May 1995
£ 2·45

ELEKTOR ELECTRONICS

THE INTERNATIONAL ELECTRONICS MAGAZINE

Focus on: Audio Standards and Test Methods

89C51 FLASH PROGRAMMER

150 watt AF power amplifier with mute

ADR-Astra digital radio

Programmable sine wave generator

MIDI analyser

Rechargeable alkaline batteries

05
9 770268 451043

AUDIO & HI-FI • COMPUTERS & MICROPROCESSORS • DESIGN IDEAS • RADIO, TELEVISION & COMMUNICATIONS • SCIENCE & TECHNOLOGY • TEST & MEASUREMENT

ELEKTOR ELECTRONICS

CONTENTS

May 1995
Volume 21
Number 233
ISSN 0268/4519

In next month's issue

- FOCUS ON: THE WORLD OF RADIO & TV AMATEURS
- DC-DC converter
- Function generator
- Current meter MAX471
- Insulated Gate Bipolar Transistors (IGBTs)
- Internet
- and others for your continued interest.

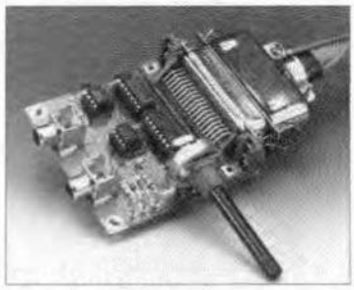
Front cover

Microcontrollers are used increasingly for switching and control applications. Consequently, there is a growing need of inexpensive programming systems. The flash programmer described in this article handles controllers from the Intel MCS-51™ family and others with on-chip flash memory (Philips, Atmel), as well as the more conventional UV-erasable types with on-chip EPROM.

Win an 8031/8051 emulator page 37

MIDI analyser - p. 8

Programmer for 87/89C51 series flash controllers - p. 14



Programmable sine wave generator - p. 26

PC-controlled audio measurement systems - p. 52

Elektor Electronics is published monthly, except in August, by Elektor Electronics (Publishing), P.O. Box 1414, Dorchester, Dorset DT2 8YH, England. The magazine is available from newsagents, bookshops, and electronics retail outlets, or on subscription at an annual (1994/95) post paid price of £28·00 in the United Kingdom; air speeded: £35·00 in Europe, £44·00 in Africa, the Middle East and South America; £46·00 in Australia, New Zealand and the Far East; and $US 59.00 in the USA and Canada. Second Class Postage paid at Rahway N.J. Postmaster: please send address corrections to Elektor Electronics, c/o Mercury Airfreight International Inc., 2323 Randolph Avenue, Avenel, New Jersey, N.J. 07001.

From the World of Electronics

Smart antenna will improve mobile communications

A new type of antenna, produced by ERA Technology, promises to improve the performance of mobile communications systems. It comprises an eight-element phased-array and a digital processing unit to control the direction and shape of the antenna pattern. An important advantage of this unit is its ability to support larger numbers of mobile users in a given area, providing significant increases in radio spectrum efficiency over the omni-directional antennas currently used in cellular mobile systems.

As the pressure on the allocation of the radio spectrum increases, the benefits of increased spectrum efficiency will pay commercial dividends. Other benefits of the new antenna technology include improved protection against multipath interference and a reduction in the interference which mobile equipment can cause to other radio services.

The smart antenna uses advanced digital signal processing algorithms and technology to implement the beam steering and control functions. A number of novel optimal beam-forming algorithms have also been developed during the course of the research.

Further research is planned to apply the new technology to third-generation mobile communications known as the Universal Mobile Telecommunications System.

Anglo-Dutch scientists link on free electron lasers

After playing an import part in the design and construction of a new Dutch facility for producing high-flux beams of rapidly tuneable free-electron laser radiation, British scientists have a 20 per cent share of its beam time. The deal gives them access to what is know as the FELIX facility at the FOM Institute for Plasma Physics at Rijnhuizen in the Netherlands. FELIX, which stands for Free-Electron Laser for Infrared eXperiments, currently produces its beams of radiation in the 6–110 μm far infrared range.

Free-electron lasers have many potential applications in physics, chemistry and materials science. Despite a history spanning nearly two decades, they have only recently begun to realize their potential. Their attractiveness to users lies in a unique combination of high power, short-pulse operation, and very wide tuning range over the visible and infrared spectrum. In addition, it is now feasible to implement advanced laser techniques in free electron lasers, in conjunction with conventional pulsed lasers.

The present FELIX configuration comprises two linear accelerators delivering electron beams of 70 A peak current into two free-electron lasers. The first laser is driven by a single accelerator (15–25 MeV), while the second laser uses the two accelerators in series (30–50 MeV). The undulators used in the two lasers originate in the UK and were made available to FELEX by SERC after completion a free-electron laser experiment.

Formex algebra configures geometric space structures

When space structures (so called because they have spatial dimensions) were first used in the 1960s, the long and laborious mathematical computations essential to create them had to be done manually. Most people have seen and perhaps admired such space structures overhead at airports and in superstore shopping centres.

In earlier days, the process of data generation for such complex structural systems was notoriously difficult. With the concurrent development of the computer, it is not surprising to learn that the possibilities of looking for a mathematical system and a suitable computer program were looked at and pursued.

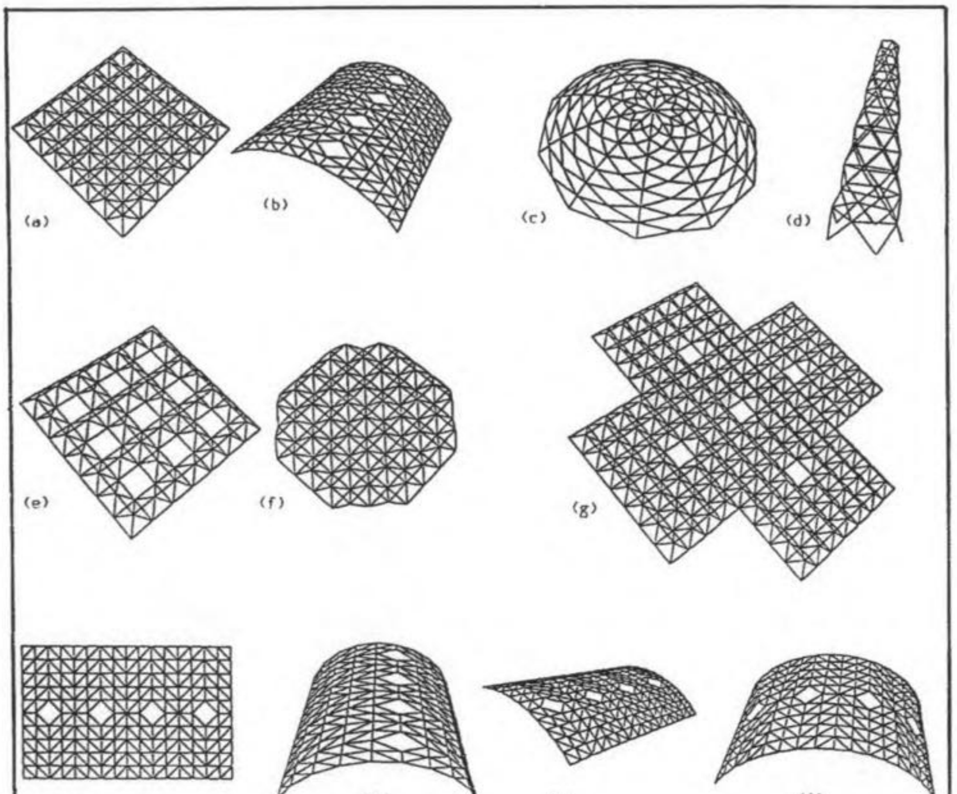
Today, the creation of space structures of every conceivable design is greatly simplified by formex algebra, which is used in conjunction with a computer language called *Formian*. The application of formex algebra to the design of space structures was pioneered by the Space Structures Research Centre at the University of Surrey.

Rudimentary ideas

The rudimentary ideas from which formex algebra has emerged were evolved in the 1970s. It is difficult to demonstrate how formex algebra can be applied to space structure design without touching on the mathematical system itself. However, in presenting some applications, the emphasis will be on the manner in which the concepts of formex algebra are used rather than the details of the formulations; no prior knowledge of formex algebra is necessary.

As one of the first steps in the design of a space structure, it is necessary to generate the data containing information regarding the elements of the structure and the manner in which they are connected together. For example, the description of the double layer grid, (a) in the illustration, can be given by the formex formulation

G=pex:rinid(7,7,2,2):rosad(1,1):[(0,0,0;2,0,0),(1,1,1;0,0,0)] + pex:rinid(6,6,2,2):rosad(2,2):(1,1,1;3,1,1).



With any computing system incorporating a suitable formex software, this formulation may be used to generate the data describing the element interconnection and joint coordination.

Typed in

When using the Formian software, the formulation is simply typed in, exactly as given, on the computer keyboard. When input, the information provide by the formulation may be used to visualize the configuration on screen or through a printout and may also be used as input data for a structural analysis program.

Once a formex formulation representing a configuration is in hand, it can be further manipulated to reflect the changing requirements that take place during the design. With the formex algebra formulation, G, for structure (a) given earlier, the double-layer grid (e) may be represented by

G1=lux[rinid(3,3,4,4):(3,3,1)]:G.

Grid (f) may be represented by:

G2=lux[lamid(7,7):0,0,0;2,0,0;0,2,0; 1,1,1)]:G,

and grid (g) by:

G3=pex:rosad[15,7:lux(7,7,1)]:G.

Two stages

In using formex to generate data representing a configuration, there is the option of dividing the data generation into two distinct stages. In the first stage, the interconnection pattern of the configuration is formulated with a convenient reference. The actual geometry of the configuration is then obtained in the second stage by a suitable geometric transformation. For instance, to represent the barrel vault in (b), first write a formex formulation representing the interconnection pattern of the structure relative to a plane Cartesian reference system. This formulation may be written as

B=pex:rin(1,4,4):lamid(2,5,):lux(2,5): rinid(2,5,1,1):[(1,0,0,1) + rosad($^1/_2$,$^1/_2$): (0,0,0;1,0)]

to represent the plane configuration in (h). The transformation of the reference system into a cylindrical one will now change the plane configuration of (h) into (b). Such a transformation may be achieved simply through standard equations that relate Cartesian and cylindrical coordinates. Also, the parameters in this geometric transformation may be varied to obtain different results. For example the barrel vault may be made less shallow, as shown in (j), or more shallow, as shown in (k). Also, the direction of the mapping may be altered to obtain the barrel vault shown in (l).

Simplified process

The storage of data in a computing system becomes very convenient, since the actual formulation may consist of just a few lines of text; the requirement to store data for a large space structure in explicit form is many hundreds of times greater than storing it in its formex form.

Formex description

Formex algebra is a mathematical system that provides a convenient means of solving problems in the areas of data generation and graphics. Although it was developed for civil engineering, it can be used in any branch of science where similar structures occur.

Space Structures Research Centre, Department of Civil Engineering, University of Surrey, Guildford, Surrey, England GU2 5HX.

Delay in interactive TV

Many consumer trials of interactive TV in the USA and Europe have been delayed or cancelled altogether. Those that have gone ahead are confined to employees of the companies running the trials. The participants in these trials are merely testing the viability of the technology, not the possible marketing benefits.

The problem is a chicken-and-egg one: it is generally reckoned that the cost of providing customers with a service that would enable them to order (video) films on demand or do their shopping is more than a hundred times what the customers are prepared to pay for the service.

In the early days, it was assumed that the cost of providing the service would be borne by an amalgamation of telephone companies and cable TV companies, and, of course, eventually, advertisers. A combination of a telephone company and a cable TV company looks attractive and logical. The former have the communication know-how and the switching facilities; the latter the high-capacity distribution networks. The telephone companies would thus be able to provide

Produced and published by **ELEKTOR ELECTRONICS (Publishing)**

Editor: Len Seymour
Technical Editor: Jan Buiting

Editorial & Administrative Offices:
P.O. Box 1414
DORCHESTER DT2 8YH
England
Telephone: (01305) 250 995 (*National*)
or +44 1305 250 995 (*International*)
Fax: (01305) 250 996 (*National*)
or +44 1305 250 996 (*International*)

U.K. Advertising Office:
3 Crescent Terrace
CHELTENHAM GL50 3PE
Telephone: (01242) 510 760
Fax: (01242) 226 626

International Advertising Office:
Elektuur BV
P.O. Box 75
6190 AB BEEK
The Netherlands
Telephone: +31 46 38 94 44
Fax: +31 46 37 01 61

Subscriptions:
World Wide Subscription Service Ltd.
Unit 4, Gibbs Reed Farm
Pashley Road
TICEHURST TN5 7HE
Telephone: (01580) 200 657 (*National*)
or +44 1580 200 657 (*International*)
Fax: (01580) 200 616 (*National*)
or +44 1580 200 616 (*International*)

Head Office:
P.O. Box 75
6190 AB BEEK
The Netherlands
Telephone: +31 46 38 94 44
Telex: 56617 (elekt nl)
Fax: +31 46 37 01 61
Managing Director:
Johan H. Boermann
Deputy Managing Director:
Menno M.J. Landman
Editor-in-Chief/Publisher:
Pierre E.L. Kersemakers
Commercial Manager:
Karel van Noordenne

Distribution:
SEYMOUR
1270 London Road
LONDON SW16 4DH

Printed in the Netherlands by NDB, Zoeterwoude

Overseas editions:
FRANCE
Elektor sarl
Les Trois Tilleuls
B.P. 59; 59850 NIEPPE
Editor: G.C.P. Raedersdorf

GERMANY
Elektor Verlag GmbH
Süsterfeld Straße 25
5100 AACHEN
Editor: E.J.A. Krempelsauer

GREECE
Elektor EPE
Karaiskaki 14
16673 Voula—ATHENA
Editor: E. Xanthoulis

INDIA
Elektor Electronics PVT Ltd
Chhotani Building
52C, Proctor Road, Grant Road (E)
BOMBAY 400 007
Editor: C.R. Chandarana

ISRAEL
Elektorcal
P O Box 41096
TEL AVIV 61410
Publisher: M. Avraham

NETHERLANDS
Elektuur BV
Peter Treckpoelstraat 2–4
6191 VK BEEK
Editor: P.H.M. Baggen

POLAND
Elektor Elektronik
02–777 Warszawa 130
Skrytka Pocztowa 271
Editor: W. Marciniak

PORTUGAL
Ferreira & Bento Lda.
R.D. Estefâni, 32-1°
1000 LISBOA
Editor: F. Ferreira de Almeida

SPAIN
Resistor Electronica Aplicada SA
P.. de la Castellana, 212–4° Izda
28046 MADRID
Editor: Agustin Gonzales Buelta

SWEDEN
Electronic Press AB
Box 5505
14105 HUDDINGE
Editor: Bill Cedrum

read/write, that is, interactive, facilities, while the cable companies (who can only provide a read-only service) would undertake the distribution on their high-capacity cables.

The feedback from the market place will probably mean that a union between the telephone and cable companies is a long way, if it comes about at all. It is more likely that these companies will go their separate ways. Because of deregulation, this will be easy in the USA, but fraught with difficulties in Europe. Although in Britain cable companies already offer a telephone service, this is still virtually unknown in Europe. At the same time, even British Telecom is (not yet) allowed to put video pictures on to their telephone network. It is doubtful whether this would be technically possible, because, although a large part of the network consists of glass fibre cabling, the connection to the subscribers' homes is still by traditional copper wire, which is totally unsuitable for wide-band signals. Even if that could be solved by compression techniques, it is a long way off interactive television, which requires not one channel, but, perhaps, as many as a few hundred to be truly interactive.

Fourth annual conference on cable telephony

The 4th annual conference on cable telephony will be held on 18–19 May at the Dorchester Hotel, London. Full details from **IBC Technical Services Ltd, Gilmoora House, 57–61 Mortimer St., London W1N 8JX. Telephone +44 171 637 4383; fax +44 171 636 1976**.

Ghost busting

Successful tests have been carried out in Britain with a new technique to improve television picture quality by reducing what is known as ghosting.

At a demonstration staged in Enfield, North London, the Independent Television Commission (ITC) unveiled experimental equipment to combat ghosting problems that can affect TV viewing in homes located in built-up and hilly areas. The ITC hopes tests with a prototype could lead to the introduction of a 'ghost-canceller' within three years. This could be in the form of special television sets or add-on units.

The experimental system requires a ghost cancellation reference (GCR) signal to be inserted into the transmitted television signal. The GCR signal has been included experimentally on selected TV programmes beamed to the south of England and Wales since January 1994, and has recently been tested on transmissions from Crystal Palace in London.

The development work has led to the international standardization of the GCR signal. Other European broadcasters are expected to begin testing the system soon and the ITC is working with the European Broadcasting Union on its evaluation.

Subject to broadcaster agreeing to transmit the signal permanently, the first European consumer ghost cancellers could be introduced in 1996. The GCR signal has been standardized for all 625-line PAL/SECAM systems.

Independent Television Commission, 33 Foley Street, London W1P 7LB.

ECOS95 turns spotlight on security

The need for security, whether security of information, premises, goods or the person is not new. Neither is the ingenuity of people in trying to overcome such security or of those charged with maintaining it. As the world becomes ever more complex through the exploitation of technology, so the competition between the two follows pace.

A major international conference to be held in Brighton, England, from 16 to 18 May will feature many of the latest developments in security technology as well as in the detection and solving of crimes. ECOS95 – the European Convention on Security and Detection – is being organized by the Institution of Electrical Engineers (IEE) and has attracted speakers from Malaysia, Japan, Holland, Switzerland, Russia, Egypt, France, Canada, Hong Kong and the UK.

