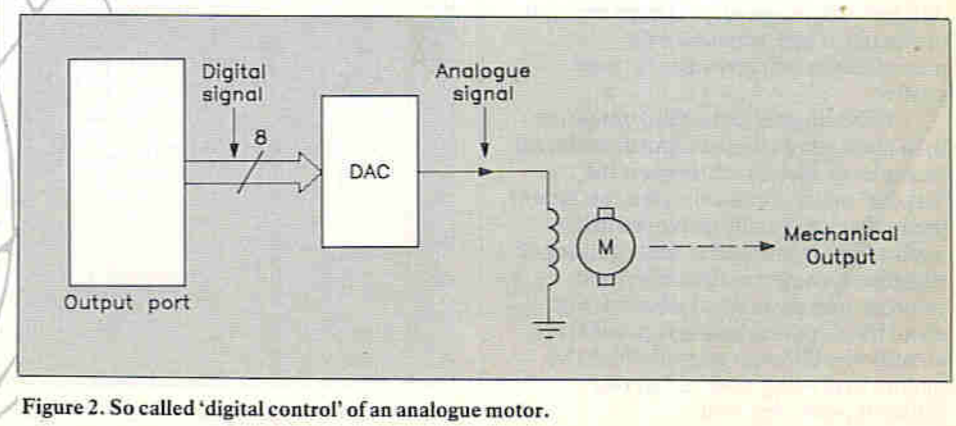


Figure 2. So called 'digital control' of an analogue motor.

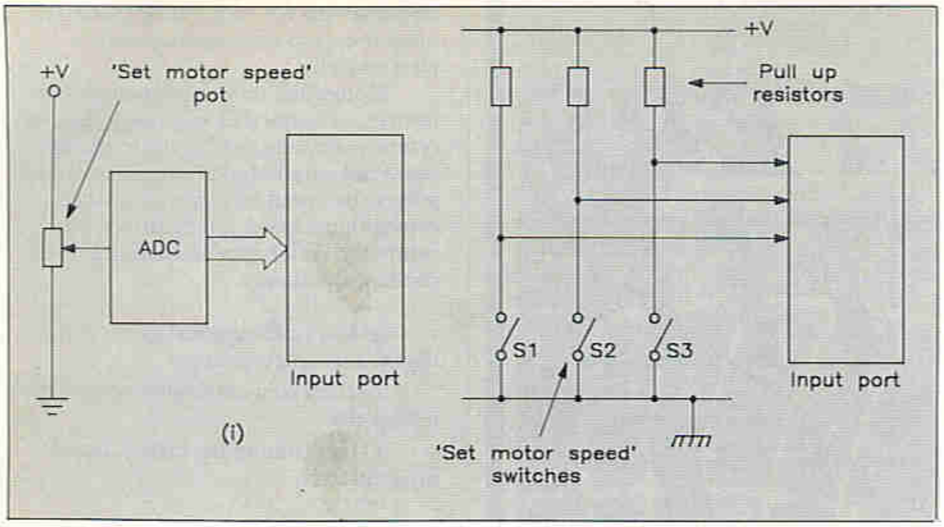


Figure 3. (i) Continuous speed control system (ii) Stepped speed control system with binary switches.

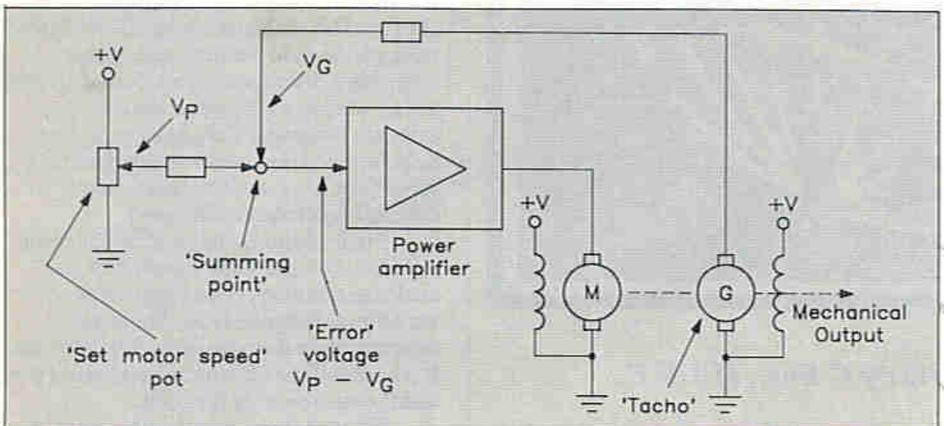


Figure 4. An analogue speed control servo system.

In the second method, (ii), the speed is selected by setting combinations of switches. For example, if three switches are used, they can be set in $2^3 = 8$ different combinations. This means that the choice of speeds is limited to this number. Whether this is adequate depends upon the application. In case it might be thought that using switches in this way to control motor speed is somewhat cumbersome, it might be worth bearing in mind that such switches might not be 'front panel' switches at all but 'embedded' in the process (whatever that is). For example, the switches might be operated in a certain sequence that depends upon the nature of the process. As each stage is completed (signified by the closing of a switch), a new motor speed (higher or lower) is selected automatically for the next stage.

Using the method of Figure 3 (i) gives almost 'stepless' speed control. If an 8-bit converter is used, this gives $2^8 = 256$ different values (speeds) between zero and maximum. Thus, rotation of the potentiometer will give smooth speed control.

This brings us to the third option, a motor that takes a digital input directly. An example of a motor of this type is the 'stepper' motor. In practice, it is this type of motor that is generally employed in the applications mentioned at the beginning of this article, namely in disk drives and printers. Bearing in mind what was said about these applications, it is possible to identify two different ways in which the motors were being used, in fact two different types of system.

(a) A speed control system, such as when controlling the rotational speed of the disk at exactly 360 rev/min.

(b) A positional control system, as when setting the read/write head correctly over the track (disk drive) or when moving the platen or print head just the right amount, as in printers.

The stepper motor is able to handle both of these tasks with great accuracy and without any really expensive electronics either. This is, of course, the prime reason for its use, apart from the obvious fact of not needing any signal conversion. Before looking at the stepper motor in principle and application, it is worth seeing the problems that exist in the two types of system mentioned and how they are handled in an analogue system.

Figure 4 shows an analogue speed control system. The motor is driven from the output of a power amplifier. The input to this amplifier is the difference between two voltages ($V_P - V_G$) and is known as the 'error' voltage. One of these voltages, namely V_P, represents 'speed demanded' and is sent in from a potentiometer. The second voltage, V_G, which only exists when the motor is running, is the output from a small generator (known as a tachometer or 'tacho'), driven by the motor.

Assume the system is at rest and then power is applied. With the potentiometer wiper anywhere other than at the bottom of the track, there is an input V_P to the amplifier, an output from the amplifier and plenty of torque to accelerate the motor. As the latter picks up speed, the tacho output V_G increases. Since the latter voltage is in anti-phase to the potentiometer voltage, the error voltage decreases. A balance is reached when the difference between the input and tacho voltages is just enough to maintain a given motor speed. Any change in the input demand will change the error voltage, up or down, until at a different speed balance is reached again. Notice that the system is self-balancing due to the negative feedback provided by the tacho. Such a system has the general title of 'speed control servo system'.

Figure 5 shows another servo system, this time for positional control. In this diagram there is a potentiometer input as before. The difference is that, in this case, it is not demanding a particular speed of the output shaft, but a particular position. The way it does this is also by using negative feedback, from a *second* potentiometer which has its voltage applied in the opposite polarity. In fact, both input and feedback potentiometers are usually centre-tapped to allow control in both directions on either side of centre. Ignoring the tacho in the diagram for the moment, the system works as follows.

The input to the amplifier is the difference between the two wiper voltages, V_1 and V_2. When both shafts are aligned

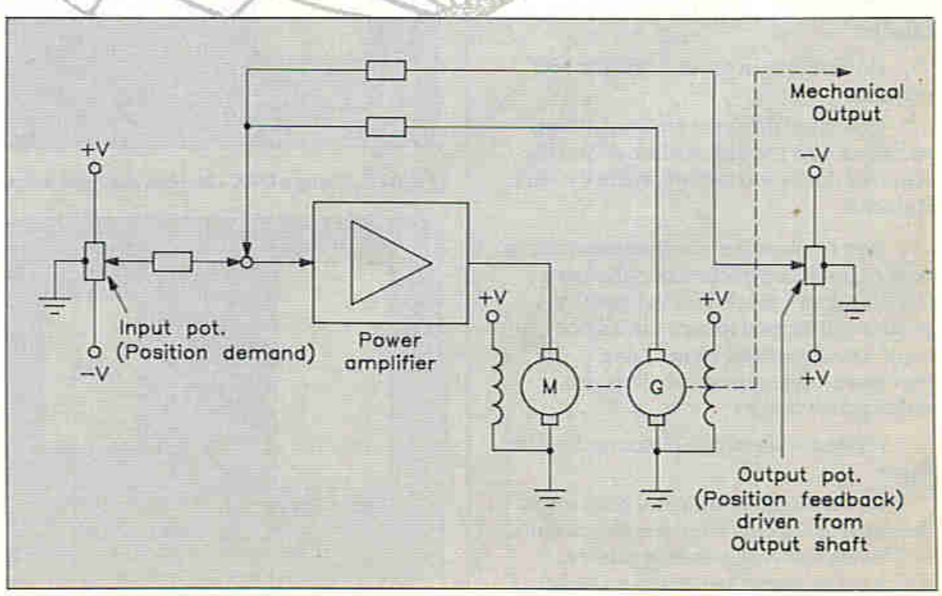


Figure 5. An analogue position control system.

the two wiper voltages are equal and opposite and cancel out. There is no input to the amplifier and no output to drive the motor. The system is at rest. Now rotate the input shaft; immediately there is an error voltage, $V_1 - V_2$, which is amplified to drive the motor. As the latter runs it drives the output shaft in such a direction that a position will be reached when, once again, input and output shafts are aligned and there is no error voltage. The system comes to rest.

But, you say, "what is the tacho doing"? Its role is not the same as in the case of the speed control system, where it had to provide a feedback voltage to maintain constant speed. In this case its function is to provide a 'damping' term, a rather more obscure function. Here's how it works.

Imagine that the tacho is omitted. It would seem that the system would still work. However, what is ignored is the effect of system inertia, or momentum if you like. When a motor is accelerating, as it does when it is racing towards its new position, it acquires momentum because of its mass. Just because it has reached its new position and no longer has a signal from the amplifier to drive it, doesn't mean that it will suddenly stop. Far from it; its momentum will carry it past the correct position. As a result of this overshoot, a feedback voltage will be derived from the wiper of the output potentiometer that will apply a torque to the motor 'in the opposite direction'. The motor stops (too late!), reverses, and heads back the way it came, once more gaining momentum. This time it overshoots from the opposite direction, probably not quite so much. The process of continually overshooting will repeat, again and again. In fact the system may become oscillatory.

What is needed is a means of decelerating the motor as it approaches its correct position, in fact a form of brake. The output from the tacho provides this, since it opposes the driving torque. As the motor gets less drive from the amplifier (approaching final position), the tacho voltage has more effect and brakes the motor. With the correct amount of feedback from the tacho, the system does a mild overshoot, returns and stops dead. Too much damping reduces the responsiveness; it becomes sluggish. Too little damping and the system tends to oscillate.

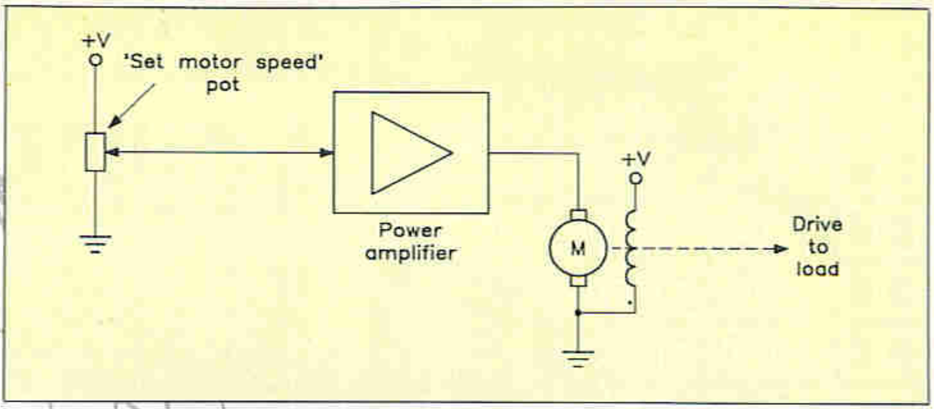


Figure 6. An open loop speed control system.

Closed and Open Loop Systems

The types of system described above are known as closed loop systems, because the feedback path 'closes' a loop. They can give great precision of control, speed or position, but are complex, require expensive components and can be tricky to set up.

By contrast, Figure 6 shows an 'open loop' speed control system. There is no feedback, so no self-balancing action. Setting the input wiper sets a certain speed. There is nothing that automatically governs this speed. Variations in load, for example, will cause the speed to vary. It is cheaper than an open loop system but not very precise.

Where does the stepper motor come in? Is it an open loop or a closed loop device? The answer is, it is an open loop device. Yet, because it works in a totally different way from an analogue motor, it is capable of excellent precision and doesn't require complex electronics to drive it. Nor is the motor itself inherently expensive.

Its secret can be understood from the fact that it is 'pulsed' in order to produce specific angles of rotation. For example, if one pulse caused the shaft to rotate '10 degrees', then in order to rotate it through exactly 40 degrees would involve nothing more than giving it four pulses. No elaborate feedback arrangements are needed.

Naturally, there are some complications and practical motors need a little more than a single pulsed input. But stepper motor controllers and drivers are available in IC form, requiring only a few external components. So now for some basic principles.

Stepper Motors

A stepper motor consists of a stator, rotor and windings. It does not rotate when power is applied but moves through a precise angle in response to a single pulse. Therefore the total angular displacement is given by:

Total angular displacement = step angle x number of pulses

Step angles vary in the range 0·45° to 90°, the most common value being 1·8° per step (giving 200 steps per revolution). A typical positional error is ± 5%.

Stepper motors can be used for linear rather than angular positioning by using one to drive a leadscrew and nut. The relation between the linear displacement and the number of step pulses is given by:

Linear displacement equals the number of steps divided by the total number of steps per revolution multiplied by the pitch of the leadscrew.

Example: if the leadscrew had 5 threads/inch, its pitch would be ⅕in., which equals 0·2in. If the step angle was 1.8°, each step pulse would produce a linear displacement of 1 ÷ 200 x 0·2 = 0·001in.

This makes the stepper motor suitable for industrial applications in such things as automatic lathes, milling and drilling machines, X-Y co-ordinate tables and other positioning mechanisms. This is in addition to the computer applications already discussed.

Advantages of Stepper Motor Systems

The major advantage of stepper motor systems is that control is 'open loop', so avoiding the problems of closed loop control. All that is needed is a counter (that can be software or hardware) that counts the number of pulses from the start. The speed of rotation is controlled by the frequency of the pulses, and the acceleration is determined by the rate of change of frequency. See Figure 7.

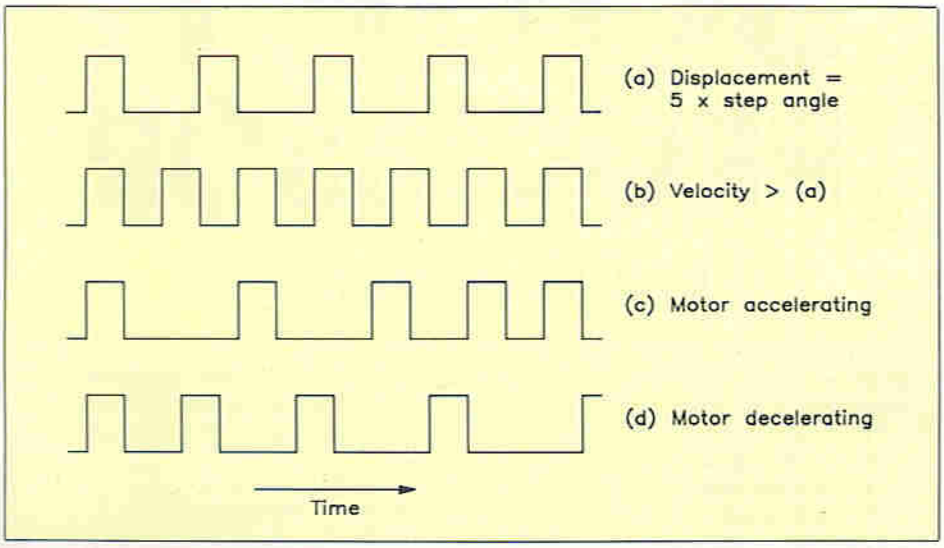


Figure 7. Pulses control position, speed and acceleration.

Stepper Motor Principles

There are three common types:

(a) Variable reluctance motor
(b) Permanent magnet motor
(c) Hybrid motor

Continued on page 38.

A GUIDE TO USING

THE MAPLIN DOLBY SURROUND SOUND PROCESSOR

by Robert Ball AMIPRE

Features:

- ★ Decodes Dolby Surround encoded material
- ★ Offers a choice of other effects from non-Dolby material
- ★ Integral amplifiers for rear channel loudspeakers
- ★ Adjustable rear channel delay
- ★ 100Hz and 10kHz boost controls
- ★ Volume and front/rear fader controls
- ★ Tape monitor
- ★ LED level meter
- ★ Infra-red remote control

Introduction

Dolby Surround is an AV (audio/visual) surround sound system for use in the domestic environment, its purpose is to enhance the experience of listening/viewing suitably encoded material. The Dolby Surround system is the domestic equivalent of the professional Dolby Stereo system used in cinema theatres to provide high quality dialogue and surround sound. A Dolby Surround decoder can be integrated into an existing AV system, however the existing system must be capable of replaying or receiving Dolby Surround encoded material.

Suitably encoded Dolby Surround material includes many of the feature films that have been produced in Dolby Stereo and subsequently released on video cassette and video disc. Encoded feature films are also broadcast via satellite television, which has stereo capability and on terrestrial television in NICAM-728 Digital Stereo. However, for any of these recording or transmission media to be sources of decodeable material, the play-back or receiving equipment must be capable of providing high quality stereo sound. This will be dealt with in greater detail later in this article.

There are two generations of domestic decoding equipment, passive and active. The passive system (first generation) is known as Dolby Surround and the active system (second generation) is known as Dolby Pro Logic. The processor reviewed here is of the passive type.

Details of how Dolby Stereo, Dolby Surround and Dolby Pro Logic systems function will not be dealt with in this review. Readers who are interested in the principles of operation should refer to the April - May 1990 issue of 'Electronics', where the Dolby Stereo system is explained in its entirety.

This review is intended to introduce the Maplin Dolby Surround Sound Processor, illustrate how it can be integrated into an AV system and give some guidelines on how the best performance can be obtained.

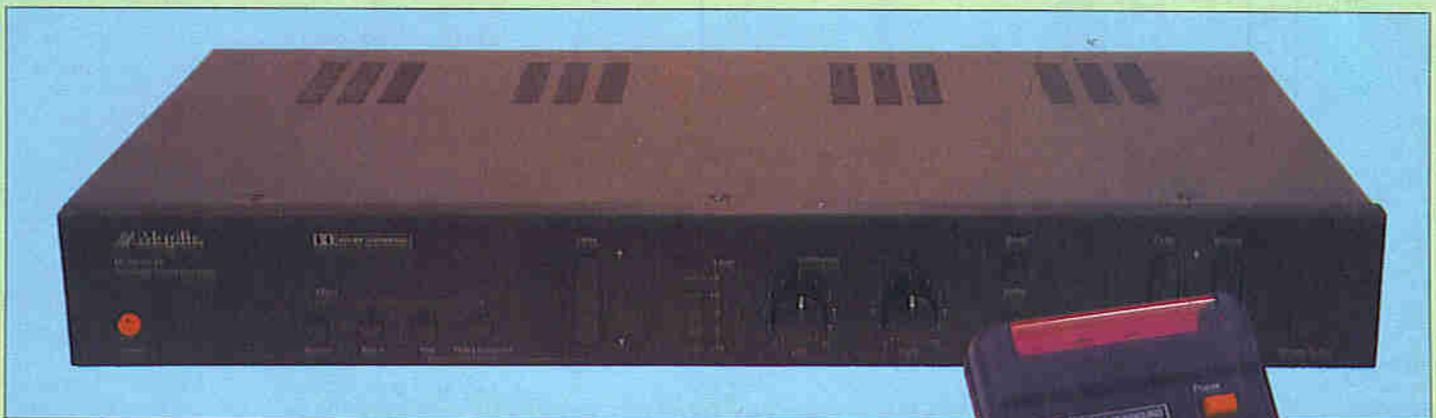


Photo 1. The Maplin Dolby Surround Sound Processor.

Specifications:
Internal Power Amplifier (Test conditions: 8Ω Load, 1 channel driven, 1kHz sinewave)

Amplifier Power:	10W RMS per channel
Total Harmonic Distortion:	0·5% (10W RMS)
	0·15% (9W RMS)
Signal Processing Circuitry	
Input Sensitivity:	250mV
Input Impedance	
Line input:	27kΩ
Tape input:	47kΩ
Output Impedance	
Line output:	<200Ω
Rear output:	<200Ω
Tape output:	<2kΩ
Signal to Noise (unweighted, 0dB = 775mV)	
Line output:	75dB
Matrix:	81dB
Hall:	65dB
Dolby Surround:	63dB
Frequency Response (−3dB +0·5dB)	
Line output:	5Hz - 300kHz
Matrix:	10Hz - 28kHz
Hall:	10Hz - 7·6kHz (delay time 14ms)
Dolby Surround:	26Hz - 7kHz (delay time 14ms)
Distortion (output level −10dB)	
Line output:	0·07%
Matrix:	0·07%
Hall:	0·15%
Dolby Surround:	0·15%
Delay time:	14ms - 32ms
Power Requirements:	240V AC 50Hz 90W
Dimensions:	419 x 52 x 231mm (WHD)
Infra-Red Remote Control	
Batteries:	2 off AA cell
Dimensions:	65 x 20 x 170mm (WHD)

Photo 2. The infra-red remote control unit.

Outwardly Appealing

The black anodised aluminium front panel gives a smart, sleek-looking appearance, which is in keeping with current fashionable trends for audio equipment. The chassis and top-cover are fabricated from mild-steel. Control and indicator designations are white and therefore easily readable. The push-button and press/release 'up/down' rocker switches are of the 'soft touch' type and have a nice positive action. The push-button switches each incorporate an LED indicator, which serves to display the

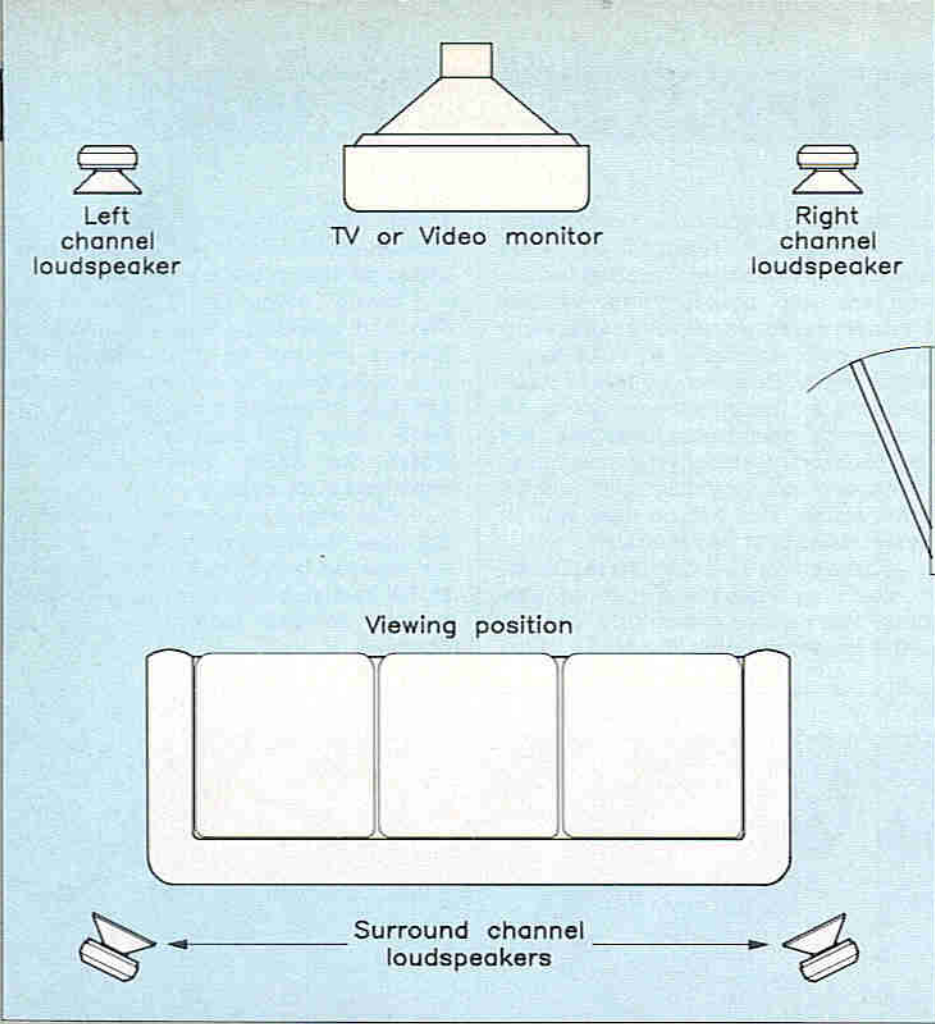


Figure 1. Typical loudspeaker placement.

selected mode or function. An interesting point is that the power on/off switch does not switch the mains supply, but instead operates a relay, which switches the low-voltage supply on the secondary of the transformer. The power switch is perhaps better regarded as an on/standby switch rather than a true on/off switch. Also on the front panel are rotary potentiometer controls for 'calibration', a 5-segment green LED VU bargraph and an infra-red window for the remote control receiver diode. Photo 1, shows the Maplin Dolby Surround Sound Processor. The infra-red remote control unit is equally as smart in appearance as the main unit and has colour coded buttons to group the controls. Photo 2, shows the infra-red remote control unit.

Features and Facilities

The Maplin Dolby Surround Sound Processor (YU59P) offers a host of facilities to anyone who wants to add Dolby Surround to their existing AV system. But before the various features are described, a quick resumé of what Dolby Surround is would be helpful: movie films encoded with Dolby Surround have extra information 'buried' into the normal stereo sound-track, which can be extracted (decoded) and reproduced through extra loudspeakers placed to the rear of the listening position. The extra surround channel provides sound effects and adds ambient sounds. The overall effect is to project the viewer/listener into the on-screen action. Typical loudspeaker placement is illustrated in Figure 1.

With the Maplin Dolby Surround Processor, selecting the 'Dolby Surround' mode activates the decoding circuitry allowing surround sound to be reproduced from suitably encoded material; the results obtained are nothing short of dramatic. Apart from decoding Dolby Surround encoded movie films the unit also provides 'Matrix' and 'Hall' modes. Selecting 'Matrix' provides a psuedo-stereo effect for use with mono programmes and movie films and selecting 'Hall' provides reverberation and ambience to stereo programmes and films that are not Dolby Surround encoded. The degree of success achieved with 'Matrix' and 'Hall' will depend on the source material and is also dependent on subjective personal taste. A 'Monitor' switch is also provided, which is used to switch to the tape monitor input sockets on the processor. This facility is provided so that when the processor is connected into the existing AV system, often via the tape input and output sockets of the integrated amplifier (a single unit having both pre-amplification and power amplification), the tape deck can still be used without having to swap cables. Each of the four modes are selected by front panel push-buttons, the current mode selected is indicated by an integral red LED on the appropriate push button. Only one of the four modes may be selected at a time.

To add the surround sound to the listening/viewing environment, it is necessary to have an extra pair of speakers in addition to those already present for the existing stereo AV system. However these speakers do not have to be particularly large or expensive as the frequency range of the surround signal is deliberately restricted. To drive the additional pair of loudspeakers, the surround sound processor incorporates a 10W RMS per channel power amplifier. The inbuilt amplifier thus obviates the need for a separate external amplifier, and removes what would otherwise be a 'hidden' cost of purchasing another amplifier. The power rating of 10W RMS per channel is perfectly sufficient for most domestic use, although the rear channel signals are available on separate 'Surround' output sockets if the user wants to employ an external amplifier.

The overall volume of the AV system may be controlled via the processor's volume control, the balance between front and rear signal levels is variable by means of a fader control. The volume and fader controls are of the press/release rocker switch type, as is the delay control.

An audio delay is introduced in the rear channel signals, the purpose of which is two-fold. Firstly, when the processor is in 'Dolby Surround' mode, the delay improves the apparent separation between front and rear channels (Haas effect). Secondly, in both 'Dolby Surround' and 'Hall' modes the delay gives the impression that the rear loudspeakers are further away than they actually are, thus increasing the apparent size of the soundfield. The delay is variable over the range 10ms to 30ms (nominally) and adjustment is achieved by means of a rocker switch located on the front panel. The delay time is adjusted according to personal taste to give the best effect.

There are two boost buttons provided which allow the 100Hz and 10kHz frequency bands of the rear channel to be boosted. The boost controls allow compensation for the rear loudspeakers

Photo 3. A Hi-Fi Stereo VCR with NICAM-728 decoder.

lacking in bass or treble. Both boost buttons may be operated simultaneously if required. If these functions are active, indication is provided by integral red LED's.

To allow the correct decoding of Dolby Surround encoded material, two calibration controls are provided, these are used to minimise the leakage of front channel dialogue into the rear surround channel. Normally these controls will be set at midpoint and adjusted slightly to the left or right to minimise dialogue leakage. These controls also allow the unit to accommodate signals of different levels and therefore minimise the introduction of unwanted noise. To indicate the level of the signal in the rear channels, a 5-segment green LED VU bargraph is provided. Ideally when rear channel signals are present, the bargraph should peak to a maximum level of 0dB. Compensation for levels exceeding the 0dB point can be made by retarding both of the calibration controls (anti-clockwise) until the correct level is achieved, conversely if the signal peaks do not reach 0dB, advance both calibration controls (clockwise) as necessary.

An infra-red remote control unit is provided to allow operation of the processor from the comfort of your favourite armchair. The remote control unit duplicates all of the front panel controls except the calibration controls and the LED VU bargraph.

It must be remembered that this unit is designed for use in an AV system and not for use in an audio only Hi-Fi (high fidelity) system. For users of audio only Hi-Fi equipment, a different type of surround sound processor should be employed to add reverberation and ambience to normal stereo recordings.

Stereo Source Considerations

It has already been stated that for Dolby Surround to be decoded, the movie film must be suitably encoded. It also goes without saying that the receiving/replaying equipment must be capable of providing high quality stereo sound if the encoded information is to be extracted. Essentially this means that if the source is a VCR (video cassette recorder) the unit must be Hi-Fi Stereo and replaying a Dolby Surround encoded stereo video cassette. If the source is a satellite receiver, the unit must be stereo and receiving a Dolby Surround encoded stereo programme. If the source is a television set or television tuner unit, it must be equipped with a NICAM-728 digital stereo sound decoder and be receiving a Dolby Surround encoded stereo programme (in an area transmitting NICAM). If the source is a VDP (video disc player) the unit must be stereo and replaying a Dolby Surround encoded stereo video disc. Photo 3 shows a Hi-Fi Stereo VHS video cassette recorder which also has a NICAM-728 decoder, thus

Photo 4. A typical AV system.

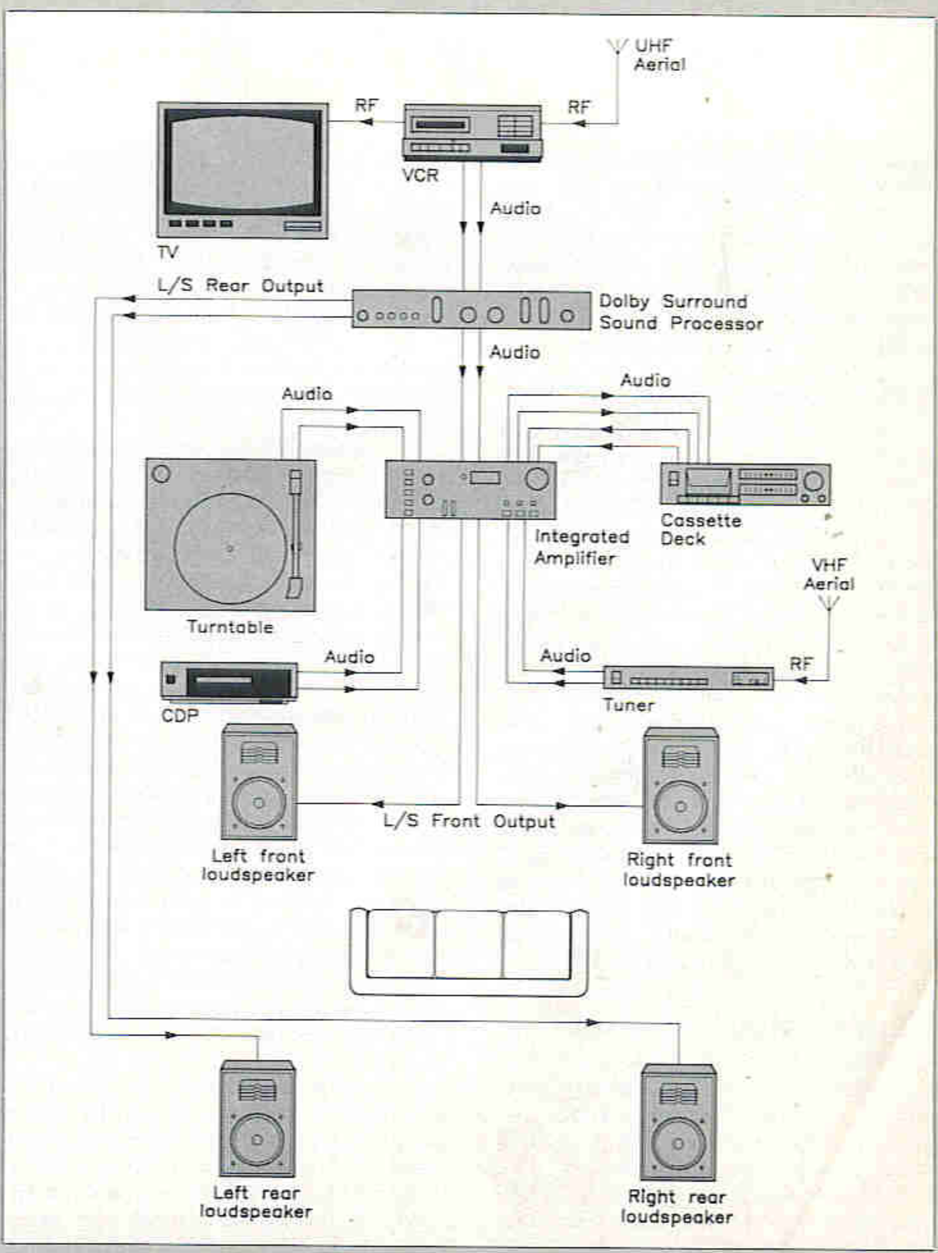


Figure 2. Diagramatic representation of installation.

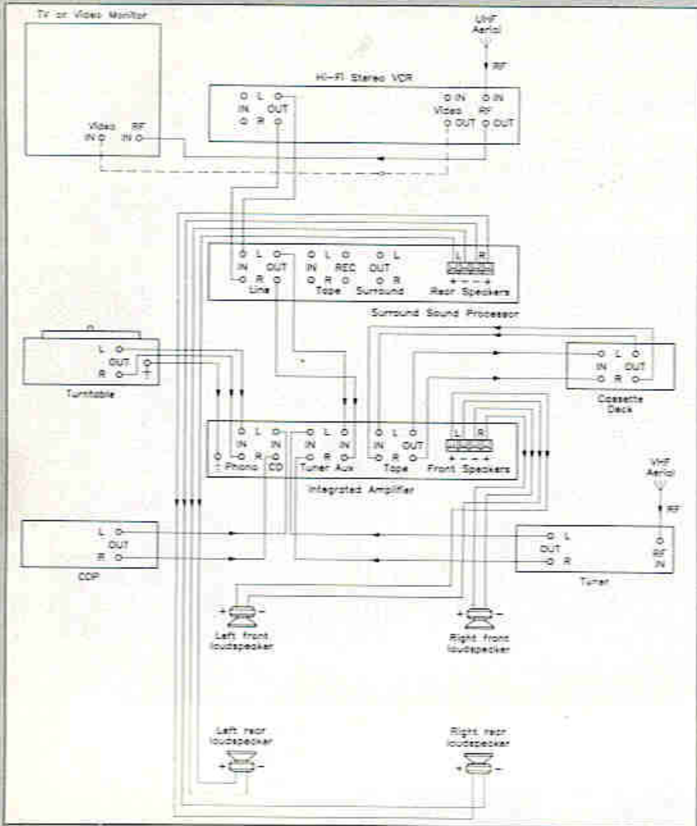


Figure 3. Wiring diagram where the processor is connected between VCR and integrated amplifier.