The conference is being organized into a number of broad topic areas covering security systems, imaging, communications and networks, intruder and perimeter protection, biometrics, explosives, fraud and X-ray imaging. Within each of these a range of specialist papers will be presented, such as speech encryption, security of fax transmissions and of networks, including telecommunications. Intruder and perimeter protection will include remote detection equipment and methods, fibre optic sensors and vehicle security.

In addition, there will be a number of general sessions as well as an associated exhibition. ECOS95 is being supported by the security and law enforcement agencies and a number of professional bodies.

Full details of the conference programme are available from the **ECOS95 Office, Conference Department, IEE< Savoy Place, London WC2R 0BL. Telephone +44 171 344 5477; Fax +44 171 497 3633**.

MIDI, electronic music & recording show

If you're into making music on your computer, get along to Future Music's MEMS (MIDI, Electronic Music and recording Show) in London's Olympia on 21–23 April. The show is not only the ultimate event to learn what's hip 'n' happenin' in the world of music and sound technology, but it's also an ace place for computer fans. There will be demonstrations and displays from some of the world's leading kit manufacturers and distributors, 'how-to' clinics, opportunities to 'try before you buy' the very latest in sound, recording and production equipment.

In addition, there will be free seminars running throughout the show, which will cover many aspects of sound production and composition. Forums will include 'Ask the hardware manufacturers' and 'Ask the software writers'.

You can save quids by booking tickets in advance. Group bookings of ten or more will pay £4·50 per ticket; advance tickets cost £5 each; tickets at the door cost £8. Call the ticket hotline on 01369 707 888.

Amanda Barry Communications, 15 Kingsmead Square, Bath BA1 2AE.

CONCEPT focuses on 'green' electronics (Call for papers)

The need to take full account of environmental impact when developing new electronic products is the focus of a major international conference to be held in Edinburgh, Scotland on 9–11 October, 1995. The joint organizers of the conference, the IEE and the IEEE, have issued a call for papers for the conference.

The title of the conference,the international CONference for Clean Electronics Products and Technology (CONCEPT), reflects efforts within the industry to take a life-cycle approach. More and more companies are looking at the environmental impact of their operations and products from the design phase through manufacturing and use to disposal and recycling.

Authors are invited to submit proposals for papers which describe case studies, research work or development of decision making tools under a series of broad headings. These range across regulatory and economic aspects, design for environmental impact, materials, the electronics manufacturing process, the effect on the environment of the use of products, end of life disposal, recycling, packaging and environmental assessment.

Authors should submit synopses of their proposals to **Louise Hudson, Conference Department, IEE, Savoy Place, London WC2R 0BL** by 12 May, 1995.

MIDI ANALYSER

Design by P. Rigaill

The unit described is intended for analysing signals that are transmitted via a MIDI link. It enables the digital information to be made visible and, if necessary, to be corrected. It can also be used to map the entire range of signal-determining data of a synthesizer.

The unit is equipped with two MIDI (Musical Instrument Digital Interface) connectors: an input and an output. Consisting of a simple processor board, an LCD module and a 12-position keyboard, it is intended to be inserted into an existing MIDI link. Although it is transparent, that is, it does not affect the data stream, it is capable of storing data transmitted across the link and make corrections if required. It is also possible for the stored data to be transmitted via the MIDI network in the form of a data block. Real-time operation is possible if the stored data are synchronous with the MIDI clock. This means that the analyser can (re)play a short piece of music. There is 32 kbyte of RAM to enable a fair amount of MIDI data to be stored. The necessary software is available on EPROM (page 70).

Circuit description

The circuit (see diagram in **Fig. 1**) is based on a Type 80C32 processor and a Type 74HC573 address latch. The clock is controlled by a 6 MHz crystal, thus providing access times that are long enough to guarantee satisfactory operation of the display. The 12-position keyboard is connected by linking K_5 to K_1.

The MIDI satisfies the usual standards: at the input (K_4) as well as the output (K_3) there are current loops.

The optoisolator, IC_7, at the input is connected directly to the serial input of the processor, IC_1.

The output is taken from pin 11 of IC_1 to K_3 via series-connected gates IC_{5e} and IC_{5d}. These gates ensure that sufficient current is provided at the output.

The necessary software is stored in EPROM IC_4, which is driven directly by pin 29 of IC_1.

Back-up for the RAM, IC_3, is provided by Bt_1.

Address decoding is effected by gates IC_{5a}, IC_{5b}, IC_{5c}, and IC_6. The RAM data are decoded from 0000_H to $7FFF_H$ via A_{15} and pins 16 and 17 of IC_1. Network R_8-C_6 prevents the RAM being addressed erroneously by the switching on of the supply voltage (which would damage any data present in the RAM).

The 2×16-character LCD is connected to K_2. The module is at address 8000_H; even addresses select the command register (8000 = write; 8002 = read), while odd addresses select the data register (8001 = write, 8003 = read).

Power is provided by a standard mains adaptor, whose output must be ≥9 V. The output is stabilized at 5 V by IC_8. Capacitors C_5 and C_7–C_{11} decouple each of the ICs for any interference on the supply lines.

Construction

The analyser is intended to be built on the double-side, through-plated PCB in **Fig. 3**. Population of the board is straightforward: the finished prototype board, complete with keyboard and display is shown in **Fig. 1**. Note that the prototype is not 100% identical with the board in Fig. 3.

The back-up battery may be a 3 V lithium type, but two penlight batteries will do just as well.

Fig. 1. Finished prototype board complete with keyboard and display.

The part of the PCB intended for the keyboard may be left in place or cut off and used independently.

Real time

Since the accepted and stored MIDI data contains the original clock information, the analyser is able to arrange the data in correct measure. At the beginning of each piece of music, the sequencer produces a start command followed by 24 timing clock pulses for each crotchet (quarter note) for the duration of the piece. At the end, there is a stop command. The software contained in the ready-programmed EPROM makes it possible for the bytes between any two timing clocks to be transmitted in the correct rhythm.

Usage

Keys 1, 2, 4 and 5 of the keyboard serve to shift the image on the display by ±1 or ±16 places in the memory.

Keys 3 and 6 enable the previous or

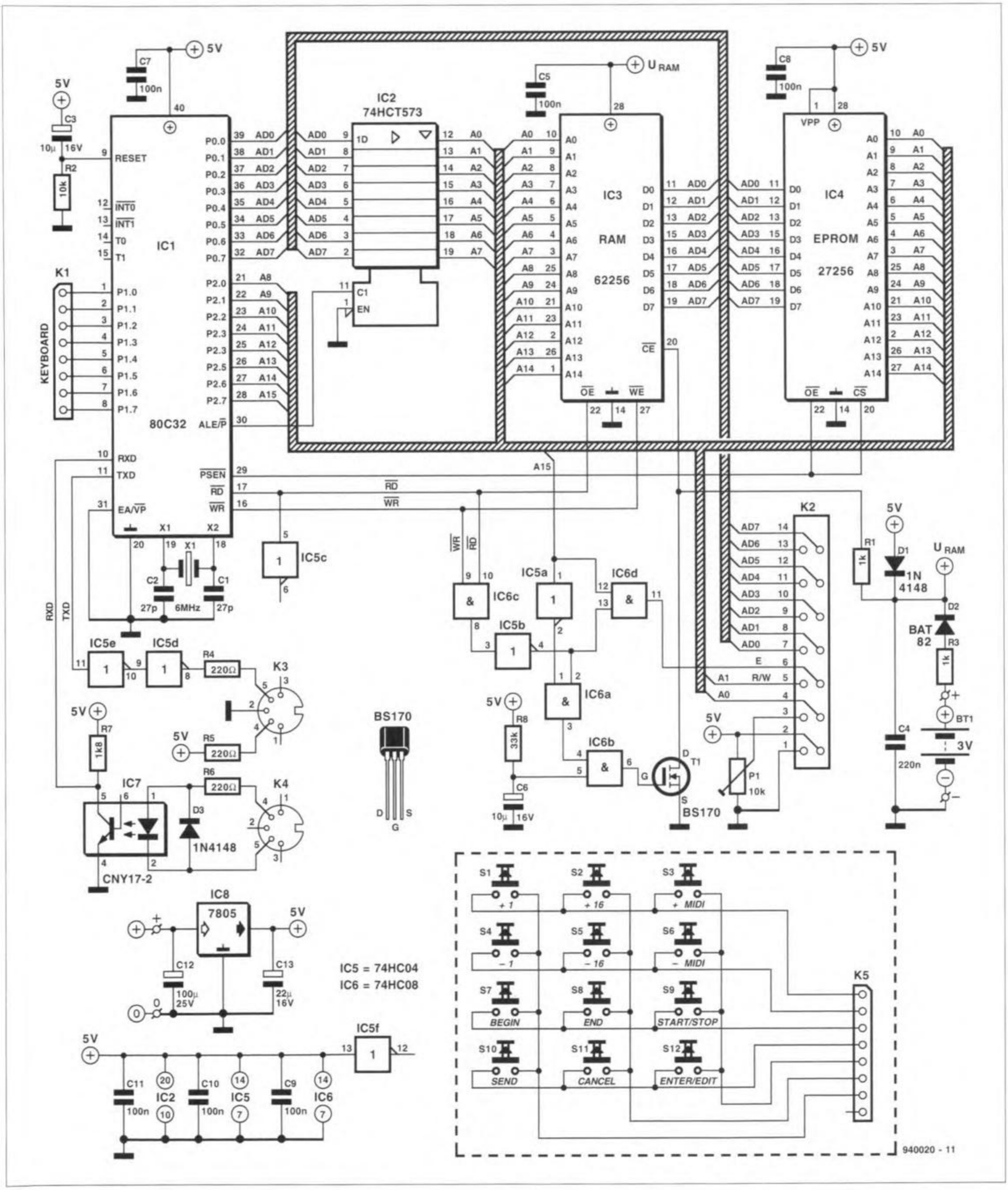


Fig. 1. Circuit diagram of the MIDI analyser.

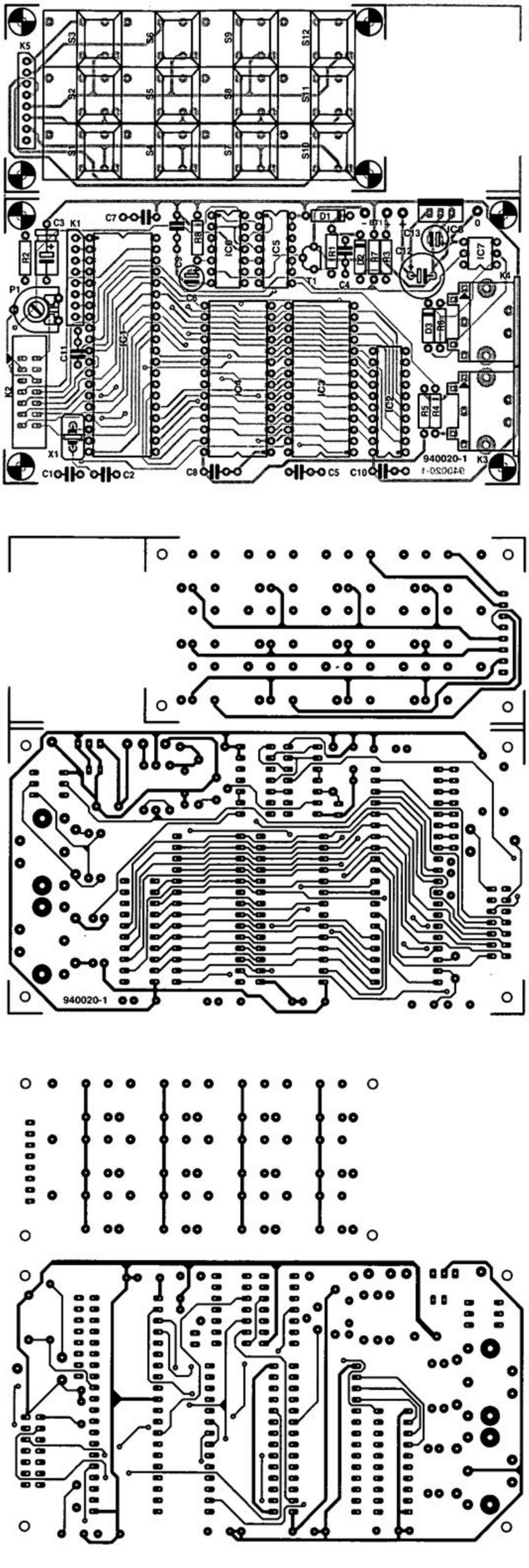


Fig. 3. Printed circuit board for the MIDI analyser (scale 3 : 4).

next MIDI command to be displayed.

Keys 7 and 8 mark the block to be transmitted,

Key 9 enables data storage to be started and stopped.

Key 11 is the cancel key.

Key 12 serves to start the data processing (edit) and enforce the commands (enter)

After the analyser has been switched on, the display will read "MIDI DATA ANALYSER". If then the 'cancel' key is pressed and held, the memory address is set to 0000. Thereupon the display reads: 0000:91 45 40 81

01NotOn G1 064.

These data are random, they depend on the content of the RAM.

The first field represents the RAM address (0000 at the start). The next four fields are the bytes associated with the next four addresses.

At 'channel messages', the relevant MIDI channel is reproduced at the beginning of the second line, followed by the function, the note, and the speed. When data arrives, the word 'data' appears on the second line. In the case of 'system messages', an inverted S appears at the start of the second line. In this function mode, which could be considered the standard mode, all received bytes are passed on without any further ado.

The memory can be consulted in a transparent manner with keys 1–6, provided, of course, that a piece has been 'recorded'.

Keys 'block begin' (7) and 'block end' (8) serve to mark the start and end of the block to be transmitted.

When key 12 is pressed, the unit changes over to the edit mode. The byte then displayed can then be altered with keys 1, 2, 4 and 5; this alteration is activated by pressing key 12 again, or cancelled by pressing key 11.

Stop/start key (9) serves to begin the storing of the data. The display will ask for confirmation. If the instruction is to be cancelled, press key 11; if it is to be continued, press key 12. In the latter case, the display reads

RECORDING

<STOP> to abort.

As soon as a byte is received, a musical note will be displayed. The data storing is terminated by pressing the start/stop key or when the memory is full. In the first case, the number of stored bytes is displayed; in the second case, the words 'memory full' are shown.

The data are transmitted when the send key (10) is pressed. The display will then show two choices:

<+1>: Real Time

<–1>: Send Block

The second line indicates that pressing key 4 will result in the transmission of the marked block. When key 1 ('+1') is pressed, the processor will search the memory for real-time information. If nothing is found, this will be communicated via the display. If such information is available, the display will indicate that the tempo, in the form of number of beats per minute (BPM), may be chosen:

Select BPM
(30–400): 120

The number of beats are selected with key 1 and key in the range 30–400. When this has been done, the data can be transmitted by pressing key 12 or cancelled by pressing key 11. Key 9 (start/stop) permits interruptions.

Parts list

Resistors:
R_1, R_3 = 1 kΩ
R_2 = 10 kΩ
R_4, R_5, R_6 = 220 Ω
R_7 = 1.8 kΩ
R_8 = 33 kΩ
P_1 = 10 kΩ preset

Capacitors:
C_1, C_2 =27 pF
C_3 = 10 μF, 16 V
C_4 = 220 nF
C_5, C_7–C_{11} = 100 nF
C_6 = 10 μF, 16 V, radial
C_{12} = 100 μF, 25 V, radial
C_{13} = 22 μF, 16 V, radial

Semiconductors:
D_1, D_3 = 1N4148
D_2 = BAT82
T_1 = BS170

Integrated Circuits:
IC_1 = 80C32
IC_2 = 74HC573
IC_4 = 27256 (see p. 70)
IC_5 = 74HC04
IC_6 = 74HC08
IC_7 = CNY17-2
IC_8 = 7805

Miscellaneous:
K_1, K_5 = 8-way SIL connector
K_2 = 14-way box header
K_3, K_4 = 5-pin 180° DIN socket for board mounting
S_1–S_{12} = digitast switch
X_1 = 6 MHz crystal
Bt_1 = 3 V battery (see text)
LCD module 2×16 characters (e.g. Sharp Type LM16A21)
PCB Order No. 940020

[940020]

PROGRAMMER FOR 87/89C51-SERIES *FLASH* CONTROLLERS

Microcontrollers are increasingly used for switching and control applications. Consequently, there is a growing need of inexpensive programming systems. The Flash programmer described in this article handles controllers from the Intel MCS-51™ family and others (Philips, Atmel) with on-chip Flash memory, as well as the more conventional UV-erasable types with on-chip EPROM.

Design by K. Walraven

IF you want to develop application circuits based on microcontrollers you need a number of tools for the trade. The programs have to be put together and debugged with the aid of the appropriate development system. Over the past few years, the cost of the relevant software has dropped considerably, enabling almost anyone to develop applications based on powerful microcontrollers. EPROM emulators, which are essential tools for software development, are available at reasonable prices. The last hurdle in the race is then programming the tested object code into the microcontroller. The Flash programmer described here is just the thing for that purpose.

The compact and easy to build programmer described here enables you to program the 87C51/87C52, 89C51/89C52/89C2051 and 87C750/87C751/87C752 controllers rapidly and at low cost. It should be noted that the programmer is **not** suitable for 87C51 versions with an **F** behind the type number. These devices differ in respect of their larger ROM size (8 KByte), RAM size and number of timers.

When processors with a Flash memory are used, clearing the internal memory takes only 10 milliseconds and a simple command. Ordinary processors, that is, types with a 'windowed' built-in EPROM, have to be exposed to ultra-violet light for at least 20 minutes before they can be programmed again.

The Flash programmer is a very flexible tool, mainly by virtue of the fact that the programming algorithms are generated by a microcontroller. In addition, electronic switches enable the signals at a number of pins of the programming socket to be adapted to different types of processor. At the heart of the circuit is the Signetics (Philips Semiconductors) type SC80C451CCA68 microcontroller, an 68-pin PLCC device sporting no fewer than 48 I/O lines.

MAIN FEATURES

Programs:	Flash and EPROM
Processors:	87C51/87C52 (UV types)
	89C51/89C52 (Flash types)
	87C750/87C751/87C752 (UV, mini)
	89C2051 (Flash mini, via adaptor)
Software:	in EPROM
	+ any terminal program
Interface:	RS232
Baudrate:	9600 baud, 8 data bits,
	no parity, 1 stop bit
File format:	Intelhex

The programmer

The complete circuit diagram of the Flash programmer is shown in **Fig. 1**. Although the circuit is fairly extensive, its operation is easy to understand. As already mentioned, the entire programming operation is controlled by IC_1, an 80C451 running at a clock speed of 14.75 MHz. This frequency is used to enable the controller to communicate with a PC at 9,600 baud (a standardized bit rate). It also allows a number of frequencies needed for the programming process to be derived easily. The controller fetches its instructions from an EPROM (IC_3) via a latch (IC_2) which separates the multiplexed address/data information. Since the programmer communicates with a PC via the RS232 port, a MAX232 line driver/interface is used. The rest of the circuit consists of a power supply and an array of electronic switches which ensure that the programming socket pins receive the right signals.

The power supply obtains its input current from an ordinary mains adaptor with an output of 12 V at about 300 mA. The (unstabilised) adaptor output voltage will typically be higher than 12 V (approx. 16 V). An LM317L voltage regulator on the programmer board (IC_5) steps the mains adaptor voltage down to 12.75 V. To make sure that the LM317L can operate correctly, check that your mains adaptor supplies at least 16 V when loaded with about 50 mA.

Diode D_2 affords a simple supply reversal protection by short-circuiting the mains adaptor output if the polarity is wrong. This may cause a fuse to blow in the adaptor, but protects the much more expensive programmer circuit.

Another regulator, IC_7, reduces the rectified voltage to a stabilized level of 5 V. All components in the circuit, with the exception of IC_8 and the programming socket, operate on this voltage. The 12.75 V supplied by IC_5 can only reach the programming socket via a number of electronic switches.

IC_1 may put the 5-V supply voltage for the controller to be programmed on the programming socket pin via transistor T_2. The programming voltage of 12.75 V for the 87C5x controllers appears at pin 31, via T_5. A 89C51 controller gets its programming voltage (12 V) via T_1. Diode D_5 ensures that the 12.75-V voltage is lowered by 0.7 V. T_4 is used to program 87C750/51/52 processors. This transistor puts 12.75 V on pin 6 of these processors. In both cases, a small ca-

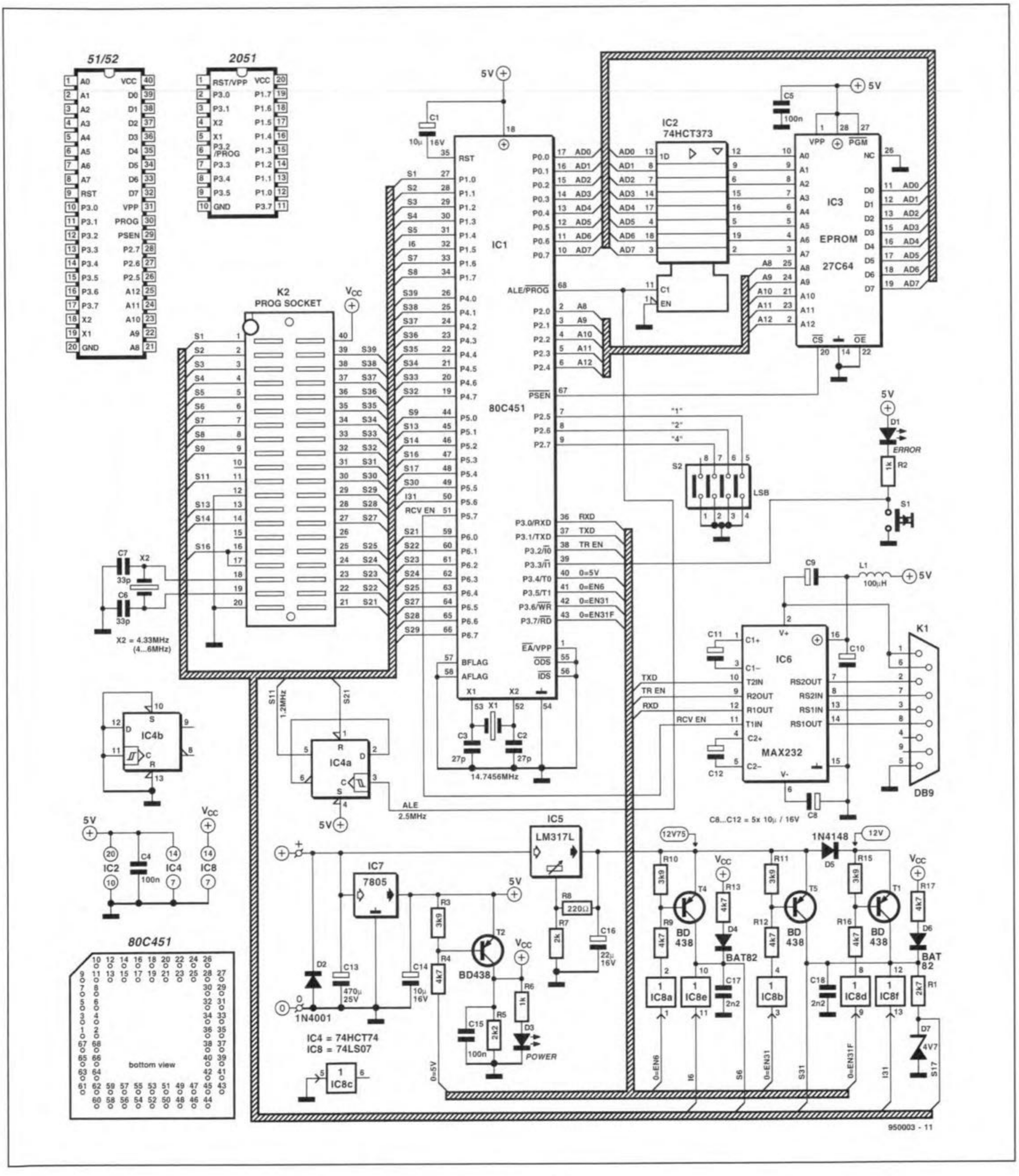


Fig. 1. Circuit diagram of the *Flash* programmer. At last, programming EPROM and Flash memory based microcontrollers from the MCS-51 family is within reach of any electronics enthusiast.

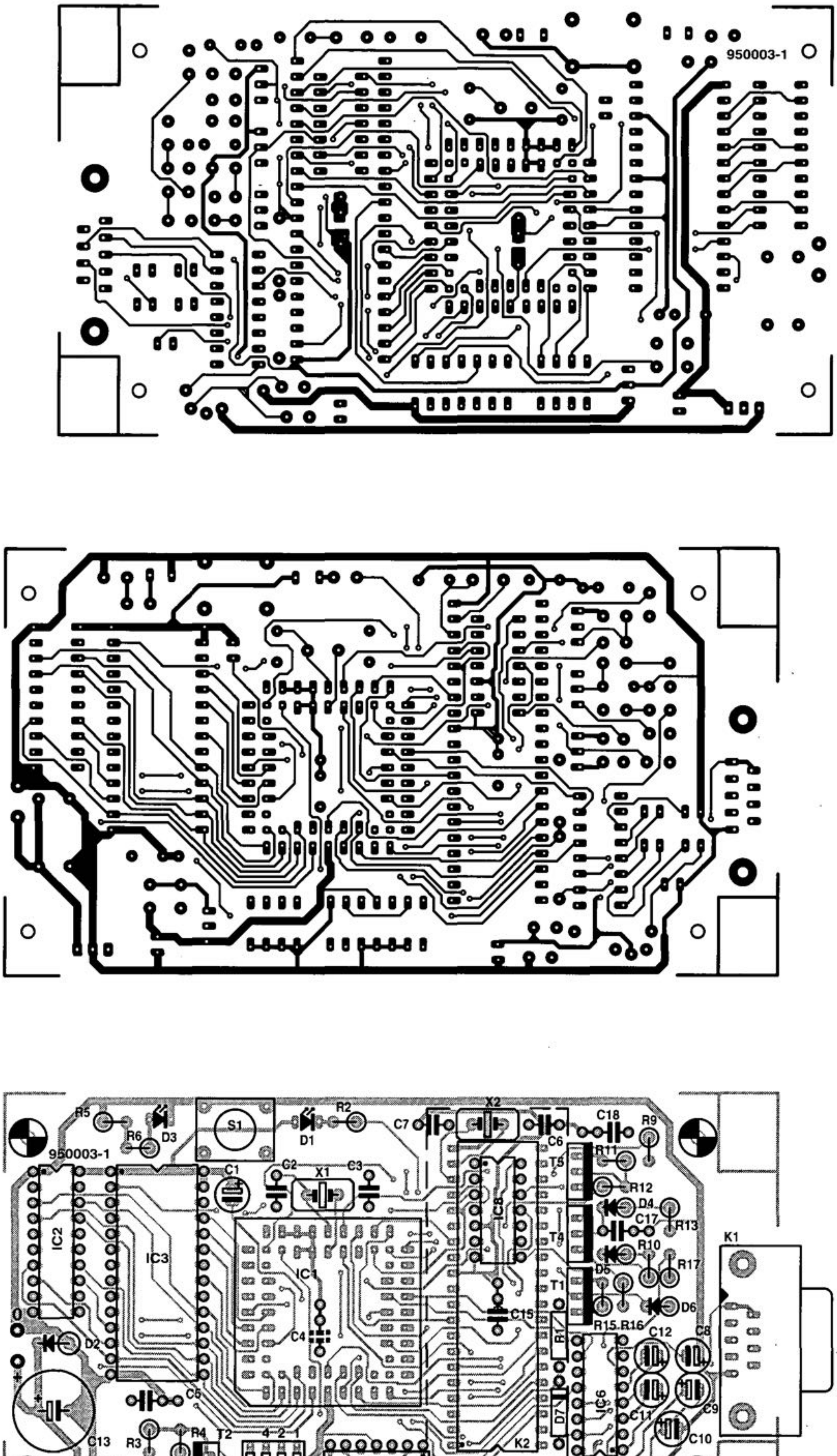


Fig. 2. Track layouts (component and solder side) and component mounting plan of the double-sided circuit board for the Flash programmer (board available ready-made, see page 70).

COMPONENTS LIST

Resistors:
R_1 = 2kΩ7
R_2,R_6 = 1kΩ
R_3,R_{10},R_{11},R_{15} = 3kΩ9
R_4,R_9,R_{12},R_{13},R_{16},R_{17} =4kΩ7
R_5 = 2kΩ2
R_7 = 2kΩ00 1%
R_8 = 220Ω

Capacitors:
C_1,C_8-C_{12},C_{14} = 10µF 16V radial
C_2,C_3 = 27pF
C_4,C_5,C_{15} = 100nF
C_6,C_7 = 33 pF
C_{13} = 470µF 25V radial
C_{16} = 22µF 16V radial
C_{17},C_{18} = 2nF2

Inductor:
L_1 = 100µH

Semiconductors:
D_1 = LED, red, high-efficiency, 3mm
D_2 = 1N4001
D_3 = LED, green, high-efficiency, 3mm
D_5 = 1N4148
D_4,D_6 = BAT82
D_7 = 4V7 400mW
T_1,T_2,T_4,T_5 = BD438 or BD140
IC_1 = 80C451CCA68 (PLCC)*
IC_2 = 74HCT373
IC_3 = 27C64 (order code 946644-1)
IC_4 = 74HCT74
IC_5 = LM317L
IC_6 = MAX232
IC_7 = 7805
IC_8 = 74LS07 or 7407

Miscellaneous:
K_1 = DB9 socket, PCB mount, angled.
K_2 = 40-pin ZIF socket with wide slots (e.g., Aries**)
S_1 = CTL3 presskey w. make contact (Multimec)*.
S_2 = 4-way DIP-switch, pref. rotary type.
X_1 = crystal 14.7456 MHz*.
X_2 = crystal 4.332 MHz* (4-6MHz).
Printed circuit board plus programmed EPROM: combination, order code 950003-C (see page 70). EPROM also available separately, see above.

* Suggested supplier: C-I Electronics, P.O. Box 22089, NL-6360-AB, Nuth, The Netherlands. Fax: (+31) 45 241877.
** Aries Electronics (Europe), Unit 3, Furtho Court, Towcester Road, Old Stratford, Milton Keynes MK19 6AQ. Tel. (01908) 260007, fax 260008.

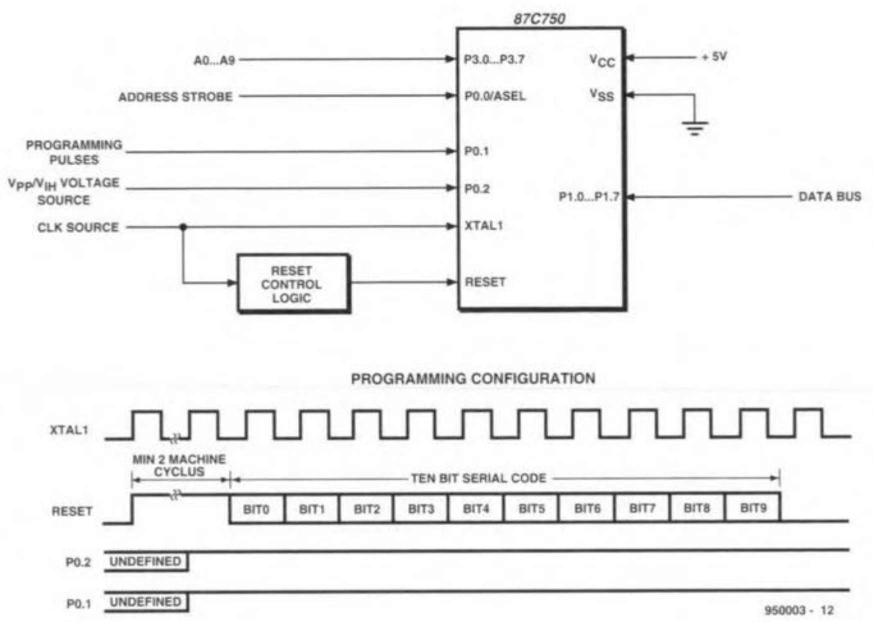


Fig. 3. Timing diagram of the programming preparation sequence for devices from the 75x family. This sequence is not required for the '51 and '52 types.

pacitor (2.2 nF, C_{17} and C_{18}) is added to slow down the rising edge of the voltage at power-on. A similar circuit is fitted on the supply line (C_{15}), only here a resistor (R_5) is connected in parallel to ensure that the capacitor voltage drops to 0 V when the voltage is switched off. LED D_3 lights when the supply voltage is present on the programming socket. It is then **not** allowed to remove or fit a controller! This is only allowed when the LED is out.

All previously mentioned supply voltages are switched by p-n-p transistors type BD438. These are driven by TTL inverters with an open collector output (IC_8) because ordinary TTL gates are unable to switch voltages up to 12 V. Here, a 74LS07 is used. If difficult to find, this IC may be replaced by an ordinary TTL 7407.

Because pins 6 and 31 of the programming socket are sometimes used as data inputs, they can be switched to ground via inverters IC_{8e} and IC_{8f}. They are pulled to V_{cc} via resistors R_{13} and R_{17}. Diodes D_4 and D_6 have been added to prevent the programming voltage from ending up on the V_{cc} line. To be able to read the level on pin 31, R_1 and D_7 have been added. The latter prevents a voltage of 12 V arriving at the input of IC_1.

How it works

When the supply is first switched on, all pins of IC_1 are at an undefined level. The outputs are not made logic 1 until the internally controlled reset sequence is finished. This sequence is initiated by the voltage on capacitor C_1. A number of precautions have to be taken because of the floating outputs during the reset sequence. A key role in this operation is played by IC_8. Short-circuits between the programming voltage and ground would occur if different inputs of IC_8 happen to be low at one time. This risk is eliminated by keeping IC_8 disabled until the reset routine is finished, and by limiting the current (an LM317 with the suffix 'L' starts to limit at about 150 mA). The delayed action of IC_8 is achieved by powering the IC via $+V_{cc}$. Remember, this voltage is not present until a programming cycle is started. An additional advantage of this approach is that the programming and supply voltage on the ZIF socket are both switched off when an error occurs during the execution of the program.

The pin functions of the ZIF socket are different for each processor subfamily. Fortunately, the programmer software is capable of doing almost all the configuration work for you. However, the initial device selection must be done manually with the aid of a DIP switch, S_2. The print on the front panel foil indicates the settings.