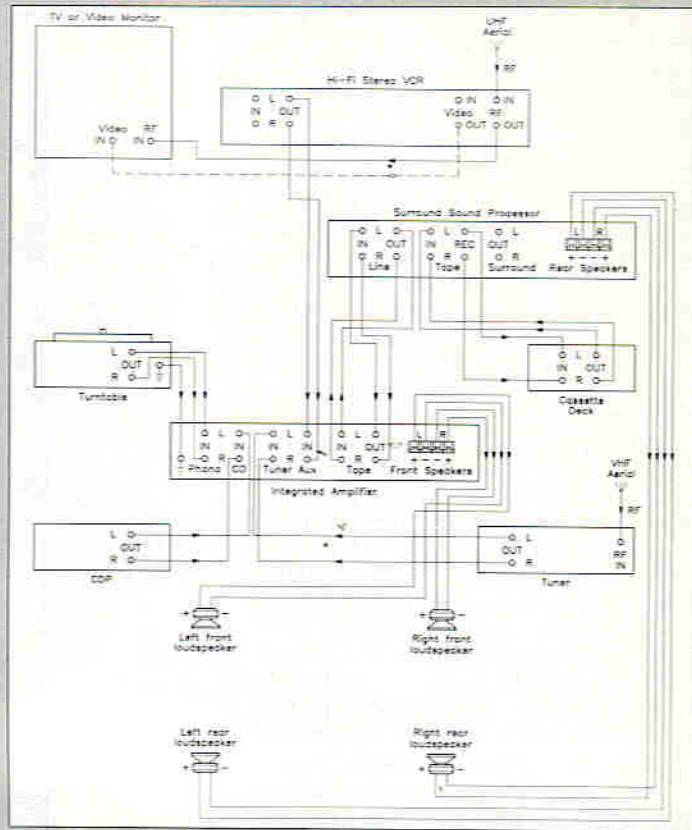


Figure 4. Wiring diagram where the processor is connected between integrated amplifier and cassette deck.

allowing Dolby Surround to be decoded from video cassette and off-air movie films. A list of over 700 movie films that have been released in Dolby Stereo is available from Maplin Electronics.

Video Cassette Labelling

In recent years the Hi-Fi Stereo logo has become more widely adopted on video cassettes, and there are indications that the Dolby logo's will be used in a more orderly fashion. Cassettes will soon say 'Dolby Surround' rather than just having the Dolby Double-D logo. Commonly the Dolby logo will be found on video cassettes with a note 'on linear tracks' below it. This indicates that the linear audio tracks (as opposed to the Hi-Fi Stereo tracks) are recorded with Dolby B noise reduction and doesn't mean that the tape is Dolby Surround encoded. The confusion continues, but in a pleasant way; there are many cassettes that do not show any indication that they may have Hi-Fi Stereo tracks, but do have them all the same! Also cassettes which have 'Dolby Stereo in Selected Theatres' on the credit list are also Dolby Surround encoded.

Connections

The hardest part of using the surround sound processor is connecting it into an existing AV system. There are many possible combinations of equipment; Satellite Receiver, NICAM TV, NICAM Stereo Tuner, Hi-Fi VCR, VDP, Pre-amplifier & Amplifier or Integrated Amplifier, FM Tuner, Cassette Tape Deck, CDP (compact disc player), Turntable, Graphic Equaliser, Surround Sound Processor and Loudspeakers. An individual system may include all or just some of these items. Photo 4 shows a typical AV system. Even with the wide range of possible combinations, there are really just three main ways of connecting the processor up.

The simplest way to use the processor is to add it between the VCR and the integrated amplifier, this type of arrangement is illustrated in Figure 2, which shows the signal paths between the various items of equipment. The main advantage is that the existing audio equipment connections are not drastically changed and installation is quick and easy. With the processor installed in this way, processing can only be performed on the audio signals from the VCR, this is not a major drawback as the other audio sources will not be Dolby Surround encoded. Since the processor only deals with the audio signals from the VCR, it can be switched into standby (or off completely) when the VCR is not in use. Figure 3 shows the actual connections between the various items of equipment. As can be seen, the tape 'IN' and 'REC' sockets on the processor are not used and pressing the 'monitor' switch will result, not surprisingly, in silence!

The second way of connecting the processor is as suggested in the leaflet supplied with the processor. With this method of connection, the processor is added 'in between' the integrated amplifier and the cassette deck (the same method of connection that many graphic equalisers use). With the integrated amplifier's 'tape monitor' switch operated, the signal path is via the processor. To monitor the output from the cassette deck, the 'monitor' button on the processor should be pressed to route the signal from the tape deck to the integrated amplifier. The main advantage being that the volume of all audio sources can be varied by operating the volume up and down buttons on the remote control unit. The unit may be bypassed completely by de-selecting the tape monitor on the integrated amplifier and the processor can be switched into standby when it is not in use. However it is always necessary to have the processor on when the cassette deck is in use. Figure 4 shows the actual connections between the various items of equipment.

The third way of connecting the unit is by making use of a group of sockets found on some integrated amplifiers, which are usually labelled as 'Equaliser', 'Link' or 'Pre-amplifier & Main amplifier Link'. These sockets will normally have a shorting link between them, thus connecting the output of the pre-amplifier directly into the main amplifier. The processor can be connected to the integrated amplifier via these sockets and thus all signals are routed through the processor. The main advantages being that normal operation of the tape monitor is retained and the volume of all audio sources can be varied from the remote control (as with the previous configuration). However, the processor must always be switched on when the audio equipment is in use, otherwise the signal path will be broken. Figure 5, shows the actual connections

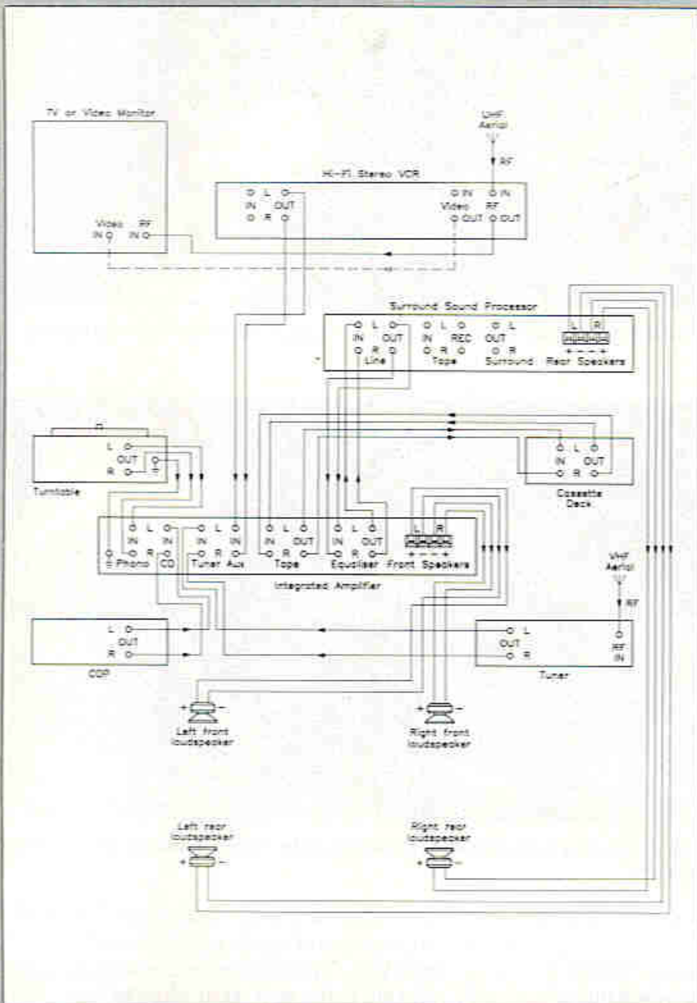


Figure 5. Wiring diagram where the processor is connected to 'equaliser' sockets.

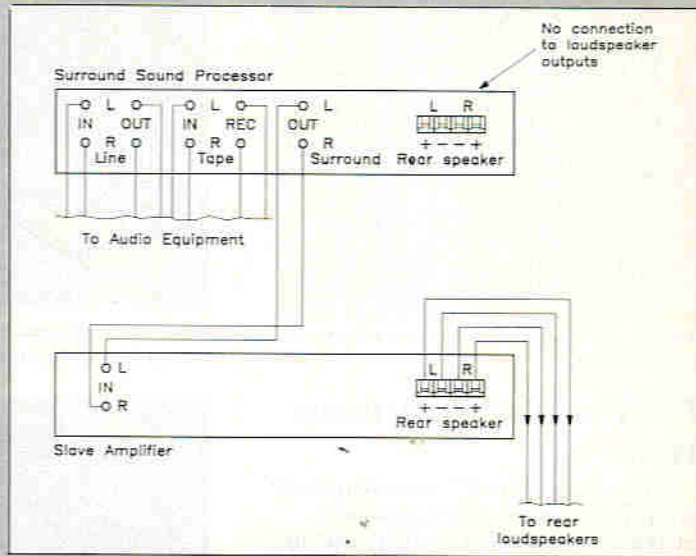


Figure 6. Using a slave amplifier with the rear channels.

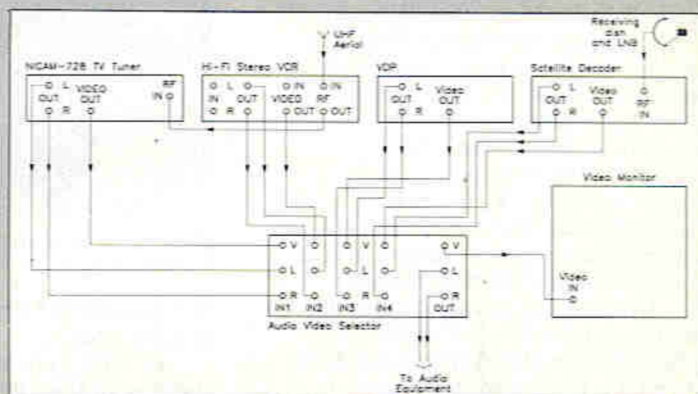


Figure 7. Connecting a number of different stereo sources to an AV system.

between the various items of equipment.

Two points worth remembering are that: 1) The processor adds rear channel surround information by processing the applied signals, it does not alter the front channel signals, apart from providing adjustment of volume and front/rear balance. 2) The 100Hz and 10kHz boost buttons provide boost to the internal rear amplifier signals only, the line level surround outputs are not affected by operation of the boost buttons. The reason being that if an external amplifier is used for the rear channels, this amplifier will have its own tone controls to provide treble or bass adjustment, thus obviating the need for 100Hz and 10kHz boost buttons.

Slaving Away

If an external slave amplifier is used to reproduce the rear surround channel signals, it should be connected to the line level surround outputs on the processor. The rear loudspeakers should then be connected to the slave amplifier instead of the processor. The connections are illustrated in Figure 6.

Adding Inputs

The examples so far have illustrated an AV system with just a VCR. If however, the user wishes to connect other sources, such as a NICAM-728 TV tuner unit, VDP or satellite decoder, provision will have to be made for selecting between the different audio and video input signals. This is best accomplished by using an audio video selector unit, otherwise a lot of lead swapping will be necessary. Figure 7, illustrates the interconnections.

In Use

Using the Dolby Surround Sound Processor is very straightforward, but it is important to correctly adjust the calibration controls. In most cases, all that is necessary is to set the calibration controls to their midpoints and when replaying a Dolby Surround encoded movie film, adjust either of the calibration controls a little to the left or the right to minimise the leakage of front channel dialogue in the rear channels. It will be found that there is a 'null' point where signal leakage is at a minimum and either side of that point signal leakage will increase. The calibration controls also allow for a varying range of input signal levels and as previously mentioned, should be used in conjunction with the level meter.

The overall volume and front/rear balance should be adjusted to give the best surround soundfield, it is tempting at first to have the rear level set high, but excessive rear levels will detract from the overall effect instead of enhancing it, not what is wanted at all! The delay control should be adjusted to give the best effect. It will probably take a little time to get used to how the controls affect the overall soundfield, but the experimentation will soon pay off, resulting in an exciting viewing and listening experience.

Conclusion

The Maplin Dolby Surround Sound Processor offers a reasonably priced introduction into the world of Dolby Surround Sound, and provides greater realism to movie films on the 'small screen'.

The Maplin Dolby Surround Sound Processor is available from regional Maplin stores and by mail order. Order as **YU59P** Surround Sound Processor price **£139.95**.

A list of movie films released in Dolby Stereo is also available. Order as **XL15R** Dolby Film List price 40p.

Acknowledgements

Ferguson Video Cassette Recorder supplied by Radio Rentals Ltd, Hockley.
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Computers in the Real World
Continued from page 31.

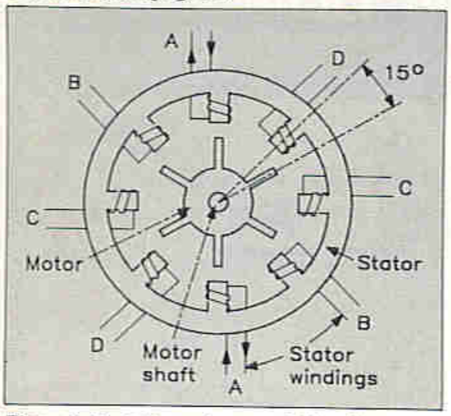


Figure 8. Basic four-phase variable reluctance stepper motor.

Photo 1. Stepper motors used in Epson printer. Left motor drives head via toothed belt; right motor drives platen via gearing.

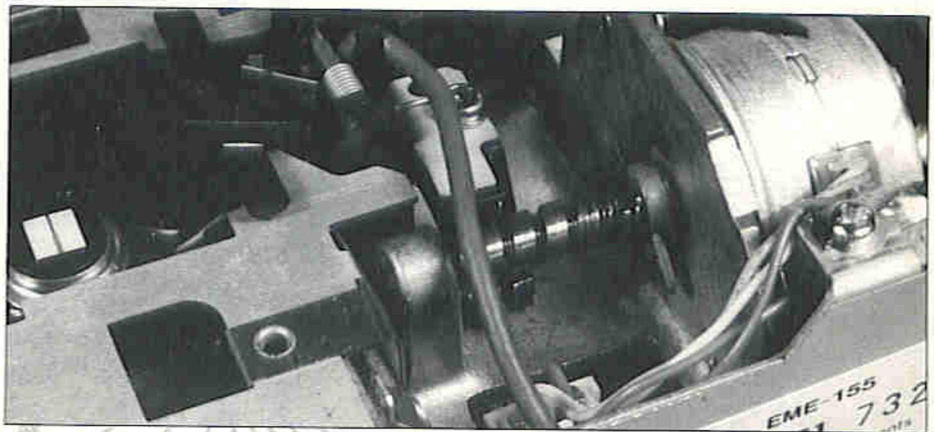


Photo 2. Stepper motor used to drive read/write head (seen at left) of disk drive. Example shows 3in. Matsushita drive used by Amstrad.

The Variable Reluctance Motor

This type (Figure 8) has a soft iron core with projecting teeth at precise angular intervals. The stator has inward projecting teeth but there are more of them so that, when two rotor teeth align with two stator teeth all the rest are misaligned (by an angle of 15° in the example shown). The stator teeth have coils wound on them, which can magnetise them if D.C. is passed through them. In the example, coils A are magnetised. If instead coils D are magnetised, the rotor moves A.C.W. (Anti-ClockWise) 15°. If coils B are magnetised, the rotor moves 15°. The rotor can be made to rotate one way or the other by sequencing the supply to the coils.

Variable reluctance motors can be used at high speed but with light loads. There is little residual magnetism to hold the rotor in position when the supply is switched off. The inertia of the rotor causes overshoot and oscillation.

The Permanent Magnet Motor

The rotor (Figure 9) is a radially magnetised permanent magnet that rotates to align itself with the field produced by the electromagnets. Using the switches A, B, C and D gives four possible field directions and, hence, four positions (90° steps) for the rotor. Half steps can be obtained if the switches are three-position with the centre position 'open'.

In Table 1 a '1' indicates a phase energised; a '0' is a de-energised phase. This table shows the switching sequence for half-step and full-step operation of a four-phase stepper motor.

The Hybrid Stepper Motor

In this type the rotor has two sets of teeth at 180° to each other and is axially magnetised, one set of teeth being North poles, the other set being South poles. Stator windings can also be either N or S, depending upon the directions of the currents in the coils. Between the rotor and stator there will be, at some places forces of attraction, and at other places forces of repulsion. These forces cause rotation from one step to the next, controlled by the sequencing of the four phases.

Stepper Motor Drives

A stepper motor moves from one step to the next by switching a D.C. supply from one set of stator windings to the next. The simplest, and cheapest, way of driving stator windings is with a single drive transistor per winding.

Figure 10 shows unipolar and bipolar circuits. The latter type produce 20 – 40% more power at low speeds than unipolar types but need two power supplies.

Figure 11 shows a stepper motor interface; this is a fairly simple type and, in practice, they can be much more complex. The motor windings are numbered 1, 2, 3 and 4.

There are three sequences:

Low power: 1, 2, 3, 4, 1 etc. (only one winding energised).

Normal: 1 & 2; 2 & 3; 3 & 4; 4 & 1, etc. (always two windings energised).

Half-step: 1 & 2; 2; 2 & 3; 3; 3 & 4; 4; 4 & 1; 1, etc. (always half-angle steps between each step).

The motor windings require a current source as their resistance is low, e.g. 0·2Ω. They also have high inductance values so that protection is needed for the switching transistors.

The timing for mode 2 (normal) is shown in Figure 12.

The truth table of Figure 12 shows that a half-byte of two O's and two 1's circulates

Phases energised	Rotor position No.	Direction
A, D	1	Clockwise ↓ / Anti-clockwise ↑
A, C	2	
B, C	3	
B, D	4	
A, D	1	

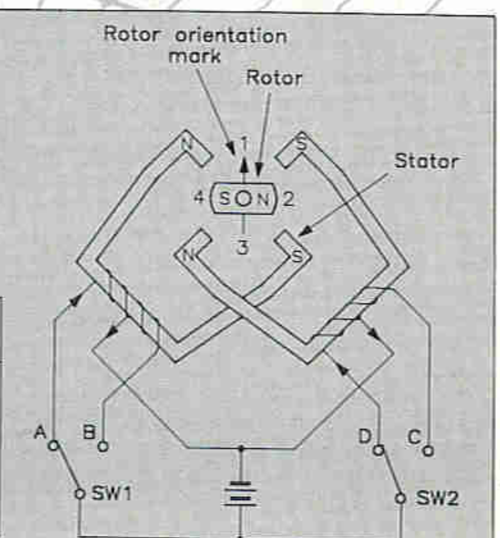


Figure 9. The four-phase, two-pole, permanent magnet stepper motor, showing phase switching to obtain CW or ACW rotation.

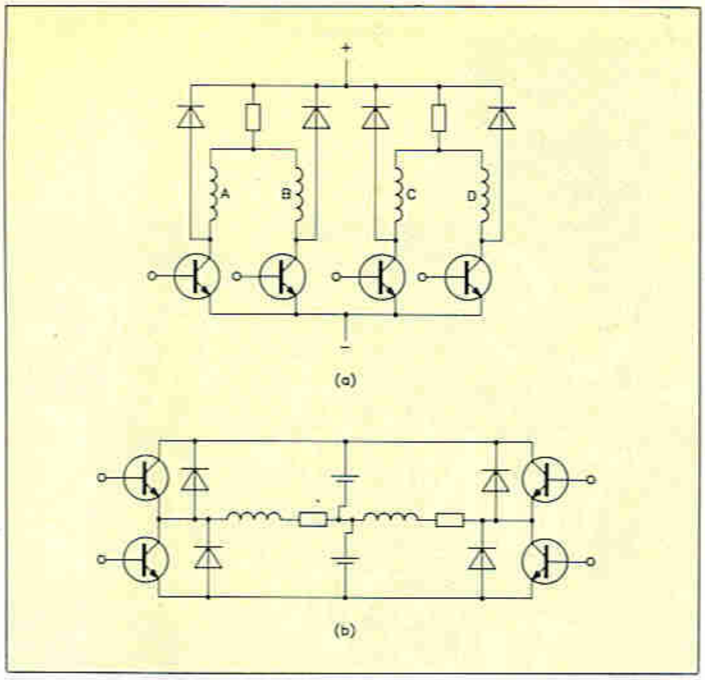


Figure 10. Drive circuits for (a) unipolar and (b) bipolar stepper motors.

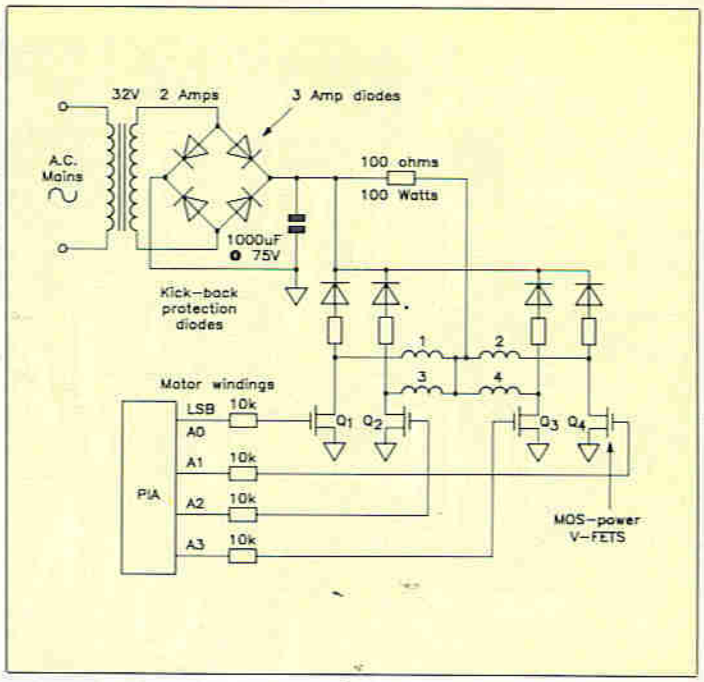


Figure 11. A stepper motor interface.

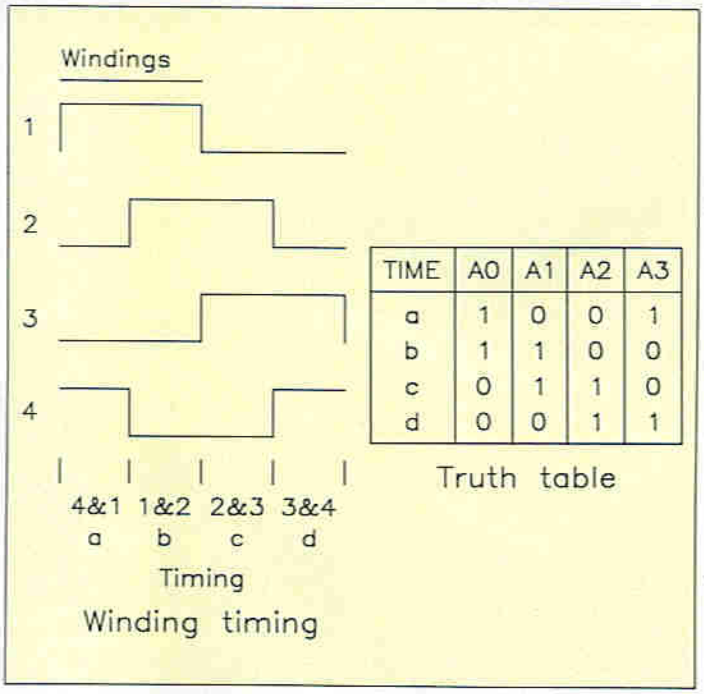


Figure 12. Winding timing diagram and truth table for interface of Figure 11.

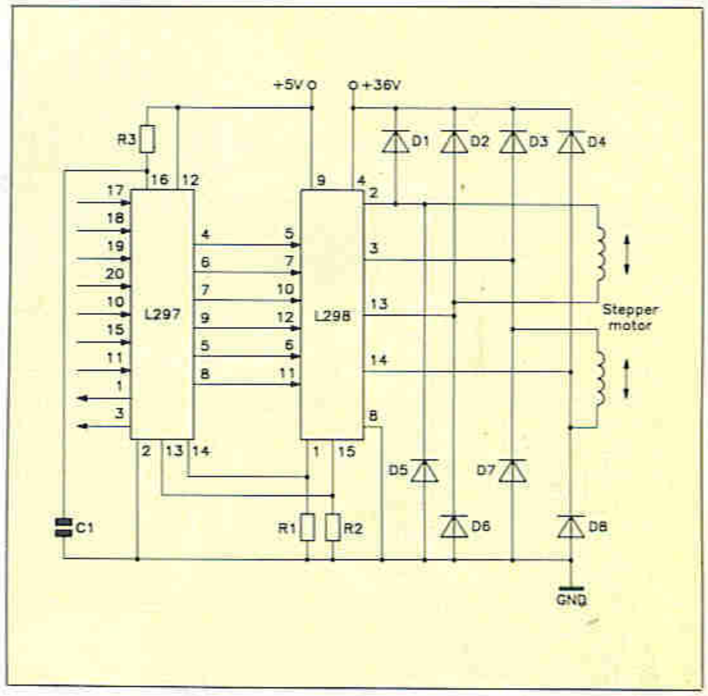


Figure 13. Application circuit for the L297 and L298 stepper motor controller and driver IC's.

continuously. This gives a clue as to how the waveforms to drive the motor can be obtained. If a 4-stage register is constructed and two successive stages are 'set' (Q=1) while the other two stages are cleared, then clock pulses will circulate the required pattern around the register. Naturally, it is possible to produce the same result by writing a machine code routine that circulates two '1's' in this way, outputting them to the PIA between shifts. The number of shifts determine the angle turned through; the frequency of the shifting determines motor speed.

Available Stepper Motor IC's

1. The SAA1027 stepper motor driver

This IC is designed to drive 4-phase unipolar stepper motors and operation is simplicity itself.

	Full step			
Step	Phase			
	A	B	C	D
1	1	0	0	1
2	1	0	1	0
3	0	1	1	0
4	0	1	0	1
1	1	0	0	1

	Half step			
Step	Phase			
	A	B	C	D
1	1	0	0	1
2	1	0	0	0
3	1	0	1	0
4	0	0	1	0
5	0	1	1	0
6	0	1	0	0
7	0	1	0	1
8	0	0	0	1
1	1	0	0	1

Table 1. Switching sequence of 4-phase stepper motor.

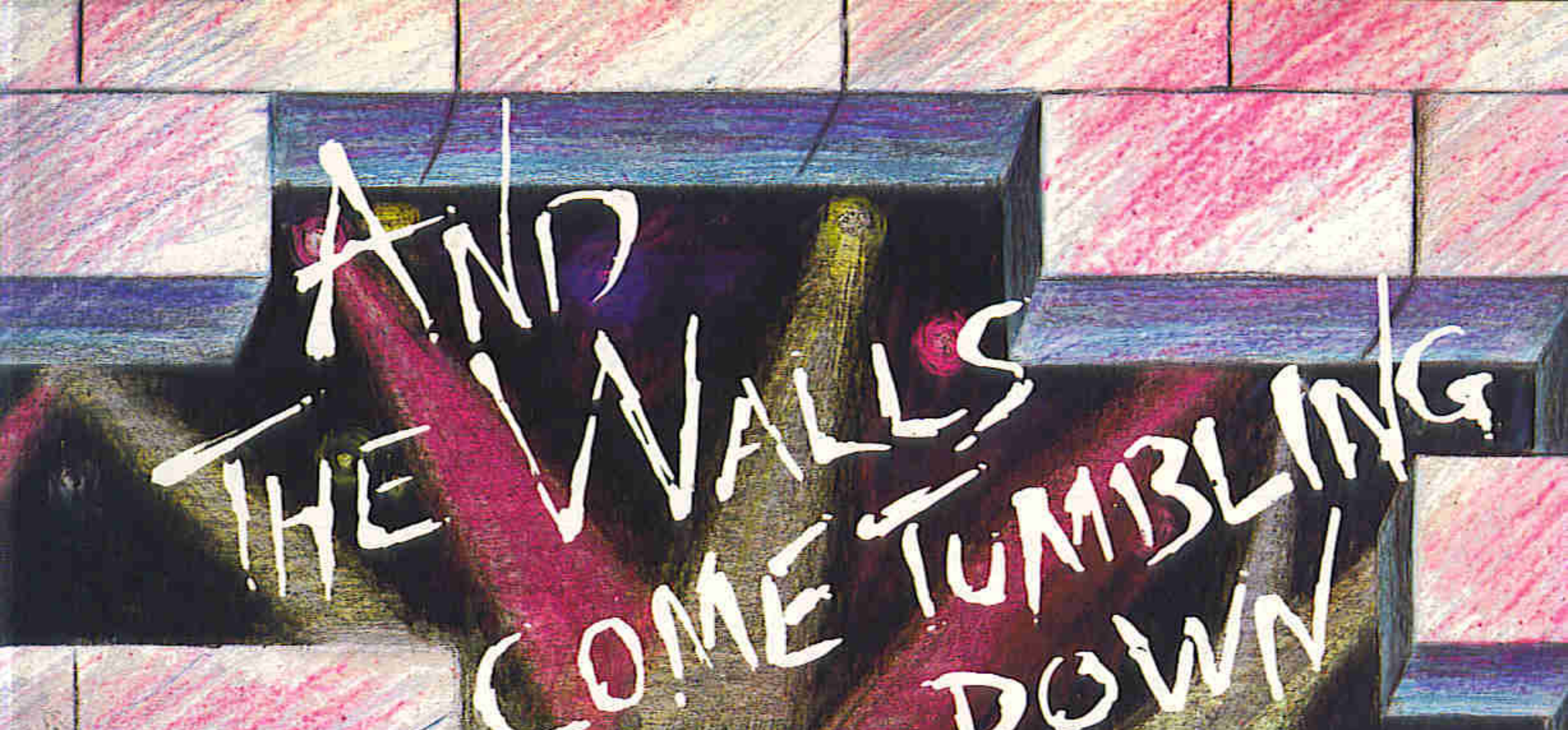
The logic level on pin 3 determines the direction of rotation. For clockwise rotation it must be low; for anti-clockwise running it should be high. Therefore, under program control the motor is easily reversed merely by writing a different logic level to the PIO port line that feeds this pin. A pulse input on pin 15 causes a step whenever the voltage transition is from logic 0 to logic 1.

2. The L297 and L298 stepper motor controller and driver

The L297 controller will work with either 2-phase bipolar or 4-phase unipolar stepper motors. It pairs well with the L298 driver for providing the drive capability to the stepper motor. The application circuit for these two IC's driving a 2-phase stepper motor is shown in Figure 13.

Full specifications and pin-outs for both of these IC's appear in the Maplin catalogue.

And the Walls Come Tumbling Down



by Alan Simpson

Acclaimed as being the worlds largest and most ambitious live musical event to be staged anywhere in the world, "The Wall" hit Berlin earlier this summer. The rock-and-roll circus took over the no-man's land between East and West Berlin, Potsdamer Platz, the site of the Berlin Wall, sealed off from the outside world for over three decades.

The old formidable wall made way for the Pink Floyd version with former leader Roger Waters driving the new production. Supporting acts are in the equally impressive class, headed by Sinead O'Connor of "Nothing Compares" fame, a leading symphony orchestra, military marching bands and the full blown Red Army choir.

Armies of workers raced against time building a 600 foot long and 60 foot high wall together with an amphitheatre seating some 200,000 people. Television coverage extended the world-wide audience by an estimated one billion.

Mark Fisher (left) and Jonathan Park with model for stage set for Roger Waters' 'The Wall', Berlin 1990.

OUT AND ABOUT

Scenes from the Rolling Stones 'Steel Wheels' U.S. tour in 1989.

Show facts and figures flowed thick and fast. Five thousand metres of electric cable powered by five megawatts of generated electricity – enough to power a whole town – 130 tons of steel scaffolding, assorted lifts, tower cranes, trucks and bulldozers. Dick Tracy also got into the act with props including stretch limo's. Apparently some 600 workers (some of whom were the original border guards) took one month constructing the set which included 2500 outsize styrofoam bricks and no less than twelve truckloads of equipment.

FREE WHEELING

Dominating the proceedings was a massive circular ferris-wheel type construction incorporating a 50ft circular screen. Inevitably the associated inflatable puppets – a teacher and a pit – were the largest ever built. The size of a six story building, they needed to be controlled by some 40 handlers, operating two 145 foot cranes. Synchronised searchlights, projection screens, lights and lasers enabled everyone to see the last few bricks being put into place and then at the climax of the show, see it torn down and reduced to a symbolic ruin. It is hardly surprising that the construction and production budget exceeded £9m.

The event was the launch pad for The Memorial Fund for Disaster Relief, an international charity set up by Leonard Cheshire VC. Once again records look set to be broken with the funds aiming to raise over £500m over the next five years. This target represents £5 for every life lost in major wars this century and the money raised will be to help relieve disasters and emergencies world-wide, both natural and man-made.

WALL TO WALL DESIGNER

Having fixed the date, the players and the charity, there only remained the appointment of the set design team. Here there was no contest. When it comes to staging mega-sized rock spectaculars, only one name matters, the UK design team Fisher Park. Over the past ten years, the partnership has been involved in such rock names and events as Pink Floyd, George Michael, Janet Jackson, Tina Turner, and The Rolling Stones. It was Fisher Park who designed the Nelson

The Rolling Stones 'Urban Jungle' European tour in 1990 shows what to expect at 'The Wall' concert.

Mandella 70th Birthday celebration, the staging of the Jean Michel Jarre concerts, and as a more lasting testament to their skills, the London Hippodrome Disco.

The engineering half of the Fisher Park team is Jonathan Park who met his partner Mark Fisher while they were both lecturing at the Architects Association. Jonathan sees his job as being unlike any other. "We have to create, in a very limited timescale, a disposable city using an art form unknown before the late 70's. It is our job to help create the Rock 'n' Roll tribal ritual which generates lavish levels of excitement and emotion. Unlike grand opera, rock audiences need a close encounter of the spectacular kind with their rock heroes".

Jonathan says he would not be in the business if he did not like rock music. He even attends concerts not designed by his company. It is hardly surprising that he sees very little of his family. "The job involves an enormous amount of travelling. You can't stage major events by remote control. You have to be there to kick backsides and hold the star's hand".

Certainly Fisher Park provides audiences with full value for money. Their sets rival those old Hollywood extravaganzas. "Everything is thrown into the event, including if it is necessary, the kitchen sink. Rock stars after all are legends in their own time" says Jonathan.

Time is one commodity always in short supply at Fisher Park and The Wall was no exception. Because of the time it took to reach agreement with all the local authorities, the design team of the multi-million pound event, were given just ten weeks to get their act together. But challenges are facts of working life at the company. "We spend all our working lives pushing the boundaries of technology, design and art. There is an insatiable public demand for ever larger scale rock concerts. It is very much a mutant art form and with new bands emerging all the time, we are always kept on our toes", says Jonathan.

MAKING RECORDS

Meanwhile, it's back to that Wall. The organisers are expecting that the world-wide sales of the video and albums of "The Wall – Live in Berlin", will establish some new records. But they will have an uphill task on their sales hands if they try to compete with the original Pink Floyd "The Wall". Issued in 1979, the record has gone on to sell well in excess of 19m copies. Similar records were set by "The Wall" video which has become one of the best selling music videos of all-time.

Now, you too can share "The Wall Live in Berlin". Maplin, you will not be surprised to learn, are lending a hand to assist the record-breaking movement. We have lined up a stunning array of "The Wall Live in Berlin" music and videos as prizes. Wait for it . . . We have six Videos, six Double Albums, six Double Cassettes and six Double CD's for you to win. Not surprisingly, we are having to line up something larger than a hat to take all your contest entries; so the first fully correct answers pulled out of the editor's wage packet will win (I wish my wage packet was that big – Ed). Post your entries to "The Wall Live in Berlin" Contest, The Editor, "Electronics – The Maplin Magazine", PO Box 3, Rayleigh, Essex, SS6 8LR. Or zapp over your entry by fax on (0702) 553935, don't forget to mark your fax THE WALL LIVE IN BERLIN CONTEST. Do let us know whether you want to win the video, album, cassette or CD version of the prize. But don't delay. In a un-typical seasonal gesture the editor has agreed to bring forward the contest closing date in order that you can enjoy the music over Christmas. So entries by 15th December 1990 please.

COMPETITION

THE WALL

Name the odd one out:
- (a) Roger Waters
- (b) Nick Mason
- (c) Mike Rutherford
- (d) David Gilmour

Who has not performed live in London this past summer?
- (a) Prince
- (b) U2
- (c) Madonna
- (d) The Rolling Stones

Which tracks are not featured on the hit album of Sinead O'Connor?
- (a) Mac Arthur Park
- (b) I Am What I Am
- (c) Can't Buy Me Love.
- (d) Nothing Compares 2 U

Which does not fit?
- (a) Dark Side of the Moon
- (b) The Wall
- (c) Wish You Were Here
- (d) Then There Were Three

Audio-Frequency Induction Loop Systems

by J.M. Woodgate B.Sc. (Eng.), C.Eng., M.I.E.E., M.A.E.S., F.Inst.S.C.E.

Part 2 – Small system practice and large system theory

Apology

For the first time since I began writing for Maplin, some printed errors, at least one of which was due to the original manuscript, unfortunately occurred in Part 1 of this series, and they are detailed at the end of this article under 'Corrigenda to Part 1'. I hope that we have found out why they occurred, and that there will not be any more.

Recap

For those who haven't read Part 1, here is what to do. First, order a copy of Issue 39 from Maplin, this minute. Done that? Good, now, while you are waiting for it, read on.

An audio frequency induction loop system (AFILS) consists of a source of signals (usually a microphone) driving an a.f. amplifier, which circulates a current in a loop of wire surrounding the area in which the system is to work. This creates an audio frequency magnetic field, which can be picked up by means of a suitable receiver (not a radio), so that the signals can be heard. In an 'AFILS for assisted hearing', the receiver is a hearing aid equipped (as most of them are) with a magnetic pick up coil ('Telecoil') and an 'M-T' (or 'M-MT-T') selector switch.