During programming, the 87C51, 87C52, 89C51 and 89C52 processors require the external quartz oscillator to run. Crystal X_2 is therefore connected directly to the ZIF socket. The 89C2051 does not require a clock signal during programming, only an occasional pulse at pin X_1. Processors of the 87C750/751/752 type obtain their clock signal via bistable IC_{4a}. This is necessary because the processor has few pins, and has to be configured by a

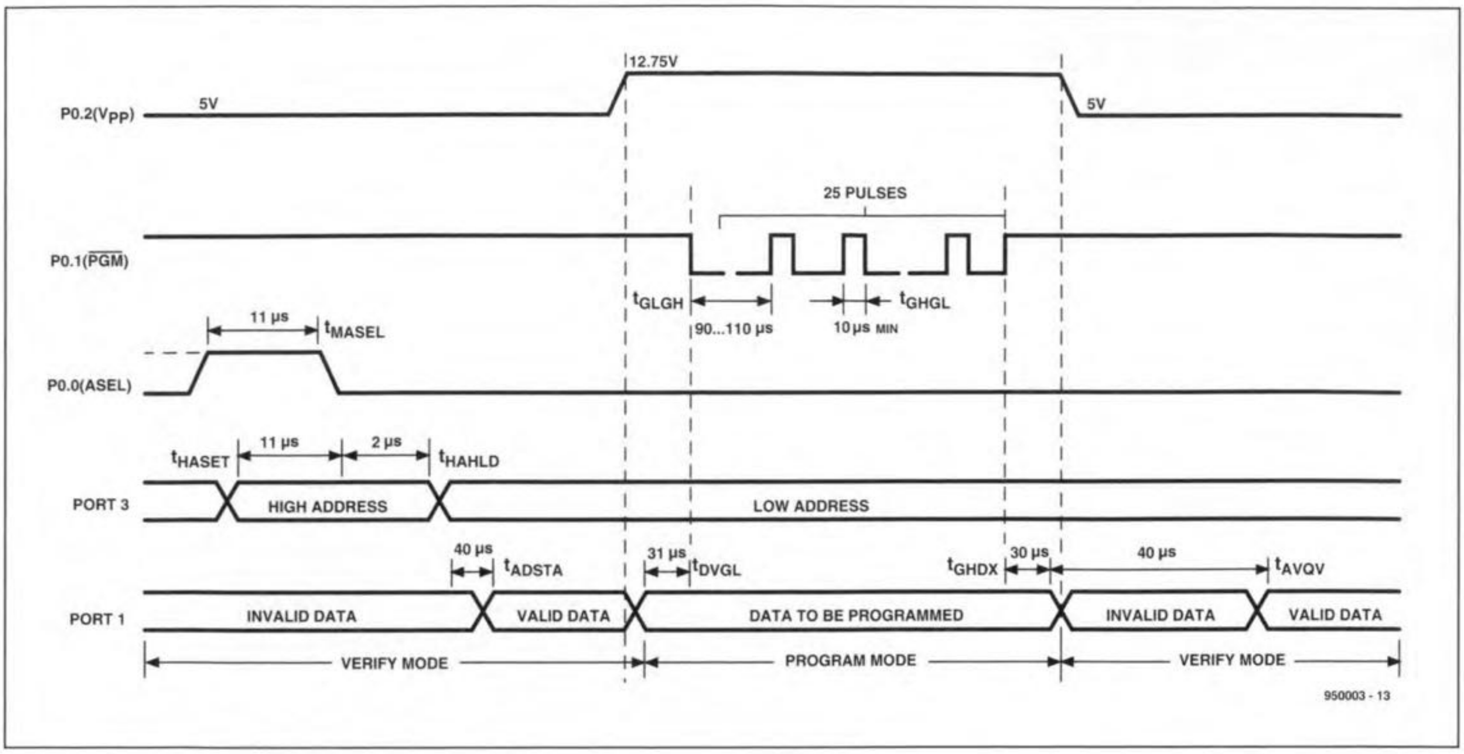


Fig. 4. The actual programming algorithm for the UV (windowed EPROM) and Flash types. With Flash devices, only one programming pulse is needed instead of 25.

10-bit serial code read via the reset input. The frequency chosen is about 1.25 MHz (1/12th of IC_1's clock frequency), which is too low for the 87C51, 87C52, 89C51 and 89C52 processors. Because some computers are too slow to handle a continuous data stream of 9,600 baud, the serial interface includes two handshaking signals, RTS and CTS.

The user interface has a compact and simple structure. LED D_1 lights when switch S_1 is pressed. To check if S_1 is pressed, the processor switches the LED off briefly, but so fast that you will not notice it. The LED is also used to convey error codes to the user. The number of flashes allows you to determine what is going on.

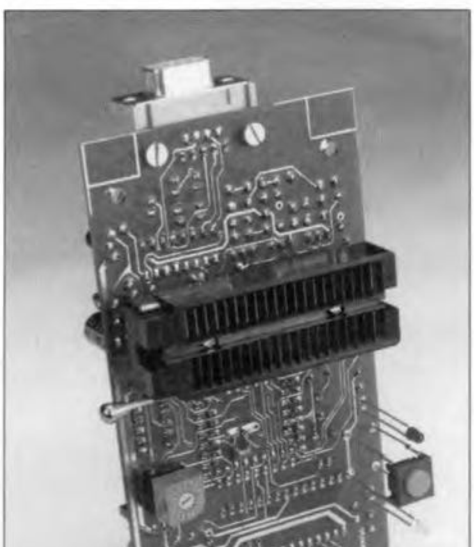


Fig. 5. View of the solder side of the board which holds, among other components, the 40-way ZIF socket.

Construction

Thanks to the compact printed circuit board designed for the project, construction is not expected to cause untoward problems. The track layouts and component mounting plan of the double-sided, through-contacted board are shown in **Fig. 2**. The board is available ready-made through our Readers Services, together with the associated EPROM.

The ZIF (zero insertion force) socket is fitted **at the solder side of the board**, and should protrude from the cover of the enclosure. The board is therefore preferably secured to the cover of the enclosure. In the prototype, the ZIF socket was inserted into a regular 40-pin IC socket which was soldered to the board. This was done to reduce the risk of damage to the expensive ZIF socket while it is being mounted. Be sure to purchase a ZIF socket with wide slots, so that 'skinny-DIP' devices can be fitted also.

The following parts are also fitted at the solder side of the board: D_1, D_3, S_1, S_2 and C_4. Mount these parts, and the ZIF socket, last. If you don't, it becomes impossible to fit, for instance, IC_8.

Getting started

Once all soldering work is finished, you can start testing the programmer. Do not insert the ICs yet in their sockets. The voltage regulators, however, should be soldered in place. Connect an input voltage of about 16 V, and use a DMM to check the presence of the regulated 12.75 V and 5 V supply voltages. Tolerances of ±5% are acceptable. Switch off the supply voltage, and insert the ICs into their sockets. Be sure they are fitted the right way around! Apply power again. Nothing should happen; the LEDs remain out. Set the DIP switch to 0 and press S_1. The power LED should light briefly, while the error LED, D_1, lights as long as S_1 is pressed. Run a communication program set to 9600,N,8,1 on your PC. Connect an RS232 cable, and look at the programmer's output on your PC. Pay attention to the handshaking. If this is not enabled, the programmer will not send anything. Any time you press S_1, data is being sent to the PC. This data consists of an empty Intelhex file, an ID (three times FF as long as there is no processor in the ZIF socket), and the value read from the DIP switch (00). Check all positions of S_2 by adjusting it, pressing S_1, and looking at the last byte on the PC screen. Only codes 0 through 7 are valid. Higher values are not used and should not be set on the switch. The setting of S_2 is very important because it determines the protocol with which the processor will be programmed. If all of this checks out, set the switch to

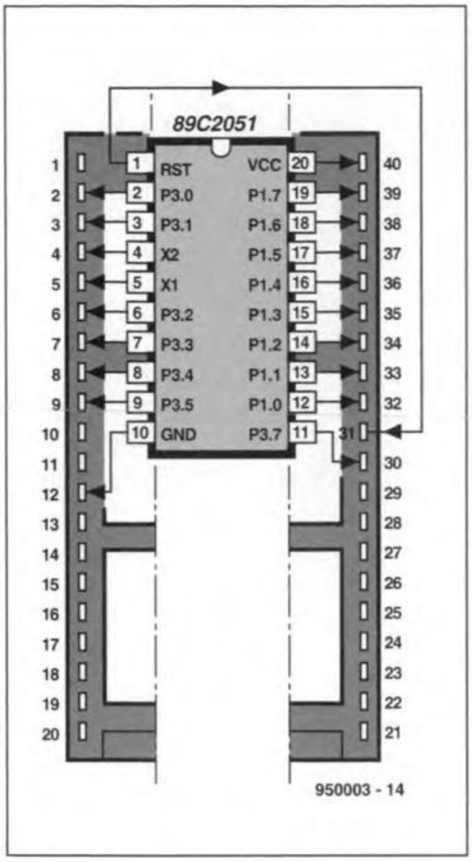


Fig. 6. To be able to program 89C2051 devices, you need this adaptor socket.

Flash processors

A Flash processor combines the advantages of an EPROM with those of a static memory. Once the data are stored safely in the flash memory, they remain there for many years even after the supply voltage has disappeared. Erasing the memory is possible without physically removing the chip from the circuit. This type of processor can be programmed with the aid of a programming voltage of 5 V or 12 V. In both cases, the same processor is used; it is only one bit which determines the level of the programming voltage! It is not possible for you, the user, to change this bit. The programming voltage may be derived from the type number: the AT 89C51xxxx is the 12-V version, and the AT89C51xxxx-5 is the 5-V version. Note that the programmer described in this article can only handle the 12-V versions.

The Flash memory can only be erased as one block. Consequently, you can not erase a section of the memory, or just one byte. A complete erasure operation takes only 10 ms, and can be repeated at least 1,000 times.

The actual programming pulse has a length between 1 µs and 110 µs. That does not mean, however, that the programming speed can be increased by varying the length of the programming pulse.

The programming pulse initiates a process in the Flash memory which eventually ensures the final storage of the data. The programmer may test if the data is stored yet by means of a data polling process. During programming, data read back from the Flash memory will be returned with bit D7 inverted with respect to the original level. As soon as the data is stored in the memory, they are also returned correctly. On average, storing a byte in Flash memory takes about 1.5 ms. The data polling operation has no effect on this duration. Incidentally, a 'ready' signal is available on pin P3.4 with the same function.

the position required for the processor you wish to program, press S_1, and insert the processor in the ZIF socket. Be sure to do this only when both LEDs are out.

Programming

The fact that all processors mentioned have roughly the same programming procedures was a considerable simplification in the design of the programmer. The main differences are:

- the 87xx types are programmed at 12.75 V, and receive 25 pulses of 100 µs;
- the 89xx types are programmed at 12 V , and receive one pulse with a length between 1 and 110 µs;
- the xx51 and xx52 types receive their data, mode and address information directly because they have enough pins;
- the 87C75x types receive their mode word serially via reset and clock. The address is multiplexed, while data are conveyed directly.
- the 89C2051 receives mode and data directly at the relevant pins. The address is supplied by an internal counter which is incremented by one on each clock pulse.

Unfortunately, a complete description of the programming algorithms is beyond the scope of this article. Interested readers are referred to the datasheets published by Intel. As an example, we take the processors from the 87C75x series. Once the supply voltage is present, the reset input is held high for at least two machine cycles. Next, P0.1 and P0.2 are pulled logic high. The 10-bit code is read (bit 0 comes first) to place the processor in the correct mode. The code is '296H' for the program as well as the verify mode. The data at the reset input is read when the clock signal is high. Changing the level is only allowed when the clock is low. Next, the reset input is made low, and remains so during the entire programming sequence. Details of the sequence are shown in **Fig. 3**.

The subsequent programming process is highly structured, and virtually identical to that for the 87(9)C51 and the 87(9)C52, as illustrated by **Fig. 4**.

Operation

The programmer is simple to operate, mainly because it has only two LEDs and one presskey. At power-on, the LEDs are off, and there is no voltage on the programming socket. Turn the DIP switch to the position required for the processor you wish to program. If you don't, the high programming voltage may arrive on the wrong processor pins, or small processors may be read several times. Next, fit the processor in the ZIF socket. Attention: pin 1 is al-

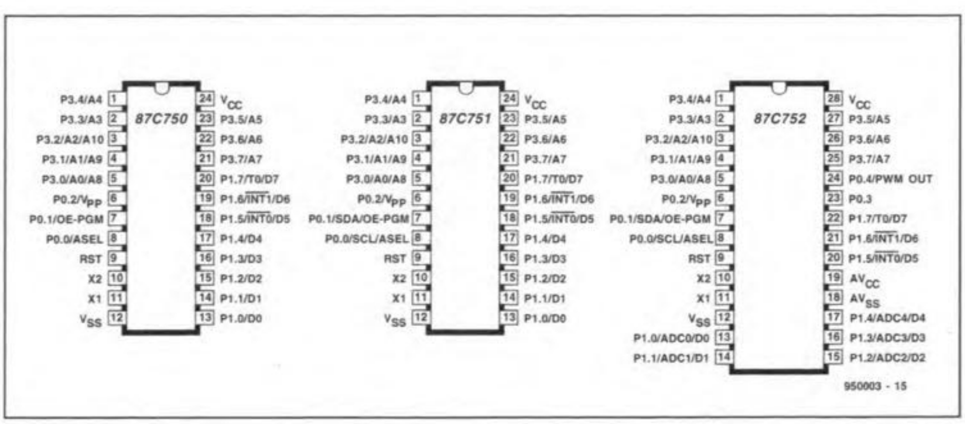


Fig. 7. Pinouts of the less well known processors, 87C750, -751 and -752.

ID byte structure

byte1:	1E Atmel 89 Intel 15 Philips
byte2:	21 means 89C2051 (Atmel) 51 means 89C51 (Atmel, Intel) 52 means 89C52 (Atmel, Intel) 92 means 87C51 (Philips) 97 means 87C52 (Philips)
byte3:	FF means high programming voltage 05 means 5 V programming voltage

PROCESSOR FAMILIES

DIP switch	Type	EPROM/Flash	RAM	pins	I/Os	Clock
0	87C51	4 K EPROM	128	40	32	0.5-3.5-33 MHz
1	87C52	8 K EPROM	256	40	32	3.5 - 24 MHz
2	87C750	1 K EPROM	64	24	19	3.5 - 40 MHz
3	87C751	2 K EPROM	64	24	19	3.5 - 60 MHz
4	87C752	2 K EPROM	64	28	21	3.5 - 16 MHz
5	89C51	4 K Flash	128	40	32	0 - 24 MHz
6	89C52	8 K Flash	256	40	32	0 - 24 MHz
7	89C2051	2 K Flash	128	20	15	0 - 24 MHz

Notes:
- The 89C2051 can only be programmed if an adaptor socket is used (see Fig. 6).
- Clock frequencies are guidance only, manufacturers are producing ever faster versions.
- Note that the 87C75x does not recognize some instructions such as MOVE, LJMP en LCALL. Do not use them with this processor; your program will not work if you do. When working with a higher programming language, be sure to use a compiler which is explicitly known to support this processor.

PROGRAM

Pin	87C5x	89C5x	87C75x	89C2051
RST	H	H	mode*, L	12 V
PSEN	L	L		
ALE/PROG	25x100 μs L	1-110 μs L		
EA/Vpp	12.75 V	12 V		
P2.6	L	L		
P2.7	H	H		
P3.2				1-110 μs L
P3.3				L
P3.4				H
P3.5				H
P3.6	H	H		
P3.7	H	H		H
P0.0/ASEL			address strobe	
P0.1/PROG			25x100 μs L	
P0.2/Vpp			12.75 V	

VERIFY

Pin	87C5x	89C5x	87C75x	89C2051
RST	H	H	mode*, L	H
PSEN	L	L		
ALE/PROG	H	H		
EA/Vpp	H	H		
P2.6	L	L		
P2.7	L	L		
P3.2				H
P3.3				L
P3.4				L
P3.5				H
P3.6	H	H		
P3.7	H	H		H
P0.0/ASEL			address strobe	
P0.1/PROG			H	
P0.2/Vpp			H	

* device mode word = 296H

ways in the same position! LED D_3 indicates the presence of the supply voltage on the ZIF socket, and D_1 flashes to signal a fault condition. The number of flashes between longer pauses indicates the type of error:

1	not blank
2	program/verify error
3	could not be erased
4	timeout error
5	stopped by user
6	data checksum error
7	data nibble is not 0 through F
8	buffer overflow

When the timeout error occurs, this means that the processor has not seen any data for more than one second. Error codes 6 and 7 indicate corruption of the Intelhex file. An overflow error is generated as soon as the amount of data in one line exceeds 16 bytes.

You can then run a blank check, or start programming straight away. It is recommended to always run a blank check first.

Blank check
Press the switch. A blank check is run, and D_3 lights. If the processor is empty, the LED goes out again, and the circuit returns to standby. Via the RS232 link, an empty Intelhex file (:00000001FF.) is sent, followed by the ID (not with 75xx devices) and the setting of the DIP switch.

If the processor is not blank, D_1 starts to flash, and the contents of the processor are sent to the PC in the form of an Intelhex file, followed by the ID and the position of the DIP switch. D_3 will remain on until the complete file has been transmitted. D_1 however will continue to flash, so that sufficient

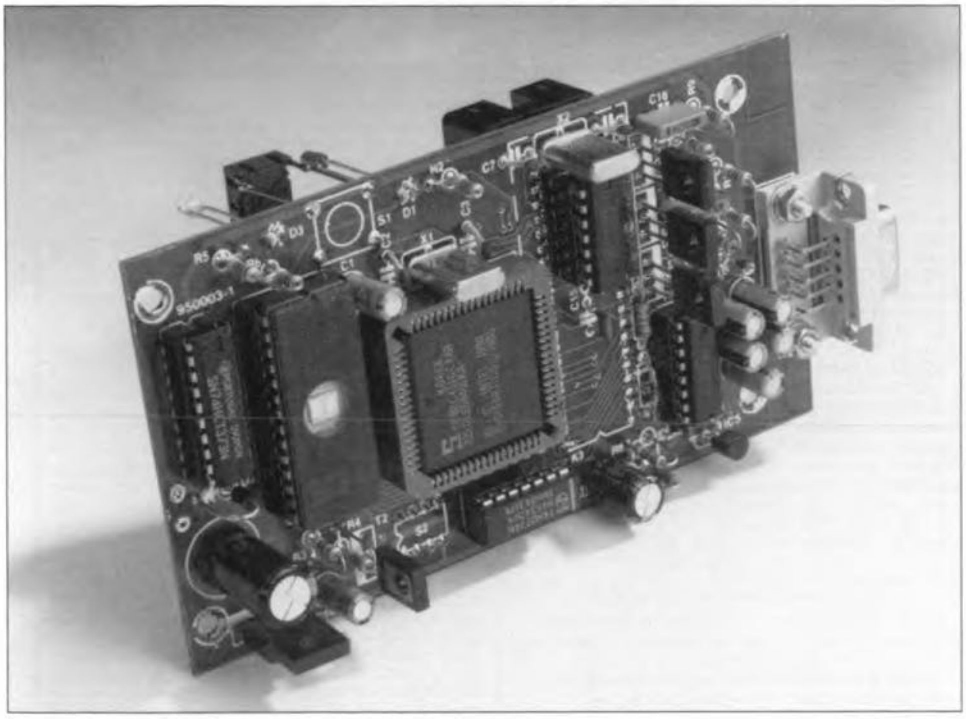


Fig. 8. View of the component side of the completed prototype board. Any computer sporting a serial RS232 interface may be used to communicate with the programmer.

time is available to see which fault condition has arisen (in this case, error number 1). To make D_1 go out, you have to press the switch once more. The standby state is signalled by both LEDs remaining off.

Should you wish to interrupt the blank check test, press the switch while D_3 is still on. The blank check is then terminated, whereupon D_1 signals error code '5'. Clear the error code by pressing the switch again.

Programming

Transmit an Intelhex file from the PC. This can be done with any communication program (such as Procomm), or simply from DOS with the aid of the COPY command. The data you transmit are automatically programmed, even if the processor is not blank. D_3 lights. If an error occurs, it is signalled by D1, and the programming sequence is halted at the faulty byte. The programming operation may be interrupted at any time by pressing the switch. D_1 will then signal the relevant error condition ('5'). Clear the error code by pressing the switch again.

Nothing is programmed when the Intelhex file consists of :00000001FF only. This string is used to erase the Flash types because the programmer software always clears Flash memories before programming them.

The software does not have a separate verify option. It is however possible to verify the contents of the processor by doing a second programming run with the same data. If the programmer does not report errors, the data in the EPROM or Flash memory matches the code to be burnt into the processor. This approach is possible because a verify operation is executed internally after programming each byte. D_1 again indicates the status of this verification process. The contents of a processor is simple to obtain by doing a blank check with the aid of a communication program.

The present programmer has no facilities to program the security bits and encryption key of the relevant processors.

The number of databytes in the Intelhex file may not exceed 16 per line. If the line is longer, an error condition is signalled because the internal buffer then overflows. The data are checked (with the aid of the checksum) **before** programming. Should the hex file contain addresses which are outside the address range of the processor, this information is still programmed at a lower 'mirror' address. (950003)

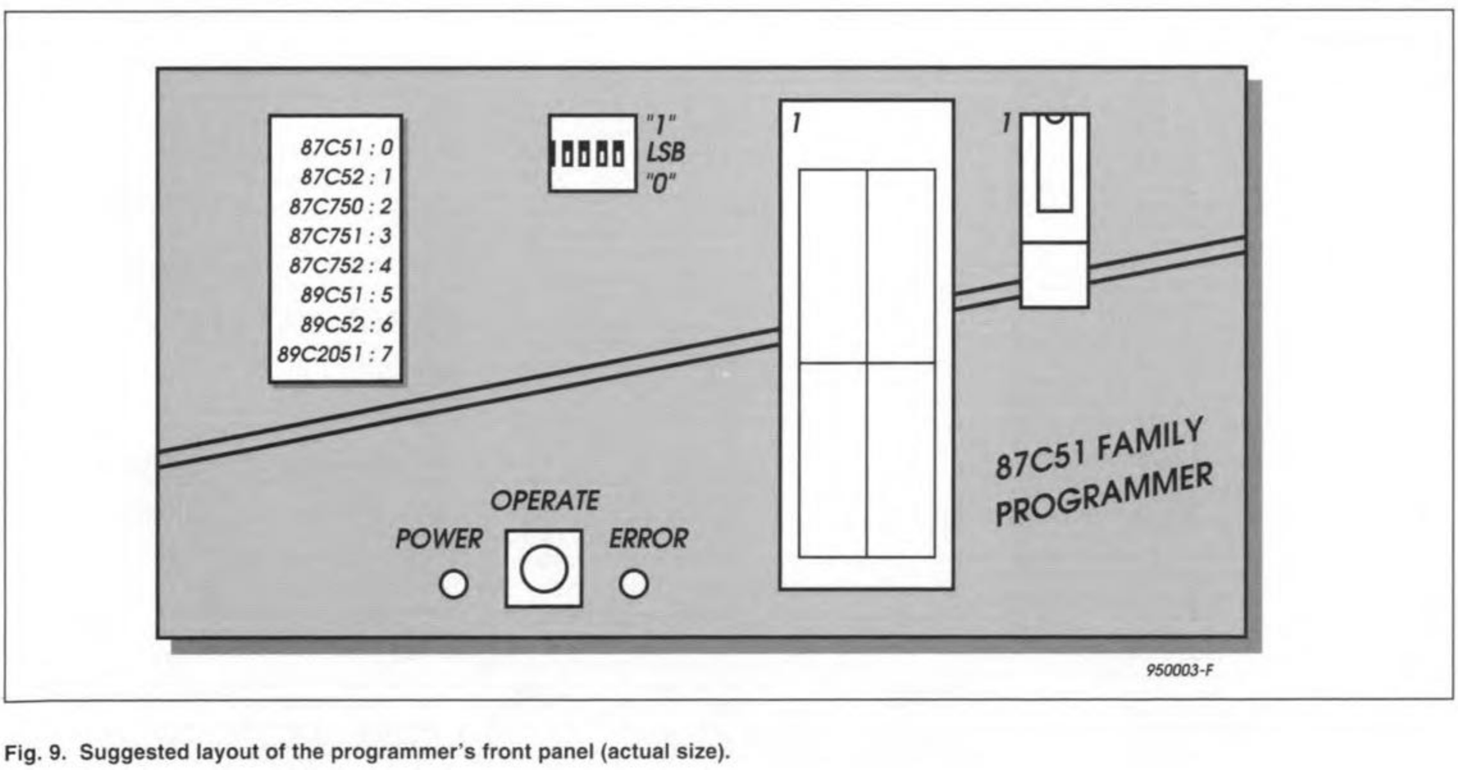
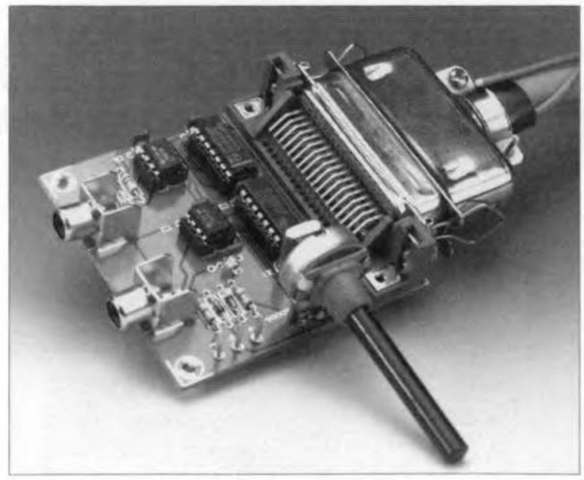


Fig. 9. Suggested layout of the programmer's front panel (actual size).

PROGRAMMABLE SINE WAVE GENERATOR



Design by J.C. Feltes

A circuit is described that, controlled by a computer, generates perfect sine wave signals.

The generator,controlled by a DOS command at the printer port of a computer, generates sine waves in the frequency range 1 Hz to 50 kHz. It is based on a monolithic chip from Micro Linear, the Type ML2036, whose block diagram is shown in **Fig. 1**.

The ML2036 uses direct digital synthesis and has been designed to generate sinusoidal signals in conjunction with a digital control. The frequency of these signals is derived from an external crystal or clock input. The frequency is programmed by a 16-bit serial data word. The output signal has an amplitude of $\pm V_{ref}$ or $\pm V_{ref}/2$. The IC also has an inhibit input which allows the sine wave output to be held at 0 V after completing the last half cycle of the sine wave, thus preventing steps in voltage. Two pins of the IC are clock outputs designed to drive other devices with one half or one eighth of the clock input frequency.

Circuit description

In **Fig. 2**, gates IC_{2a}, IC_{2b}, and IC_{2c}, buffer the control signals from the computer: AutoFeed, bit 3; Select, bit 1; and Data Strobe, bit 0. The levels of these signals may be altered (by software) via computer registers $37A_H$ (LPT_1) and$27A_H$ (LPT_2). The signals are inverted because they are active low, whereas the ML2036 needs active high signals.

The ML2036 is controlled by serial data SID; serial clock signal SCK at whose instruction each bit in the data word is read; and strobe signal LATI which ensures that the input data are stored in the register. These signals are generated in the computer and applied to the present circuit via the Centronics.

As mentioned before, the frequency of the sine wave output is programmed by a 16-bit serial data word. The relationship between the output frequency, crystal frequency and the data word is

$$f_o = f_c \cdot \text{data}/2^{23} \qquad \text{[Hz]}.$$

The sine wave output is available at pin 10 of IC_1, from where it is applied to comparator IC_3. During the positive half of the signal, the output of IC_3 is high; during the negative half, low. Network R_1-D_1 converts the output of IC_3 into a rectangular signal that switches from 0 V to +5 V. This signal is brought to TTL standard (as regards level and edge steepness) by IC_{2d}, which is available at K_2. The fanout at this connector is 5.

The output of IC_1 is also applied, via P_1, to voltage follower IC_4, which buffers the output. The signal at K_3 is a symmetrical sinusoidal signal whose amplitude may be varied between 0 V and 5 V with P_1.

The power supply is conventional: the unregulated input voltage is stabilized by two zener diodes, D_2 and D_3, to a symmetrical ±5 V output.

Construction

The generator is intended to be built on the printed-circuit board shown in **Fig. 3**. Commence the construction by fitting K_1 with two suitable M3 screws, nuts and washers, followed by the necessary solder pins, sockets K_2 and K_3 (use gold-plated types if at all possible) and the IC sockets. Cut the spindle of potentiometer P_1 to the required length and solder the component directly to the board. The potentiometer may be replaced by a link between pin 10 of IC_1 and pin 3 of IC_4 if a variable output level is not required.

The crystal frequency may be anywhere between 3 MHz and 12 MHz; in the prototype a 6 MHz type was used. If a crystal with a frequency outside this range is used, the software must be adapted as required.

Software

The BASIC routine is contained on diskette EPS956005 (see p. 70). The diskette contains two further programs. SINUS.EXE enables the sine wave generator to be accessed and set via

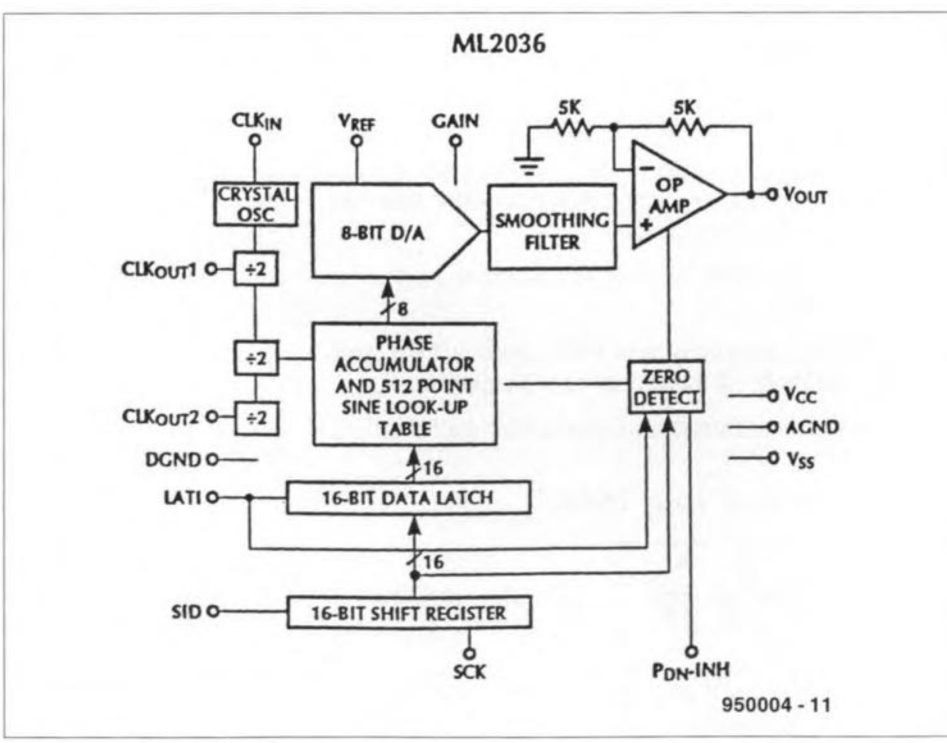


Fig. 1. Block diagram of sine wave generator IC Type ML2036.

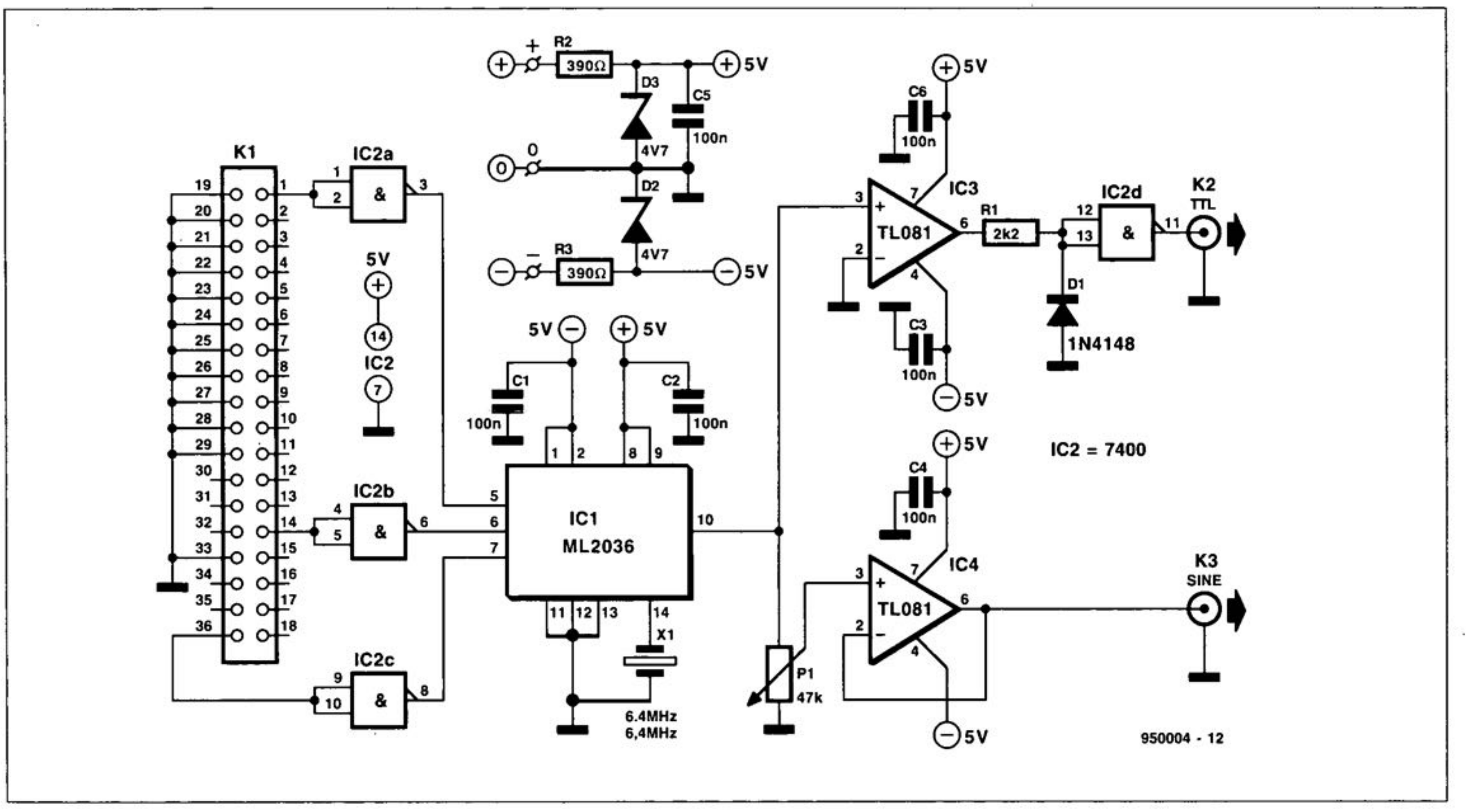


Fig. 2. Circuit diagram of the programmable sine wave generator.

the command line (note that in this program the crystal frequency is fixed). The SINUS command needs two parameters: the number of the printer port (1 or 2) and the sinusoidal frequency in hertz.