The Household Loop Amplifier System

This design is based partly on the TBA 810P audio amplifier i.c., using the Maplin p.c. board BR02C. If, by the time you read this, these have been replaced by a newer device and board, the new parts should still be suitable. In fact, with precautions, other devices such as the TDA2003 could also be used. It should be emphasised that the complete amplifier system has very high voltage gain (about 75dB) and current gain (about 150dB!), so that great care must be taken to avoid spurious feedback and unwanted coupling in the circuitry. It is essential to follow the detailed comments on this, as they appear at the appropriate places in the text. As is usual with equipment designs in my articles, there should be enough information for experienced constructors and even advanced learners, but full constructional details are not included and you should not tackle one of these projects unless you are confident of success. If you do hit trouble of a particularly fascinating kind, you can always write to the Kindly Editor. (Satisfaction will be striven for, but cannot be guaranteed!).

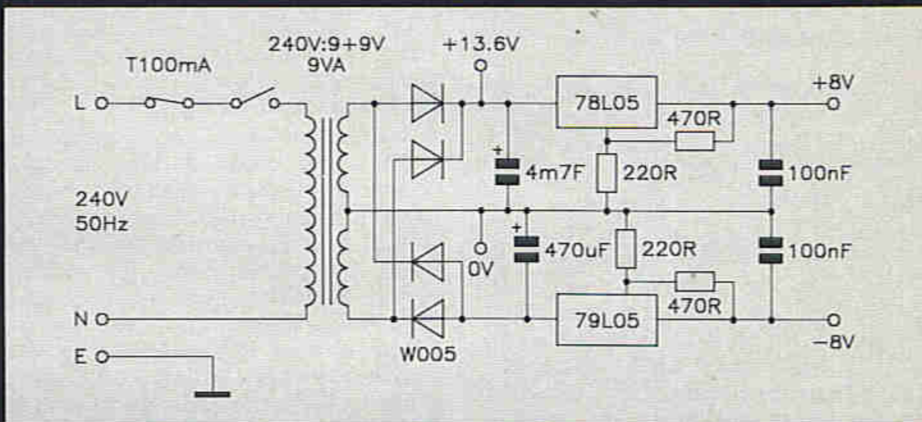


Figure 6. Power supply circuit for the Household Loop Amplifier. The +13.6V and 0V terminals should be as close to the terminals of the 4m7F (4700μF) capacitor as possible. The 0V connection must go ONLY to the final amplifier board.

The Final Amplifier

I am trying to get this term adopted for 'an audio amplifier which drives an output transducer' (loudspeaker, headphone, induction loop, etc.), because the present term 'power amplifier' suggests that power output is what is wanted. In fact, power is only needed to the extent that most output transducers do not work very well. For example, 99% of the power input to a loudspeaker is wasted as heat, and, as we shall see later, the optimum design of a *large* induction loop may involve making the resistance very low, so that very little actual power is dissipated in it.

Power Supply

In our present case, what we need is current, because the magnetic field strength produced by the loop is proportional to the current in the loop wire. The TBA810 can produce 7W in a 2Ω load with a 14.4V supply, corresponding to a current of $\sqrt{(7/2)}$ = 1.87A. Unfortunately, we can't get 14.4V from the recommended WB11M transformer, and the next larger YK28F gives too much, apart from not fitting into the box XY43W that I wanted to use. The WB11M gives 13.6V d.c. with no signal input, and this is sufficient for our purpose. We need also positive and negative regulated low-current supplies for the preamplifier, which can be obtained from a bootstrapped 78L05 and 79L05 devices, using a W005 bridge rectifier as two full wave rectifiers.

To prevent loss of regulation when the rectifier output voltage is dragged down by the final amplifier current on programme peaks, the regulated voltages are set at ±8V. The transformer can be protected against damage due to short-circuits, by means of a primary fuse, provided it is a 100mA 'anti-surge' type (UJ92A). Don't be tempted to take risks by fitting a higher rating fuse or no fuse at all, nor in the associated area of safety earthing. The amplifier should have a 3-core mains lead, with the metal case connected to the safety earth conductor. I always cover up all the bare mains connections inside the box as well, so that fingers or tools cannot touch the live parts even with the lid off! It is noteworthy that, in this respect, Class I equipment (earthed) is less safe than Class II (double insulated), because you are much more likely to touch live mains and earth simultaneously on a Class I unit, than you are to touch both poles of the mains simultaneously on a Class II unit. In any

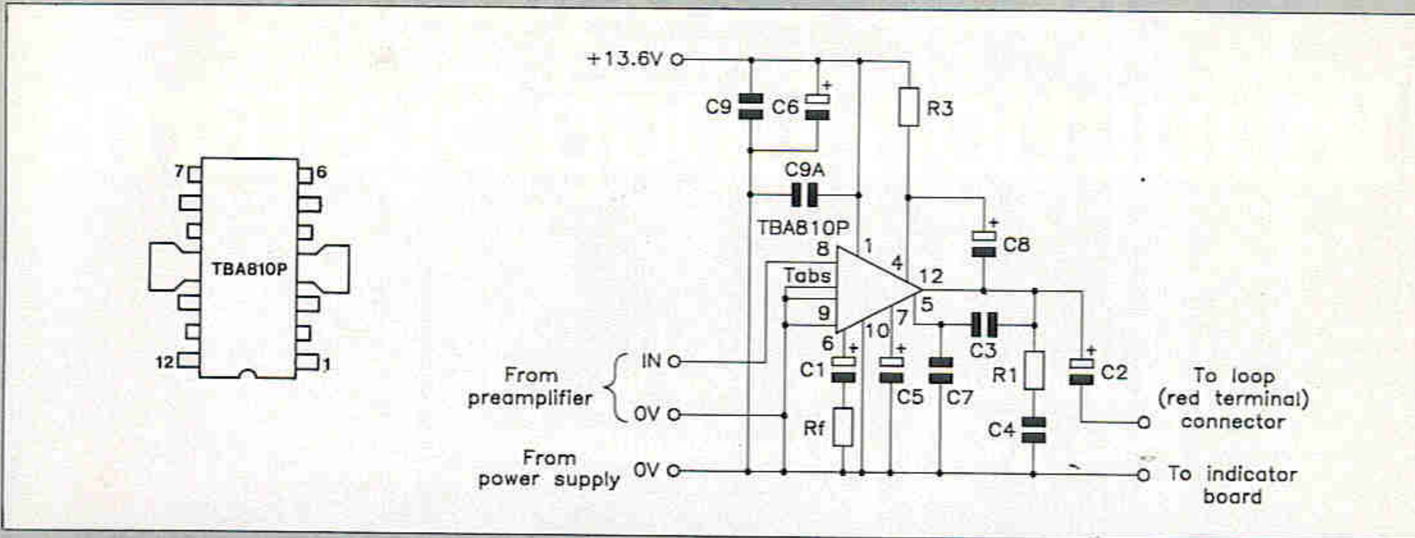


Figure 7. Final amplifier circuit.

case, most Class II electronic equipment has to have all the mains terminals covered up in order to meet safety standards. The complete circuit of the power supply is given in Figure 6.

The TBA810 Board

This can be made up exactly as described in the catalogue, with the addition of a 100nF ceramic capacitor (YR75S) with short wires connected directly from pin 1 of the TBA810 to the common rail of the copper side of the board. This is because improvements in the chip manufacture have tended to lead to r.f. oscillation at some output levels when the chip is hot. The 100nF capacitor C9 on the board has too much lead inductance to prevent this. Note that the board won't work by itself, without a temporary d.c. path (e.g. a 10kΩ resistor) across the input terminals, because the volume control shown on the circuit diagram in the catalogue is on the pre-amplifier board in this design. The complete final amplifier circuit is shown in Figure 7.

The Pre-amplifier

In order to pick up the sound of the television or radio without too much 'bathroom' reverberation (so try it, and listen to what I mean!), the microphone should be very close to the loudspeaker. Here, the sound level is high, so the amplifier gain does not need to be very great. However, it is also very desirable for the same system to pick up and relay the sound of the telephone, doorbell, cat at the back door, burglar under the bed, etc., which are not so loud. But we can't have the hearing aid user (hereinafter referred to as 'Grandma') running backwards and forwards to adjust the volume control all the time. We need a volume control to set the current in the loop so that it suits the size of the particular loop being used, and the position that Grandma sets the volume control of her hearing aid, but we then want *Automatic Gain Control* (A.G.C.) to compensate for the different loudness levels of the wanted sounds.

A.G.C. Techniques

There are now several techniques available for audio A.G.C. including programmable-gain op-amps, voltage-controlled amplifier (VCA) chips, the FET used as a voltage-controlled resistor and, simple but a little crude, the ordinary bipolar transistor used as a current-controlled resistor. This last was the technique used in millions of cassette recorders (before dedicated IC's were available), and is the one I have chosen. It is simpler and cheaper than all the others to execute, and the only penalty is some distortion, most of which is the relatively harmless second harmonic kind, and which can be kept to a low level by optimum design.

The action of the circuit can be understood with reference to Figure 8. There is no D.C. collector voltage on TR1, and for small signals it behaves as a resistor whose value is determined by the base current flowing through R2. It thus acts as a potential divider with R1. In order to keep distortion low, the signal level at the collector should not exceed about 30mV r.m.s. On the other hand, if the signal level is set very low to minimise distortion, the signal-to-noise ratio is poor. This signal is amplified in A1 and rectified to produce a smoothed D.C. voltage across C1. This voltage then produces the current through

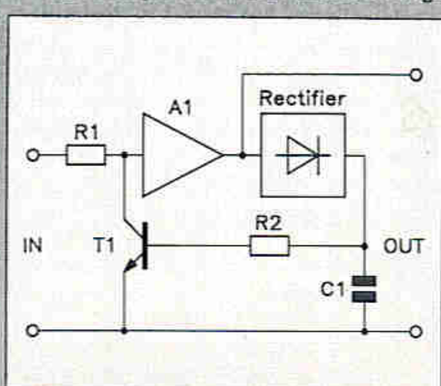


Figure 8. Basic audio A.G.C. circuit using a bipolar transistor as a current-controlled resistor.

R2. An increase in input signal level thus produces an increased current through R2, which turns on the transistor more, so that the signal fed to A1 does not increase in level as much as the input signal did. It must increase somewhat of course, in order to produce the higher current in R2, so that we may expect the circuit to give an output level which rises with input level but by a far less extent. The larger the value of R2, the more output voltage change occurs, but if R2 is too low, the distortion rises.

There is an optimum value for R2, which gives minimum distortion for a particular input level, and an acceptable, indeed desirable, change in output level. The reason that this is desirable is that Grandma wants, to a certain extent, to know whether the original sound is loud or soft, and if they are all converted to the same loudness by the A.G.C., then the subjective effect is not pleasant. Since C1 is charged by the rectifier circuit, which can have a low output impedance, but discharges through R2 and the base-emitter resistance of TR1, which can be several thousand ohms, the A.G.C. has a fast 'attack time', i.e. it responds quickly to a sudden loud sound, and a slow 'fall time', i.e. the gain increases slowly after a loud sound stops. This prevents undesirable 'pumping' effects in the reproduced sound, and is a GOOD THING.

The Gain Budget

The final amplifier board needs about 60mV input for full output, and the electret microphones that are most convenient for use in this system produce about 3mV at 90dB sound pressure level. So it would appear that the preamplifier needs a gain in the region of 20 times. However, the signal rectifier needs much more input than 60mV, in order to overcome the diode voltage drops. At the same time, we must not have more than 30mV or so at the transistor collector for any likely value of sound pressure level at the microphone. This can be achieved by a gain of 13.5 times before the A.G.C. transistor, and a gain of

122 after it. The signal at the second stage output then needs to be attenuated about 7 times in order to feed the final amplifier correctly.

The gain of the amplifier for low level signals is very high, so it is essential to minimise the bandwidth to that required for audio signals. Therefore, capacitors are fitted across the feedback resistors of both the TL072 amplifiers, giving each stage a −3dB bandwidth of just over 10kHz. It is also essential that, while direct connections to the power supply are made for the ±8V supplies, the 0V connection is made to the signal input terminal of the TBA810 board and *not* to the power supply! An error here will cause violent distortion and instability.

Input Circuits and Connectors

It is necessary to provide a very smooth and well-decoupled supply voltage for the preamplifier in the electret microphone, and it is also helpful to provide inputs for a self-contained microphone (with an internal battery) and for direct injection (yes, just like DI) of line-level signals from a radio or cassette recorder, thus eliminating the room reverberation problem mentioned earlier, and providing a much higher quality signal. These objectives can be achieved by providing a 3.5mm mono jack for an electret microphone capsule such as FS43W (very inexpensive but you provide your own case and grille), and a 5-pin 180° DIN socket for a self-contained microphone or line-level source.

Alternatively, the microphone connections on the DIN socket can be transferred to a 6.3mm (¼in.) jack for one of the tie-clip microphones LB69A etc., and the 3.5mm jack with the D.C. voltage feed eliminated. The microphone can be fixed near the loudspeaker of the television or radio by means of a clip, or the Velcromounts (FE45Y) are very effective. The complete circuit of the preamplifier and its input connectors is shown in Figure 9, and this, regarded as an audio A.G.C. module or circuit function on its own, can be used in hundreds of other applications. In the interests of stability and low hum and distortion, no part of the circuit (excluding the green/yellow wire of the mains lead), should be connected to the case, except the 0V wire at one of the input sockets (most conveniently the 3.5mm jack if JK02C is used).

The Indicator Board

It is highly desirable to be able to monitor the operation of the system independently of Grandma's hearing aid, which may be switched off or have a flat battery at the most inconvenient times. The first need is for a power supply indicator, but it is also very helpful to monitor the actual current through the loop (in case it has gone open-circuit), as well as the sound quality of the amplifier output signal.

For the current monitor, a transistor detector working on the voltage drop across a very low-value resistor in series with the loop is convenient and inexpensive. If this is used with a low-current LED as an indicator, it is possible, by connecting a capacitor across the LED and its current-limiting resistor, to give the indicator 'peak programme meter' (PPM) characteristics, i.e. like the A.G.C. circuit it responds quickly to signal peaks and the indication dies away slowly afterwards. This sort of indication is much more useful, and less annoying, than a madly-flashing fast-response indicator would be. The current-sensing resistor must have a low value compared with the loop impedance, because it takes up output capability ('power'), and a value of 0.2Ω, giving about 0.4V on current peaks, is suitable. This is most conveniently made from five 1Ω (M1R) resistors in parallel. The transistor detector is used in the common-base mode and runs, like the low-current LED used for the power indicator, from the −8V supply. The circuit of the indicator board is shown in Figure 10. The germanium diode provides temperature compensation for the base bias voltage of the detector transistor, and the preset resistor is adjusted when the system has been set up. When the amplifier volume control has been adjusted so that Grandma is satisfied with the sound level that the system produces in her hearing aid, the preset resistor can be adjusted so that the LED just flashes on peaks of signal, rather than being on all the time.

It should not be necessary for Grandma to adjust the volume control of her hearing aid (from the position that she normally sets it to) when she uses the loop system. To have to do this means that the loop system is not set up properly, and is very inconvenient for Grandma. Sound quality is easily monitored by connecting headphones across the 0.2Ω resistor. A series resistor is necessary, otherwise acoustic feedback may well occur between the headphones and the system microphone, which could damage your hearing if you are unlucky. The circuit diagram shows the jack (FK03D) wired such that the two earphones of a lightweight stereo headphone (YM38R) are connected in series, and the 56Ω resistor is suitable for these. You may have to increase this value in some cases. The jack socket body must be insulated from the metal amplifier case by means of thin plastic washers, otherwise there is a spurious earth path which may cause distortion, and only one earphone will work!

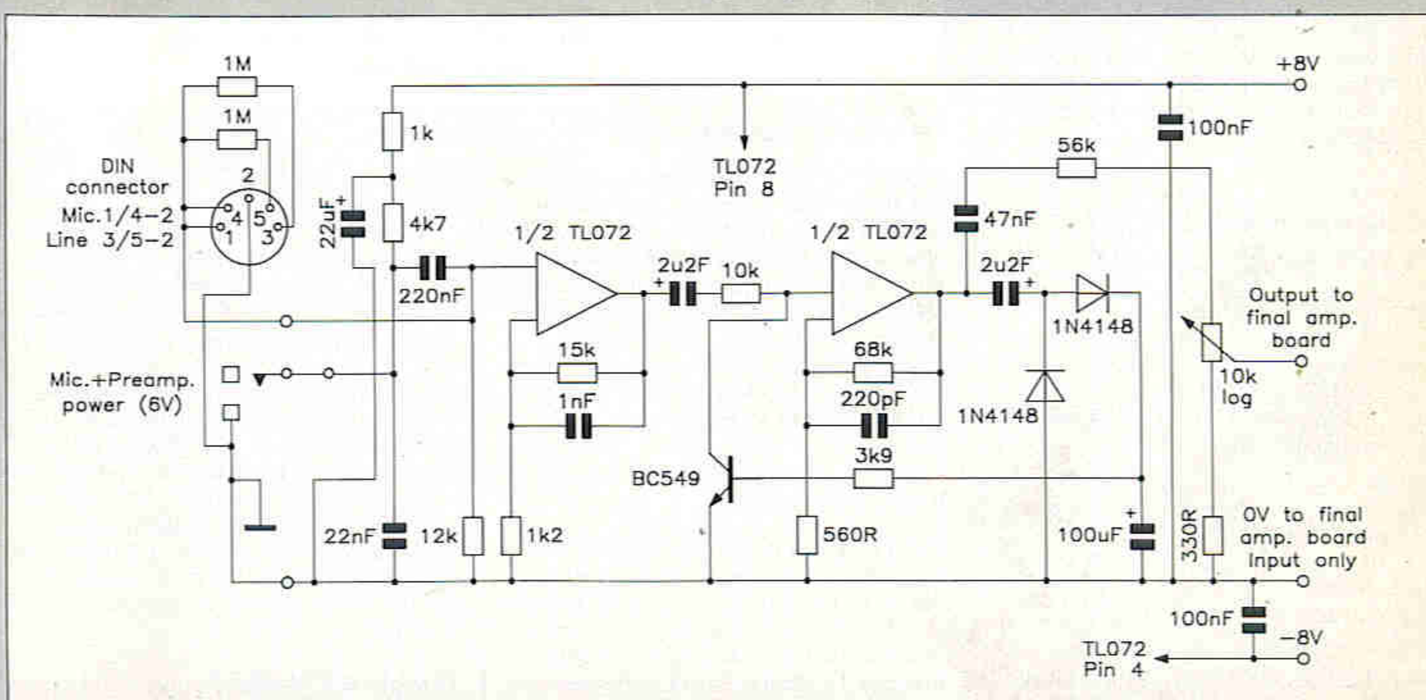


Figure 9. Pre-amplifier with a.g.c., and input circuit connections. The 22nF capacitor across the input is to prevent r.f. pick-up. The 330Ω resistor in series with the volume control preserves a (weak) signal even if the control is set to minimum by mistake.

Connecting to the Loop

Since the loop is made of one stranded wire (7/0.2 or 10/0.1), the most convenient connector is BW72P, 'Lever term. 2-way', which can be mounted on the back of the box. The red terminal goes to the final amplifier output, while the black terminal goes to the 0.2Ω resistor on the indicator board, and the other end of this resistor goes to the earthy output terminal of the final amplifier. Now these wires carry enough current to produce magnetic feedback to the early stages of the amplifier, so they should be twisted together tightly so that their magnetic fields cancel. The loop wire itself can be run around the floor, providing no-one could trip over it, or at ceiling level. The resistance should be no less than 2Ω; it can be greater if the loop is less than about 20m in perimeter.

Measured Performance

The prototype amplifier was measured with a test load consisting of a 2Ω 7W resistor in series with a 54μH air-cored coil (30 turns of thick wire wound on a large coffee jar!), and the output was measured across the internal 0.2Ω current-sensing resistor, thus accurately representing the output current. The inductance and resistance represent a loop of 7/0.2mm equipment wire, of about 25m perimeter (PA45Y, if you don't want at least two joints in the run!), which is ambitious for this amplifier, but the measured results are quite encouraging. For a smaller room, a 10m loop of 10/0.1mm wire (BL46A) is suitable. Remember that wire thicker than 7/0.2 will restrict the frequency response, as well as driving the TBA810 into short-circuit protection mode, and don't coil up any spare loop wire, as this will obviously increase the inductance a great deal! The same applies if you try to use more than one turn. The results are shown below, and practical tests, with volunteer 'Grandmas', show that the system performs quite well (better than some commercially available systems).

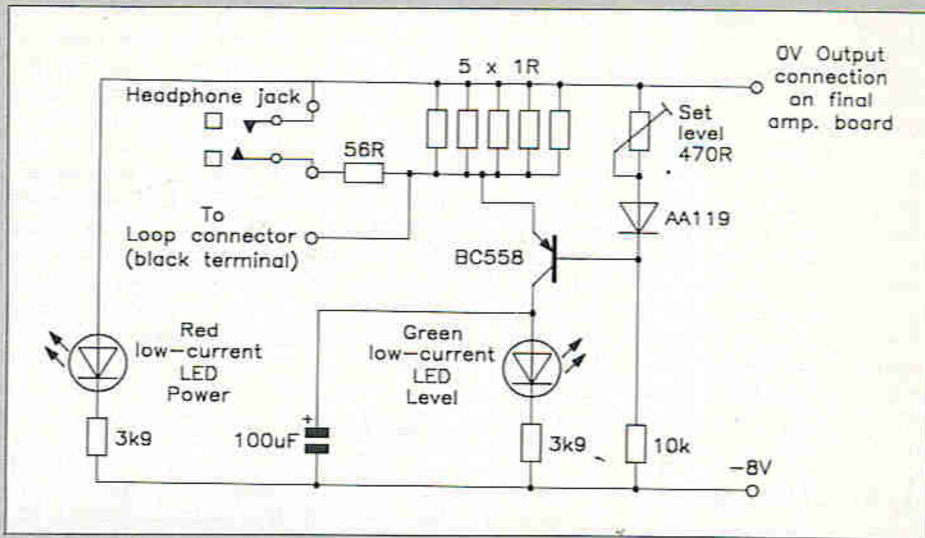


Figure 10. Indicator board circuit. The 'sleeve' connection and body of the headphone jack should not be connected to the case. Insulating washers should be fitted.

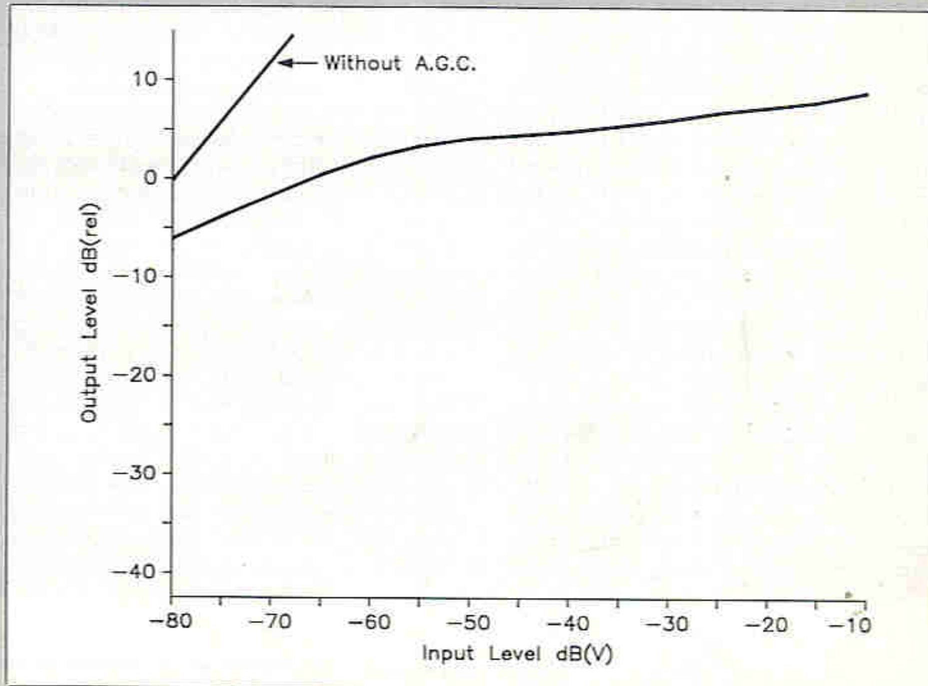


Figure 11a. A.G.C. control characteristic of the pre-amplifier.

Measurements on the Prototype Household Loop Amplifier

Conditions
Mains supply: 250V 50Hz.
Load: 2Ω in series with 54μH. Signals to the microphone input.

Results
Minimum input voltage for 'just clipping' at output: 350μV.
Output current at clipping: 1.25A continuous (1kHz).
Output current at 10% THD: 1.67A continuous (1kHz).

Note: Because the power supply voltage drops from 13.6V with no signal to 10.7V at clipping, the output current on programme peaks is considerably higher than the above figures suggest; approximately 2A can be obtained. This gives a magnetic field strength of about $0.3Am^{-1}$ at the centre of a loop of 25m perimeter, which is capable of giving satisfactory results in the absence of magnetic interference (e.g. from a vacuum cleaner motor). For small loops, field strengths in excess of the values given in BS6083-4 can be obtained (see Part 1 for a discussion about these).

A.G.C. Action and Harmonic Distortion
Refer to Figure 11.

For this test, the volume control was set to 20dB attenuation, to avoid overloading the final amplifier. The A.G.C. range is about 50dB for a 7dB change in output level, while for a 10dB change (approximately a 2:1 change in subjective loudness for a person not suffering from the hearing defect known as 'recruitment'), the input range is nearly 60dB.

Frequency Response
Refer to Figure 12.

For this, and subsequent tests, the input signal level was set to about 8mV, so that the A.G.C. was in operation (and thus producing distortion), and the volume control set to give an output current of 1A.

Noise: Measured by the CCIR-weighted quasi-peak method (not CCIR-ARM), the signal-to-noise ratio was 47.5dB. Not CD quality but not too bad. The A-weighted noise measured 53.4dB, but this is misleading because the magnetic coupling to the hearing aid, and the aid frequency response itself, both produce treble boost, so the lower value is probably more realistic.

Distortion: At the fixed input level of 8mV, the total harmonic distortion results were:
40Hz 3.0% 100Hz 1.5% 315Hz 0.5%
1kHz 0.8% 6.3kHz 0.9% 10kHz 1.8%

Except for the 315Hz measurement, these values include noise up to 22kHz. The distortion at 1kHz was investigated further with a wave-analyser, with the following results:

2nd 0.7% 3rd 0.3% 4th 0.04%
5th 0.1% 6th 0.03%

It can be seen that the distortion is mostly second harmonic, which does not sound so bad as odd-order.

Larger Loop Systems

Traditionally, loop systems in public buildings such as churches and halls have used one large loop, running more or less around the perimeter of the floor area. However, this technique produces rather more overspill than is usually desirable. For example, a passing Grandma may be able to hear what is going on inside the building quite well, and it could be the Council discussing next year's Community Charge! For many churches, of course, which stand in their own grounds, this is not a great problem.

We have already seen that the simple voltage-drive technique, with an amplifier designed for a low-impedance load, can be used for quite large loops, provided that the loop wire is no thicker than 7/0.2, (or 1/0.6 as the real top limit). However, amplifiers for sound reinforcement systems often have a 100V (or some lower voltage) line output, and it may well be desirable to use the same type of amplifier for the loop system as is used for normal sound reinforcement. Note 'the same type'; using one amplifier for both purposes is full of snags and is best not attempted unless you are very confident.

Now, unless of very high output, these line amplifiers require quite a high load impedance. For example, a 50W 100V line amplifier requires a load impedance of $100^2/50 = 200\Omega$. We could get this from a loop of many turns of very thin wire, but it would be very prone to breaking. It is clearly necessary to use a multi-turn loop of reasonably thick wire, but this is bound to have far more inductance than resistance at 5kHz, so how can you get a flat frequency response? Well, you could just turn up the treble boost on the amplifier, hoping that it won't overload and that the frequency response correction is somewhere near right. But I hope you won't, because a simple equaliser, consisting of a capacitor and resistor in parallel, allows the required frequency response to be achieved and the loop to be driven directly from the 100V line amplifier without an expensive 'matching transformer'. The equaliser is inserted at by far the most convenient place, too. Not inside the amplifier, nor in a screening box between the preamp or mixer and the final amplifier, but between the amplifier and the loop itself, using an ordinary junction box. The author (and the industry) is indebted to Mr Frank Poperwell for suggesting this technique, but any errors in the mathematical analysis and its conclusions are entirely the author's responsibility.

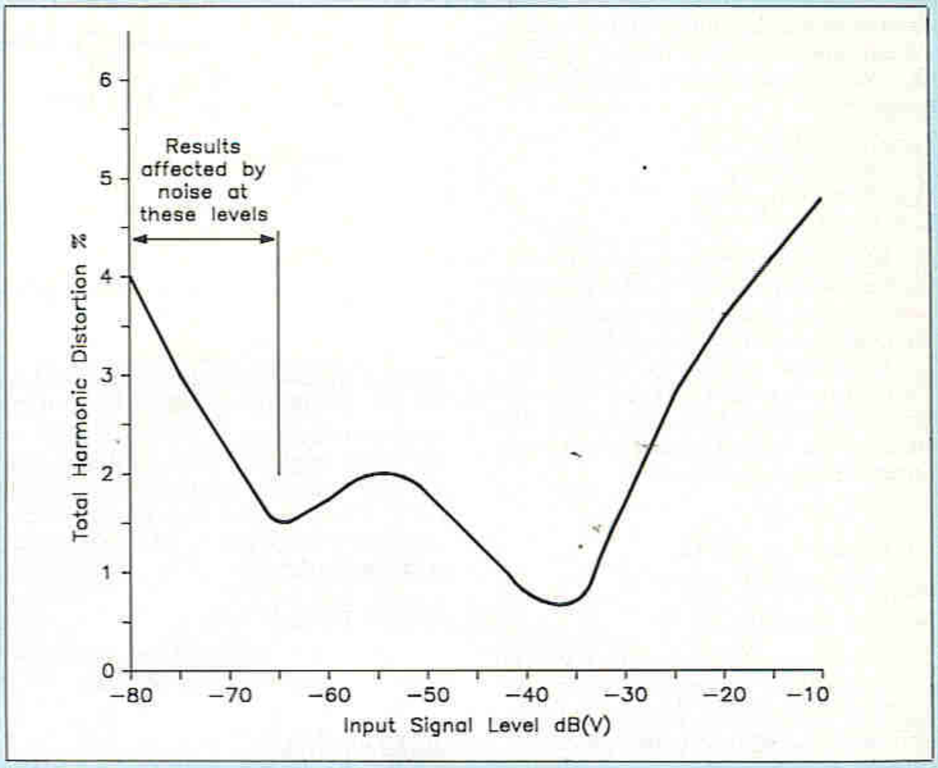


Figure 11b. Total harmonic distortion as a function of input signal level for the pre-amplifier. The distortion is mostly 2nd harmonic, which is relatively harmless in terms of sound quality.

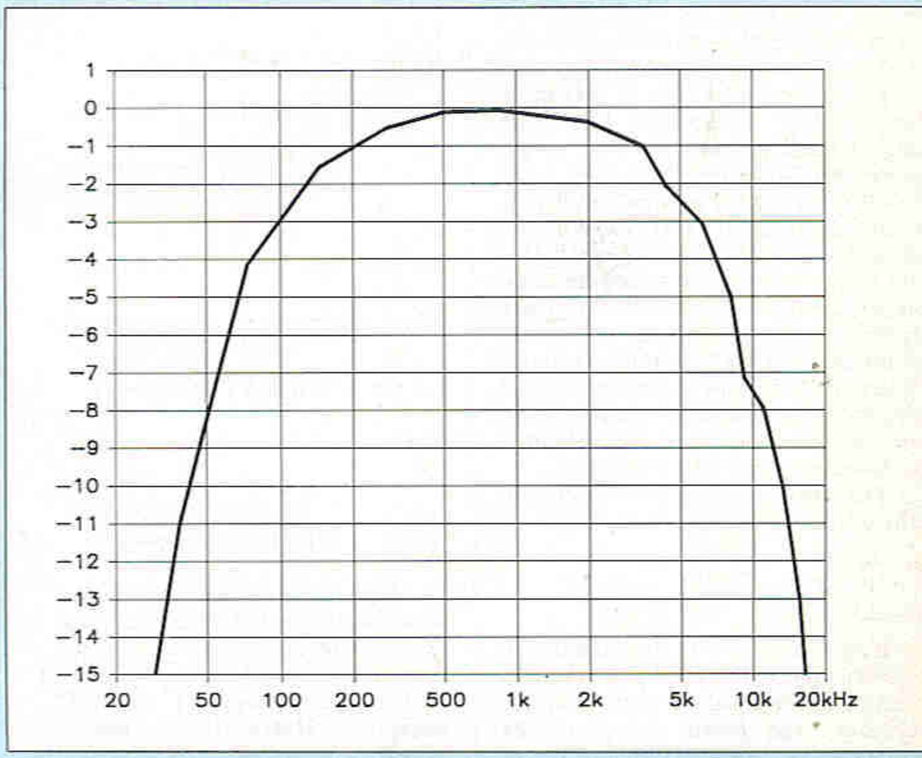


Figure 12. Frequency response of the output current of the Household Loop Amplifier. Load 2Ω+ 54μH. 0dB=1A.

The relevant circuit is shown in Figure 13. One way of looking at the problem is to consider that we could 'get rid of' the unwanted inductance by resonating it with a series capacitor. By itself, this is disastrous: you get an enormous peak in the frequency response at the resonant frequency, and a very low load impedance for the amplifier, which will object strongly! We have to damp the resonance, so that its good effects are spread over a larger frequency range and the very low input impedance raised. Overall, we are looking for a response without a peak ('maximally flat'), which is −3dB at 5kHz. This 'response' is the current produced by a constant input voltage, which is what the amplifier produces, and the ratio of current to voltage is the input admittance (reciprocal of impedance) of the equaliser plus loop. The complete mathematical analysis is given in the appendix, and will probably be included in the British Standard Code of Practice for AFILS. The results are simple equations for the circuit values, in terms of the loop inductance (which is fixed by the dimensions and the number of turns only) and the upper −3dB frequency f_c, here taken as the optimum value of 5kHz. The equations are:

$R_1 = 2\pi f_c L_1/2.652$
$C1 = 4.121/4\pi^2 f_c^2 L_1$
$R_2 + R_3 = 2\pi f_c L_1/3.751$
$Z_1 = 1.29\pi f_c L_1$

We can use these equations as follows. Let I be the current we would need in a 1-turn loop of size a metres to obtain the required maximum magnetic field strength H_o. Then for an n-turn loop, we need I/n. The inductance of the 1-turn loop is 8a μH, so that of the n-turn loop is $L_1 = 8n^2a\ \mu H$ (nearly). The voltage required to drive a current I/n through Z_1 is thus:

$V = Z_1 \times I/n$

$= 0.162na\ I$, if $f_c = 5kHz$

But $I = \pi aH_o/2\sqrt{2}$ (see Part 1)

So $V = 0.178na^2H_o$

We have to choose n so that V is as close to 100V (or whatever line voltage the amplifier is designed for) as possible, without exceeding it. For example, if a = 15m and $H_o = 0.56Am^{-1}$,

$V = 22.4n$,

so that n = 4 is optimum. For a lower field strength, *more* turns are better: quite the reverse of what you would expect. For $0.35Am^{-1}$, 7 turns are required.

For the 4-turn loop, $Z_1 = 39\Omega$ and a 250W amplifier is necessary, but if the magnetic interference level is low enough, the reduced field strength is likely to be perfectly satisfactory, and the 7-turn loop has an impedance of 119Ω, requiring an amplifier of just 84W. It has to be said that good results have been reported for loops equalised by this method even when using amplifiers of lower power in acoustically and magnetically 'quiet' conditions, but it is likely that 50W is the minimum practical power rating. Amplifier overload causes all sorts of problems, including potential interference, and must be avoided.

The equaliser component values are easily calculated. For the 7-turn loop:

$R_1 = 69.7\Omega$
$R_2 = 49.3\Omega$
$C_1 = 0.71\mu F$

If the resistance of the loop is less than the above value of R_2, a fixed resistor R_3 is necessary in the equaliser to make up the difference. The power ratings of the resistors can be calculated from the current flowing, remembering that the average r.m.s. current is only about one-fifth of the maximum current, i.e. 0.71A for a 100V/84W amplifier. Thus, R_1 should be a 2W resistor and R_3 need be no more than 1.5W. However, they may burn up if the loop is short-circuited, so take precautions in mounting them.

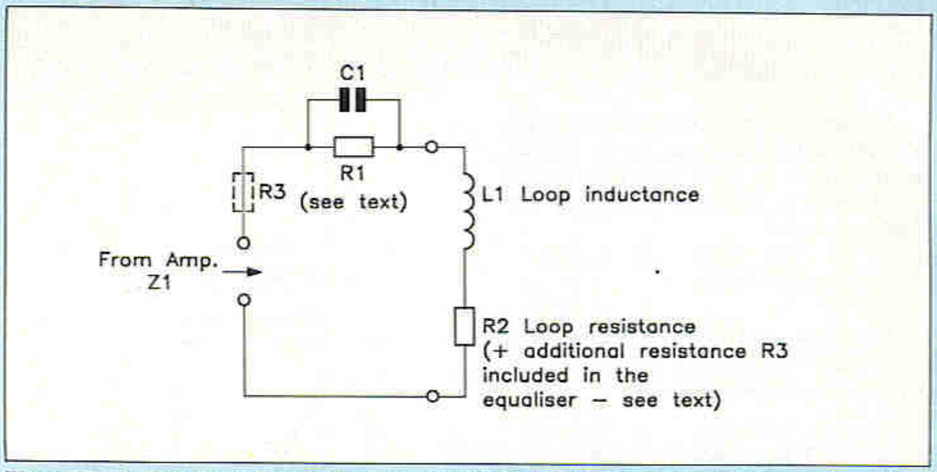


Figure 13. High-level equaliser for feeding a multi-turn loop from a sound reinforcement amplifier with 100V (or other) line output. Z1 is the input impedance of the equalised loop.

Next Time

In Part 3 we really will look at current drive, and some other ideas as well.

Appendix

High-Level Equaliser

Referring to Figure 13

$1/Z_1 = I/V =$
$1/\{(R_2+sL) + R_1/sC(R_1+1/sC)\}$

Let $CR_1 = T_1$ and $L/R_2 = T_2$. Then:

$$1/Z_1 = \frac{R_2(1+sT_1)}{R_2(1+sT_1)(1+sT_2)+R_1}$$

Let $R_1 = \alpha R_2$ and $T_2 = \beta T_1$. Then:

$$1/Z_1 = \frac{1+sT_1}{(1+\alpha)+s(1+\beta)T_1+s^2\beta^2T_1^2}$$

Let $\gamma = 1/1+\alpha$ and $T = \sqrt{(\beta\gamma)}T_1$.

Then:

$$1/Z_1 = \frac{\gamma\{1+sT/\sqrt{(\beta\gamma)}\}}{1+sT\gamma(1+\beta)/\sqrt{(\beta\gamma)}+s^2T^2}$$

For a maximally flat response, the squares of the moduli of numerator and denominator, after substituting $s = j\omega$, must be Butterworth polynomials, i.e. of the forms, in this case, $1+T^2$ and $1+T^4$ respectively. Hence $\sqrt{(\beta\gamma)} = 1$ and $\gamma(1+\beta)/\sqrt{(\beta\gamma)} = \sqrt{2}$, i.e. $\beta = 1+\sqrt{2}$, $\gamma = \sqrt{2}-1$ and $\alpha = \sqrt{2}$.
Also $T_1 = T/\sqrt{(\beta\gamma)} = T$.

At the upper band-limit frequency ω_c,

$(1+\omega_c^2T^2)/(1+\omega_c^4T^4) = 1/2$

Therefore, $\omega_c^2T^2 = 1+\sqrt{2}$

so that
$T = T_1 = \sqrt{(1+\sqrt{2})}/\omega_c = 1.554/\omega_c$

and $T_1 = (1+\sqrt{2})\sqrt{(1+\sqrt{2})}/\omega_c = 3.751/\omega_c$

Thus, $R_2 = \omega_c L/(1+\sqrt{2})\sqrt{(1+\sqrt{2})}$
$= \omega_c L/3.751$

$R_1 = \sqrt{2}\omega_c L/(1+\sqrt{2})\sqrt{(1+\sqrt{2})}$
$= \omega_c L/2.652 = \sqrt{2}R_2$

and $C = \{\sqrt{(1+\sqrt{2})}/\omega_c\}$

$\{(1+\sqrt{2})\sqrt{(1+\sqrt{2})}/\sqrt{2}\omega_c L\}$
$= (1+\sqrt{2})^2/\sqrt{2}\omega_c^2L = 4.121/\omega_c^2L$

Corrigenda to Part 1

Second line on page 37 column one, sentence should read: It can be shown that this technique reduces the field strength outside the area of the loops quite significantly, but it also produces a 'comb filter' frequency response for the field strength inside the loops, with a null every 100Hz (Figure 4).

The first equation on page 37 should read $H_o = I/2r$. The second equation at bottom of same (first) column should have been $H_o = 2I\sqrt{(a^2+b^2)}/\pi ab$. Equation at end of first paragraph of second column should be $H_o = 2\sqrt{2}/\pi a$.

On page 39 column 1, the first equation after the heading 'Inductance of the Loop' should be $L = (\mu_o/\pi)\ \{b\ \ln(2a/\emptyset) + a\ \ln(2b/\emptyset)\}$. The equation at the bottom of column 2 on the same page should read $a = 2\sqrt{2}I/\pi H_o s$. Finally there are several mathematical symbols in the text of column 3, page 39 which are enclosed in single quotation marks but shouldn't be.

AMPLIFIER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film

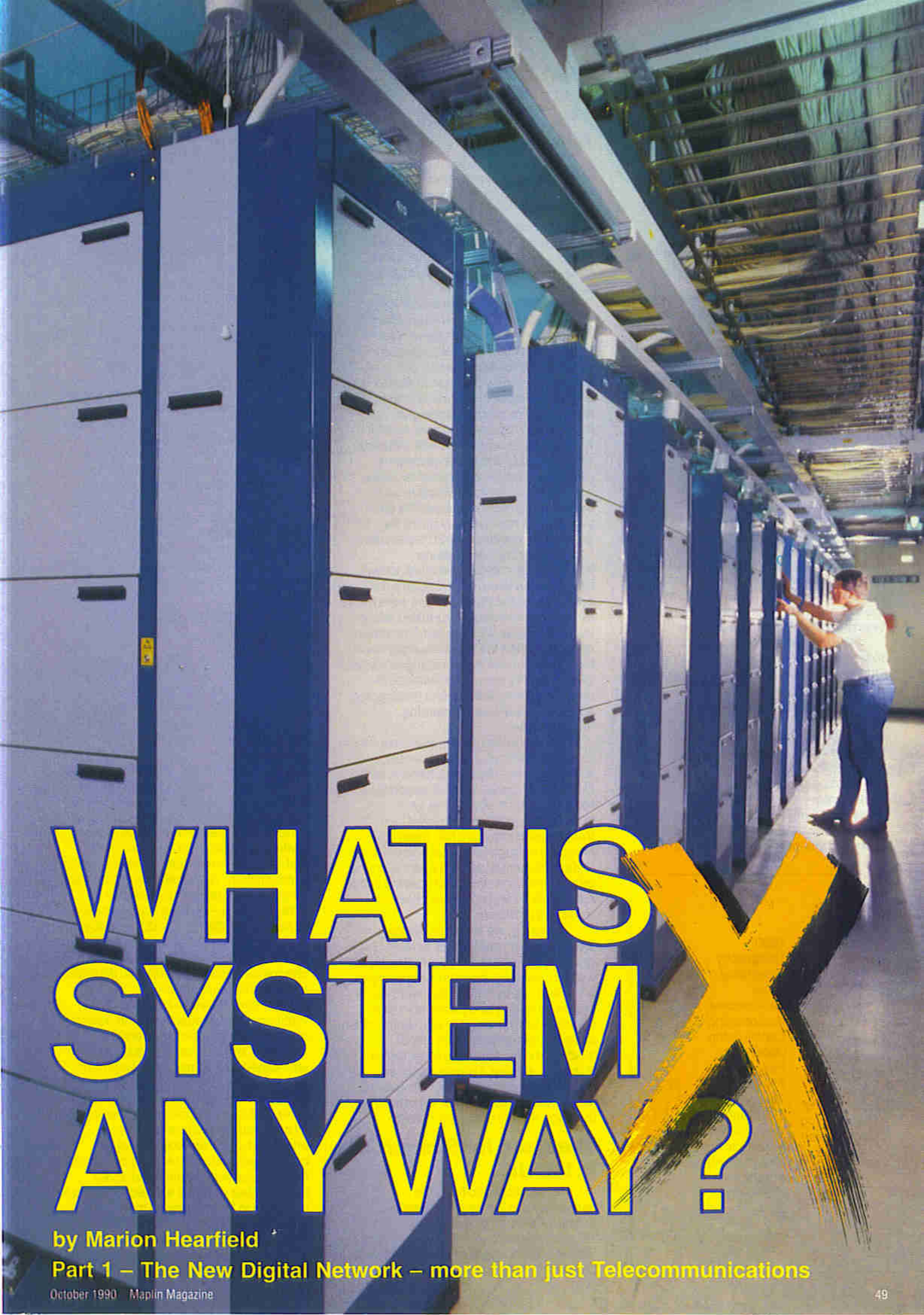
R1	1R	1	(M1R)
R3	100R	1	(M100R)
Rf	68R	1	(M68R)
R2	470k Pot Log	1	(FW27E)

CAPACITORS

C1	100μF 10V Axial Electrolytic	1	(FB48C)
C2	1000μF 16V Axial Electrolytic	1	(FB82D)
C3	1nF Poly Layer	1	(WW22Y)
C4,9	100nF Polyester	2	(BX76H)
C5,6,8	100μF 35V Axial Electrolytic	3	(FB49D)
C7	4n7F Poly Layer	1	(WW26D)
C9A	100nF Ceramic (see text)	1	(BX03D)

MISCELLANEOUS

	Printed circuit board	1	(BR02C)



WHAT IS SYSTEM X ANYWAY?

by Marion Hearfield

Part 1 – The New Digital Network – more than just Telecommunications

Some of us have had to reprint our stationery because BT decided to change our telephone number. We have also been able to click our telephone tone/pulse switches to 'tone', resulting in the delightful musical peeps which now accompany our keying-out, and a much faster connection to our called number. Why? Because we have at last been hauled into the new communications age by the installation of a local digital exchange.

This article concentrates on why a new network was needed, how it has been implemented, and what services it will offer you. The next will describe the network's architecture and operation.

The photograph which accompanies this article shows a typical exchange installation. Like most modern technology, it is difficult to tell from the outside what it is. Only the tidy bundles of cable, lying neatly in their overhead baskets, give a small clue that an awful lot of something is being sent somewhere else.

The idea of changing a country's telephone system from analogue to digital is a daunting one. The huge investment in nationwide equipment means that it must be operational for a large number of years in order to justify the initial cost. Some of the equipment being replaced today has been in use for forty years or more.

The major constraint of the old systems is their analogue operation. During the decades when speech was the only signal which needed to be transmitted, this was not a problem. Modern communications however need a worldwide, digital network which can integrate voice, data, text and video transmissions.

Just look at the list of services which have become available in the last ten years, and compare it with the original telephone service of 100 years ago, and even with that of just ten years ago.

1890	1930	1980	1990
Telegraphy	Telegraphy	Telegraphy	Telegraphy
Telephony	Telephony	Telephony	Telephony
	Telex	Telex	Telex
		Radiophone	Radiophone
		Radiopaging	Radiopaging
		Confravision	Confravision
		Datel	Datel
		Fax	Fax
			Telecommand
			Telemetry
			Viewdata
			Viewphone
			Electronic funds transfer
			Electronic office
			Prestel
			X-Stream

The International Marketplace

As the technology improved and allowed the non-voice features to become available, business users wanted them. National PTT's wanted to offer them, and there was really no reason why they should not. Customers were happy, PTT's and equipment manufacturers made more money (usually); business flourished.

But only within each country. In western Europe, with fourteen different PTT's, this became a problem. Well-established international analogue telephone links could pass a call effortlessly from Inverness to Rome, but any other kind of transmission came up against national boundaries and international inconsistencies. And for multi-national companies with offices in all major cities, the public telephone networks offered very limited facilities.

World demand for voice-based systems is growing by about 4% each year; but for non-voice systems the growth rate is more than 20% per year.

The process of agreeing international standards is much slower than the time taken to introduce a new product, and the new services were made available in individual countries before such standards could be set. The natural result has been that terminal equipment designed to meet the signalling standards of one country cannot be installed in another country without first undergoing expensive and time-consuming modifications.

But manufacturers in every country in the world had had, out of necessity, started to make digital equipment. In the UK, the first digital network switch was called System X. Achieving an internationally accepted digital transmission standard has been a painful progress through technical and political minefields, and System X was the first to be designed specifically to operate within agreed ISDN and common-channel signalling protocols.

However, we are all on our way towards the internationally agreed ISDN (Integrated Services Digital Network). As well as providing the international digital links required by business, it will offer improved quality and facilities – faster connection times, higher fax transmission speeds, high definition full colour electronic mail, home banking, home shopping, on-line database access, dial-up video conferencing, centralised meter-reading for utilities. Whether you think you need it or not, it's coming.

The Need for System X in the UK

The political and financial background to the development of System X is more complicated than any TV soap opera script. System X resulted from studies carried out in the late 1960's to determine the future requirements for the UK's telecommunications network. Upgrading the analogue network had been carried out piecemeal, and a plan was needed which would take the UK's public telephone network into the next century.

It was clear even then that the emerging microelectronics technology would be able to provide systems with greater reliability and at lower costs. A family of exchanges was envisaged which would incorporate digital switching and transmission, stored-program control and common-channel switching.

By the time development had started on System X, the UK had 25 million telephones connected by 16 million telephone lines to over 6,000 local exchanges. (By 1988 this had increased to 30 million telephones and 20 million connections.) The local exchanges were grouped into 400 switching centres, which were connected by nearly one million circuits to the main network, which itself had 300,000 circuits. Replacing the switching for this lot would be a bit more complicated than re-wiring the kitchen (regular readers will also be relieved to know that ours is now finished). Doing it whilst still being able to plug in the kettle at any socket would be even harder.

The Development of System X

The actual development of System X started in the mid-1970's and was carried out jointly by BT, GEC, Plessey and, for a short time, STC. It was described by one of BT's departmental directors at the time as "a challenge unmatched in the previous history of UK telecommunications". How right he was.

The fundamental principle was to design basic hardware modules which, given different software, could be joined up to control the different types of digital switching required for local, long-distance, and international voice and data transmission. The same modules would be grouped to replace exchanges of different sizes – existing installations varied from 5,000 to 50,000 lines. And in the longer term, enhancements and improvements could be installed via changed software with minimal disruption to the working network.

BT could have abandoned its traditional British suppliers, and bought its replacement exchanges overseas. Certainly, by the time GPT (as it is now) was experiencing major development problems with System X, Canadian and Swedish telecommunications companies were able to supply proven systems. But all of this started before BT was privatised, and the idea of a major nationalised industry re-equipping itself from abroad was unthinkable. Even though other systems existed, they would have had to be changed to meet UK standards. The UK was one of the first to develop digital technology; it was important to maintain this lead. It was also important to retain the skills available in the 'home-grown' telecommunications industry.

Plessey and GEC were put in the unusual position of co-operating, with BT, on the design and development of the system, but competing to supply BT with finished exchanges. Development and manufacturing would take place at different sites, so standardisation was essential. Design and development rules covered circuit design techniques, hardware interfaces, software language, components, power supplies and converters, enclosures, and construction methods. With a development project as large as this one, it was inevitable that new technology and techniques would become available during the process; they were incorporated or exploited where possible.

To begin with, work concentrated on replacing the main trunk and tandem exchanges, to provide a backbone national network (see Figure 1). Eventually, the network would extend to the existing 6,500 local exchanges. These would be replaced by fewer and more flexible System X exchanges, each capable of covering larger geographical areas – this explains why some of your STD codes have changed too.

Lack of software expertise in the early stages of development meant that some of the initial software was inefficient, creating many problems later. Early installations were fraught with failures, some caused by software bugs, others by insufficient connections to the existing analogue exchanges. Local 'down times' of several hours occurred, calls were cut off, and would-be callers received the busy tone from businesses whose sales staff were wondering why the phones didn't ring.

As with any development project, delays occurred and costs spiralled. By March 1985, the System X cost per exchange line was estimated to be around £200 (BT's initial order was for 2.6 million lines). Then Thorn-Ericsson was given a £100 million contract by a now privatised BT to supply similar systems, with a cost-per-line reported as being about £170. The cost-reduction chase was on.

By the end of 1985, System X was reported to be fifteen months and over one million lines behind schedule. Both Plessey and GEC had closed down some of their manufacturing plants with a joint loss of over 1,500 jobs. The first four exchanges had been shipped, although only 2,000 lines were actually connected to customers. (Not surprisingly, one of the earliest installations was at the exchange covering BT's research station.)

But the development companies had finally solved the major software problems which had slowed the project down for so long. Then staff at the Plessey site developed a special digital switch which could reduce the largest System X exchange (a 20,000 erlang switch) from forty racks to four. They also started to use surface mounting for assembling line cards. Since the line cards form nearly 50% of the cost for local exchanges, any savings here make a significant change to the overall exchange cost.

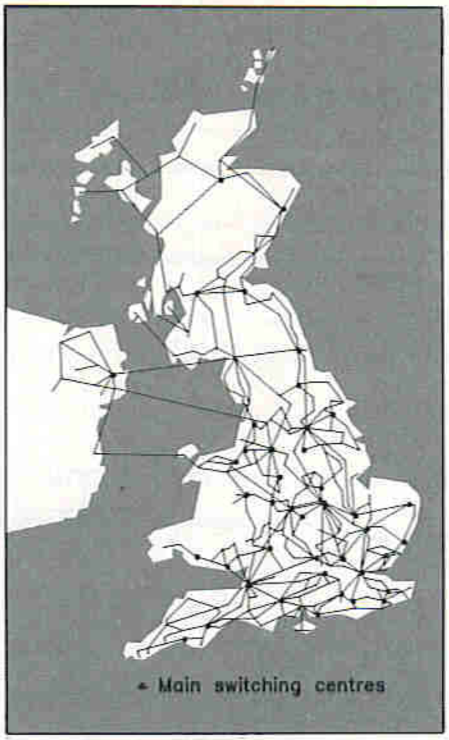


Figure 1. The digital trunk network nearing completion.

Encouraged by these sudden breakthroughs, BT announced a completion date of 1988 instead of 1992 for full conversion of the UK trunk network using 57 System X computerised telephone exchanges.

Installation of local digital exchanges was carried on in parallel with the trunk network. All the electro-mechanical exchanges will be replaced, together with the small electronic (TXE2) local exchanges. This work is due to be completed by the end of the 1990's, and the total cost will have been more than £5 billion. But offset against this cost, which would have been incurred anyway since we needed a new public network, is the fact that once the network is installed and running, the maintenance and accommodation costs will be much less than they are at present. Already, System X is proving to be extremely reliable, and is performing better than the international standards specify.

The 500 or so larger existing electronic exchanges will remain in use for some years yet, although they will be enhanced to give the high-speed call set-up achieved by common channel signalling, and to allow itemised billing.

Some business areas, such as in the City and Dockland development area in London, already have direct digital links, provided as private wires rather than as part of the public network. However, for the rest of us, the completion of a door-to-door digital public telecommunications network cannot be expected until well into the next century, so don't throw your modems away yet!

Push Button Telephones and Star Services

So what advantages does the new network offer? Here is a summary of the ones currently available, and there are more to come.

Push-button telephones offer benefits to both the system user and system supplier. In the old days, the speed of rotary dial pulses had to be controlled so that the exchange equipment could detect each series of pulses accurately. An average 10-digit number would take 15-20 seconds to dial, tying up line and exchange equipment for that length of time even before the call was established. And we don't start paying until we start talking.

Keying a number on a push-button telephone takes less than half the time. The fingering is easier; we don't have to squint around and through scratched plastic to make sure we are putting our fingers in the right hole. The increased call connection speed means that BT gets a much more efficient use of the equipment and a larger income, because non-paying connection time is reduced, and the number of calls per hour is significantly increased.

Assuming that you have got a push-button telephone, there are a number of features that you can now use, or will be able to when the network is complete. These are called Star services, and are available to residential as well as business users. The '✱' ('star') and '#' (inexplicably called a 'gate' by BT) symbols on telephone keypads enable you to access the facilities offered by digital exchanges. (Because of the delay in implementing the much publicised new telecommunications network, BT installed dedicated minicomputers in analogue exchanges to offer some of the services which would eventually be available in the new digital exchanges. So some of these features were actually available before the exchanges were replaced.)

Some of the services already existed on analogue exchanges via the human operator, and these are still available automatically. Others are familiar to those of us who use modern PBX's but they are new to the public network, and have to be arranged and paid for on a quarterly basis.

They are:

RE-DIAL – which allows you to call the previously-keyed number using one key, rather than re-keying the whole number. This feature may be included in your telephone as well; it is certainly a useful feature, since most of us make more than half our telephone calls to less than ten other numbers.

ALARM CALL – where you can key into the local exchange using '✱55#' and program the required time (current cost 10p per call).

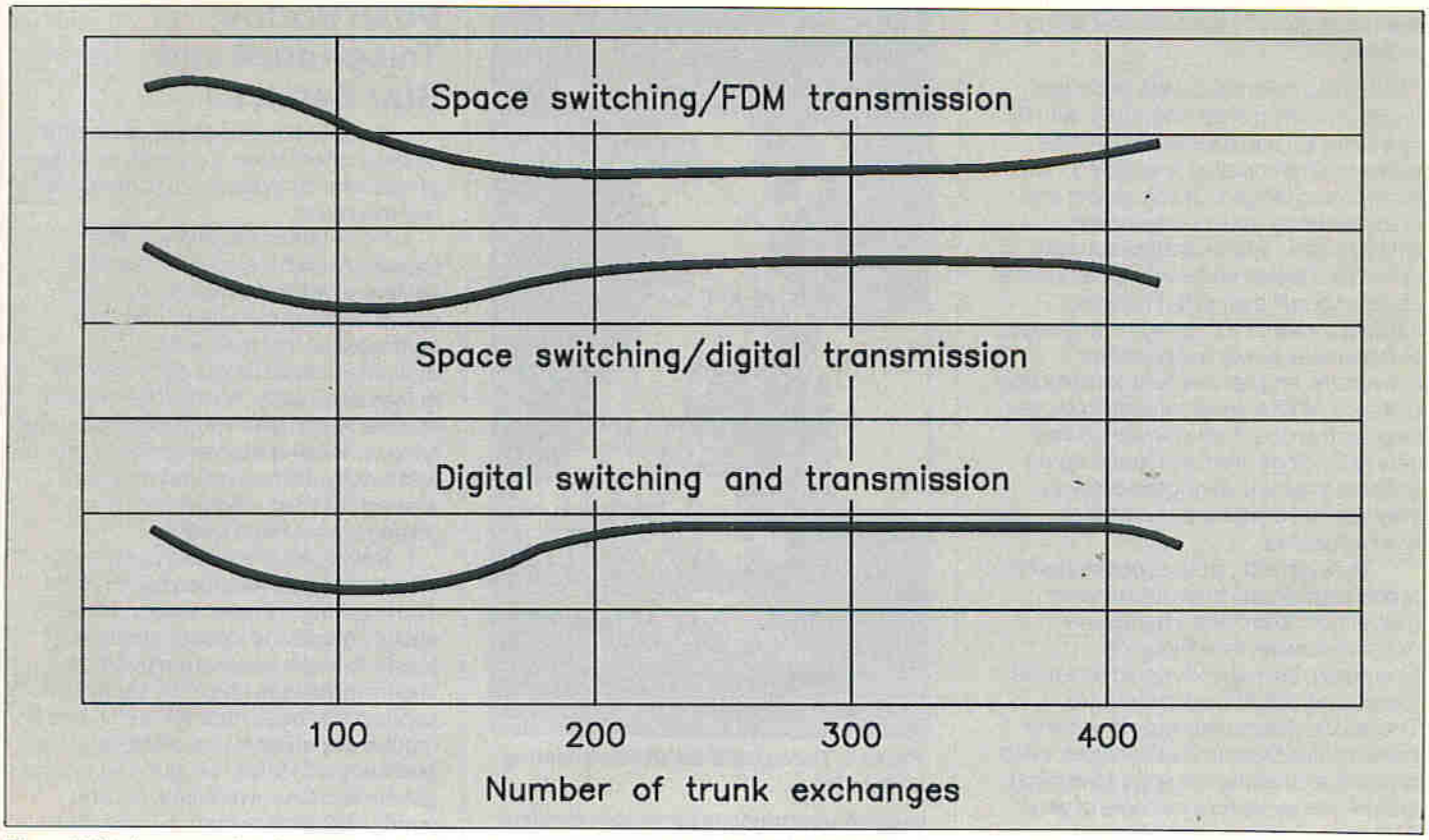


Figure 2. The low cost of an integrated digital network is shown relative to alternative systems.

CALL COST – where, by prefixing the called number with the code '✱40#', you will automatically be called by the exchange at the end of the call and advised of its duration and cost (current cost 5p per call).

ITEMISED BILLING (at last!) if requested – where calls occupying more than ten units can be listed separately in the quarterly bill, which shows the time the call was made, the called number, and the cost of the call. This facility is long overdue, and was particularly welcomed by our own design consultancy company, since it allows us to charge our clients the actual cost of calls as well as threaten our teenage children with a bill.

The new facilities (currently chargeable as shown) are:

SHORT-CODE DIALLING (£4 per quarter) – which enables you to program and store a series of frequently-used telephone numbers and implement them by keying an abbreviated code. Some telephones of course allow this as part of the telephone's features, rather than the exchange's, but an exchange-based short code will mean that all your household telephones can access the short code, rather than having to be programmed separately.

CALL DIVERSION (£7 per quarter) – the most popular service at the moment – which allows you to program an alternative telephone number to which callers will be diverted (either automatically or under user-determined conditions).

NEW CALL INDICATION (£4 per quarter) – where you can be beeped during a call if a subsequent incoming call is received (you can then put the first call on hold and talk to the second caller).

THREE-WAY CALLING (£4 per quarter) – where you can put an existing call on hold, call another number, and either switch between calls or combine all three conversations.

CALL BARRING (£7 per quarter) – which enables you to selectively bar outgoing, or incoming, or both, calls. Full service is resumed via a PIN issued by BT. For incoming calls, you can also temporarily 'switch off' the telephone and the exchange will play a recorded announcement to incoming callers.

I came across two other thoughtful facilities when reading the System X literature. One is the capability of an operator to trace malicious calls. The other is for disabled subscribers, who are presumed to be in distress and are automatically re-routed to a preset number if their outgoing call is not keyed within a known number of seconds. There is no mention of these facilities in BT's general feature list, but if you need them it would be worth asking your local operator if they are available from your telephone.

Other features are aimed at mobile users or businesses with PBX's, but if you are a business user you probably know about them anyway.

A Postscript – Teleshopping

If you have been on holiday in France and have wondered why advertising hoardings sometimes just display a telephone number, you have been experiencing the influence of Minitel. In an inspired marketing campaign, the French PTT gave terminals to anyone who wanted to access the range of services offered by Minitel, from airline bookings to grocery orders. Having provided the terminals, the PTT sat back and counted the income from customers all over France using the telephone to do their shopping.

Similar interactive systems are on their way in the UK, although no mention is yet being made of free terminals. (France started charging for Minitel terminals as soon as people started demanding them.) One teleshopping company is already signing up home users well ahead of BT's ability to do the same. You have been warned!

Acknowledgements: thanks are due to British Telecom and GPT for providing some of the technical and statistical information contained in this article, and to the BT Picture Library for the photographs.

And for the future? This is BT's published list: Time will tell what they all mean!

- Ring back
- Credit call
- Message call
- Automatic freephone
- Conference call
- Diary service
- Universal access number
- Personal ringing

Square One

A First Course in the Theory and Practice of Electronics

Part 4 by Graham Dixey CEng., M.I.E.E.

The Semiconductor Diode

The diode is the simplest type of semiconductor device. For obvious reasons, it is known as a 'two terminal' device. In spite of its simplicity it is able to perform a number of very useful functions. Some of these will be explored in this article. The circuit symbol for a diode is shown in Figure 1 (a). Sometimes this symbol may be drawn 'filled in'. Nonetheless, it is always instantly recognisable by its distinctive shape. The two terminals are known as the anode and cathode, these being identified in this figure. Since there are some variants of the basic device, minor variations will be found in the symbol to indicate these. For example, the zener diode has its cathode bar shown cranked downwards, as seen in Figure 1 (b). There is another diode known as a DIAC, that also has its own symbol. However, this type is rather specialised and will not be discussed here.

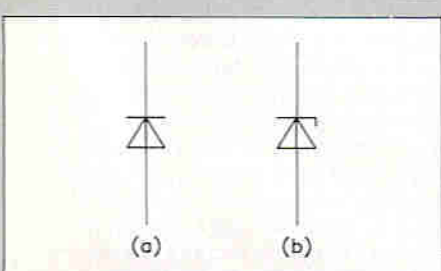


Figure 1. (a) Basic diode symbol and (b) symbol for a zener diode.

Diving back into history for the moment, the diode was the first type of 'valve' to be developed. It was called a valve because it only allowed current to flow through it in one direction. This direction corresponded to the case when an applied voltage made the anode positive with respect to the cathode. The word valve is used in many instances to describe a device which allows the flow of some quantity in one direction only. Examples that may spring to mind are the valve in a car tyre or a valve for controlling water flow. With the advent of semiconductors, the original meaning of the word valve tended to be forgotten, and what had previously been known as a diode valve became a semiconductor diode, or just plain diode. Enough of history!

Diode Construction

There are two main types of diode, known as point contact diodes and junction diodes respectively. The latter are by far the more numerous, although the older point contact types still have their occasional uses. Junction diodes may be made from either of the usual semiconductor materials, germanium or silicon. The latter are more robust and have lower 'leakage currents', but the germanium diode has a lower 'forward voltage', which can be useful on occasions. These terms can be better understood by looking at Figure 2.

This shows the forward and reverse characteristics for a diode. The forward characteristic is obtained by plotting the relationship between the voltage and current when the anode is positive with respect to cathode. This is the normal conducting direction. The reverse characteristic really speaks for itself. Ideally, in the latter case, there would be no current flow no matter how large the applied voltage. In practice a very small leakage current flows, often of little practical significance. What is of the utmost importance however is the fact that, if the reverse voltage is made large enough, the diode breaks down suddenly, a large reverse current flows and the diode burns out. It does this because of the combination of high voltage and high current at the instant of breakdown, and this means high power, so "phut"! It is worth mentioning at this point that a diode must always be chosen to have an adequate reverse voltage rating to avoid this catastrophic event ever taking place. The reverse breakdown voltage for a diode may lie anywhere between a few tens of volts and several thousand volts, according to type.

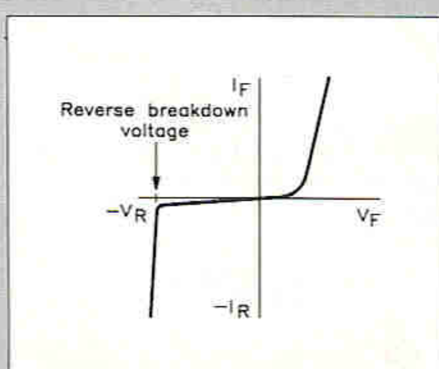


Figure 2. The forward and reverse characteristics for a diode.

Now looking at the forward characteristic, what do we learn? The device starts to conduct at a very low forward voltage and the current then rises rapidly. The diode must be designed to be able to pass a particular maximum value of the latter and this may have to be borne in mind when choosing a diode for a particular application. To put actual figures to the forward voltage to produce conduction, it is necessary to consider the cases of germanium and silicon diodes separately. These figures are nominal only, but it is usually accepted that a germanium diode conducts for a forward voltage of about 0·2V, while for a silicon diode the figure is about 0·6V.

Diode Ratings

Obviously it is necessary to specify a diode in terms of the maximum reverse voltage that can be applied to it and the maximum current that it will be expected to pass. The former is quite easy; it is specified as the Peak Inverse Voltage (PIV). The forward current may be specified in two ways. These are the 'peak forward current' (I_P) and the 'average forward current' (I_F), both of which are fairly obvious in meaning. Another factor that may be specified is the reverse current (I_R), at a given reverse voltage (and at a given temperature, usually 25°C, since this current is temperature dependent). The following data will give some idea of how these values may vary from one device to another. See Table 1.

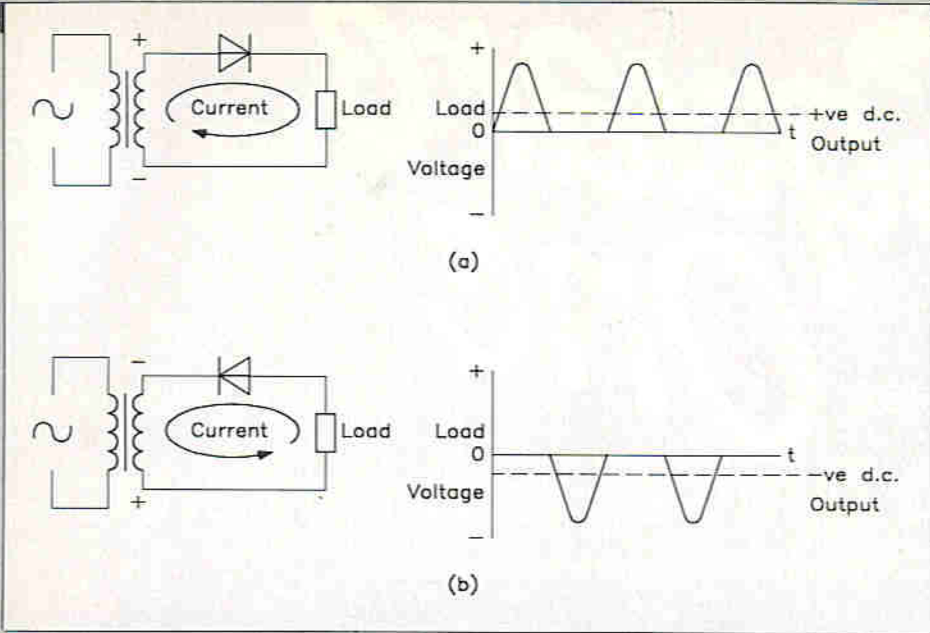


Figure 3. Half-wave rectifiers (a) positive voltage output, (b) negative voltage output.

Device Name	Construction	PIV (V)	Max I_F	Max I_R
OA90	Ge point contact	30V	10mA	<1·1mA @ 30V
1N914	Si	100V	75mA	<25nA @ 25V
1N4001	Si	50V	1A	<10μA @ 50V
1N5408	Si	1000V	3A	<10μA @ 1000V

Table 1. Diode ratings.

Rectification

The fundamental point from the foregoing is that the diode only conducts when the anode is positive with respect to the cathode. This means that, if the applied voltage is alternating, current will only flow on alternate half-cycles. In Figure 3(a), a mains transformer develops a low level voltage across its secondary winding, which is applied to a diode and a resistor 'load'. Figure 3(b) shows essentially the same circuit, the only difference being that the diode is reversed. As a result, in the first case the diode will conduct when the upper terminal of the secondary is positive with respect to the lower one. In the second case the reverse will be true. Notice the direction of current flow and the waveforms that appear across the load in each situation. The load current, and hence load voltage, consists entirely of either all of the positive half-cycles only, or all of the negative half-cycles only.

An alternating input to the diode has produced a unidirectional output. The A.C. input is said to have been 'rectified' by the diode; the latter is then also known as a 'rectifier'. This does not necessarily mean that it is physically different from any other diode, just that this term describes its application. Having said that, there are diodes that are produced for the specific purpose of rectifying alternating supplies. A rectifier in which only alternate half-cycles produce any output is known as a 'half-wave' rectifier, and consists of one diode.

The output shown is not pure D.C., but a mixture of A.C. and D.C. We could determine the value of the latter by connecting a D.C. voltmeter across the load resistor. It would read the *average* value of the pulsating waveform. This would be found to be roughly 'one third of the peak value' (more accurately, peak value/pi). A peak secondary voltage of, say, 21V would give an effective D.C. output of about 7V. Not very efficient! Not only that, but there is a great deal of A.C. left. This wouldn't matter much if the aim of the rectifier was simply to charge batteries, but would be quite useless for powering electronic equipment. However, before considering how to eliminate the unwanted A.C. content, it would be more productive to see how the efficiency could be improved first. What must be done is, in some way, to make use of the other half-cycles rather than just eliminating them. Two methods for doing this are shown in Figure 4. They both use what are known as 'full-wave' rectifiers.

The circuit of Figure 4(a) is known as a 'bi-phase' rectifier because there are two secondary voltages from the transformer that are equal in magnitude (voltage value) but *opposite* in phase. As a result, diode D1 will conduct when the top of the secondary is positive, current flowing *down* through the load as shown, while diode D2 will conduct on the next half-cycle, when the bottom of the secondary winding is positive. Current will then flow through the load, also in the *downward* direction. The output now consists of positive half-cycles without any 'gaps' between them, and the average value of the load voltage has doubled. The rectifier comprises the two diodes, which may occasionally be found combined into one component but are usually separate components.

This latter type of full wave rectifier is not much used these days, although it has had a long history. The use of a 'push-pull' transformer secondary achieves full wave rectification with only two diodes, which used to be an important consideration – originally they would have been thermionic valves (most often two diodes sharing a common glass envelope or 'tube'), or, later on, a pair of the earliest semiconductor devices, either of which were invariably bulky and/or expensive. Nowadays it can be argued that this method is no longer to be seen because it needs a mains transformer with either a centre-tap or a double secondary, which may be expensive. This isn't much of an argument, since most small mains transformers have double secondaries anyway. The real reason for the lack of popularity of this circuit type is the

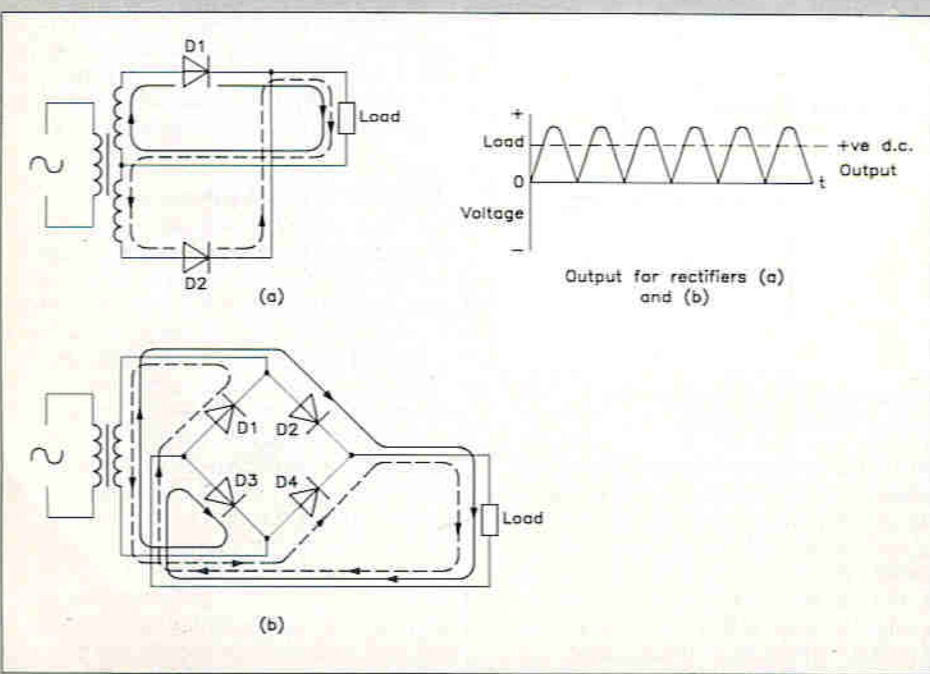


Figure 4. Full-wave rectifiers (a) bi-phase and (b) bridge.