Command SINUS.EXE 1 1000 <<cr>> arranges a sinusoidal signal of 1000 Hz at the output of the sine wave generator connected to LPT1.

Command SINUS <<cr>> requests the printer port used and the desired frequency. The set frequency is then shown as a text on the display.

SINUS/h provides an auxiliary menu that tells how SINUS.EXE should be used. Additionally, the set frequency is indicated.

SINUS/? indicates the set frequency only.

The third program is SIN-GEN.EXE, which shows a keyboard on the display. Alongside this is the sine wave voltage in a graphic window; the exact frequency of this signal is also indicated. **Figure 4** shows how this DOS program relates with the user. Individual preferences may be selected from the various options on the menu bar. This program also has an auxiliary menu that gives the required information.

Parts list

Resistors:

R_1 = 2.2 kΩ
R_2, R_3 = 390 Ω
P_1 = 47 k, linear potentiometer

Capacitors:

C_1–C_6 = 100 nF

Semiconductors:

D_1 = 1N4148
D_2, D_3 = zener 4.7 V, 500 mW

Integrated Circuits:

IC_1 = ML2036CP (Micro Linear)
IC_2 = 7400
IC_3, IC_4 = TL081

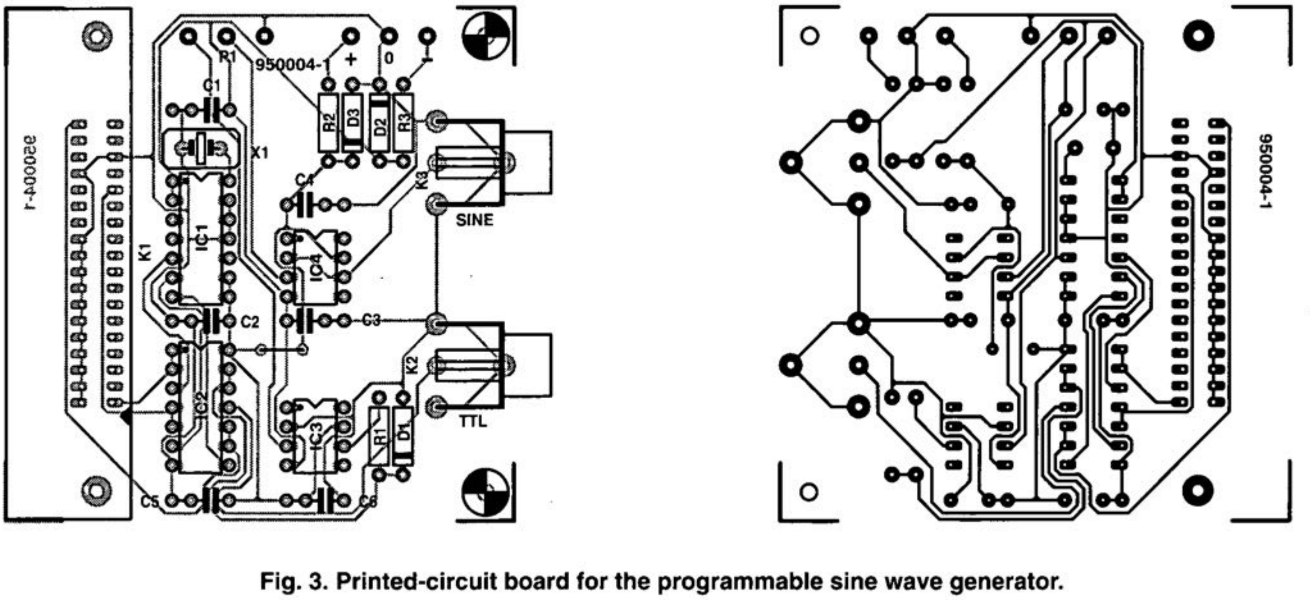


Fig. 3. Printed-circuit board for the programmable sine wave generator.

Miscellaneous:
K_1 = 90° Centronics socket for board mounting
K_2, K_3 = 90° audio socket for board mounting
X_1 = crystal, 3–12 MHz
Diskette, 3.5 in, MS-DOS, Order No. EPS956005 (see p. 70)*
PCB Order No. 950004 (see p. 70)*

*These items may be ordered as one package Order No. 950004-C (see p. 70)

Reference: 'Micro Linear 2035 & 2036 Programmable Sine Wave Generators', *Elektor Electronics*, October 1993, p. 40

[950004]

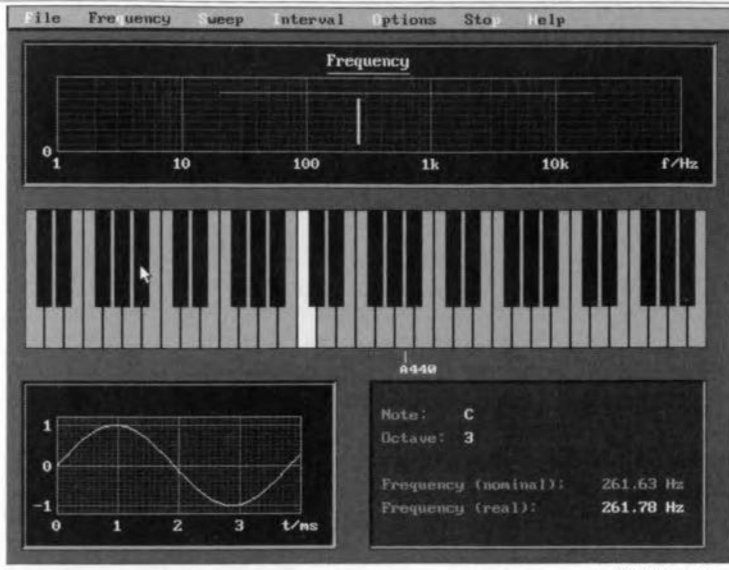


Fig. 4. Illustrating how the SIN.GEN.EXE program communicates with the user.

How the sine wave is created

A complete sine wave is synthesized by Direct Digital Synthesis (DDS) from 2^{21} phases or fractions. The 16-bit data register decides how many of these fractions will be enclosed by two successive clocks. In this way, low values in the 16-bit register produce low frequencies since many of the phases are scanned. High 16-bit values result in high frequencies since many phases are ignored.

In the adjacent flow diagram, the adder and the latch form the phase accumulator that contains the actual phase status. The accumulator is clocked at $f_{clk(in)}/4$. The value stored in the data latch is added to the phase accumulator every four cycles of $f_{clk(in)}$. The frequency of the analogue output is equal to the rate at which the accumulator overflows and is given by:

$$f_a = [f_{clk(in)} \times (D_{15} - D_0)_{DEC}]/2^{23}.$$

Thus, the length of each phase caused by a change of 1 in the content of the register is given by:

$$\Delta f = f_{clk(in)}/2^{23}.$$

For instance, if the crystal frequency, f_c, is equal to the clock, $f_{clk(in)}$ = 4.194304 MHz,

$$\Delta f = 0.5 \text{ Hz}$$

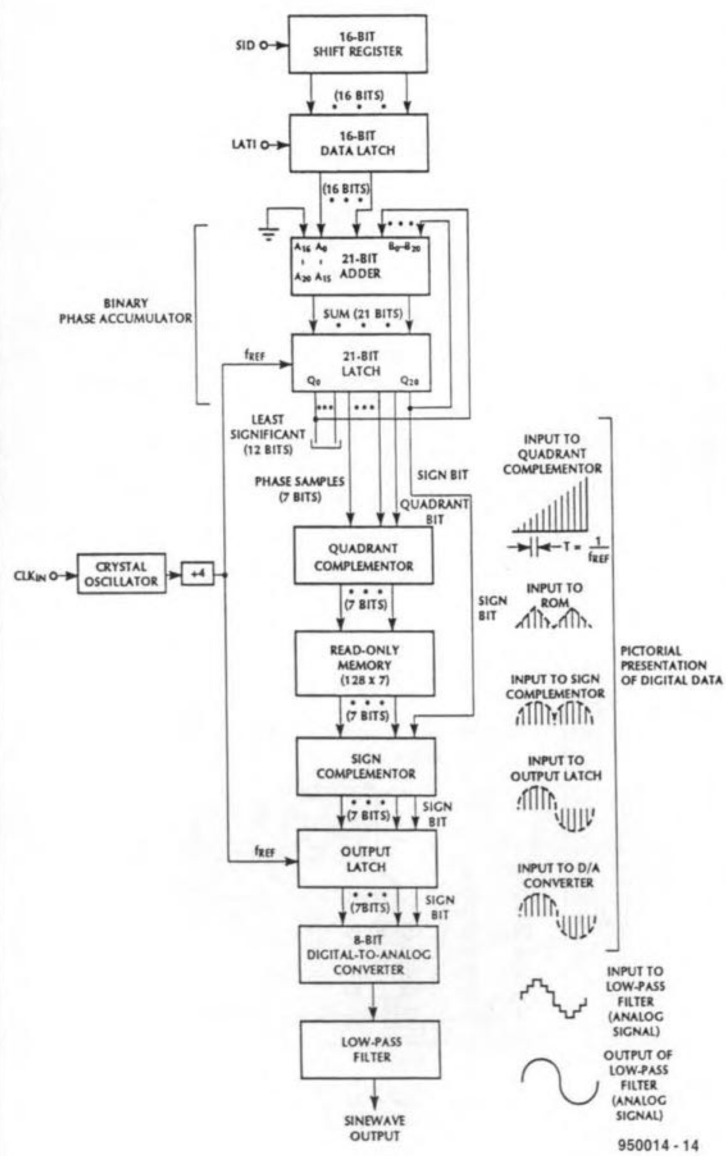
and

$$f_{a(max)} = f_{clk(in)} \times 2^{16}/2^{21} = 32.768 \text{ kHz}.$$

Only the eight highest valued of the 21 bits are processed further; the highest valued bit is the polarity symbol.

Apart from symmetry around the time axis, a sine wave also has symmetry with respect to the peak values of the waveform. With the aid of the second highest bit, the quadrant complementor translates the 7-bit phase value in such a manner that a sine look-up table with values for 128 phase steps between 0 and $\pi/2$ suffices to compute the output amplitude.

The next step in the process is the reemployment of the polarity symbol to ascertain whether a positive or negative half wave must be output (sign complementor).



THE DIGITAL SOLUTION

Part 6 – Data storage media and data processing

By Owen Bishop

In this series we look closely at digital electronics, what it is, what it does, how it works, and its promise for the future.

Storing binary data in electronic devices is relatively expensive and there may be a loss of the data when the power is switched off. For storage of large quantities of data at minimal expense, we rely chiefly on magnetic or optical methods, sometimes a combination of both. Ideally, the medium should allow the data to be stored permanently, with the option of amending it or adding to it, as required, and access time should be short. We examine various techniques in detail and see how they measure up to the ideal.

Magnetic storage

The most commonly used medium for mass storage data is a film of magnetic material, usually coated on to a disk or tape. Within minute local regions of the coating, known as *domains*, all the magnetic particles are arranged in the same direction. Thus, the domain acts as a tiny magnet. On an unrecorded disk or tape, the domains are magnetized in many different directions, at random, so the overall effect is that the film is unmagnetized (see **Fig. 48a**). If a magnetic field with strength greater than a certain threshold value is applied to the film, all the domains in that region will 'flip' so that they are all magnetized in the same direction (see **Fig. 48b**). The domains retain their orientation after the magnetic field has been removed. A minute permanent magnet is created in that region. The disk is spun and data recorded in concentric tracks, as a series of magnetic regions, oriented one way to represent 0 and in the opposite direction to represent 1.

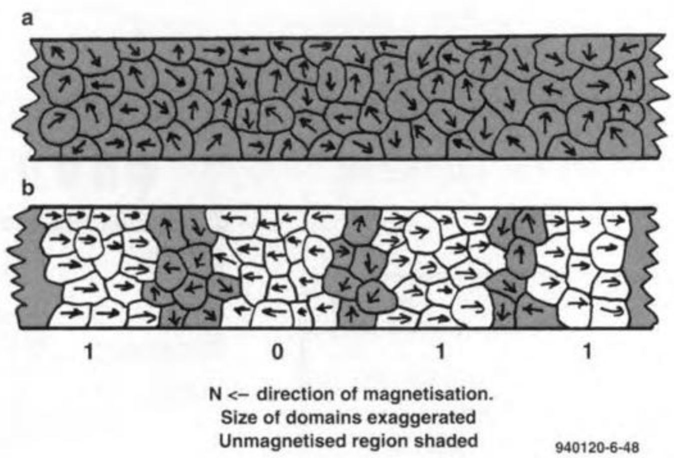


Fig. 48

In a *diskette* (or *floppy disk*) drive, the disk is removable and enclosed in a stiff plastic case, with a metal shutter which slides back automatically when the disk is inserted into the drive to expose part of the disk surface to the head. A typical floppy disk can store up to 1.4 megabytes of data. The disk spins at about 360 revolutions per minute but, even so, it still takes an appreciable time for the magnetic head to be moved to the correct track and sector. Data can be read at rates of several hundred bits per second. A typical access time is 200 milliseconds, which is much slower than the 25–150 *nano*seconds of RAM or ROM. For this reason, diskettes are suitable for permanent storage, but not for accessing data during the running of a computer program.

A *hard disk* drive has one or more aluminium disks with magnetic coating, attached to the same spindle. It spins at high speed (about 3600 revolutions per minute). This results in an aerodynamic effect in which the magnetic head 'floats' in a film of air, very close to the disk. Because the head is close to but not actually touching the magnetic film, it is possible for the tracks to be closer together than on a diskette. Because of the high speed of rotation, the reversals of direction of magnetization are closer together. In other words, the bits are more densely packed on the disk, with the result that a hard disk stores much more information than a diskette. Some have a storage of 1 gigabyte (a thousand million bytes) or more. The high speed of rotation and high density of storage mean that access time is only about 20 milliseconds. This makes the hard disk suitable for storing working data in a computer. Data can be transferred from hard disk to RAM at speeds which do not appreciably impede the action of the microprocessor. The hard disk is completely sealed in its case when it is manufactured to exclude dust and smoke particles which would damage the surface of the disk.

A *Bernoulli* drive records data on a flexible magnetic disk, but makes use of aerodynamic principles to bring the head close to the disk without actually touching it. The principle is similar to that of the hard disk drive, except that the head is fixed and the disk is drawn upward toward the head. As with the hard disk, the closer the head, the smaller the magnetic regions and the more data can be packed on to the disk. A Bernoulli drive stores 35–150 megabytes per disk, depending on the disk diameter.

Compact disks

In a compact disk, or CD, data is stored as a series of dimples on the surface of a reflective metal layer, a dimple corresponding to a 1 and the absence of a dimple corresponding to a 0. At present, CDs are read-only and are often referred to as CD-ROMs. Data is recorded on the master disk with a laser beam and is subsequently read from the copy disk with a laser of low power. A CD stores about 600 megabytes of data, so it is equivalent to 400 double-density 3.5" floppy disks. But data transfer from a CD is so fast that high-quality digital sound (16-bit resolution, with a sampling rate of 44.1 kHz) can be played from a sound card in a computer. A CD can store 100 high-resolution photographic images in full colour and can provide the data for on-screen motion-pictures. A major British supplier, RS Components Ltd, now issues its component catalogue (over 1700 closely-packed pages, fully illustrated in colour) as a CD. The CD also includes the company's data sheets, and programs to allow the user to access the catalogue in a number of different ways, making it easy to find, assess, and order components. Comprehensive reference books are now being published in CD form, as are elaborate computer games with brilliant animated graphics and sound effects.

Optical disks

Optical disks depend on changes in the reflectivity of the surface of a plastic disk when it is subjected to the action of a laser beam. It is later read by using a lower-power laser beam and measuring the amount of reflected light. In one type of optical disk the plastic is made more reflective by the laser action. Such a disk is read-only. Another type of optical disk initially has a crystalline surface with high reflectivity. The beam used for writing destroys the crystalline structure and reduces reflectivity. This type of disk can be written to repeatedly; the laser is energized at high power where it is to destroy the crystalline structure and produce a new spot, or at medium power where it is to melt and recrystallize the surface to erase a previously recorded spot.

Magneto-optical disks

The disk is made from a magneto-optical material. When saving data, the disk is spun in a magnetic field. A pulse from a laser melts the material, and its molecules orientate themselves in the field. The material immediately cools, leaving the small spot permanently magnetized, representing a 1. Magnetization alters the reflective properties of the surface of the disk, which can be detected during reading as a change in the direction of polarization of a reflected laser beam. This type of disk can be written to repeatedly and stores between 120 Mbytes and 1 Gbyte, depending on the disk diameter.

Current research is directed towards cramming on to the disk as much data as possible. One limitation in increasing data density is the diameter of the laser beam. With a red laser, the type normally used in disk writing and reading, the minimum practicable diameter is 0.8 μm. Researchers at IBM have now produced a laser that emits blue light. Blue light has a wavelength about half that of red light, so the minimum diameter of the beam is 0.4 μm. This means that approximately four times as much data can be stored on the disk, making it possible to store up to 6.5 gigabytes on a 5.25-inch magneto-optical disk.