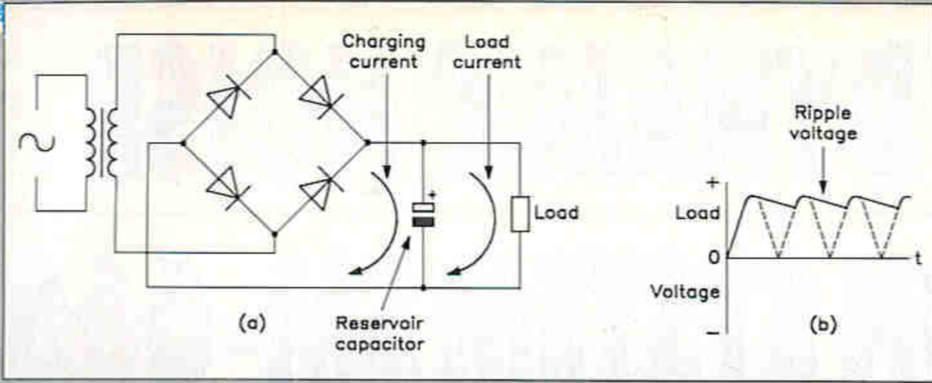


Figure 5. (a) Adding a reservoir capacitor, (b) how ripple voltage is developed.

availability of cheap bridge rectifiers. Which brings us to Figure 4(b).

The bridge rectifier consists of four diodes which, nowadays, are invariably integrated into a single package. Of the four pins on this package, two are marked with the A.C. symbol and are connected to the ends of the transformer secondary winding. The other two are marked with plus and minus signs and give the polarity of the output. Those readers who built last issue's power supply project will be familiar with this device already.

In use, two diodes always conduct together. In the circuit shown, D2 and D3 conduct whenever the top of the secondary winding is positive; diodes D1 and D4 conduct on the other half-cycles. The output looks exactly the same as for the previous full-wave circuit, of course. If there is a disadvantage of the bridge rectifier, it is that two diodes conducting at once means that there are two forward voltages in series. Since the diode volt drops subtract from the available output, this reduces the latter slightly. It is possible to lose anything between about 1·2V and 2·2V across a bridge rectifier. In use this rarely has much practical significance.

Since the output of a rectifier contains A.C. as well as D.C., it is necessary to remove the former. This is done, to a large extent, by connecting a very large electrolytic capacitor, known as the 'reservoir' capacitor across the rectifier output. This is shown in Figure 5(a), the action being explained by Figure 5(b).

When the power is initially switched on, the first positive half-cycle out of the rectifier charges the reservoir capacitor up to approximately the peak value of the rectified voltage. If no current was drawn by the load, the voltage across the reservoir capacitor, and hence across the load also, would be absolutely constant. In other words, the output would be pure D.C. Naturally, the load will draw some current, which must come from somewhere. It is drawn out of the reservoir capacitor. As a result, the terminal voltage of this capacitor dips slightly, until the next positive half-cycle comes along and 'tops it up'. The result is an output voltage on which there is superimposed a voltage variation known as 'ripple'. The greater the load current drawn, the greater will be the ripple voltage. Formerly, it was standard practice to follow the reservoir capacitor with a low pass filter, either a combination of inductance and capacitance, or resistance and capacitance. This practice is less used nowadays. IC regulators are so cheap that it is actually much cheaper to provide a regulator after the reservoir capacitor. Not only is the ripple totally eliminated by the regulating action, but the output voltage is constant with changes of supply voltage or load current.

Reference to (the previous) Part three in this series will serve to illustrate the use of the above philosophy. In the power supply design included in that article, a bridge rectifier was used to develop D.C. voltages of equal magnitude across two reservoir capacitors. One supply was positive, the other negative. These were then used as inputs to chip regulators to develop the stable D.C. output voltages.

The Zener Diode

The zener diode characteristic is shown in Figure 6. At first sight, this doesn't look too different from that of a normal diode. Essentially it isn't in 'shape' very different at all. Where it does differ is in the magnitude of the reverse voltage that causes breakdown to occur. It is very low, anywhere between about 3 to 24 volts being a typical range. Another point to notice is that, when it breaks down, there is a very large flow of reverse current for an almost constant value of the breakdown voltage. In other words, the characteristic is almost vertical.

At this stage it may be remembered that it was the combination of reverse voltage and reverse current that caused the destruction of the rectifier type of diode! Why doesn't it happen here? The answer is that it can if you let it, it just happens to be easily avoidable because of the low value of breakdown voltage.

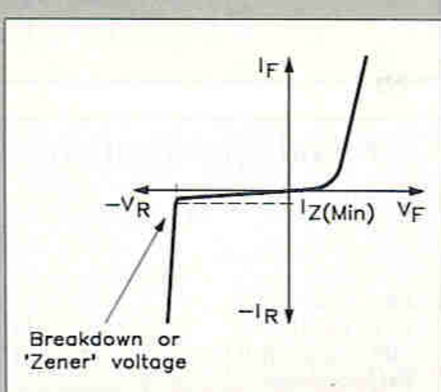


Figure 6. The zener diode characteristic.

To take a few figures to illustrate this point:

Case A: a rectifier diode breaks down at a reverse voltage of 200V and passes a reverse current of 0·1A. The power in the diode at this time is the product of voltage and current, which equals 200 x 0·1 = 20W. Unless this is an exceptionally beefy, high power device, the result will be instant destruction.

Case B: a zener diode breaks down at the reverse voltage of 5V and also passes a reverse current of 0·1A. The power in the diode in this case is equal to 5 x 0·1 = 0·5W. This is well within the capabilities of most zener diodes so no harm is done. The only real difference between a 'normal' diode and a zener diode lies in the physical construction (which means more than its mechanical arrangement), that causes the latter to break down at a very low reverse voltage. It is then only necessary to ensure that, when breakdown does occur (a deliberate event in this device), the reverse current is limited to a safe amount. This is easily done by the use of a series resistor, often known as the 'ballast' resistor as shown in Figure 7.

The circuit of Figure 7 is known as a 'shunt regulator' because the zener diode (the regulator) is in 'shunt', that is, in parallel with the load. The zener diode is connected so as to conduct when the positive voltage on the cathode exceeds the breakdown or zener voltage. Thus, the normal operation of the diode is in the breakdown region. The voltage across the diode is known as V_Z and the current through it is I_Z. Figure 7 shows that there are two other voltages and two other currents. The D.C. input voltage V_S is, naturally, greater than V_Z and the difference between the input voltage and the zener diode voltage is $V_S - V_Z$. This is the voltage that is dropped across the ballast resistor R_S. A current I_L flows in the load, so that the total current that is drawn from the supply is the sum of I_Z and I_L, namely $I_Z + I_L$.

The output voltage across the load, namely V_Z, is supposed to be constant, since that is the object of the circuit. The input voltage V_S may be assumed, for our present convenience anyway, to be nominally constant also. With these assumptions, the voltage across R_S must also be more or less constant, which means that the current in this resistor is constant also. This current is 'shared'

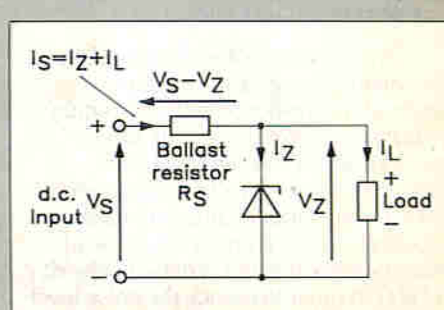


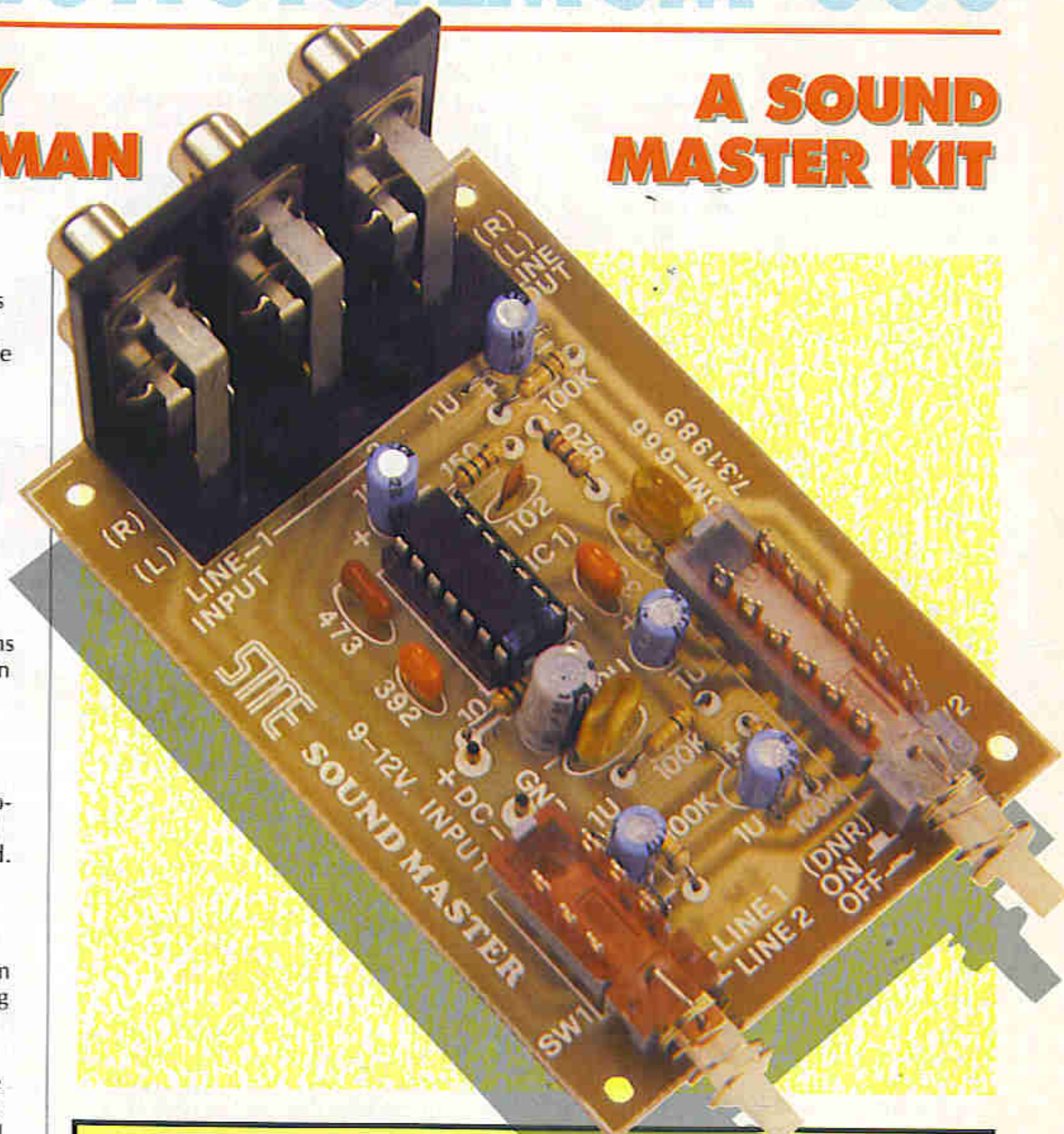
Figure 7. The zener diode shunt regulator.

Continued on page 61.

STEREO DYNAMIC NOISE REDUCTION SYSTEM SM-666

REVIEWED BY DAVE GOODMAN

A SOUND MASTER KIT



Introduction

Based on National Semiconductor's LM1894 IC, the Dynamic Noise Reduction (DNR) System reduces audible noise already present in source material such as pre-recorded cassette tapes and FM broadcasts. To ensure correct operation, line-input signal levels of 300mV or greater are required by the processor and the DNR section can be switched in or out of service with little insertion loss. Processing is dynamic – according to the bandwidth of the input signal – as opposed to encoded or companding type noise reduction systems like the well known Dolby B employed in many domestic hi-fi's.

In order to explain the operation of the processor, we need to delve into the realms of 'auditory masking' and 'psycho-acoustics' – a subjective area concerned with hearing and the perception of sound.

The LM1894 DNR processor IC depends on three basic operating principles:

1. Audible noise is proportional to system bandwidth. In other words, decreasing the system bandwidth will reduce the level of noise.
2. The required signal is able to mask the noise when the signal-to-noise ratio is high enough. If the noise level is equal to (or greater than) the wanted signal, then the signal will be lost as the noise level 'masks' the signal.
3. The human ear is unable to detect distortion in less than one millisecond. For a time period less than 1ms, the ear acts as an integrator and becomes less able to detect distortion.

Dynamic Noise Reduction entails continually changing the system bandwidth in response to the *frequency and amplitude* contents of the source material. Spectral weighting filters inserted in the system control path ensure that the bandwidth increases enough to pass any music present. It would be of little use restricting the system bandwidth to 1kHz in order to reduce the noise level when the source material bandwidth extends to 20kHz, for example!

★ Can reduce tape noise and FM hiss by 10dB
★ Compatible with Dolby B encoded tapes
★ Two switchable (stereo) phono inputs
★ DNR defeat and phono outputs

General Specifications

Supply voltage range:	9 to 12V DC	Input impedance:	20kΩ
Average supply current:	20mA max.	Noise reduction:	−10dB @ 20kHz −5dB @ 5kHz −4dB @ 1kHz
Max. allowable input signal:	3·5V r.m.s. @ 1kHz	Total harmonic distortion, 300mV input for full bandwidth of 20kHz:	<0·04%
Min. input signal for flat frequency response:	300mV		

Listen (with Mother)

Whilst a particular sound is being heard (perceived?), the listener becomes less able to hear another sound – a bit like the sensation of pain, it would seem (have you ever hit your thumb with a hammer and then dropped the hammer onto your toe?). Soundwise, this effect is known as 'auditory masking' and becomes particularly important when noise is 'masking' the ability of the listener to hear tones. In the ear, the hearing mechanism incorporates a membrane with nerve endings spaced along its length, instead of all being 'clumped' together at one end, and each nerve is responsive to one point in the sound spectrum. This tends to ensure that the ability to hear at one frequency is not masked by a sound at another frequency when the signal frequencies are well separated – ingenious, no? On the other hand, white noise has spectral components at all frequencies which stimulate many of these nerve endings, all together at the same time – masking therefore occurs. The effect of this is particularly noticeable with single tones – fortunately, music waveforms exhibit good noise masking qualities concentrated around the most sensitive region of our hearing, between 700Hz and 1kHz, and can be up to 30dB more effective than single tones.

Chips Under Pressure

Signals recorded with a Sound Pressure Level (SPL) averaging 40dB allow the full audio bandwidth to be used without any noise becoming apparent, but speech and solo musical instrument signals exhibit noise due to their similarity to single tones.

If signal sources were able to continuously remain at high Sound Pressure Levels then noise reduction systems would not be necessary. Of course, this could hardly be the case as a momentary drop in SPL results in the noise becoming un-masked and hence able to be heard. DNR systems must therefore be able to track the source programme dynamics very closely if noise is to be kept below audible levels. Also, a sudden increase or transient in the source programme must be met with by a quick response from the system if distortion is to be prevented. For the LM1894, a variable cut-off frequency, low-pass filter (LPF) is inserted into each audio channel, as can be seen in the block diagram of Figure 1. The control path is generated from the stereo source material dynamics and is filtered and detected before driving the filter stages.

Based on the principle that the human ear is unable to detect distortion for less than 1ms, signals having sufficient energy to mask noise will open up the system bandwidth by driving the low-pass filter cut-off point above 30kHz in less than 1ms. Conversely, the system bandwidth is lowered by driving the LP filters down to 1kHz, taking

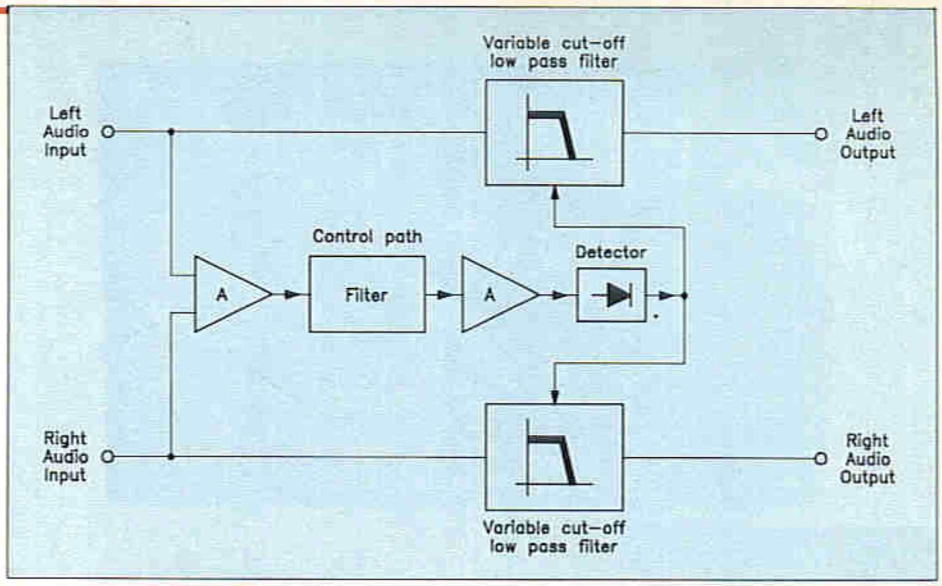


Figure 1. LM1894 block diagram.

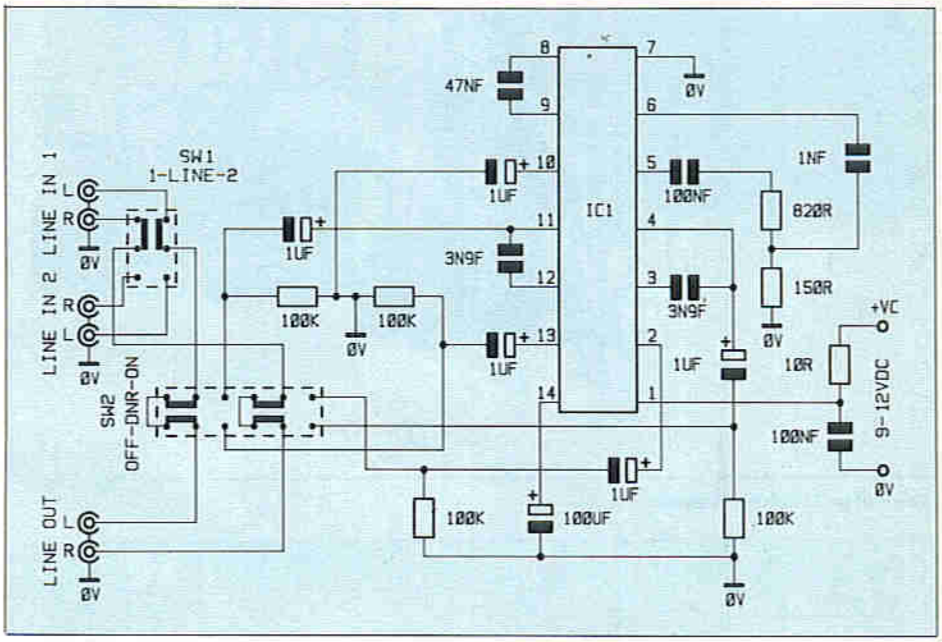


Figure 2. Circuit.

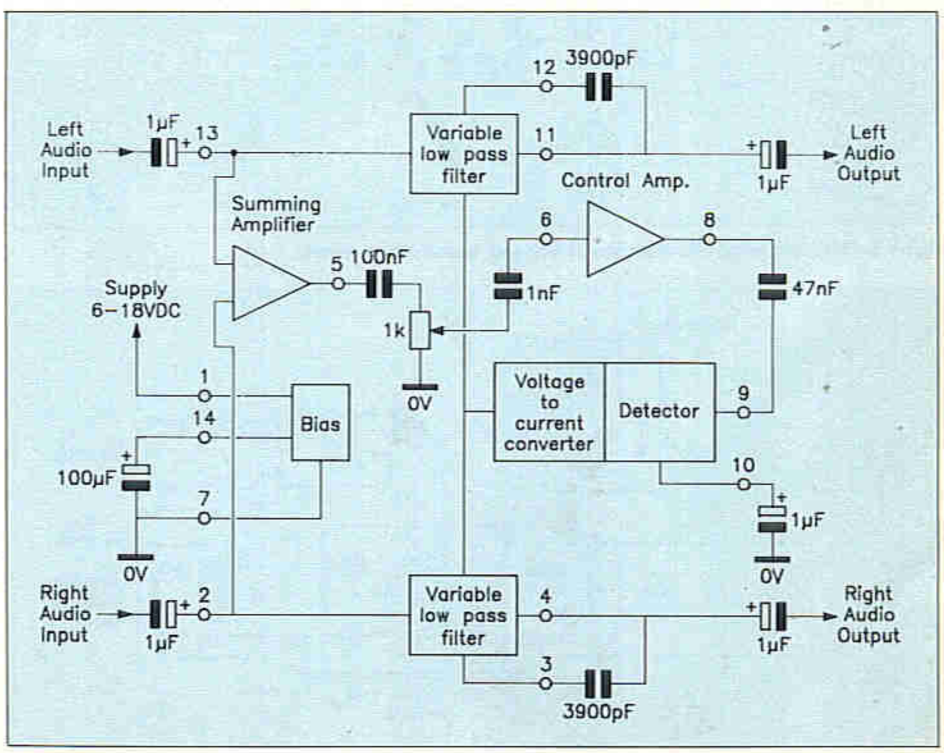
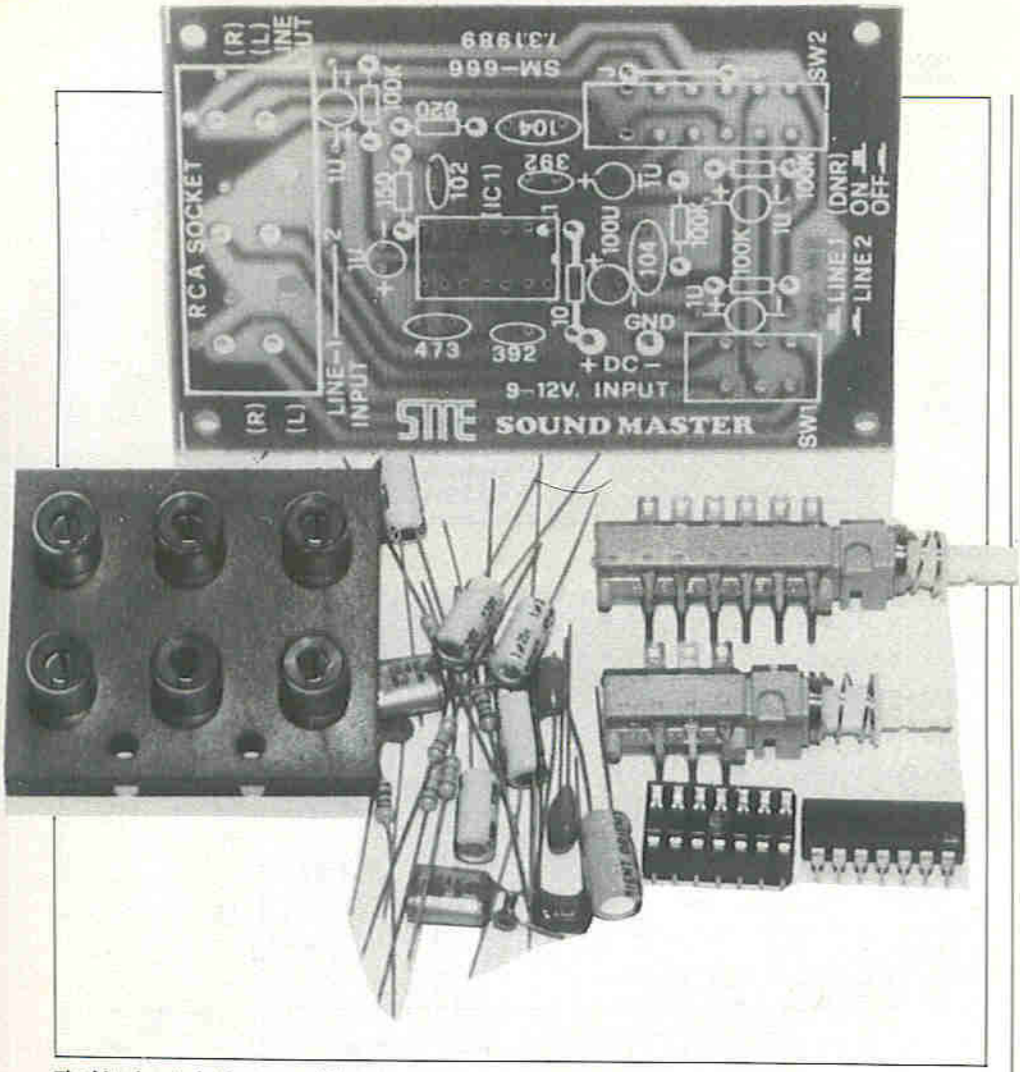


Figure 3. System schematic.



The kit of parts before assembly.

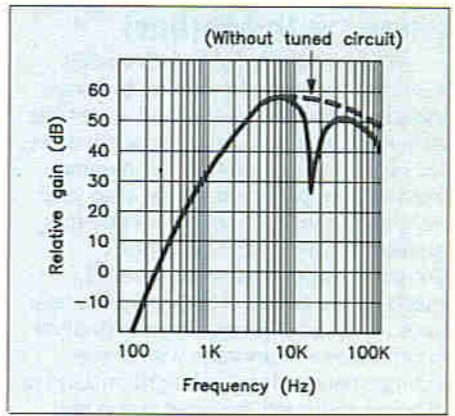


Figure 4. Notch filter response.

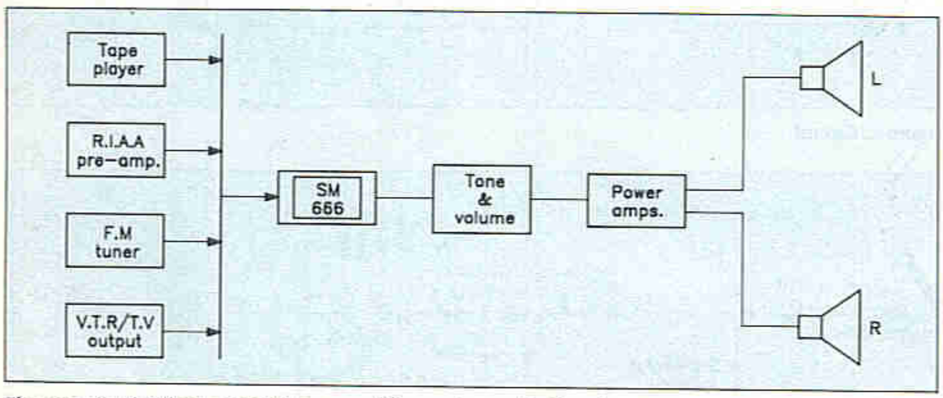


Figure 5. DNR following initial pre-amplifier and equalisation stages.

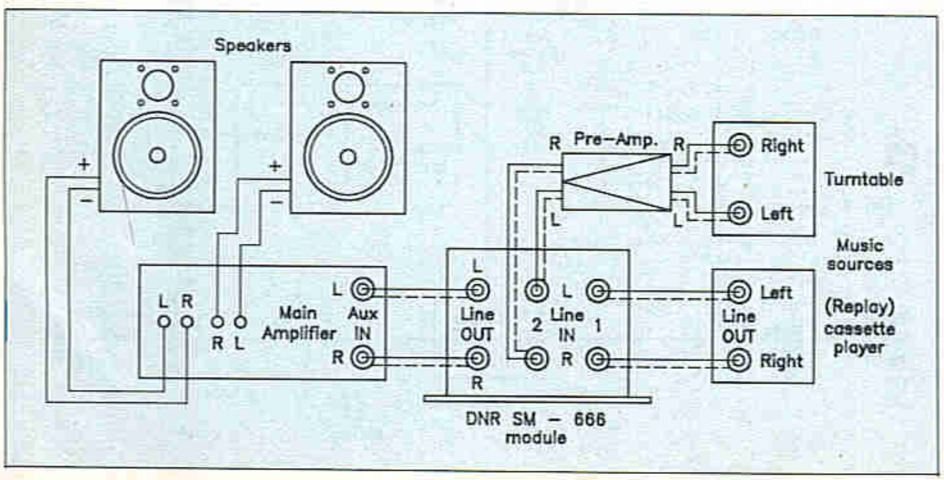


Figure 6. Fitting the DNR in your system.

approximately 60ms to do so, thus allowing enough time for the music 'ambience' to remain without allowing noise through.

Circuit Description

Figure 2 shows the circuit diagram of the DNR, but this is too confusing to see how the system operates, therefore a block schematic diagram showing essential operations is included, Figure 3. Signal inputs and outputs to/from the LM1894 are AC coupled and the layout of both audio signal paths is symmetrical. In fact, there are three signal paths: left and right stereo audio channels, and a bandwidth control path. This control path generates a bandwidth control signal which simulates the human ear's sensitivity to noise in the presence of a tone. Signals from both left and right channels are summed together to centralise the stereo image, and then connected to the potential divider (shown as a potentiometer) at pin 5. The actual values of these resistors have been chosen in order to allow the noise level from a cassette tape to *just* open the bandwidth of the voltage controlled LPF's. The control path gain is approximately 60dB, as set by the control amplifier and peak detector stage gains, and is high enough to ensure that the LPF bandwidth can be opened (increased) by very low noise threshold levels. Two capacitors, of values 100nF and 1nF at pins 5 and 6 respectively, attenuate low frequency transients – below 1·6kHz – and determine the control path frequency weighting characteristic at 6kHz. This prevents signal transients of high amplitude and low frequency content (below 1kHz), such as bass drum etc., from momentarily activating the control path detector and 'pumping' the audio bandwidth.

For signals that have a high frequency content the control path sensitivity is increased at the rate of 12dB per octave and allows for the fact that the harmonic content of program material falls off rapidly with increasing frequency. The 47nF capacitor is chosen for the 12dB per octave slope characteristic here, and Figure 4 shows the control path gain peaking at 6kHz (dotted line). Signal energy content between 1kHz and 6kHz will therefore cause the audio bandwidth to extend to 30kHz, thus allowing any high frequency signal component through the audio path. LP filters in the audio path are of the variable transconductance type with linearising diodes for minimum distortion levels. When the transconductance of the stage changes, then for a fixed value of capacitance between pins 11 and 12, or 3 and 4, the filter's cut-off point will change with a 6dB per octave slope characteristic. As a point of interest, decreasing the value of the 3n9F capacitor will increase the maximum audio bandwidth, set here at 965Hz to 34kHz, and vice-versa!

Finally, the capacitor coupled to the peak detector at pin 10 forms the control path tracking envelope. What this means is that a fast attack response time is required to cater for fast transient signals, and yet a relatively slow decay time is needed to maintain the 'ambience' quality of music signals, which is taken care of by this stage.

Construction

The PCB legend has been labelled with the actual values of the components to be fitted at these positions, which is different to the usual Maplin technique where the devices are numbered and listed with reference to a parts list. If you are used to this then the component value identification method used with this kit may give rise to some confusion! The 'three figure' markings of some capacitors may be obscure unless you are familiar with the convention – for example '392'. Here the third digit, in this case 2, indicates the number of noughts or zeros that must be added to the value indicated by the first two numbers. The final number is always in picofarads (pF). Here '39' plus two 0's = '3900pF'; this can also be written as 3n9F (3.9nF), and 0.0039µF. '473' is '47' plus three 0's or '47000pF' (also 47nF and 0.047µF), etc.

The longest lead of the radial electrolytics is usually the positive (+) lead, the other is identified by a black stripe and minus (–) sign on the body. These must be inserted into the PCB with regard to correct polarity, the hole for the positive lead is marked with a '+' on the board. The resistors are easier and should be matched against the appropriate figures on the board, e.g., 150Ω to '150', 100kΩ to '100K' etc. Positions of electrolytics are indicated by circles, non-electrolytics by lozenges and resistors by rectangles.

Fit a wire link between the two holes marked 'J' besides SW2. Links can be fashioned from 24 s.w.g. tinned copper wire or left over component lead off-cuts.

Building this project is very simple and should not create any problems for anyone with average construction skills. A methodical approach and attention to detail, especially when soldering, is the main requirement while building projects and kits and it is recommended that the track side of the PCB is well cleaned with a suitable solvent when complete.

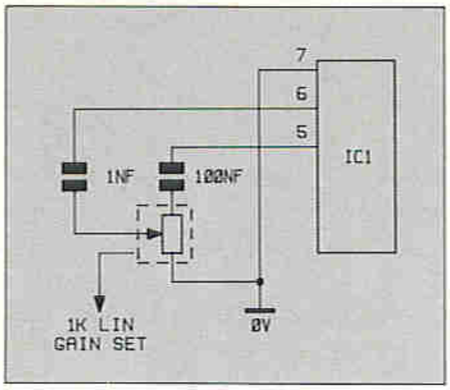


Figure 7. Variable noise threshold.

Using the DNR

The DNR requires an average signal input level of 300mV or more and should *not* be coupled directly to tape heads or magnetic pickup cartridges, the output level from these is far too low. Figure 5 shows the DNR following initial pre-amplifier and equalisation stages from such signal sources. The pre-amplifiers need to have a RIAA (for magnetic cartridge) or NAB (for tape) equalised frequency response. The reason for this is that the un-corrected tonal characteristics will change the signal noise floor (basic level) and hence the bandwidth of the DNR. Installing the DNR at a position before any tone controls will reduce the system noise at maximum boost settings, particularly with treble boost controls. If the signal levels available at the chosen installation position for the DNR in the system are very small (below line level), then a further separate preceding pre-amp will be needed to raise the level. Various types are available from the range of Maplin 'Mixer' Modules, see the projects section of our catalogue for details.

Power Requirements and First Time Trial

A well smoothed and regulated power supply of +9 to 12V is required for the module to operate and is connected to the two supply pins marked + DC – on the PCB. Current consumption is 17mA on average, but can vary up to twice this figure depending on the actual supply voltage. Connect the module into your system as indicated in Figure 6, apply power to the DNR and play a cassette tape

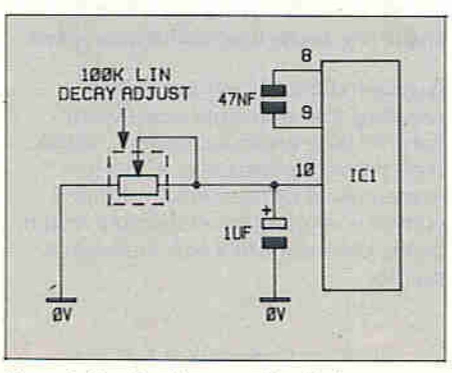


Figure 8. Varying the control path time constant.

or FM radio through the system. Just how the module performs is best evaluated by switching the DNR 'in' and 'out' repeatedly with SW1 and listening to a variety of source programme material.

Improving the Improvements!

Some solo instruments and speech signals may produce a 'noise pumping' effect as the signal level changes, or the source signal to noise ratio may be so poor that masking will not be completely effective. Under these conditions the noise floor threshold needs to be changed to accommodate these varying input sources. This is best achieved by replacing the two sensitivity control resistors between pins 5 and 6 with a 1kΩ potentiometer or preset, see Figure 7. A really poor signal to noise ratio could be improved by changing the control path detector's envelope decay characteristic with a resistance placed across the envelope timing capacitor at pin 10,

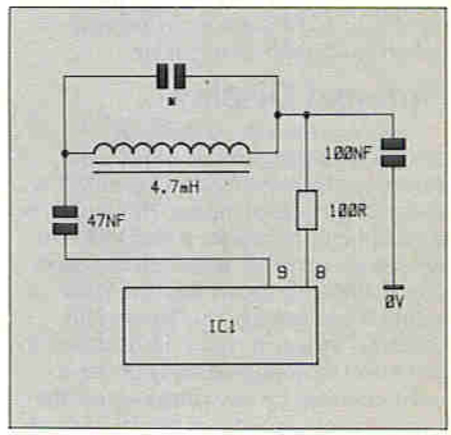


Figure 9. Reject filter.

allowing it to discharge faster. For a variable decay time connect a 47kΩ to 100kΩ potentiometer between pin 10 and 0V, as shown in Figure 8, and which may be used in conjunction with the 1kΩ noise threshold control to even greater effect.

Feeling Dejected?

Audio sources such as TV's and FM radios can present particular problems due to the presence of part of a 15kHz timebase signal (TV) or 19kHz pilot tone (stereo FM) which manages to find its way through into the main audio signal proper. These can be detected by the DNR control path and upset the noise threshold setting, consequently requiring adjustments to the noise level adjustment resistors. If a different audio source is then played through the system, or the timebase/pilot signals are no longer present, then source noise will become audible again, requiring further changes to be made to the noise level setting. However, a simple reject filter can be inserted between the control amplifier and detector input, as shown in Figure 9, and attenuation of these frequencies to –15dB is possible. Figure 4 shows the expected response of this filter on the control path and the value of capacitor 'C' needs to be 15nF for FM correction and 22nF for TV audio correction.

Signal Switching

Switch SW1 has been included to allow the DNR system to be bypassed when required. Although this method is fine in practice, there will tend to be noticeable changes in programme quality whilst operating the switch, like insertion loss, occasional clicks and pops if input and output connections are referenced to different DC levels! The only way to avoid this is to ensure that all four inputs are AC coupled only and linked to ground with high value leakage resistors. An alternative method for defeating the DNR operation without changing any switch positions is to merely ground pin 9 to 0V, as shown in Figure 10a. Here the control path is disabled by this action forcing the audio filters open to their full 34kHz bandwidth. A further method is suggested in Figure 10b; here the detector filter input is connected via a 1kΩ resistor to the +V supply rail and this forces an increase in system bandwidth up to 50kHz.

Optional Display

You can include an optional bandwidth display circuit which will show how the control path of the DNR is responding to signal inputs. This facility is not absolutely necessary if the DNR is to be used as part of an audio sound system or hi-fi, and only shows how the DNR system is performing. This bandwidth indicator, shown in Figure 11, is based on the LM3915 bargraph display driver IC and is operated by the voltage across the detector filter capacitor at pin 10, but note that the display will not operate if the previously mentioned decay modification is implemented. The LEDs light in 3dB increments of input voltage, and component values used are based on the minimum audio bandwidth setting needed to operate the first LED; the full audio bandwidth voltage should then light the last LED. In practice, adjusting the sensitivity (1kΩ pot) so that the second LED just lights – with no input signal – should produce the best results. Apart from the ten LED's there are only four other components and so the display circuit can easily be accommodated on a small piece of veroboard.

The optional display can be used as an indicator to show how the DNR system is responding to input signal level. For a truly flat frequency response, the display should indicate maximum – the bargraph *does not* indicate signal level, but the instantaneous bandwidth of the two filters, as sampled from the control path peak detector output at pin 10. This control voltage is then converted into a control current to modulate both transconductance amplifier blocks which form the basis of the filters. In short, if the 150Ω and 820Ω resistors are changed for a 1kΩ potentiometer as Figure 7, the bargraph display is a visual aid to adjusting the bandwidth sensitivity to cater for different music sources, which have different noise floors. If this is implemented then essentially the 1kΩ control is adjusted for a minimum reading on the bargraph while *only* the noise is present.

DNR is a trademark of National Semiconductor Corporation.

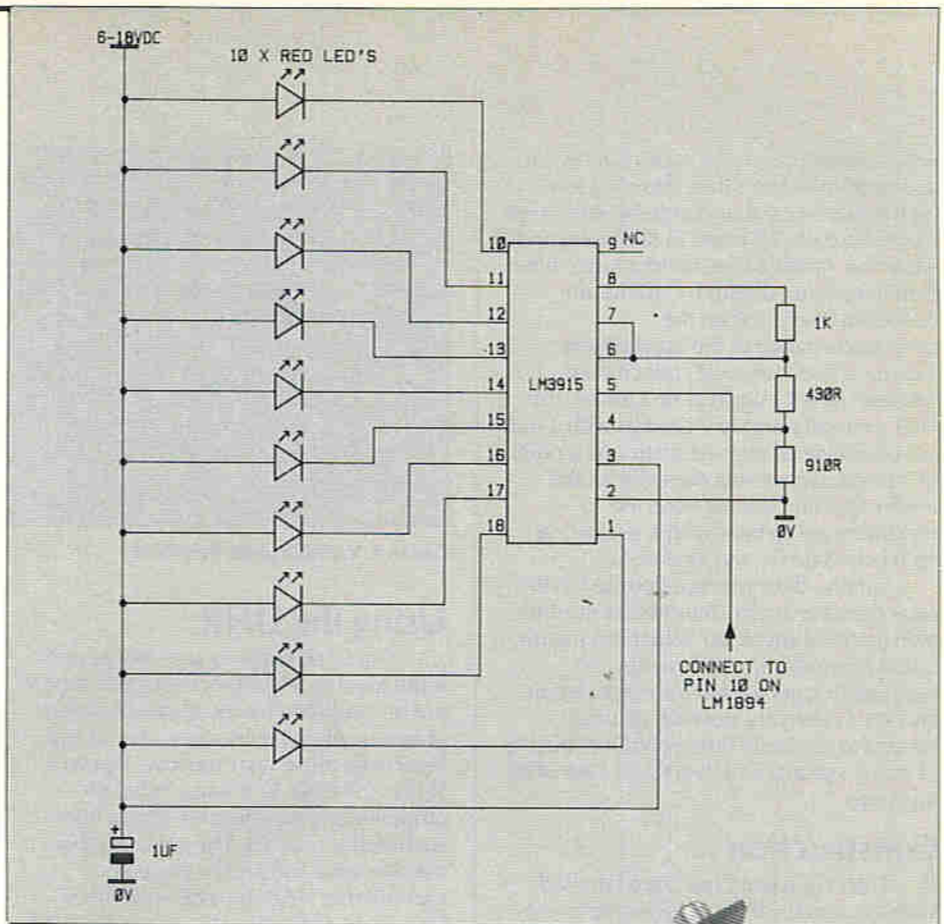


Figure 11. Bandwidth indicator.

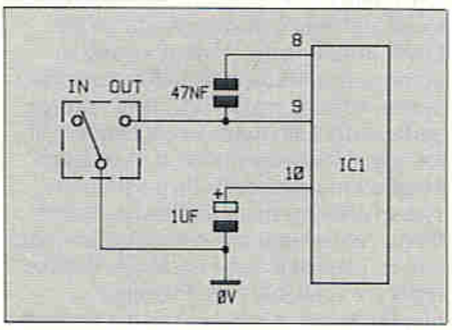


Figure 10a. 34kHz bandwidth bypass system.

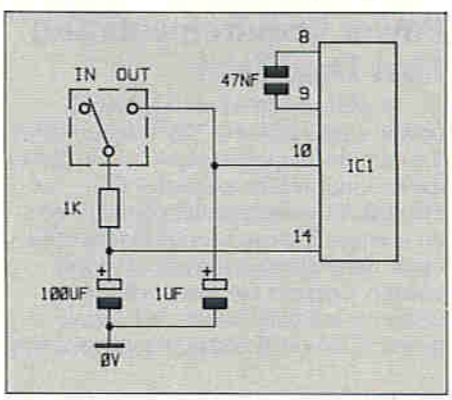
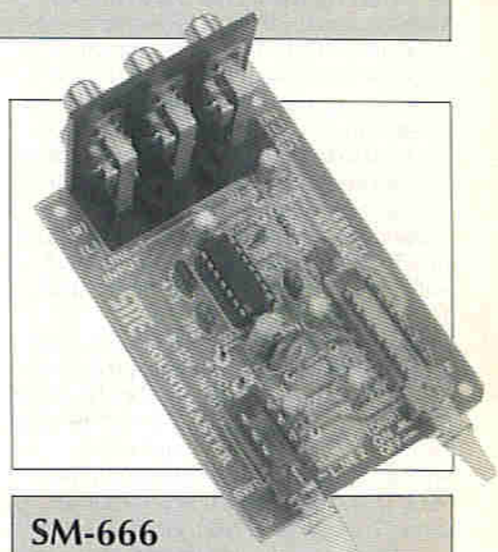


Figure 10b. 50kHz bandwidth bypass system.



SM-666 PARTS LIST

RESISTORS ⅙ Watt 5%	
10Ω	1
150Ω	1
820Ω	1
100kΩ	4
CAPACITORS	
1000pF ceramic	1
3900pF 5% polyester	2
47nF 5% polyester	1
100nF 10% radial polystyrene	2
1µF 25V radial electrolytic	5
100µF 16V radial electrolytic	1
SEMICONDUCTORS	
LM1894 (IC1)	1
MISCELLANEOUS	
14-pin DIL socket	1
6-way phono socket	1
2-pole latch switch	1
4-pole latch switch	1
1·5mm terminal pins	2
SM-666 PCB	1

Available in kit form only:
Order As LP21X (DNR Filter) Price £14.95

SQUARE 1 continued from page 55.

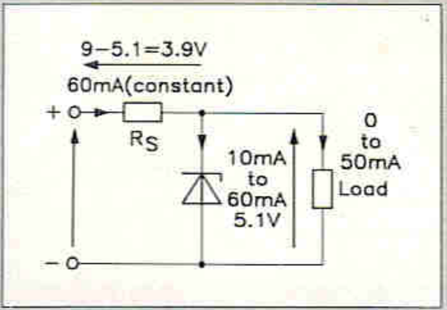


Figure 8. The zener shunt regulator – design example.

between the zener diode and the load. As one takes more, the other takes less. That is the key to the success of the circuit.

It now becomes possible to calculate a value for the resistor R_S if the values of I_Z and I_L are known. The problem is knowing what value to assign to I_Z. The criteria are:

(a) I_Z must never fall below the minimum value shown on the graph of Figure 6 as $I_{Z(MIN)}$.

(b) At the other extreme it must never exceed the value that corresponds to the power rating of the device. This value of maximum diode current can be found by dividing the diode power rating by the zener voltage.

It might be helpful to take a specific design example. Assume a 5·1V zener diode is to be used for a simple 50mA supply, where the unregulated D.C. input is 9V. The circuit of Figure 7 is shown again in Figure 8, with the voltage and current values added.

It is fairly safe to take 10mA as the value for $I_{Z(MIN)}$. What is then necessary to assume (or know) is the likely variation in load current. Suppose that the figure of 50mA is the maximum value and that, in practice, it could be anything between zero and 50mA. This straight away provides some useful information:

(i) The total supply current will be $10 + 50 = 60$mA.

(ii) When the load current is zero, the diode will have to pass the whole of this current, namely 60mA.

(iii) When the load current is the full 50mA, the zener diode current will be just 10mA.

From (i) it is possible to calculate the value of R_S.

$R_S = (V_S - V_Z)/(I_Z + I_L)$,
$= (9 - 5{\cdot}1)/(0{\cdot}010 = 0{\cdot}050)$ {putting the currents in A}
$= 3{\cdot}9/0{\cdot}060$
$= 65$ ohms (practical value = 68 ohms)

It is important to decide the power rating of this resistor as well. This can be done by using $P = I^2 \times R$, for example. In which case, the power rating of $0{\cdot}06^2 \times 68 = 0{\cdot}245$W. Thus, a 1/4W resistor will suffice.

From (ii) it is possible to determine the power rating of the diode. This is equal to 'zener voltage x maximum zener current'. Thus, power rating is $5{\cdot}1 \times 0{\cdot}06 = 0{\cdot}306$W. A 500mW (0·5W) device will be adequate.

This is about the simplest design procedure; it can be much more complicated if one has to consider the effects of possible input voltage variations too. The aim of the above is to give an insight into the operation of the shunt regulator circuit.

Miscellaneous Uses of Diodes

In spite of its basically simple nature, the diode appears in a great variety of different circuits, some of which make direct use of its 'one-way' property, some of which use some other, perhaps less obvious, characteristic. A few such varied applications will now be described.

The Diode as a Protection Device

Because of its unilateral conduction quality, the diode can be regarded as a 'voltage sensitive switch'. By this it is meant that it will only conduct when the polarity of applied voltage is correct – anode positive with respect to cathode. This allows it to behave as if it weren't there at all when reverse biased (cathode positive), since the resistance in the reverse direction is extremely large (many Megohms) compared with the forward resistance, which may be measured in ohms only. Use is made of this fact in the circuit of Figure 9.

Here the diode is connected in parallel with a relay coil. Although the primary purpose of this coil is to allow the flow of direct current through it to develop a strong magnetic field to attract the armature, it nonetheless behaves as a high value inductor. It will have all of the properties of an inductor including one that is covered by Lenz's law. This latter states, in simple terms, that when the current flowing in an inductor tends to change (for example when it is switched on or off), an EMF is induced in the coil. Thus, if the current is falling the induced EMF tries to sustain it; if the current is rising, the induced EMF opposes this growth. Now back to the relay coil of Figure 9!

Suppose that a sudden positive voltage 'step' is applied to the transistor base; the transistor will start to conduct, current will flow in the relay coil, and the relay will 'pull in'; that is, its armature will move, causing the switch contacts to close. The rising current will be opposed by the induced EMF just mentioned. Apart from delaying the moment when the relay operates, this action is of no great consequence.

However, when this base drive is suddenly removed, the current in the coil will immediately try to fall to zero, an action that will be opposed by an induced EMF of opposite polarity to the previous one. In the absence of the diode shown, this induced EMF would cause the

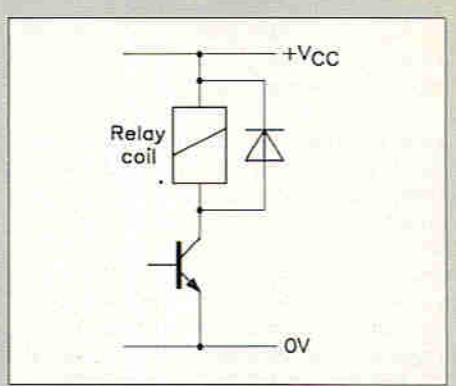


Figure 9. Diode used to protect transistor with inductive load.

collector voltage to rise well above the normal supply value (V_{cc}). This sudden rise of voltage at the collector could be well in excess of the rated value for the transistor, causing sudden failure.

How does the diode prevent this from happening? It does so because, whereas it is normally reverse biased, as soon as the collector voltage exceeds V_{cc} by about 0·6V (the amount required to cause a silicon diode to conduct in the forward direction), the diode resistance changes from a very high to a very low value. This low resistance absorbs the magnetic energy stored in the coil and the induced EMF collapses.

This action may perhaps be better understood by considering the coil as the electrical equivalent of the 'flywheel'. The latter, when rotating, acquires momentum. As a result, it cannot stop instantly. The energy that it contains has to be absorbed by some form of brake. Any tendency to stop it while in motion is opposed by its momentum. Similarly, when it is at rest, a suddenly applied force (a step) will not cause it to reach full speed immediately. Its inertia opposes the accelerating force, slowing the process down.

A second way of using a protection diode, this time to protect an entire circuit, is shown in Figure 10. The intention here is to guard against any failure resulting from an attempt (accidental of course!) to connect the supply to the circuit the wrong way round. Sometimes even a momentary reversal, such as might occur sometimes when changing the battery, can cause a catastrophe. A diode connected in series with the D.C. supply so that it normally conducts when all is well will obviously not conduct if the supply is the wrong way round. Since no current can flow under the latter conditions, no harm can

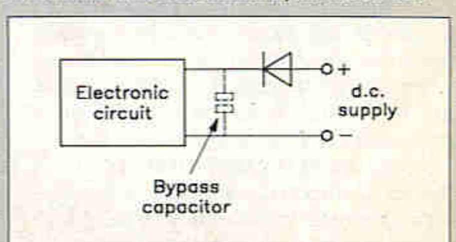


Figure 10. Using a diode to protect a circuit from supply reversal.

be done. Simple but effective. Sometimes it may be necessary to fit an electrolytic capacitor as shown, as a bypass.

It is possible to use diodes so as to make a circuit totally independent of the supply polarity. All that is needed is a diode bridge, just like the one described earlier for rectification. The difference is that, this time, both input and output are D.C.! This is shown in Figure 11. It may be remembered that, when it was used as a rectifier of A.C., the polarity of the D.C. output was the same for both polarities of the A.C. input. Thus, by the same rules, the output will have the same polarity for either polarity of a D.C. input. Again a bypass capacitor may be necessary sometimes.

The final use of a protective diode is shown in Figure 12. Here, two diodes are connected 'back to back' (also known as 'inverse parallel') across the terminals of a moving coil meter. It is assumed that when the meter is reading its full-scale value, the voltage across the meter coil is less than 0·6V. Neither diode will then conduct. Now suppose that the meter is used incorrectly by passing an excessively high current through it. This can easily happen in a thoughtless moment. Since moving coil meters are quite delicate instruments, such a large overload could damage it – an expensive mistake! However, if the voltage across the coil reaches the diode conduction value of 0·6V *before* any damage is done, the now conducting diode will bypass any excess current safely away from the meter coil. By including two diodes as shown, the meter is protected from overloads in either direction.

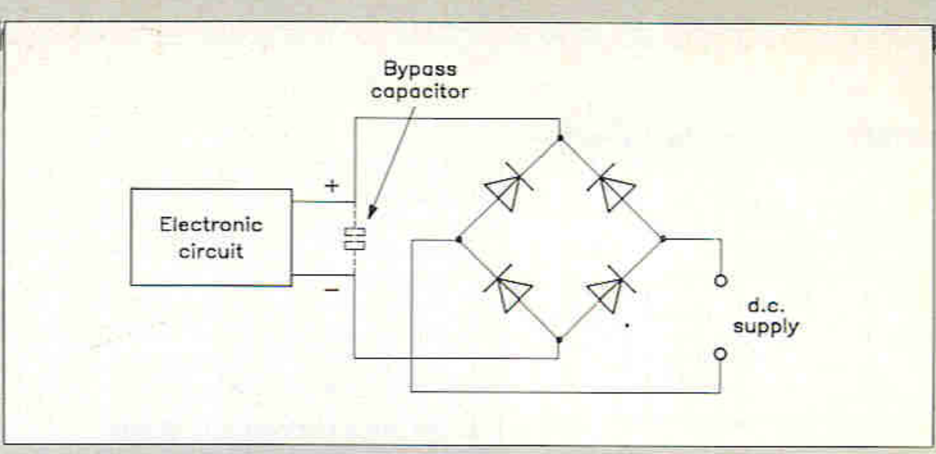


Figure 11. Method for making circuit independent of supply polarity.

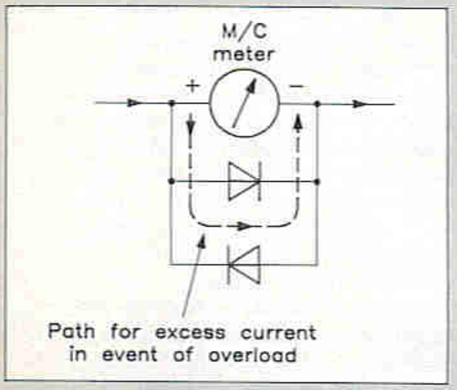


Figure 12. Using diodes to protect a moving-coil meter.

The Varicap Diode

The ability of a TV or radio receiver to 'tune in' a particular station depends upon the selective property of a circuit comprising inductance and capacitance, and known as a 'resonant' circuit. The inductance may be in series with the capacitance or in parallel with it, depending upon the specific tuning arrangements. In general, a parallel LC circuit is used to develop maximum signal voltage at the required station frequency. Any particular combination of L and C gives a resonant frequency that corresponds to some station frequency. To change stations, it is necessary to change either the value of L or the value of C. In the case of continuous tuning, as with most radio receivers (except those in cars) a variable capacitor is the usual method. In the case of push-button selection of a limited number of stations (car radios and TV's), either fixed values of capacitance or inductance could be switch-selected. The TV 'turret' tuners of the past are a good example.

Current television practice adopts a somewhat different approach. It is, of course, still necessary to change the value of, say a capacitor in order to change stations. The difference is that the capacitor is not identifiable as an 'actual' capacitor, but is a diode that, when reverse-biased, behaves as a capacitor of small value. Furthermore, the capacitance value is dependent upon the value of this reverse voltage. This makes it possible to control a tuned circuit by varying the D.C. voltage applied to a 'varicap' diode (as it is called). In practice, each channel to be tuned in has its own preset resistor. When the receiver is first put into use, the wiper position for each selected channel is adjusted to tune in the required channel.

Figure 13 shows part of a TV tuner circuit in which the varicap diodes and their controlling potentiometers can be recognised. Note the special symbol for this type of diode, a combination of capacitor and diode symbols.

It is possible to go on at great length on the many ways in which diodes can be used. Space limitations, apart from any other considerations, preclude such a possibility. It is hoped that, by now, the reader has acquired a sound understanding of the diode's basic behaviour, enough to help him/her to appreciate its value and perhaps to recognise what its particular function is when it is seen in use.

The Project

The project this time has a degree of usefulness out of proportion to its extreme simplicity. It will take no time at all to make, so there is no excuse for not having it ready to use, by the time that the next issue of your favourite magazine comes out! Of course, if you already have a logic probe, then that will do all that this one will do and more besides. It isn't a serious servicing tool but a simple indicator of the two possible logic states, logic 0 and logic 1. As you may now gather, the next issue will start to look at the basics of digital electronics.

For the moment though here is yet another type of diode, the light emitting diode, or LED. This diode is so constructed that, when it is forward biased (with about 2V across it) it emits

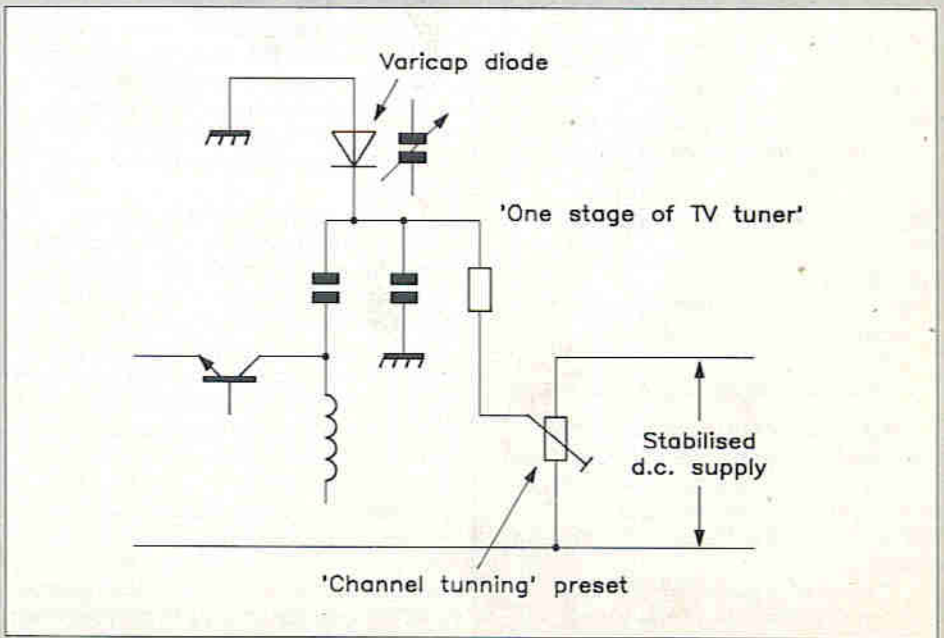


Figure 13. Varicap diodes are used in TV tuners.

visible light. It is usual to house the semiconductor device in a coloured plastic 'bulb' in order to use it as an indicator. Red, yellow and green are the usual colours. To limit the current through it to a safe value, a current limiting resistor is always included in series with it.

The circuit for this ultra simple logic probe is shown in Figure 14. It uses only one IC, three resistors, one capacitor, two LED's (of different colours), a 14-pin DIL socket for the IC and a small piece of stripboard. A pair of crocodile clips is needed for hooking it into the 5V supply of the test circuit and a probe of sorts is required. The probe was a gold plated PCB pin attached by means of two wire loops fitted to the board and soldered in place, this is illustrated in photo 1. Anyone who fancies boxing it up is welcome to do so. However, it isn't really necessary; after all, there are no more than 5 volts on the circuit board. To test it after completion proceed as follows.

Connect a 5V supply to the clips. Touch the probe tip to +5V and LED1 should light up. Touch the probe tip to the 0V line and LED2 should light up instead. They should never both be on together. That's all.

The circuit works as follows. The 7404 IC contains six logic level inverters. Only two are used for the project, wired in series. Thus, a signal applied to the input of the first inverter (pin 1) appears inverted at pin 2 and is inverted again by the second inverter to reappear at its original logic level at pin 4. When a logic 0 level appears at the output of an inverter, it causes the LED at that output to light up. Thus a logic 0 input causes LED2 to light up (since two inversions gives a logic 0 output from the second inverter), while LED1 is unlit (since the output from the first inverter is logic 1). If, instead, the input is logic 1, then LED1 (red) will light up and LED2 will be unlit. In this way, a simple indication of logic level is obtained, sufficient for investigating the basic properties of a wide range of digital circuits.

A limitation of this circuit, a result of its extreme simplicity, is that the logic 1 LED is always lit, even when the probe is not on the test point. However, touching it on logic 0 (e.g. the 0V supply line) should then cause it to go out and the logic 0 LED to light instead. Furthermore, if the probe is connected to a slow enough pulse train, such as is used when testing counters and registers, the two LED's will flash alternately at the pulse rate. Thus, this 'logic probe' is adequate for the type of testing envisaged.

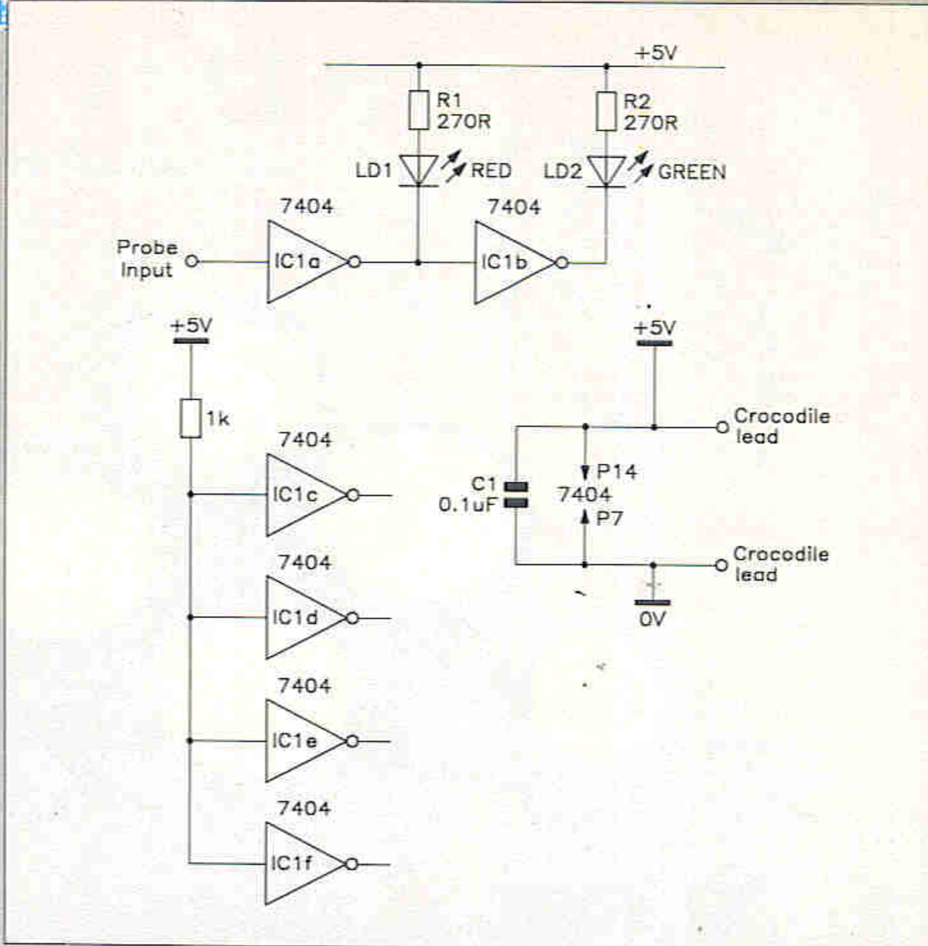


Figure 14. A simple logic probe project.

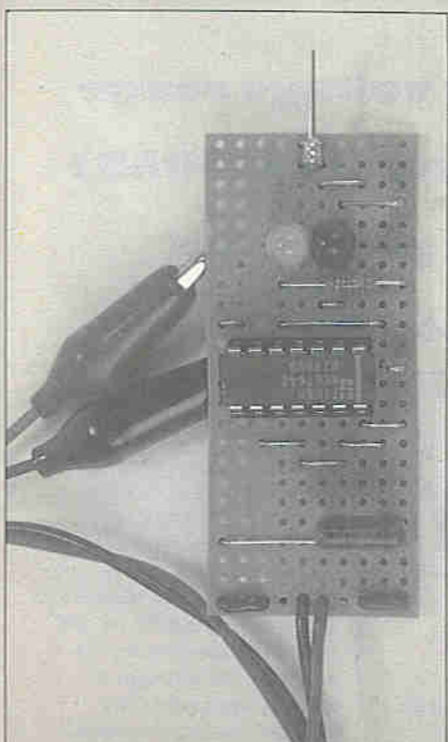


Photo 1. The assembled logic level indicator.

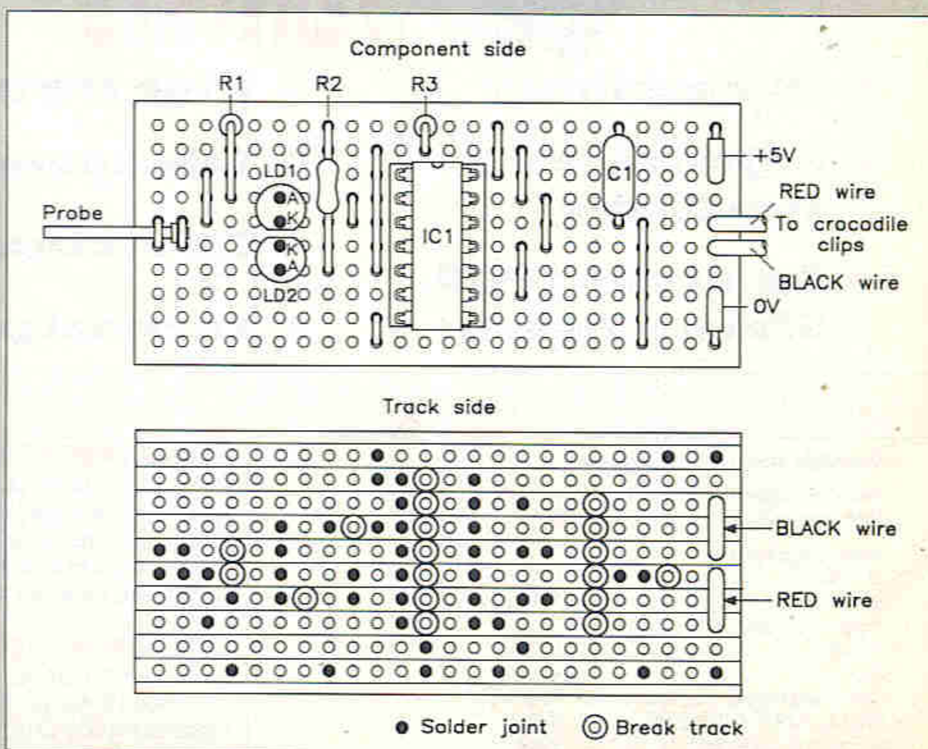
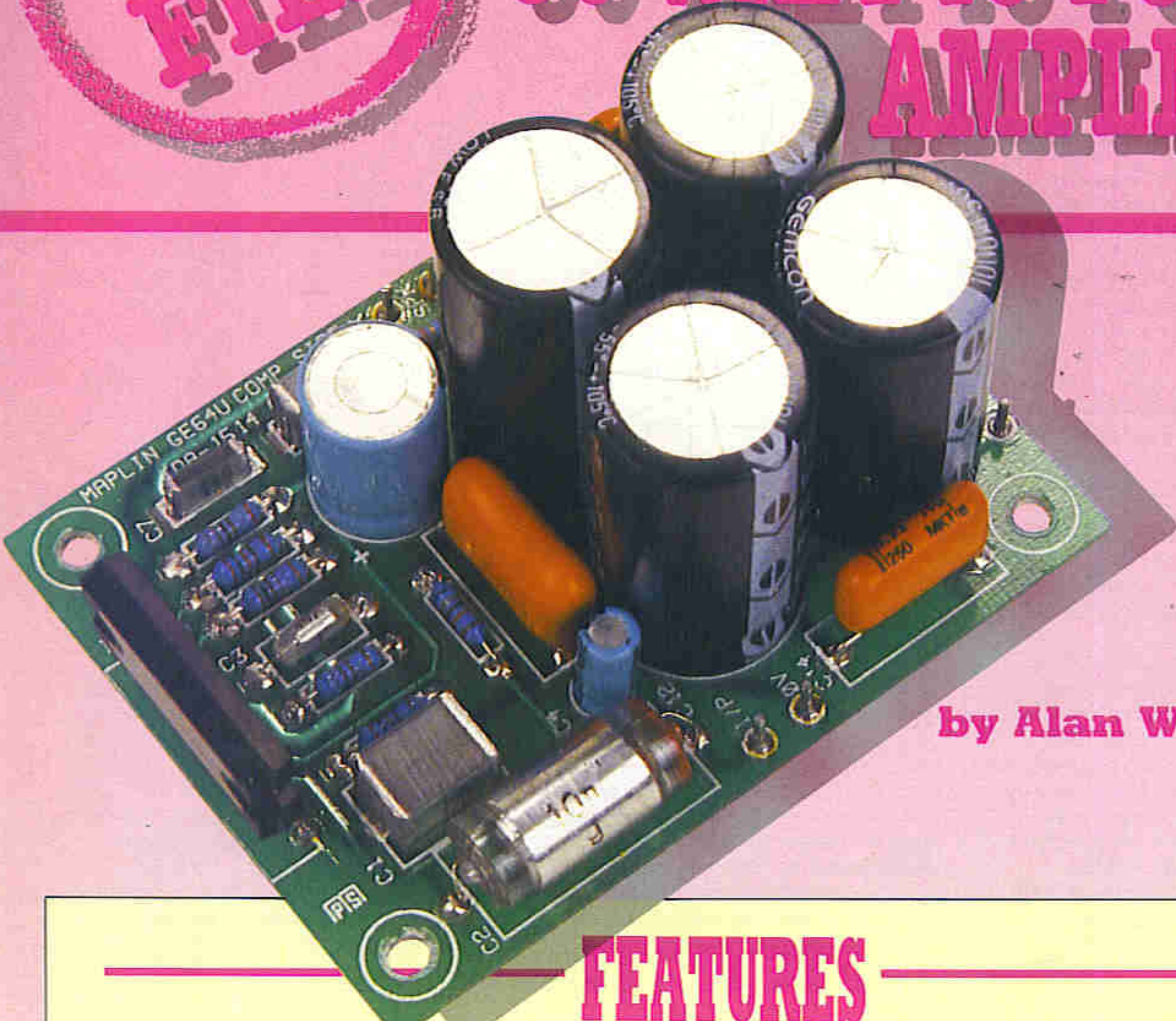


Figure 15. Stripboard layout for logic probe project.

DATA FILE

TDA1514A 50 WATT IC POWER AMPLIFIER



by Alan Williamson

FEATURES

- ★ Star earthing
- ★ Adjustable input sensitivity
- ★ Small size (PCB, 57mm x 76mm)
- ★ Wide supply voltage range
- ★ Safe operating area (SOAR)
- ★ Short circuit protection
- ★ Thermal protection

Specification of Prototype

Test Conditions: $V_P = \pm 27{\cdot}5V$, R3 = 20kΩ, input signal 1kHz sinewave.

Rated output power	
Rl = 8Ω:	39W rms
Rl = 4Ω:	78W rms
Supply current (full output)	
Rl = 8Ω:	2·2A
Rl = 4Ω:	4·4A
Quiescent supply current:	55·2mA
THD at -3dB of full output:	0·1%
Signal to noise ratio:	99dB
Frequency response:	20Hz to 25kHz -3dB

Introduction

The TDA1514A is a high quality power amplifier IC which is com-atible with a wide range of source material, including compact disc.

IC Description

The TDA1514A is supplied in a 9-pin SIL plastic package (SOT131A). The amplifier IC is used with a symmetrical power supply, with or without the bootstrap. The IC features protection against AC and DC short circuits when used with symmetrical supplies. The IC also includes an output mute circuit preventing 'clicks' and 'pops' during switch on and switch off, eliminating the possibility of damage to delicate speakers. The amplifier is also protected against thermal runaway and includes SOAR (Safe

Operating Area Region) protection making the device almost indestructible. An internal block diagram of the device is shown in Figure 1, and the pin out details are shown in Figure 2. Graph 1 shows the SOAR protection characteristics and Graph 2 shows the thermal derating curve.

Circuit Description

The amplifier uses the circuit shown in Figure 3. The circuit also includes the previously mentioned bootstrap. If the circuit is used without the bootstrap, pin 7 must be connected to pin 6 and the associated components (R4, R5, C5, C6) removed; the power output will be reduced by approximately 4 watts.

The input impedance of the ampli-fier module is 20kΩ, determined by R1. The gain of the amplifier is adjustable over the range 20dB to 46dB, determined by R2 and R3. See Table 2 to select the required input sensitivity.

Input Sensitivity	R3 Value
300mV	36kΩ
500mV	20kΩ
1V	10kΩ
2V	5k1Ω

Table 2.