The digital solution

The earlier reference to CDs and CD-ROMs illustrates some of the advantages of digital methods in contrast to analogue methods. A music CD is smaller than its counterpart, the 12-inch vinyl disk, yet it stores more information and with greater precision. Not only is the playing time slightly longer, but the quality is superior. There are some who may dispute this for subjective reasons, but most people listening to a recording of a symphony orchestra will agree that, quite apart from the improved reproduction of the characteristic sounds of each instrument, it is far easier to pick out the musical parts and the individual instruments. The data that leads to this improved fidelity, and so gives this improved perception of the orchestra, is additional to the data recorded by analogue techniques.

A second advantage of the digital technique is that the data is less likely to be affected by external influences. Analogue circuitry is especially prone to electromagnetic interference, the effects of ambient temperature, and noise, to name but three sources. Digital circuitry, with its robust two-level operation, is immune to all except the most gross interference and distortion. Data stored on a hard disk is virtually immutable and is transferable to RAM by digital circuits with every bit retaining its correct value. Analogue techniques lack this precise reproducibility.

Returning to the example of the music CD, the data stored there is an exact bit-by-bit copy of that recorded on the master disk. The music sounds exactly the same, and *is* exactly the same, on all copies. There is virtually no wear (a bugbear of the vinyl disk), so that the thousandth playing is precisely the same as the first. Moreover, we do not experience the 'colouring' of the sound owing to resonances and other influences of the disk surface, the stylus, the cartridge and the playing arm that occur in an analogue system. The 0s and 1s detected on the surface of the CD are faithfully processed by digital circuits as far as the digital-to-analogue converter in the player.

A CD is less liable to mechanical damage than a vinyl disk, but this is more a consequence of its physical structure. However, if a CD has minor blemishes on its surface, these need not have such a devastating effect as they do on a vinyl disk. This is because of the ability of digital systems to process the data in between reading it from the disk and sending it to the DAC. Music data is not simply recorded as a series of sample values converted to binary by an ADC. After conversion, the data is processed or coded. One reason for doing this is to compress it, making it possible to store more data on a disk of given diameter, or on a tape of given length. Data compression is not applicable to analogue signals. Another aspect of coding is error checking. A surface flaw or a speck of dirt on the surface of a vinyl disk produces a clearly heard 'pop' or 'click'. Similarly, a flaw in the coating of an analogue audio tape causes 'dropout', a temporary but noticeable loss of the signal. By contrast, as the data is read from a CD, it is continually being examined to check that it is error-free. If an error is detected, the digital circuits take immediate steps to correct it. Sudden peaks or troughs in the output signal are smoothed over. Gaps in the signal are patched by error-removing circuitry. These corrections are made as the data is being fed through from the reading head to the DAC and the listener is usually unaware that anything untoward has happened. Digital techniques allow for error checking and error-correction in a way that is quite impossible with analogue methods.

Much of what has been said in the context of the music CD applies also to other applications of digital techniques. In telecommunications, for example, the use of digital methods of data compression and error-checking make it possible to achieve reliable high-speed transmission of data across world-wide networks. Many telephone conversations (including computers talking to computers) may be transmitted along a single cable (usually an optical fibre cable) as a result of data compression.

During the earlier discussion, there have been several references to the *processing* of digital signals (or data). We can process analogue data, too. For example, we can amplify it, filter it, and modulate it. But digital data lends itself to processing in ways that are not possible with analogue data. Error-checking and correction has been mentioned as an instance of this. The discussion above has also implied that data processing is often done in *real time*, that is to say, while the data is actually being generated. This is not always the case; for example, the data from a scientific experiment can be stored on a disk as a data file and subsequently analysed by statistical software. But, when a music CD is played, or when we send a fax, data is being processed in real time. The music issues from the loudspeaker, or the fax issues from the remote fax machine, only a fraction of a second after the data is generated.

A CD player reads 44 100 16-bit samples per second and each sample has to be processed before being sent to the DAC. Similarly, a fax machine transmits 9600 bits or more per second. Rapid processing is a leading requirement. Other essential features are freedom from error, minimal circuit size and complexity, and lowest possible cost. We now look at digital signal processing to see how these aims may be achieved.

Signal conversion

Most applications of electronics begin with the analogue input from a sensor (microphone, thermistor, photocell) and finish with an analogue output to an effector (motor, lamp, loudspeaker). The choice is whether to link the sensor and effector by analogue circuits or whether to set up at least part of the link in digital for. For the reasons given earlier, and for several other reasons, it is usually preferable to convert the signal from the sensor to digital form at as early an age as possible, to process it digitally, and then to leave the conversion from digital to analogue to the last possible stage. Although this necessarily requires two extra stages, ADC and DAC, the advantages of digital as opposed

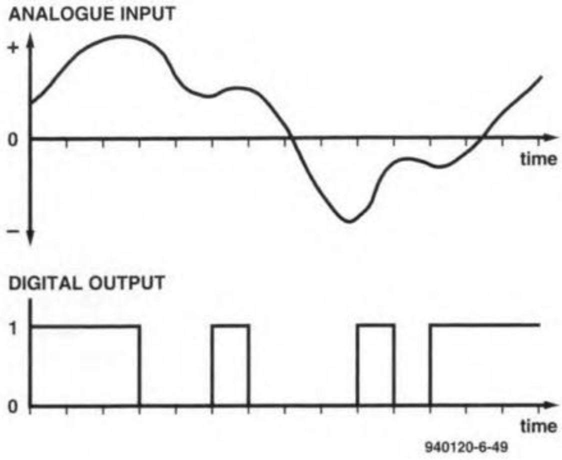


Fig. 49

to analogue processing make the double conversion worth while.

Various types of conversion circuit were described in Part 4, of which the flash converter was the fastest. In many applications, conversion speed is not critical, but in real-time processing of signals from CDs, for example, or in telecommunications, conversion can become a bottleneck. A flash converter may operate at 20 MHz or more, which gives enough time for a sample to be taken and processed before the next sample is due for conversion. Flash converters are ideal for real-time processing of many kinds of data, including data related to instrumentation and process control. But flash conversion depends upon having a resistor chain with n equal resistors, where n is the number of different digital values that the converter produces. For example, an 8-bit flash ADC produces 256 digital values, from 0 to 255, which means that 256 identical resistors are required. For accuracy, these have to be trimmed by laser during manufacture of the chip, which makes the device expensive. For 12-bit flash conversion, 4096 resistors are needed, so it is not surprising that such devices are rare, if not unobtainable. For audio reproduction of CD quality, we need 16-bit conversion, so an alternative to flash conversion must be sought.

Sigma-delta ADCs employ a different approach to conversion. Instead of converting each sample, such an ADC converts the *difference* between each sample and its predecessor. It does not attempt to *measure* the difference, but simply determines whether the difference is positive or negative. In **Fig. 49**, the output of the converter is high when the input signal is increasing and low when it is decreasing. We require only a 1-bit converter, which makes it very simple to construct, and very fast in action.

With a 1-bit conversion, it is essential that consecutive samples differ by relatively small amounts. This can be ensured by sampling at a high rate, so that the sample value has too little time to change appreciably. Applying the Nyquist theory, we normally sample data at twice the rate of the highest frequency that we wish to reproduce. The highest frequency that the human can detect is about 20 kHz, which is why the standard sampling rate for digital audio is 44.1 kHz (or 48 kHz in some systems). A sigma-delta ADC uses a sampling rate of 100 kHz or more, and this is referred to as *oversampling*. A block diagram of a sigma-delta ADC is shown in **Fig. 50**. The summation block subtracts the output of the converter from the incoming analogue signal. Thus, the subtraction is either one unit, if the output is 1, or nothing, if the output is 0. Taking a bit-by-bit view of the operation, this would not seem to have the desired effect. The action of the integrator is an es-

Errors in digital signals

The most elementary form of error checking is the *parity bit*. In *even parity*, an extra bit is added to a binary number or code to make the number of 1s in it even. For example, if the number is 0100100, there is an even number of 1s, so the parity bit is 0, and this is added (usually as the most significant bit) to give 00100100. But a number such as 0110100 has an odd number of 1s, so it needs another 1 to make parity even, and the result is 10110100. A digital circuit can easily check that there is an even number of 1s in a number and, if not, to signal an error.

In analogue systems, noise or glitches in the signal have an equal effect wherever they occur. Digital signals are more subject to position error. For example, given the number 01011010 (90 decimal), an error which causes the least significant bit to change from 0 to 1 causes the value to change from 90 to 91 decimal, which may not be important. But if the error occurs in the most significant digit, so that the number becomes 11011010, the value becomes 218 decimal, which is a serious error.

Parity bit checking detects an error in a single bit, but can not tell us which bit it is. If parity checking is the only method in use, the safe solution is for the data to be sent again. In processing a stream of data from a CD, for example, this method is unworkable, so more sophisticated techniques are employed. In a digital audio signal, which has a high sampling rate, it is unlikely that there will be large differences between two successive samples. If an error causes the difference between a sample and the one before it to exceed a given amount, the circuit rejects the erroneous sample and repeats the previous sample instead. Alternatively, the mean of the two samples before and after the erroneous sample may be substituted. In this way, the error is patched over and the listener hears nothing amiss.

There are several other correcting routines, some of them very complex indeed, to allow for errors in a sequence of samples caused by a serious disk or tape defect. These involve *interleaving*, in which the consecutive bits of a block of data are not recorded in their proper order in the block. Instead, they are 'scrambled' in a particular way, bits from one block being recorded with bits from another block. If a large defect occurs, it will affect a few bits from each of several blocks, and it is more likely that the content of each block can be reconstructed, so reducing or even eliminating the effect of the error. The reproducing equipment has complementary circuits to unscramble the interleaved data.

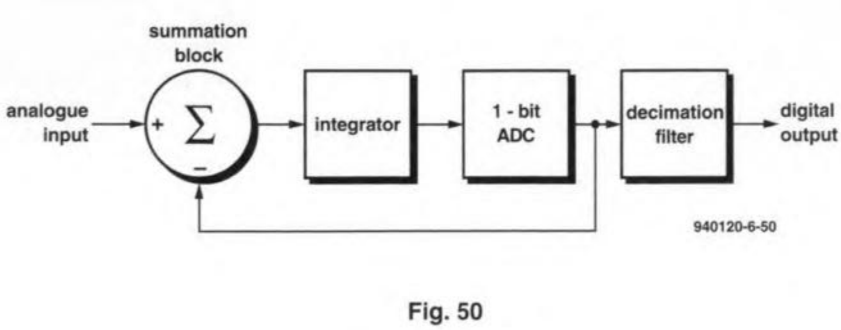


Fig. 50

sential part of the conversion. It is easier to understand what happens if we imagine the integrator to be in the feedback section of the loop. Then, if the input signal is increasing, a succession of 1s results in an increasing output from the integrator, to be subtracted from the increasing in-put voltage. At each sampling, the difference output of the summation block becomes a negative 'step' and the converter gives a zero output. The action of the integrator is thus to keep track of the most recent value of the input. Putting the integrator in a different part of the loop, as we have done in Fig. 50, makes it more difficult to explain how the loop works, but makes no difference to its operation. It has advantages for the construction of the converter chip.

Oversampling gives a converted signal that changes at a higher frequency than subsequent stages require. The digital decimation filter reduces the signal frequency to more acceptable levels. In effect, it averages out the digital values over a given number of bits. For example, the output of a ×5 oversampled signal is split into groups of five successive bits:

... 01001 10011 11011 00100 11001 ...

The decimator examines each group to determine which is the majority value:

... 0 1 1 0 1 ...

This is the decimated output at the normal sampling rate.

[940120-VI]

Answers to Test Yourself 5

1. 0101.
2. See **Fig. 47**.

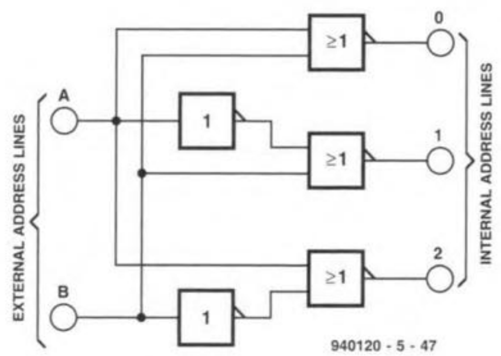
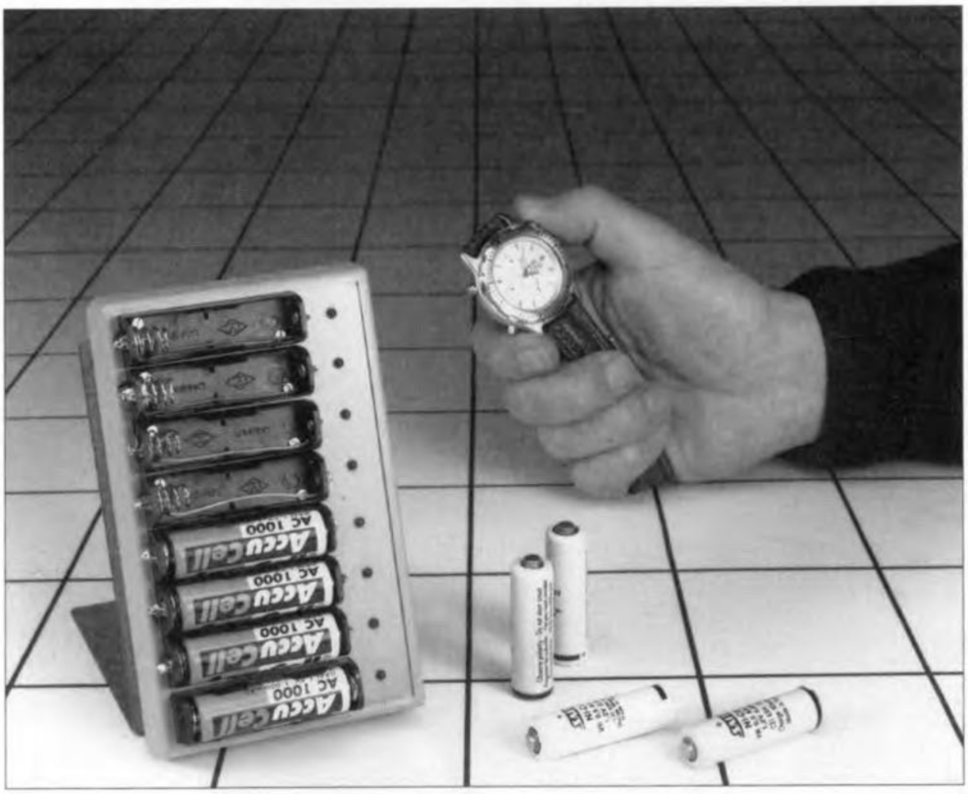


Fig. 47

NiCd BATTERY-QUALITY TESTER

The capacity of a fully charged NiCd (nickel-cadmium) battery can be measured fairly accurately by draining the battery with a fixed current until a certain cell voltage is reached. The product of current and time then yields the actual capacity. In the design presented here, the discharging and voltage measurement operations run under the control of a microprocessor. Up to eight cells may be tested in one go.



Design by A. Rietjens

THE actual capacity of a NiCd battery is hardly ever the value printed on the label. That is unfortunate and regrettable, but it is an actual fact. As cells become older and have a history of many charging and discharging cycles, it is normal for their capacity to drop gradually. Two other aspects also play a role. Firstly, there is, obviously, a limit to the life of NiCd cells. Depending on the type and the way it is used, a NiCd cell gives up the ghost after 500 to 1,000 charging/discharging cycles. It is then almost impossible to charge such cells, and their capacity seems to drop to almost nil. Secondly, there is the so-called 'memory' effect: cells which are always charged again after a relatively short time 'memorize' that their capacity not fully used, and show up a significant decrease in performance after a while. This can lead to embarrassing situations where you are caught out with a totally drained battery pack. For instance, you may suddenly discover (too late) that the batteries in your photo flasher, although dutifully charged overnight, have insufficient capacity for the 100-odd flashes you would hope to get. Instead, you get only 20 flashes before the 'battery low' symbol appears. That can be very awkward during a photo session, for instance, at a wedding.

Unfortunately, the reduction in battery capacity is difficult to measure exactly. It can not be measured by just looking at the cell voltage, unless, of course, the battery is defective. The only reliable way to gauge the battery capacity is to first charge it completely, and then discharge it (again, completely) at an accurately known current until it is 'flat'. The amount of current and the length of the discharging period enables the actual capacity to be computed. There is no other way.

Microcontroller

Another problem with testing NiCd batteries is that in nearly all applications a number of series connected cells are used. These are inevitably charged and discharged together. If a battery pack of, say, four cells suddenly fails, it could be that all four cells have reached the end of their life. It is also possible, however, that only one of them has broken down early. To prevent the whole pack being chucked away for recycling, it is important to be able to test each cell in a pack **individually**.

Unfortunately, that is a time consuming activity, particularly when the battery pack consists of six or even eight cells. Because 'power users' of NiCd cells are on the rise these days (just look at the number of portable computers, camcorders, cordless telephones, etc.), something has to be done to speed up capacity testing of such battery packs. A solution was found in the use of a microcontroller type ST6, for which a small program was written to suit the application. This program allows the hardware to be kept very simple, as illustrated by the circuit diagram in **Fig. 1**. Apart from the microcontroller (which is available ready-programmed through the Readers Services), the tester only contains eight load resistors, eight indicator LEDs, a buzzer and an 8-MHz quartz crystal for the internal oscillator. This handful of parts is sufficient to test up to eight NiCd cells in one go, but still independently.