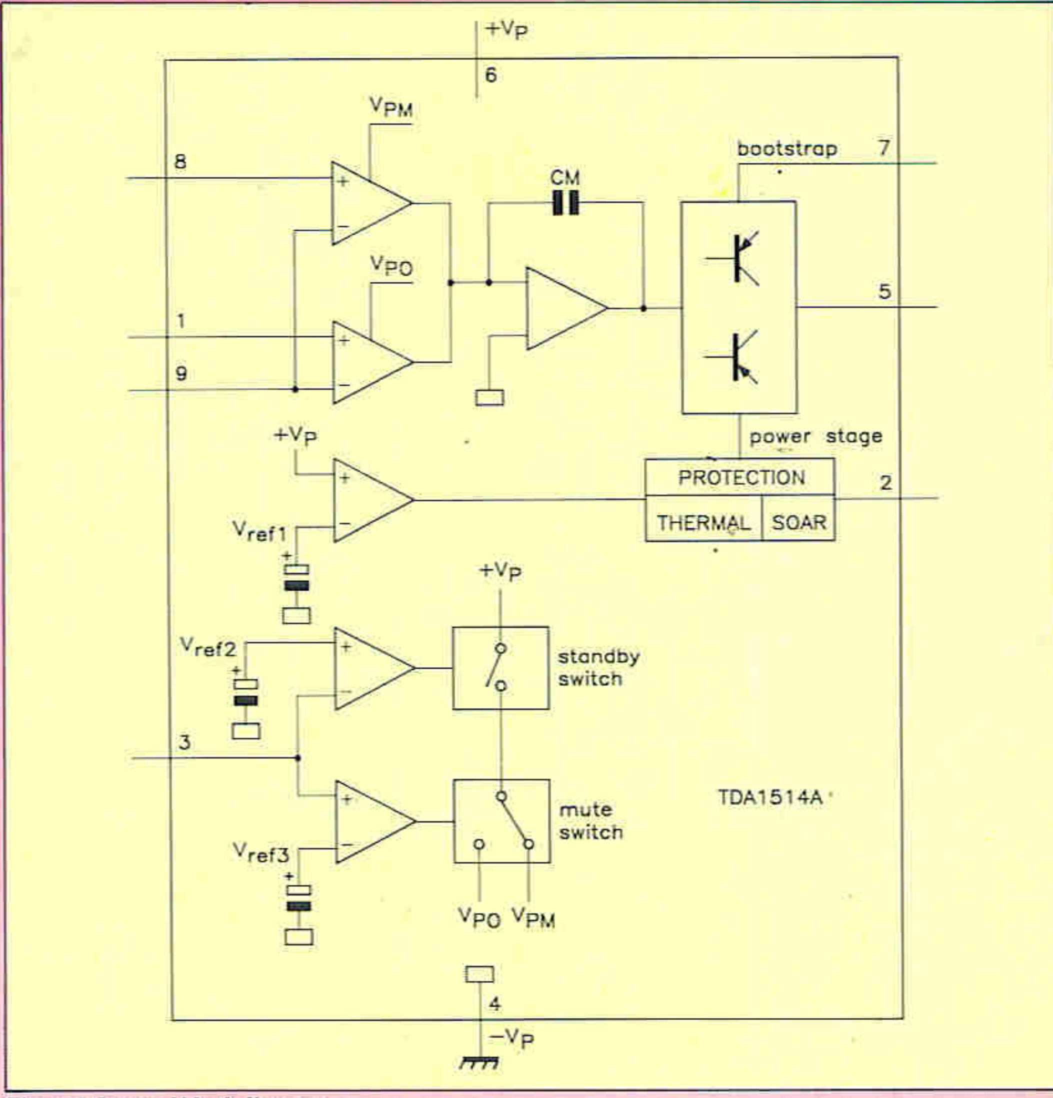


Figure 1. Internal block diagram.

Parameter	Conditions	Symbol	Min	Typ	Max
Supply voltage range pin 6 & pin 4 wrt pin 8		V_P	±9V	—	±30V
Total quiescent current	$V_P = \pm27{\cdot}5V$	I_{tot}	—	60mA	—
Output power (rms)	THD = −60dB $V_P = \pm27{\cdot}5V$ $R_L = 8\Omega$	P_o	—	40W	—
	$V_P = \pm23V$ $R_L = 4\Omega$	P_o	—	60W	—
Closed loop voltage gain	Determined externally	G_c	—	30dB	—
Input resistance	Determined externally	R_i	—	20kΩ	—
Signal plus noise to noise ratio	$P_O = 50mW$	(S + N) ÷ N	—	82dB	—
Supply voltage ripple rejection	$f = 100Hz$	SVRR	—	72dB	—

wrt = with respect to.

Table 1. Quick Reference Data.

IC Power Supply Requirements

The IC has a wide supply range, the minimum requirements being ±9V to a maximum of ±30V. To deliver an 50W into a 4Ω the amplifier requires a 23V supply and to deliver 40W into 8Ω the amplifier requires a 27·5V supply; see Table 1.

Printed Circuit Board

A high quality glass fibre printed circuit board, with a silk screened component legend to aid construction is available, the order code is GE64U. The PCB has a 0V star earth to minimise current return interaction from the speaker, amplifier and input, this will also help to maximise sound quality. Figure 4 shows the PCB and component legend. Figure 5 shows the required leg shape and drilling details for mounting the TDA1514A IC onto a heatsink.

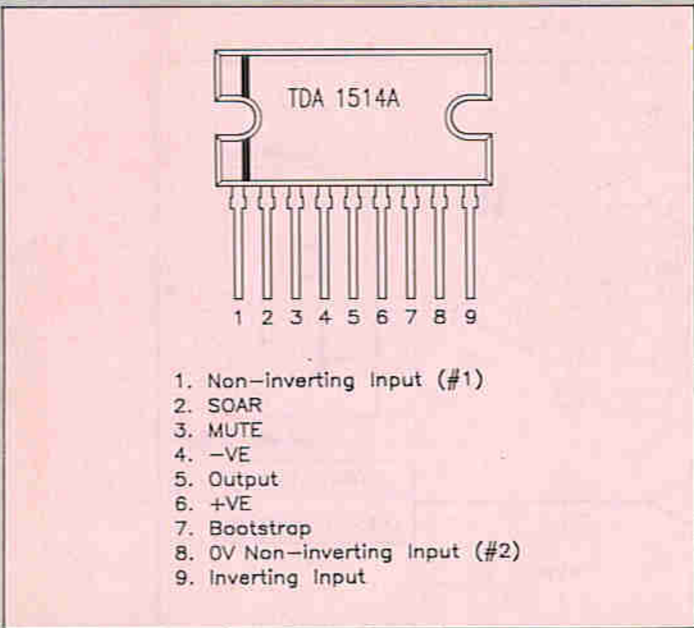


Figure 2. IC pinout.

Wiring details

Figure 6 shows the module with the minimum power supply required, which consists of a transformer and bridge rectifier. Figure 7 shows the module wired up to an existing supply, note the speaker 0V returns to the main power supply.

Power supply

The amplifier requires a supply of ±27·5V, preferably smoothed. The absolute maximum voltage being ±30V, which should be avoided if driving a load of less than 8Ω. The supply must be capable of delivering peak currents of 8 amps or more, otherwise distortion will occur at full volume during loud passages and bass transients.

Thermal rating

Note: Thermal resistance is shown written $K.W^{-1} \equiv K/W$ (kelvin per watt).

The theoretical maximum power of dissipation for an output power of 40 Watts is:

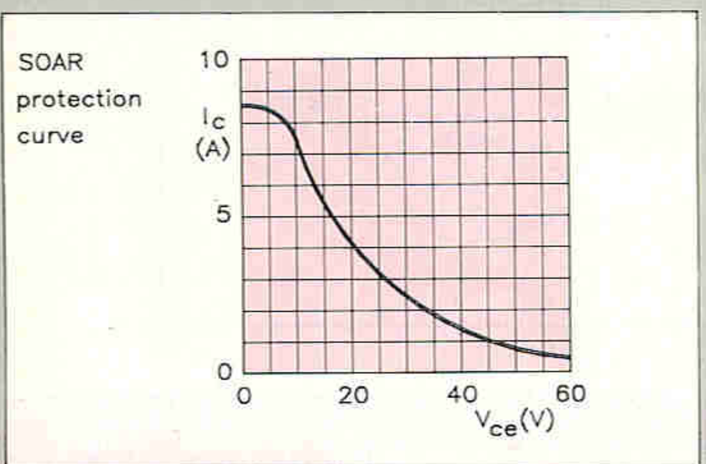
$V_P^2 \div (2 \times \pi^2 \times R_L) = 19W$
where $V_P = \pm 27{\cdot}5V$; $R_L = 8\Omega$.

Thermal resistance from junction to base: $R_{th\ j\text{-}mb} = 1K.W^{-1}$

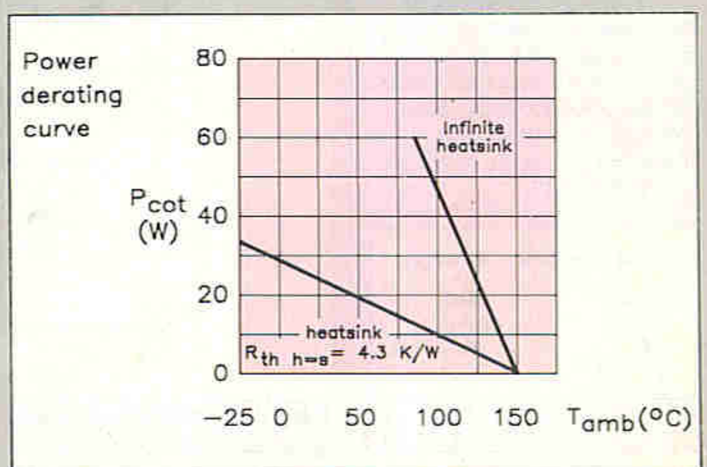
With an ambient temperature of 50°C and a maximum junction temperature of 150°C, the total thermal resistance is:

$R_{th-j} = (150 - 50) \div 19 = 5{\cdot}3K.W^{-1}$

Taking into account power dissipation of the package ($1K^{-W}$):



Graph 1. SOAR protection.



Graph 2. Thermal derating.

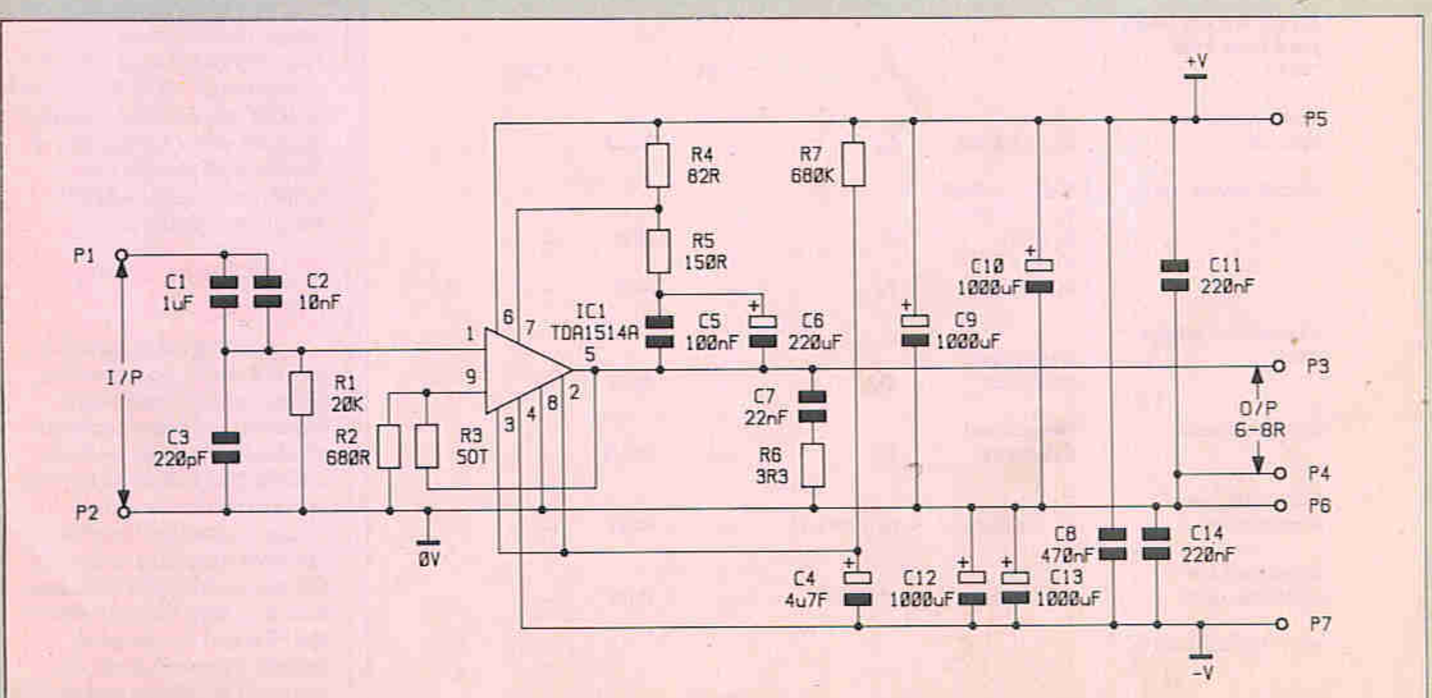


Figure 3. Application circuit.

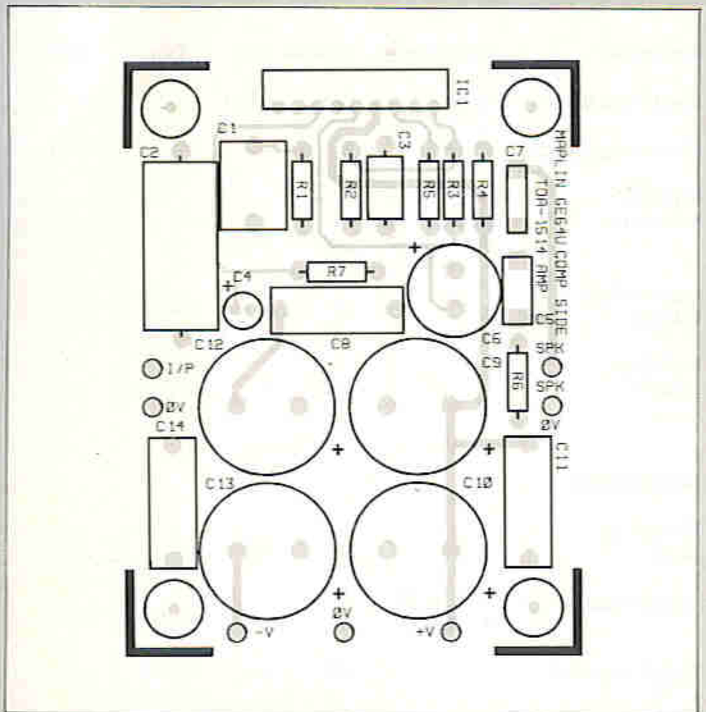


Figure 4. PCB component legend.

Kit

A complete kit of parts with four different value R3's is available, kit number LP43W. The amplifier will require a heatsink (not included in the kit) with a thermal rating of $4{\cdot}3K.W^{-1}$ or less; heatsink type 2E ($2{\cdot}1K.W^{-1}$) is ideal, the order code is HQ70M. An insulating washer must be used between the IC and heatsink if the heatsink is electrically connected to earth, 0V or any supply line, a suitable washer is available the order code is UL74R. The TDA1514A should be secured using either M3 bolts or No. 4 self-tapping screws of suitable length. Care should be exercised to avoid overtightening fixings, which will result in damage to the plastic package.

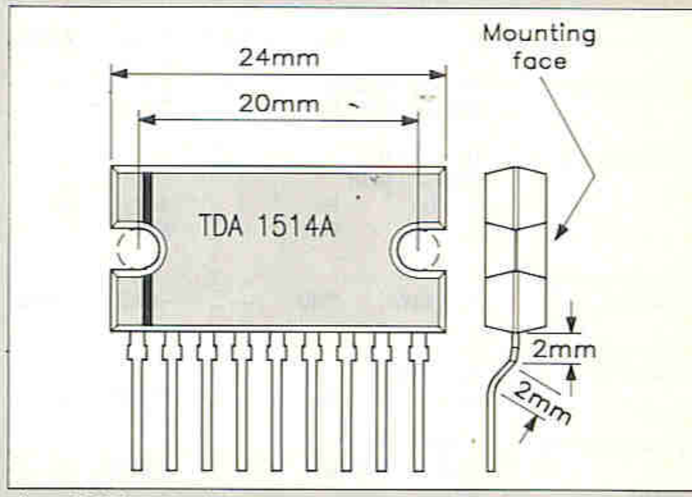
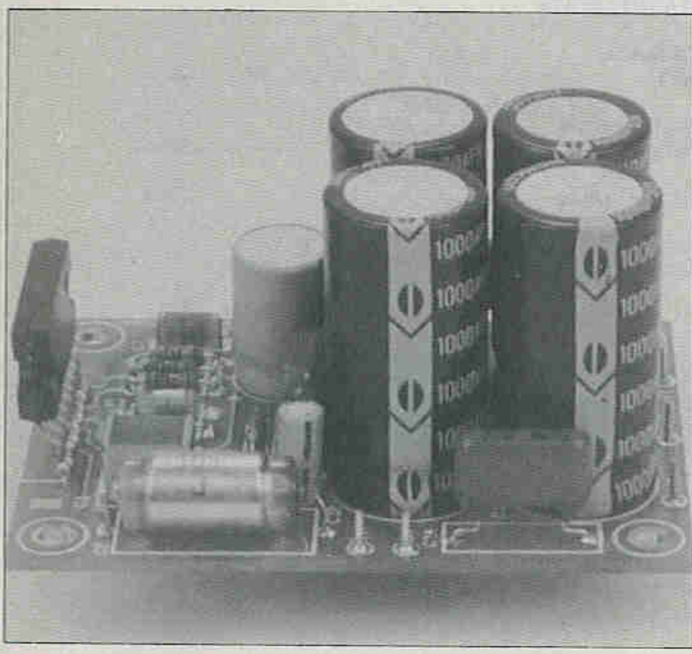


Figure 5. IC leg shaping, heatsink mounting



$$5{\cdot}3 - 1 = 4{\cdot}3K.W^{-1}$$

A heatsink of thermal resistance $4{\cdot}3k.W^{-1}$ or less is required.

Applications

The TDA1514A amplifier kit was designed to provide a small, low cost Hi-Fi equipment to be easilly built.

Applications include active speak-ers; achieved by removing the cross-over unit from the speaker enclosure and powering each speaker directly from an amplifier. A suitable electronic crossover is then fitted between the preamplifier and power amplifiers.

Other applications for the TDA1514A module range from building a small 'midi' amplifier to replacing blown power amp IC's that are difficult to obtain, by simply picking up the signal at the preamp output, and using the existing supply.

The amplifier module is also ideal for use in high power car stereo applications, a switch mode power supply would be required to step up the 12V car supply to $\pm 25V$. See 'Switched Mode Power Conversion — The Secrets Revealed' articles and watch this space for a practical design.

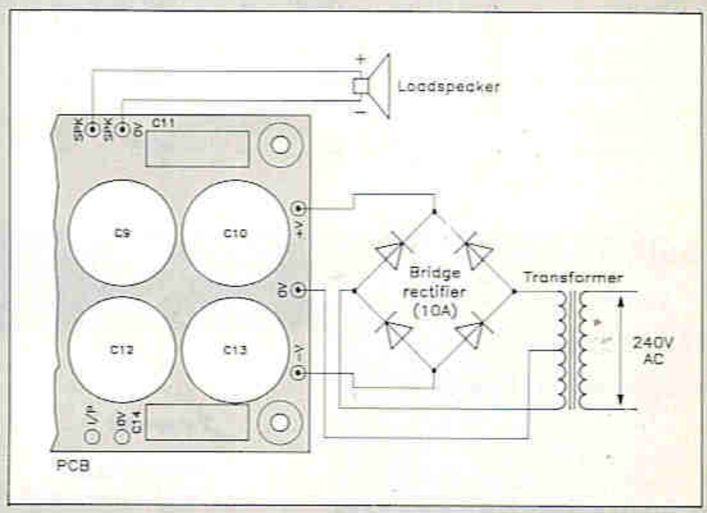


Figure 6. Minimum supply requirements.

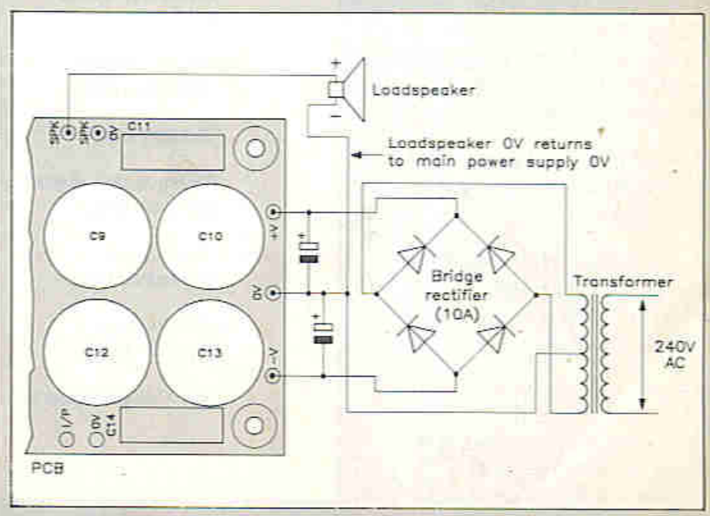


Figure 7. Using an existing supply.

Parameter	Conditions	Symbol	Min	Typ	Max
Supply voltage range pin 6 & pin 4 wrt pin 8	V_P	±9V	—	±30V	
Maximum output current (peak value)	I_{OM} max	6·4A	—	—	
OPERATING STATE					
Input voltage pin 3 to pin 4	$V_{3\text{-}4}$	6V	—	7V	
Total quiescent current	Rl = ∞	I_{tot}	30mA	60mA	90mA
Output power	THD = −60dB	P_o	37W	40W	—
	THD = −20dB	P_o	—	51W	—
Output power	V_P = ±23V THD = −60dB Rl = 8Ω	Po	—	28W	—
	Rl = 4Ω	Po	—	50W	—
Total harmonic distortion	P_o = 32W	THD	—	−90dB	−80dB
Intermodulation distortion	P_o = 32W note 1	d_{im}	—	−80dB	—
Power bandwith 25kHz	−3dB THD = −60dB Hz	B	—	20Hz to	—
Slew rate		dV/dt	—	$10V.\mu s^{-1}$	—
Closed loop voltage gain	note 2	G_c	—	30dB	—
Open loop voltage gain		G_o	—	85dB	—

Parameter	Conditions	Symbol	Min	Typ	Max
Input impedance	note 3	Z_i	1MΩ	—	—
Output impedance		Z_o	—	—	0·1Ω
Signal to noise ratio	note 4 Po = 50mW	S/N	80dB	—	—
Output offset voltage		V_o	—	2mV	—
Input bias current		I_i	—	−0·1µA	—
MUTE STATE					
Voltage on pin 3		$V_{3\ 4}$	2V	—	4·5V
Output voltage	$V_{I(rms)}$=2V f=1kHz	V_o	—	100µV	—
Ripple rejection		RR	—	70dB	—
STANDBY STATE					
Voltage on pin 3		$V_{3\ 4}$	0V	—	1V
Total quiescent current		I_{tot}	—	20mA	—
Ripple rejection		RR	—	70dB	—
Supply voltage to obtain standby state		$\pm V_P$	4·5V	—	7·0V

Table 3. IC Data.

Notes to characteristics

1. Measured with two superimposed signals of 50Hz and 7kHz with an amplitude ratio of 4:1.

2. The closed loop gain is determined by the resistors R2 & R3, the gain is adjustable between 20 and 40dB.

3. The input impedance of the circuit is determined by the bias resistor R1.

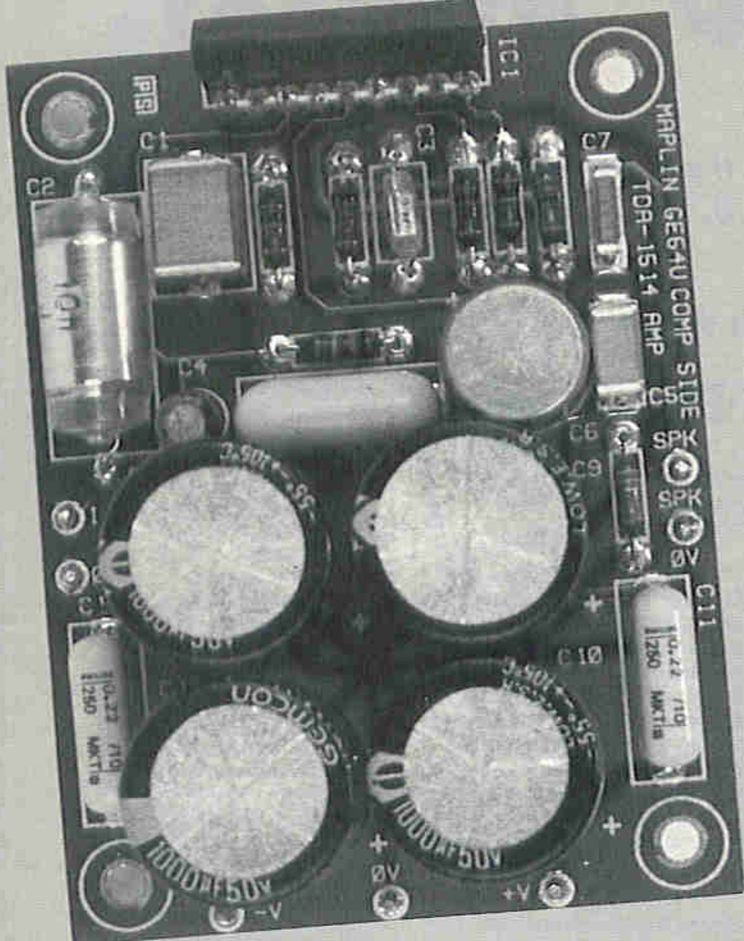
4. The noise output voltage is mea-sured in a bandwith of 20Hz to 20KHz with a source resistance of 2kΩ.

5. The quiescent current into pin 2 determines the minimum supply volt-age at which the mute function remains in operation, $V_p - V_n = I_{2\,tot} \times R4 + V_{m\,(on)\,max}$.

Parameter	Symbol	Min	Max
Supply Voltage pin 6 & pin 4 wrt pin 8	V_P	—	±30V
Bootstrap voltage pin 7 wrt pin 4	V_{bstr}	—	70V
Output current repetitive peak	I_0	—	8A
Storage temp	T_{stg}	−65	+150°C
Thermal shut-down protection time	t_{pr}	—	1 hour
Short circuit protection time	t_{sc}	—	10 minutes
Mute voltage pin 3 wrt pin 4	V_m	—	7V

wrt = with respect to.

Table 4. Maximum ratings.



TDA1514 POWER AMP PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	20k	1	(M20K)
R2	680R	1	(M680R)
R3	36k (See text)	1	(M36K)
R3	20k (See text)	1	(M20K)
R3	10k (See text)	1	(M10K)
R3	5k1 (See text)	1	(M5K1)
R4	82R	1	(M82R)
R5	150R	1	(M150R)
R6	3R3	1	(M3R3)
R7	680k	1	(M680K)
CAPACITORS			
C1	1μF Polylayer	1	(WW53H)
C2	10nF Polystyrene	1	(BX92A)
C3	220pF Polystyrene	1	(BX30H)
C4	4μ7F 63V PC Electrolytic	1	(FF03D)
C5	100nF Polylayer	1	(WW41U)
C6	220μF 63V PC Electrolytic	1	(FF14Q)
C7	22nF Polylayer	1	(WW33L)
C8	470nF Polyester	1	(BX80B)
C9,10,12,13	1000μF 50V SMPS	4	(JL57M)
C11,14	220nF Polyester	2	(BX78K)
SEMICONDUCTOR			
IC1	TDA1514A	1	(UK75S)
MISCELLANEOUS			
P1-7	Pin 2145	1 Pkt	(FL24B)
	PC Board	1	(GE64U)
	Constructors Guide	1	(XH79L)
OPTIONAL (not in kit)			
	Insulator T0218	1	(UL74R)
	Heatsink Type 2E	1	(HQ70M)
	Self-tapping screw No.4 x ⅜ in.	1 Pkt	(BF65V)

The above items, excluding Optional, are available as a kit.
Order As LP43W (TDA1514 Power Amp) Price £15.95
The following item is also available separately:
PC Board **Order As GE64U Price £4.95.**

MAPLIN'S TOP TWENTY KITS

THIS MONTH	LAST MONTH		DESCRIPTION OF KIT	ORDER CODE	KIT PRICE	DETAILS IN PROJECT BOOK
1.	(2)	♠	150W MOSFET Amplifier	LW51F	£19.95	Best of E&MM
2.	(1)	♠	Digital Watch	FS18U	£1.98	Catalogue
3.	(4)	♠	Live Wire Detector	LK63T	£3.95	14 (XA14Q)
4.	(3)	♠	Car Battery Monitor	LK42V	£8.95	37 (XA37S)
5.	(5)	♦	I/R Prox. Detector	LM13P	£9.95	20 (XA20W)
6.	(10)	♠	Car Burglar Alarm	LW78K	£9.95	Comp 2 (XC02C)
7.	(12)	♠	PWM Motor Driver	LK54J	£9.95	12 (XA12N)
8.	(16)	♠	LM386 Kit	LM76H	£3.75	29 (XA29G)
9.	(6)	♠	Partylite	LW93B	£9.95	Best of E&MM
10.	(11)	♠	8W Amplifier	LW36P	£5.95	Catalogue
11.	(14)	♠	U/Sonic Car Alarm	LK75S	£19.95	15 (XA15R)
12.	(-)	RE-ENTRY	TDA7000 Radio MKII	LM55K	£19.95	27 (XA27E)
13.	(7)	♠	Siren Sound Generator	LM42V	£4.25	26 (XA26D)
14.	(8)	♠	Mini Metal Detector	LM35Q	£5.25	25 (XA25C)
15.	(-)	RE-ENTRY	Car Digital Tacho	LK79L	£19.95	37 (XA37S)
16.	(9)	♠	TDA2822 Stereo Power Amp	LP03D	£6.45	34 (XA34M)
17.	(15)	♠	15W Amplifier	YQ43W	£6.75	Catalogue
18.	(-)	NEW ENTRY	Audio Controlled Switch	LP29G	£5.95	38 (XA38R)
19.	(-)	RE-ENTRY	Laser & PSU	LM72P	£99.95	29 (XA29G)
20.	(13)	♠	Watt Watcher	LM57M	£3.95	27 (XA27E)

Over 150 other kits also available. All kits supplied with instructions.
The descriptions above are necessarily short. Please ensure you know exactly what the kit is and what it comprises before ordering, by checking the appropriate Project Book mentioned in the list above.

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A readers forum for your views and comments. If you want to contribute, write to the Editor, 'Electronics – The Maplin Magazine', P.O. Box 3, Rayleigh, Essex, SS6 8LR.

Pie in the Sky

Dear Sir,
I am very disappointed that the mag has descended to doing a 'puff' about Sky Television just to get a prize offer. It's not worthy of you. We expect articles with a technical bias and an unbiased assessment. Even ones describing kits to unscramble 'film' channels! Instead we got a very biased view of Sky – no mention of the higher technical standards demanded of BSB by the British authorities; the fact that Sky is foreign owned and is still losing money apart from the fact that if the BBC could demand £10 per month, so effectively doubling the licence fee, there would be an uproar. So information describing the technical differences between Astra and BSB would be welcome, but an assessment of their programmes is best left to specialist magazines and experienced TV journalists.
S. Casey, Stockport.

The article made no claims to be technical, it was intended as a general interest article. I'm afraid we cannot publish details of kits to descramble satellite TV channels, if we did 'Electronics' would not be around for very long! The merits of DMAC transmissions over PAL were not at issue here, but it is hoped to deal with satellite TV transmission in a technical article. Perhaps other readers would like to comment on the general feature articles that we publish and the competitions we run. After all we don't have to include free to enter competitions with good prizes...

Feedback on Switched Mode PSU's

Dear Sir,
With reference to your request for feedback on switch mode power supplies I have the following suggestion for a project. To enable to charge a 10 cell Ni-Cd battery pack a voltage of around 24 volts is required, the only way to obtain this from a car battery is via a voltage doubler. After browsing a data sheet for a μA78S40 switching regulator a unit was constructed using a 78S40, L200 and a LED in series with the output to indicate correct operation. This unit will now charge up to 10 cells at 40mA from about 7 volts input (the current falls off at 7 volts). I have considered rebuilding this unit but have been unable to obtain any 78S40's from your stores recently. How about a decent project for all those radio amateurs, radio control modellers whose only source of power is a car battery.
A. Lether, Preston.

I must admit that we haven't thought of using a switched mode doubler circuit to charge Ni-Cd's from a 12V supply, it is a good idea and has been passed onto the lab. The supply problem of the μA78S40 is due to National Semiconductor Corporation halting production to improve the specification of this device; by now supplies should be back to normal.

★★★★★★★★★★★★★★★★★★★★★★

S·T·A·R L·E·T·T·E·R

This issue, Dr S. W. Bateson from Cleveland receives the Star Letter Award of a £5 Maplin Gift Token for his letter on the SSM2015 Microphone Preamplifier.

Microphone Preamplifier

Dear Sir,
I am writing regarding two errors in this otherwise very useful project. The first is I think a simple misreading of the data sheets; resistors R2 and R3 should be the usual phantom powering value of 6k8, not 68k, since phantom powered microphones consume up to about 2mA each. The second is more interesting and concerns the values chosen for the input coupling capacitors C4 and C5. The value (1μF) appears to have been chosen in conjunction with R5 and R4 in order to form a low frequency – 3dB point of 16Hz, as is usual with line level amplifiers. This means that at 16Hz the apparent impedance of the capacitor is also 10k, so the source impedance as seen by the amplifier rises at low frequencies. This causes a significant increase in low frequency noise from two sources: 1. voltage (Johnson) noise; $V_N = (4kTRB)^{0.5}$ where R is the input impedance seen by the amplifier; and 2. current noise from the 2015 (up to 2pA $Hz^{0.5}$) appearing across the input impedance. These facts mean that the impedance of the input capacitors should be very small over the whole operating bandwidth, that is, the capacitors should be as large as possible. 100μF, 63V would be satisfactory. If phantom powering in unused and direct coupling is acceptable (e.g. with dynamic microphones) it is better to omit (bypass) C4 and C5 altogether. The result will be a dramatic reduction in low frequency noise. The SSM2015 is capable of excellent performance and I feel that your kits and the article should be amended to take advantage of this. It may also be worth mentioning that the best performance will be obtained from a separate stable and low noise power supply since it is all to easy to couple power supply noise into circuit ground lines.

Gavin Cheeseman from the Design Laboratory replies:
I would like to thank Dr Bateson for taking interest in the project and for raising the above points. Firstly, I would like to point out that the Datafile series of projects are primarily building block circuits and are designed as basic modules which may be modified to suit a variety of applications. With this in mind the projects are generally not optimised for any particular use and some components are given arbitrary values.
Although, the module will work satisfactorily in many applications using the components supplied in the kit, the point regarding the value of input capacitors C4 and C5 is very valid and as Dr Bateson suggests, where appropriate these may be linked out when the phantom powering part of the circuit is not used. Larger value capacitors could also be fitted but this may produce undesirable results by coupling unwanted low frequency signals into the system from some audio sources.

★★★★★★★★★★★★★★★★★★★★★★

Thumbs Up for Maplin Service!

Dear Sir,
Last week a friend lent me some recent copies of the Maplin Electronics magazine in which I was surprised to read letters from customers complaining of the Maplin service. I buy electronic components, purely for amateur use as I am retired, from several mail order firms and I can say without contradiction that Maplin are top of the league table for speed and quality of service. A couple of weeks ago I ordered an oscilloscope from you using one of your pre-paid envelopes and was amazed when the item arrived four days later which included the weekend. Again only last week I ordered some items again by post and the parcel arrived in less than 48 hours. Compare this service with that of a firm in Surrey which took nearly four weeks to supply items which were held in stock. However the record for slow delivery goes to a company in Bristol which on one occasion took eleven weeks again to supply components WHICH WERE HELD IN STOCK. There are times when Maplin run out of stock of certain items and this can be quite frustrating especially as Murphy's law ensures that the missing item is the one most urgently required but even Maplin cannot be held responsible for delays in deliveries from its own suppliers. Should you receive any more complaining letters about your service you are at liberty to quote this letter at the senders.
R. Christian, Cheshire.

Thank you for taking the time to write in, it is always a pleasure to receive complimentary letters!

Beginners' Projects Wanted

Dear Editor,
I read 'Electronics – The Maplin Magazine' everytime it is published but I soon realised that as I am only 13, a lot of the projects are either too hard and a bit over my head or are no use at all to me. This is a disappointment as the rest of the Magazine is VERY good. I hope I am not the only one complaining on this point, but I feel that you might get a lot more younger readers reading your magazine if you included some simpler projects.
M. Bridgstock, Cambs.

Maintaining a good balance of projects, which cover abilities from beginner to expert, is not easy. But we always endeavour to provide the best balance possible. The content of the magazine is determined by the feedback we receive from our readership, so if you have any project ideas; whether aimed at beginner or expert, please let us know. 'Electronics – The Maplin Magazine' is YOUR magazine, help us to help you by writing in.

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Specifically, this book aims to provide the reader with a unique collection of practical working circuits together with supporting information so that circuits can be produced in the shortest possible time and without recourse to theoretical texts. All circuits described have been thoroughly tested and, as far as possible, a range of commonly available low cost components has been adopted in each case.

The Maplin Electronic Circuits Handbook assumes that the reader has at least an elementary understanding of electrical principles and, in particular, is familiar with common electrical units and quantities. Provided the reader has this knowledge and can perform simple arithmetic calculations, there should be no difficulty in following the mathematics presented here. A selection of the most popular Maplin projects are described in the text and readers can be confident of obtaining the components from Maplin.

Contents:
Passive components; Semiconductors; Power supplies; Amplifiers; Operational amplifiers; Logic circuits; Timers; Computer interfacing; Tools and test equipment; Circuit construction; Selected Maplin projects; Appendices.

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VARIOUS

BABY MONITORS Hear baby's every sound on an FM radio. Matchbox size, long range. Ready for use. £10 each. Mrs M. Hicks, 24 Brewster Road, Boston, Lincs PE21 0DY. Tel: 0205 362003.

GIANT CLEAROUT! Components, IC's, capacitors, resistors – everything must go. Send S.A.E. for list to M.J.D. Blenheim, Walton Lane, Bosham, West Sussex PO18 8QF.

MICROPROFESSOR-1B TRAINING SYSTEM plus applications board, switch/lamp unit and books £150. Kevin Edwards, 19 The Square, Tatsfield, Kent TN16 2AS.

BECKMAN DIGITAL MULTIMETER, fully shockproof and waterproof, £100. Transistor Tester £20. Also various ATA communications and various books for sale. Tel: 051 356 1112.

THANDAR PFM 200A frequency counter 0-200MHz – £30. Veleman K2623 30V/3 Amp variable Power Supply kit, ready assembled, with toroidal – £15. Matsui Video Player, brand new condition – £50. Polaroid Instamatic, spanking new, used only with first film pack – £12. Urgent need of cash. Tel: Derek on 081 690 5159.

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FOR SALE Advance OS1000 dual trace oscilloscope. Complete with manual and probes. Excellent condition. £175 o.n.o. Tel: 0206 226968, evenings only.

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CIRKIT DFM7 Mk II Digital Frequency Meter, 6 months old, £35. Also Denco GP coils, blue range 3-4-5, £6 slightly used. Seon Smyth, "De Porres", 67 East Princes Street, Helensburgh, Scotland G84 7DG.

MAPLIN STEREO DISCO UNIT – unfinished project – needs assembly and wiring. Over £400 of parts including 12V Varispeed decks, 150W Mosfet Amps etc. Offers around £225 + carriage. Tel: 0272 248679 (day).

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FOR SALE I have some periodic tables available. Probably the best available – atomic number, mass, density, discovery date and more!!! Send 3 x 20p coins and A4 SAE. S. Yousaf, 137 The Crescent, Slough, Berkshire SL1 2LF.

FLUKE Digital Meter £50. HV Probe £16. Megger £100. AVO Multiminor £16. Tel: 081 665 2913, 6-8 p.m.

MUSICAL

MAPLIN MATINEE ELECTRONIC ORGANS 2 for sale. Professionally built – £350 each. Also Yamaha PSS-570 keyboard – £150 o.n.o. Tel: Leeds 0532 673251.

MAPLIN MATINEE ELECTRONIC ORGAN in perfect condition. Built by the designer of this instrument. Full specification including 2 × 49 note keyboards, pedal board, 30 rhythms, Draw bar and pre-programmed voicing plus many other effects. Ideal beginners organ. Genuine reason for sale. £175 ono. Tel 0702 339204

COMPUTERS

PRINTER FOR SALE Epson MX80 compatible printer, excellent condition, RS232 & Centronics interface, ideal 1st time buyers printer, with ribbon and brand new spare. £65 working. Tel: Lee on 0702 352638. Buyer collects.

ELEKTOR JUNIOR COMPUTER Uncased but working, with power supply and books. Also Maplin Electronics piano, working but needs tidying up – buyer collects. Offers on Bristol 0272 350054.

190W COMPUTER PSU +5.2V at 6.75A, +13.2V at 0.5A, -13.2V at 1.0A; Fan Cooled, Switched Mode, £30.00 o.n.o. Also various Maplin HQ Mixer Kits, unused. Tel: Adam on 0777 702264 after 5p.m. for details.

WANTED

CIRCUIT, GEN AND SWITCH REAR positions for Monitor-keyboard of Televideo model 950 (USA). Write G2DHV QTHR or phone 081 300 1649.

ANY NAB TYPE Broadcast cartridges for Sonifex cart machines. Must be cheap as possible as for enthusiastic student working in hospital radio. Tel: Alex on 081 777 6241.

ANYONE OUT THERE with a project to tap RS232 serial from the expansion port on an Amstrad PCW? James M. Clark, Flat 169 Hunter House, Walker, Newcastle Upon Tyne NE6 3XH.

MANUAL or any other information on M-basic and Adler Alphatronic P2. Also any software for above. Contact Kimber, 27 Walton Close, Worthing, West Sussex BN13 2BJ.

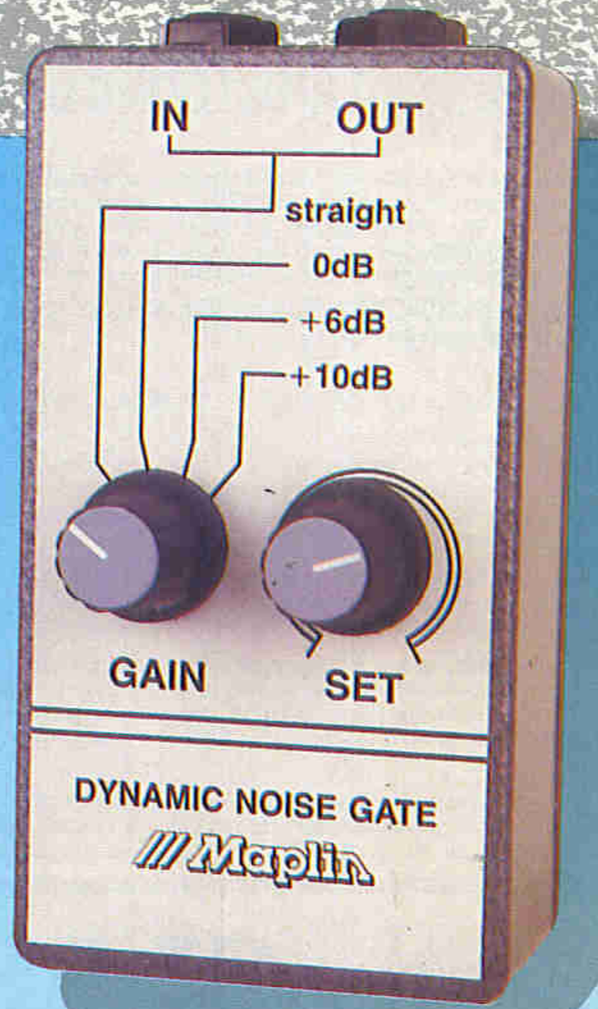
WANTED! Circuit diagram/PCB layout to convert Maplin Wind Speed Module into LED digital readout, if possible with memory. Please send information to Richard Davies, Blackthorns, 16 Dingle Lane, Crundale, Haverfordwest, Dyfed SA62 4DJ.

MY FORTEL CCDHP time base corrector needs repair. Service manual or circuit diagram or any other help wanted for suitable fee. Tel: 0642 583075.

PLASTIC FUNCTION KEYSTRIP wanted for BBC B. Also wanted: Aviator on disc (need not be packaged) and copy of Micro User July 1989. Tel: 0754 67890 after 4 p.m.

2nd Time Around

DYNAMIC NOISE Gate



Revised and Updated by Alan Williamson
Original Design by Dave Roffey

FEATURES

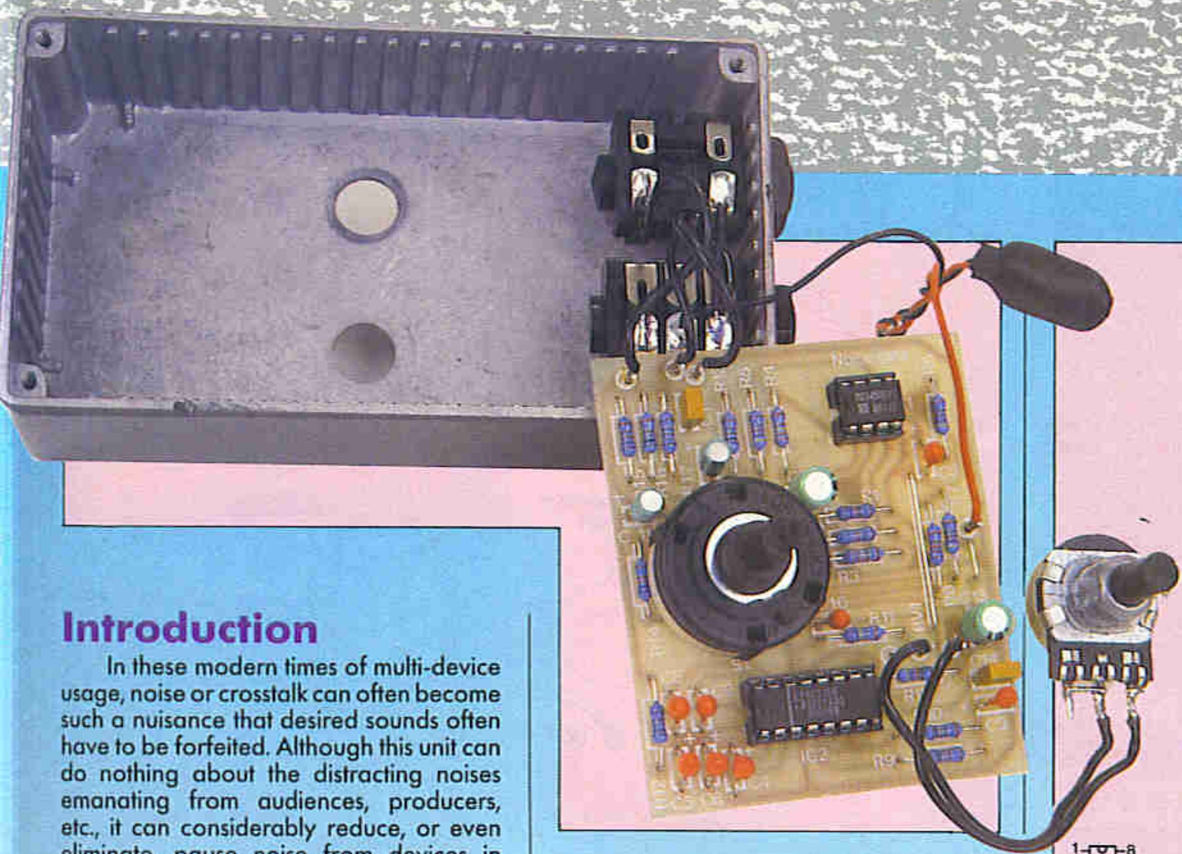
* Provides automatic shutdown of unwanted noise during 'pause' conditions
* Compander technique eliminates 'signal snapping'
* User adjustable characteristics for high or low level network insertion
* Can be used in multi-instrument layouts for instant unit shutdown on changeovers
* Can effectively cancel crosstalk in multi-microphone set-ups
* No circuit trimming required
* Can be used in its own right as an effect to create soft attack 'bowing' characteristic

Specification

Supply:	9V
Current consumption:	6mA
Frequency response:	20Hz - 20kHz
Input (min):	10mV rms
Input (max):	2V rms
Output (max):	1·6V rms

Foreword

Particular projects from the Maplin range have proven to be very popular over the years; but unfortunately, technology and components often change, thus resulting with these projects becoming obsolete. Some of our early publications such as the 'Best of E&MM', will soon be out of print as many of the published projects and kits are now discontinued. However, some projects remain available and will be re-published (with updates and improvements as necessary) under the series title of "2nd Time Around".



Introduction

In these modern times of multi-device usage, noise or crosstalk can often become such a nuisance that desired sounds often have to be forfeited. Although this unit can do nothing about the distracting noises emanating from audiences, producers, etc., it can considerably reduce, or even eliminate, pause noise from devices in which it is otherwise impossible to improve on signal to noise ratios.

Several types of noise gate are available for this kind of noise elimination – 'snap-off' units – programmable types – externally controlled units - low level expansion devices, etc. In order that the unit may have as wide a range of applications as possible, the low level expansion or dynamic gate has been selected. This type tends to be less critical in set-up and general use, giving a more musically acceptable sound entrance and exit than the 'sudden shutdown' units.

Having a wide range of user adjustable characteristics, it should find many a useful working place with, for example, guitar/organ/keyboards levels to mixer desk/P.A./recording levels. Not only can it be used for its main purpose, that of closing down noise or unwanted signals below a selected level, but as an effect in its own right, creating soft attack, bowing type characteristics.

Noise gates have been in use for considerably longer than most people would imagine. In fact, the first application of these devices was in the 1930's, when

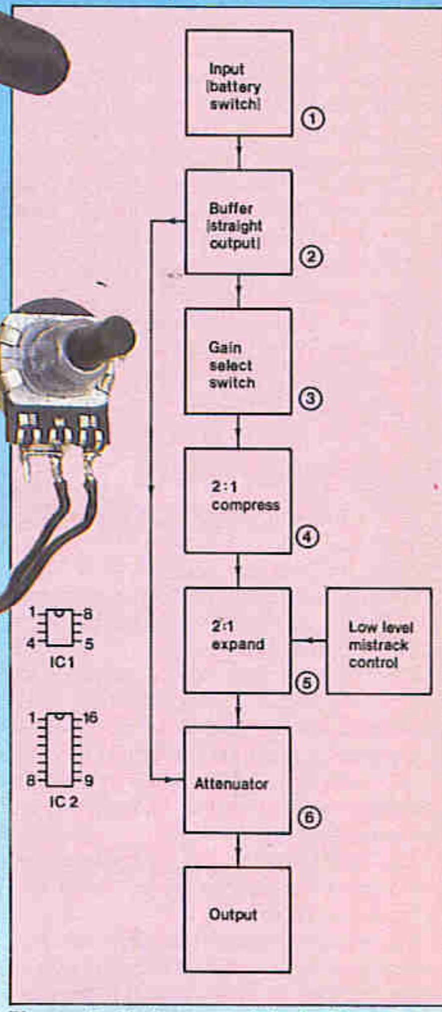


Figure 1. Noise gate block diagram.

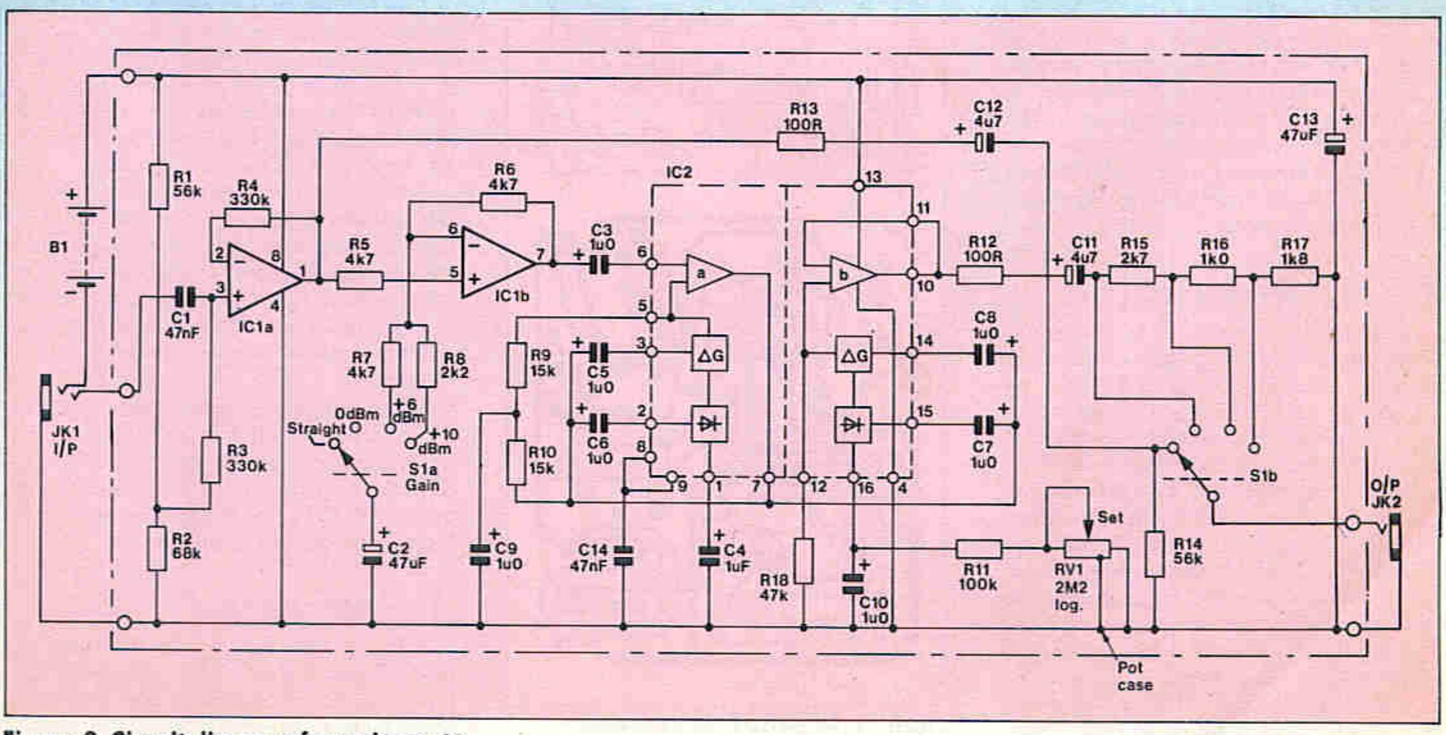


Figure 2. Circuit diagram for noise gate.

they were used to reduce unwanted cracks and pops from soundtrack film that had become dusty or scratched.

The most popular use for the dynamic noise gate is that of removing undesirable noise during pauses in live or recorded performance. To achieve this, the threshold (the level at which the incoming signal operates the gate) would normally be set just under the required signal level.

The unit is easily constructed and does not require any 'setting up'. As a self contained unit, it can be used in different applications at will, although being very compact it may be easily panel mounted for 'one-off' noise conditioning.

Basic Functions

Referring to Figure 1, take note of the following:

1. When a mono jack plug is inserted into the stereo jack socket (JK1), the plug will short the socket ring and sleeve contacts together to effect power up.
2. A high input impedance stage is used to reduce any input device loading. The output of this stage serves also as the unity gain stage for pre-post comparisons.
3. A switched gain stage provides adjustable sensitivity and allows flexible device usage.
4. The first stage of the dynamic noise gate's active circuity consists of a fixed 2:1 compression network.
5. The second stage of this active circuitry consists of a 1:2 expansion network with a simple resistive control element (potentiometer) for adjustable low level mistracking. This action is responsible for the total characteristics of the unit.
6. A passive attenuator is switched in conjunction with block 3, so that an overall 1:1 input/output level is maintained. A straight-through route is included in the switch position which allows for comparison tests. For those of you interested in such details, phase inversion between input and output does not occur.

Circuit Description

Referring to Figure 2, audio signals enter the non-inverting input of IC1a via coupling capacitor C1. IC1a is biased by R1 and R2 via R3, the input impedance is (R1 // R2) + R3. The voltage set by R1 and R2 ensures that outputs from IC1a & IC1b will be evenly clipped at maximum levels. (At lower than normal battery levels maximum positive excursions will be greater than available negative excursions due to internal circuitry.) R4, connected in the negative feedback line, equalises input biasing (R3 = R4) for minimum output offset in IC1a.

IC1b is also used in the non-inverting mode and has gain levels of 0dB, +6dB and +10dB, selected via S1a, which introduces feedback reducing resistors R7 and R8.

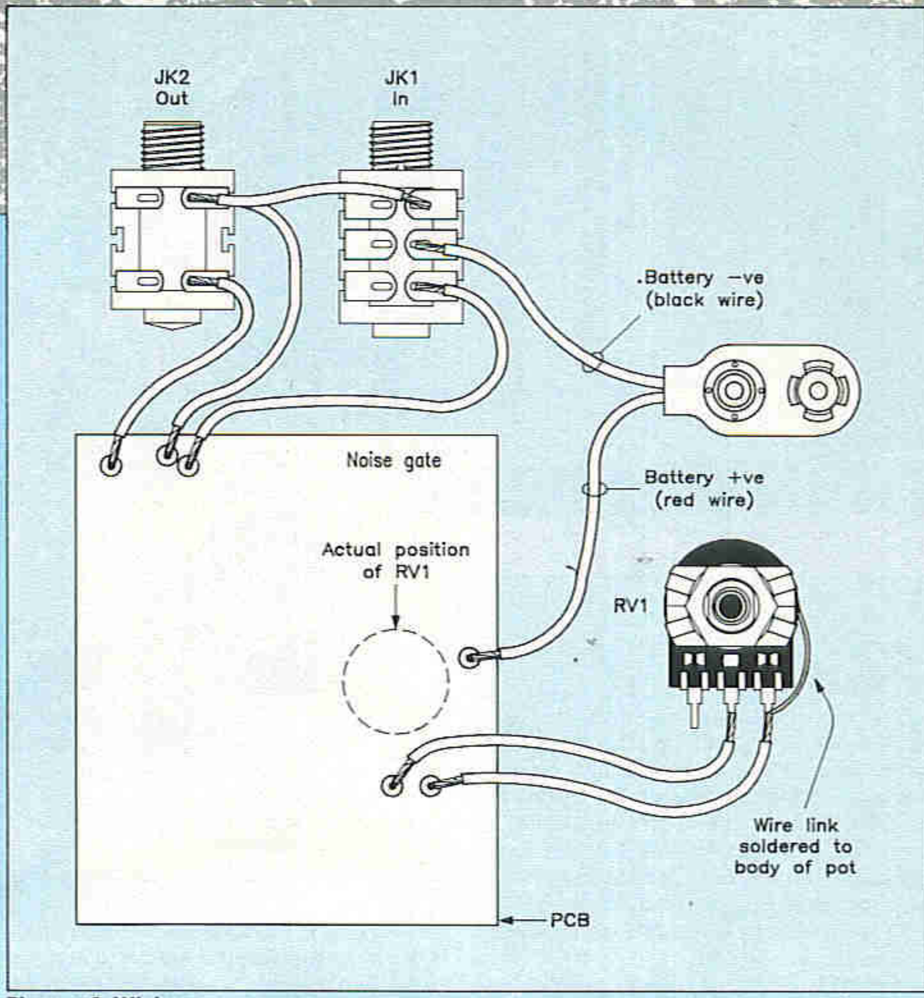


Figure 4. Wiring.

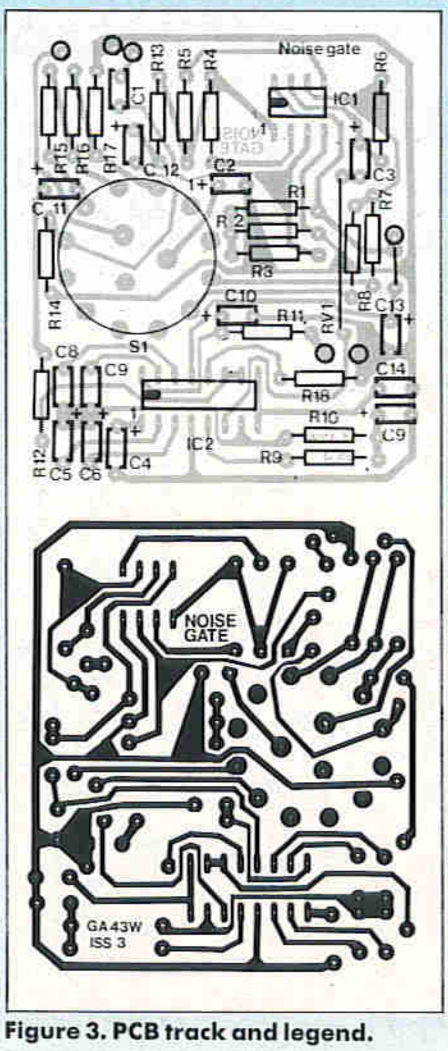


Figure 3. PCB track and legend.

Entry to the Compander section is now effected via C3, with compression achieved through IC2a and expansion by IC2b. All components in this section have been selected for the best overall performance in terms of frequency, speed of operation and distortion when bearing its 'musical' application in mind.

In the expansion section, RV1 is the resistive control element and operates by giving an increasingly false representation of a lower level of rectified signal, this voltage appears across C10; increasing the attenuation rate is achieved by decreasing RV1's resistance.

This increasing attenuation characteristic creates low level expansion and is used to reduce the dynamic range of any signal within its domain, with a consequent drop in noise level. At higher signal levels RV1 becomes less effective in its role and allows a return towards original signal dynamics.

Returning to normality, S1b selects the resistors (R15-R17) that are required to achieve an overall device gain of unity. S1a in conjunction with S1c also selects the output of IC1a, enabling pre and post 'gate' comparisons to be made.

Construction

Please refer Figure 3 for the layout of the PCB and to the Constructors Guide for hints and tips on soldering and constructional techniques when building the board.

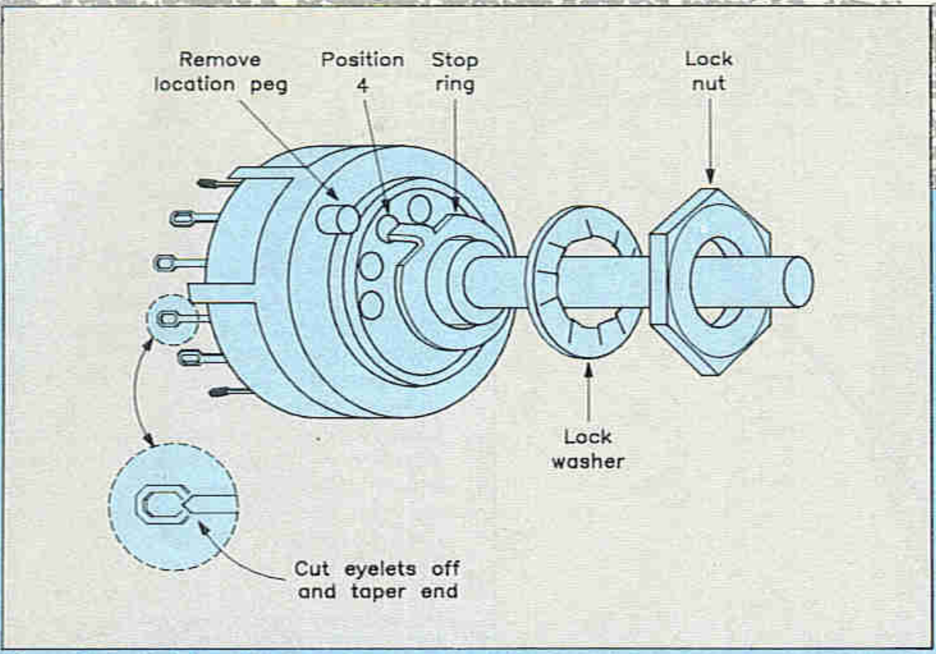


Figure 5. Switch location peg.

The rest of the construction details relate to the cased unit, for panel mounting all that is necessary is to 'flylead' the input, output and RV1 wiring. Circuit board fixing is effected by the rotary switch on which the board is mounted. PCB assembly should begin with the resistors, the offcuts from the resistors are used to make two wire links, which should be fitted next. The capacitors and ICs may now be fitted. Double-check polarised capacitor and IC orientation. Two lengths of hook-up wire are required for connection to RV1, these are made from the hook-up wire provided, remember that since RV1 is mounted above the board, components beneath it should lay flush to the board.

Solder the battery connector with the positive lead connected to the pin next to R7, and the negative lead to the centre tag of the stereo socket. See Figure 4.

Then fit the rotary switch. This switch is normally obtained with solder type eyelet tags which need simple modifications for PCB fitting. Cut these eyelets off, as close to the solder hole as possible using a pair of snips, taper the ends to assist in aligning and fitting into the board, you will also need to cut off the location peg on the switch, see Figure 5. Next, the stop ring needs to be moved, turn the switch to the furthest position anticlockwise and fit the stop ring tag in hole number 4.

Using the hook-up wire, 0V, input and output leads can be connected to the board.

After making sure all components have been fitted correctly, a few quick checks with a multimeter will verify correct basic operation. Switch the meter to the 100mA DC current range. Connect the negative lead of the meter to the battery 0V, this is identified from the symbols on the side of the battery, connect the +V terminal of the battery clip onto the battery. With the remaining meter probe, briefly touch the battery clip 0V terminal, if a reading of less than 10mA shows all is well. Remove the meter from the circuit. If you obtain a reading of more than 15mA, something is wrong! Check to see if you have any solder blobs shorting out the PCB tracks, also check to see if you have inadvertently inserted the IC's with the wrong orientation, or is there an incorrectly fitted tantalum capacitor? If a capacitor or IC has been incorrectly fitted and power applied to the circuit, replace the offending component(s) even if it still works, otherwise it may become faulty after a short time. Voltage checks on the outputs of each IC with a meter set to the 20V range will confirm correct operation of the circuit, pins 1 & 7 on IC2 should be approximately 5V and pins 7 & 12 of IC2 approximately 3·6V. (These voltages are related to internal references that remain constant, and therefore allow for battery voltage reduction.)

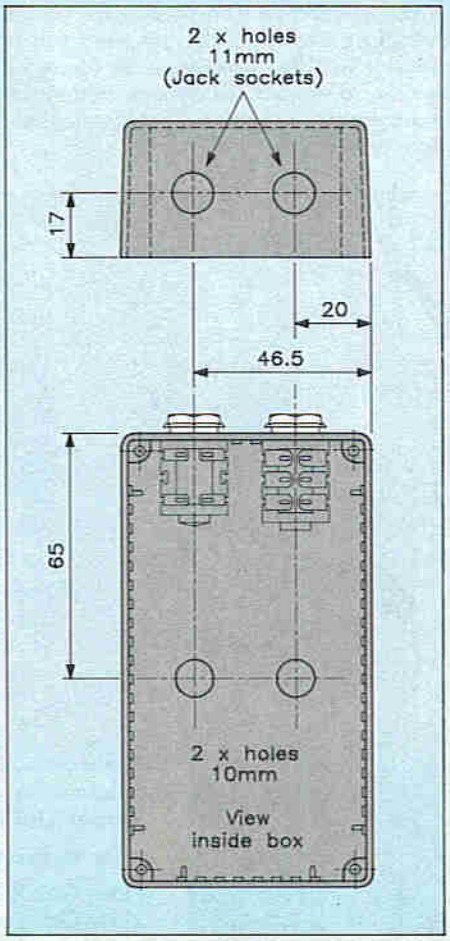


Figure 6. Drilling instructions.

All that remains now is to mount the unit in a suitable case. The metal case suggested in the parts list is ideal since it is small, easily workable, and provides good screening. Two appropriate holes in both the top and the side are all that are needed, see Figure 6 for drilling instructions. The whole assembly is then fitted into the main case body and secured by the switch, fitting the jack sockets into the case first will make assembly easier, ensure the jack sockets do not short out against the lid of the case when fitted. A wire link soldered to the body of RV1, with the other end attached to the 0V pin of the potentiometer, will provide case screening, see Figure 4.

Operation and Application

To obtain maximum usage of the unit, its functional characteristics should be fully understood. This can be achieved mainly by studying the response curves. The 1:1 gain slope (Figure 7) reference allows you to visualise the deviation from the normal input/output characteristics. The curve closest to the 1:1 gain slope shows the input/output of the device when set in any gain position with RV1 set for minimum effect (clockwise). Note that input signals or noise below −60dBm (horizontal axis) will be attenuated reducing its effective level as the output deviates rapidly away from the 1:1 gain slope towards −85dBm (vertical axis), This is the operating region of the unit. Signals above −60dBm will have a virtually normal dynamic range. The unit will completely shut down below −38dBm with RV1 set to maximum (this setting will shut off most extraneous noises).

Since the compander section uses rectified signal levels in its operation, speed of recognition of these levels becomes a compromise between several factors, one of which is the loading of the circuitry by RV1. It should, therefore, be remembered that the unit will take a finite time to attack and decay and that these items bear a direct relationship to the threshold level (that level selected at which deviation from normal characteristics occurs), and change in amplitude of the input signal. For example, with the noise gate set at 0dB gain and input levels gated from infinity to 0dBm, attack times will vary for minimum to maximum threshold settings.

Before continuing with typical applications, it is pointed out that this unit is designed for 'pause' noise reduction, and has no "magic ingredient" for reducing any noise present in actual signals.

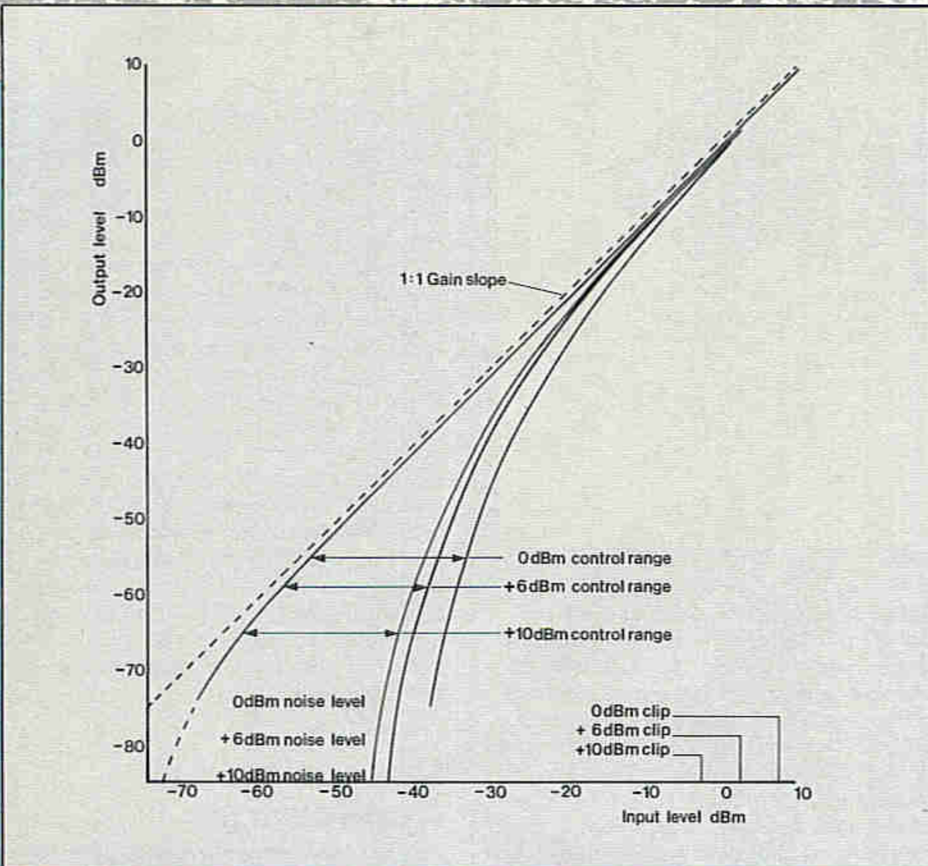


Figure 7. Response curves for noise gate.

Wherever noise exists, in-line connection of the noise gate should discriminate and further separate it from the required signal. Probably the lowest levels encountered will be from microphones and low output guitars (low Z mics will have to be transformed or the unit modified to suit). In the case of microphones, crosstalk (pick-up from sources other than that intended) will be the problem to cure. The +10dBm setting will allow maximum dynamic range with fast attack to be achieved when dealing with low level crosstalk (drum kit miking, awkward placement instrument miking, outside recording in windy conditions excluded).

Typical Applications

Guitarists working in cramped conditions may encounter induced hum from closely situated amps, especially via single coil pick-ups. This often embarrassing situation can be alleviated by using the noise gate between guitar and amplifier, adjusting the threshold control for the best overall effect. Another interesting use of the noise gate is for changing the attack characteristics of an instrument. If the threshold level is taken to extremes, soft attack type sounds can be created. Similar treatment can be applied to special effects units (Echo, Chorus, Phasing, Distortion, etc.), by using the noise gate between the last unit and its main amplification. The +6dB mode will be the norm in this application, since typical peak levels may introduce clipping.

Multi-instrument set-ups can often make the background noise unacceptable, considering that all units probably remain set at the required output level when only one or two instruments are actually being used at any time. Fitting individual noise gates to the noticeably 'noisey' instruments will automatically shut off their outputs when not in use.

Extreme levels of noise, or higher input levels, can be coped with in the 0dB mode, making this setting suitable for most line and mixer desk levels.

It is also possible to reduce fixed delay times of instruments or reverberation treatments by suitable setting of the noise gate control parameters.

NOISE GATE PARTS LIST

RESISTORS: 0·6W 1% Metal Film			
R1,14	56k	2	(M56K)
R2	68k	1	(M68K)
R3,4	330k	2	(M330K)
R5,6,7	4k7	3	(M4K7)
R8	2k2	1	(M2K2)
R9,10	15k	2	(M15K)
R11	100k	1	(M100K)
R12,13	100Ω	2	(M100R)
R15	2k7	1	(M2K7)
R16	1k	1	(M1K)
R17	1k8	1	(M1K8)
R18	47k	1	(M47K)
RV1	2M2 Pot Log	1	(FW29G)
CAPACITORS			
C1,14	47nF Monocap	2	(YY10L)
C2,13	47µF 16V Minelect	2	(YY37S)
C3-10	1µF 35V Tantalum	8	(WW60Q)
C11,12	4µ7F 35V Minelect	2	(YY33L)
SEMICONDUCTORS			
IC1	1458C	1	(QH46A)
IC2	NE571	1	(YY87U)

MISCELLANEOUS			
S1	Switch 3-Pole 4-Way Rotary	1	(FH44X)
SK1	Jack Socket Stereo ¼in	1	(HF92A)
SK2	Jack Socket Mono ¼in	1	(HF90X)
	PP3 Clip	1	(HF28F)
	Pin 2145	1 Pkt	(FL24B)
	DIL Socket 8-Pin	1	(BL17T)
	DIL Socket 16-Pin	1	(BL19V)
	Hook-up Wire 7/0·2mm Black	1 Reel	(BL00A)
	PC Board	1	(GA43W)
	Constructors Guide	1	(XH79L)
OPTIONAL (not in kit)			
	Front Panel	1	(JR87U)
	Box DCM5004	1	(LH71N)
	Collet Knob Low Cost	2	(YG40T)
	LC Cap Grey	2	(QY03D)
	Stick-on-Feet Large	1 Pkt	(FW38R)

The above items, excluding Optional, are avaliable as a kit:
Order LK43W (Noise Gate Kit) Price £9.95

The following items are also available separately:

Noise Gate PCB **Order As GA43W Price £1.48**
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XA36P price £1.45.

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