So how does the test protocol work? The tester has eight inputs to which a NiCd cell may be connected. Each cell is loaded by a resistor with a value of 1.2 Ω (R_1 through R_8), which causes a (nominal) current of 1 A to flow. The

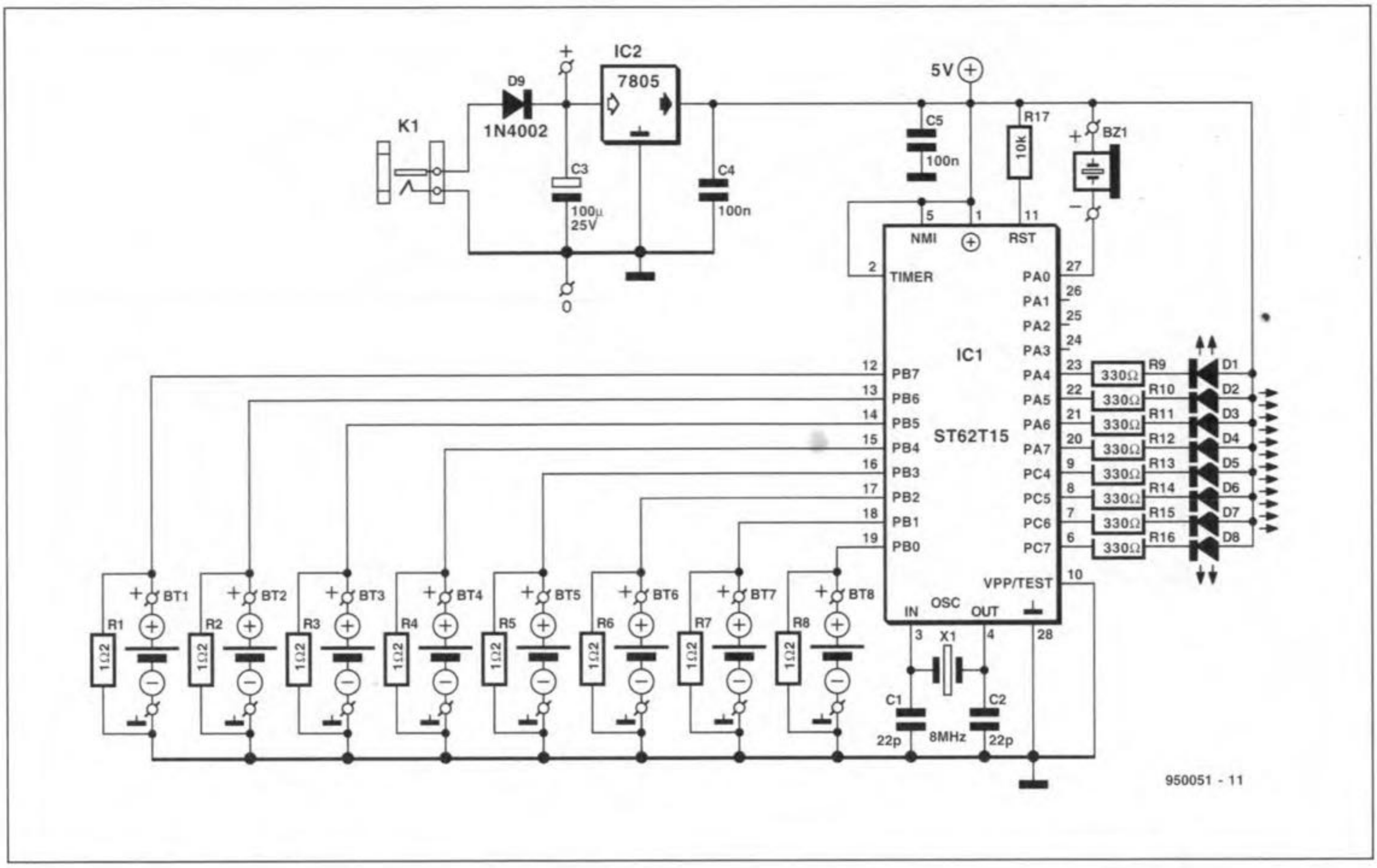


Fig. 1. Circuit diagram of the NiCd battery-quality tester. The hardware remains limited to a microcontroller, eight load resistors and an equal number of LEDs. The buzzer sounds whenever a cell is completely discharged.

cells to be tested are fully charged before this test. Discharging starts immediately when a cell is fitted into the holder. During discharging, the cell voltage is continuously monitored. As soon as the terminal voltage of any of the cells drops below 0.9 V, the buzzer is briefly actuated to alert the user to the fact that one of the cells is empty. At the same time, the LED with the holder contact lights, which pinpoints the empty cell. LED D_1 belongs with cell BT_1, LED D_2 with BT_2, and so on. Finally, the software arranges for contacts with no battery installed to be marked by a flashing LED.

Procedure

In addition to the present tester, you need a stopwatch (or an ordinary clock), and pen and paper. The tester is switched on; all LEDs will start to flash. Next, the (charged) batteries are installed in the holder. The LEDs which belong with the holder contacts keep flashing until the battery makes proper contact with the spring terminals. Note the time, or start the stopwatch.

When the buzzer sounds, check which LED lights, and make a note of the cell number and the time. The relevant cell is immediately removed from the holder to prevent deep discharging and possible damage. At each buzzer signal that follows after some time, remove the relevant cell and note the time.

Once all cells are discharged, the times noted are multiplied by the (known) discharging current (here, 1 A). This yields the current cell capacity. So, if a cell is discharged in, for instance, 30 minutes, it has a capacity of 500 mAh.

Two final remarks. If the test indicates that the battery pack has a much lower capacitance than stated by the manufacturer, although it is still relatively new, there is no need to draw negative conclusions as yet. The reduced capacity may be indicative of the previously mentioned memory effect. In many cases, cells may be 'refreshed' by doing a few complete charging and discharging cycles.

The tester is also suitable for checking the actual performance of 'no-name' cells which are offered at low prices. Other than with those of well-known manufacturers, these inexpensive cells may have widely different capacities. Sometimes these cells are excellent value for money, but there are also types whose actual capacity falls way short of the claimed or even printed value.

If you want to test cells with a capacity other than 500 mAh, it is recommended to change the load resistors such that the discharging current is $2C$ at the most, where C is the nominal capacity printed on the battery.

Construction

The construction of the battery tester is made easy by the availability (through our Readers Services) of a ready-made printed circuit board. The copper layout and component overlay of this board are given in **Fig. 2**.

Actually building up the tester is a matter of routine, and should take you no more than an hour or so. The microcontroller is preferably mounted in an IC socket. Resistors R_1 through R_8 run fairly hot during discharging, and should be mounted a few millimetres above the board surface. The finished printed circuit board used to build our prototype of the tester is shown in **Fig. 3**.

The board has eight screw-type PCB terminal blocks to which the contacts of the battery holder are connected. For easy access to the cells, the battery

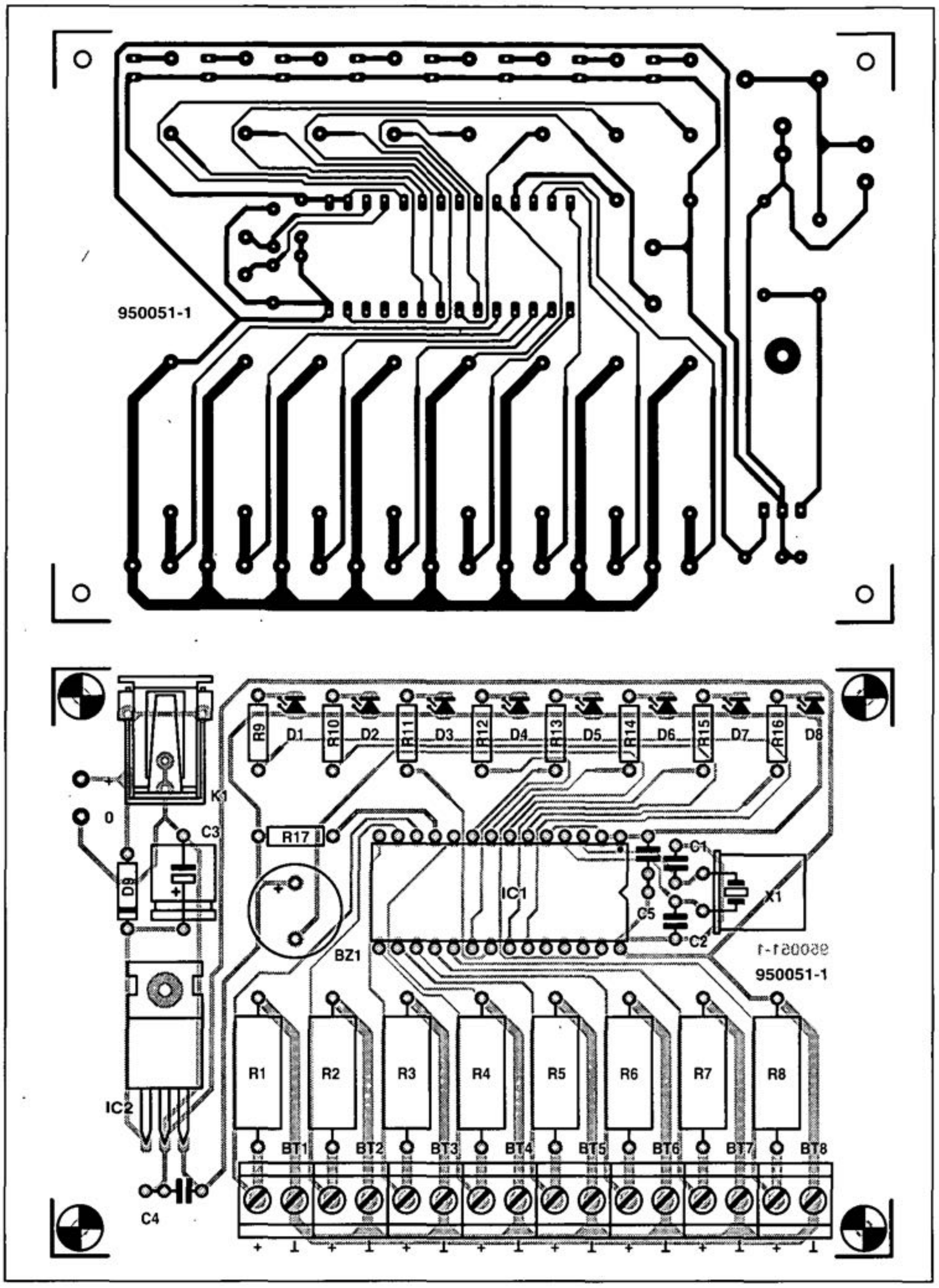


Fig. 2. Printed circuit board designed for the tester (board available ready-made, see page 70).

COMPONENTS LIST

Resistors:
R_1-R_8 = 1Ω2 5 watt
R_9-R_{16} = 330Ω
R_{17} = 10kΩ

Capacitors:
C_1,C_2 = 22pF
C_3 = 100µF 25V
C_4,C_5 = 100nF

Semiconductors:
D_1-D_8 = LED 3 mm
D_9 = 1N4002
IC_1 = ST62T15 (order code 956506-1)
IC_2 = 7805

Miscellaneous:
K_1 = adaptor socket, PCB mount.
Bz_1 = 5-V buzzer.
Bt_1-Bt_8 = PCB terminal block + battery holder.
X_1 = 8 MHz crystal.
Printed circuit board and programmed ST62T15 controller: available as a set, order code 950051-C (see page 70).

holder is best fitted on the case cover. Because the LEDs should also be easily visible during testing, they are initially mounted in sockets (for which a length of single-row pin socket strip was cut into small pieces). Once the circuit is functioning properly, the LEDs are pulled from their sockets and fitted into the cover. A trunk of flexible wire then provides the link with the board.

Figure 4 gives an indication of the construction of the prototype. The battery holders are of the 'penlight' type, because that is the most frequently used type of NiCd cell. The case used was supplied by Pactec, and has a size of about 9×14.5×3 cm. Obviously, every constructor is free to choose a fitting enclosure for the tester.

The tester is best powered by an ordinary mains adaptor capable of supplying a direct voltage between 9 V and 12 V at a current of about 100 mA. This will hardly ever pose problems. Voltage regulation is not necessary, because that function is built into the tester (IC_2). Diode D_9 protects the tester if the adaptor voltage is connected the wrong way around.

(950051)

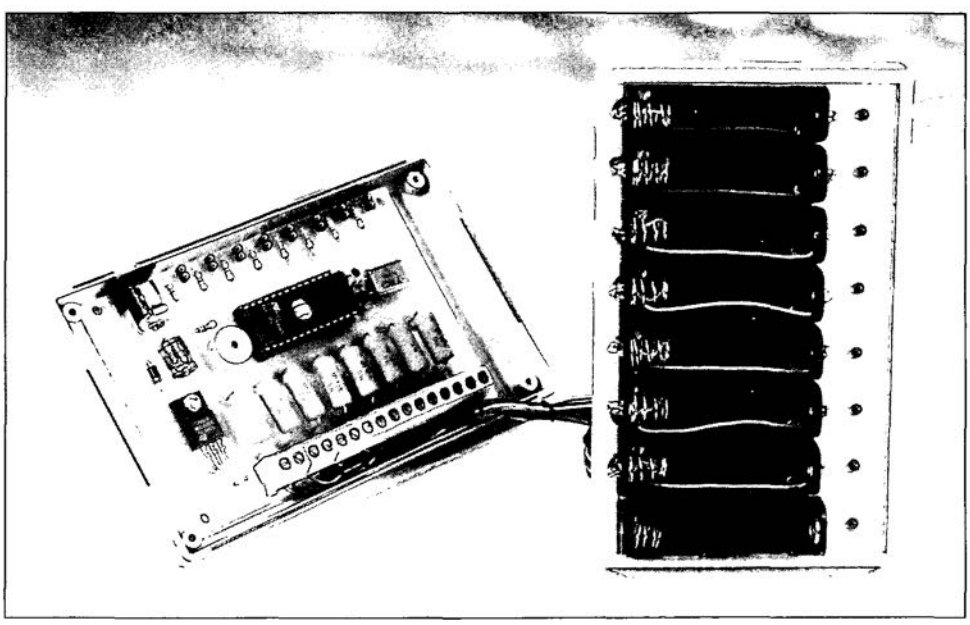


Fig. 3. The battery holders and LED indicators are obviously fitted at the outside of the case. They are connected to the circuit board via two trunks of flexible wire.

ASTRA DIGITAL RADIO (ADR) - 1

Radio stations so far have been using analogue frequency modulation (FM) to broadcast via subcarriers of TV programmes beamed down by the Astra satellites. A new system, called ADR (Astra Digital Radio) has seen rapid acceptance both with the receiver industry and large TV broadcasters. The all-digital system enables twice as many sound programmes to be transmitted than the old analogue system. Instead of two subcarriers for a stereo programme, ADR only requires one for all the relevant sound information, plus additional services. This allows a single Astra transponder to carry one TV programme and no fewer than 12 ADR channels. DMX (Digital Music Express) is the first pay radio station to make use of ADR.

By G. Kleine

THE future of digital radio has begun on the Astra satellites. Unnoticed by large audiences, broadcasters such as the German ARD are actually using ADR to feed programmes to local FM transmitters for re-conversion into analogue stereo FM. This is done to cut on the cost of long cable and microwave links throughout the country. But obviously a large number of people in possession of a satellite TV dish will also be able to receive these ADR programmes directly from the Astra satellite, and so reap the benefits of stereo radio with digital quality. As such, ADR is a serious competitor of the Panda-Wegener analogue compression/expansion system.

In contrast with the DSR (digital satellite radio) system employed on Germany's (now defunct) TV-SAT2 high-power DBS TV satellite, ADR does get acclaim from broadcasters and receiver manufacturers alike. The first ADR home receivers are announced for the Cebit Fair at prices between £200 and £350.

The key feature of ADR is a data reduction system called **Musicam**, which allows the audio data rate of a stereo signal to be reduced to 192 kbit/s. The Musicam system is also used for the DCC (digital compact cassette) from Philips/Matsushita, and the MiniDisc (MD) from Sony. As far as the sound quality is concerned, ADR is comparable to these systems. Even if ADR just falls short of the sound quality offered by systems without data reduction (CD, DAT and DSR), it still guarantees a vast improvement over VHF FM stereo and analogue satellite radio. The designation 'digital' is therefore not a hoax.

The first instalment of this article aims at giving an overview of the principles underlying the ADR system, and introduces the basics of the transmission method and data encoding used. In addition to an ADR programme overview, a brief description is presented of the technology used in the first generation of ADR receivers. The second instalment in next month's issue will go into details of the channel encoding and the signal processing at the transmitter and the receiver side.

The baseband signal

As illustrated in **Fig. 1a**, a regular satellite TV baseband signal contains a composite video signal and the accompanying (mono) sound channel, which is frequency modulated on a subcarrier at 6.5 MHz. Starting at 7.2 MHz, the baseband also contains narrow-band FM subcarriers which are used to convey additional sound broadcasts (which may or may not be related to the TV programme). These subcarriers are located in a raster of 180 kHz.

Ever since the introduction of stereo TV sound, the first two subcarriers, at 7.02 MHz and 7.2 MHz, have been popular for this application. Alternatively, they are often used as simultaneous comment channels for different languages. Whereas the main (mono) sound subcarrier at 6.5 MHz is modulated at a deviation of ±140 kHz, the narrow-band carriers have a deviation of only ±50 kHz. To improve the signal-to-noise ratio in these channels, Wegener's Panda-1 compander system is used.

The baseband of an Astra transponder with full ADR occupancy is shown in **Fig. 1b**. The mono subcarrier is no longer used, and leaves room for five ADR subcarriers. The sound with the TV

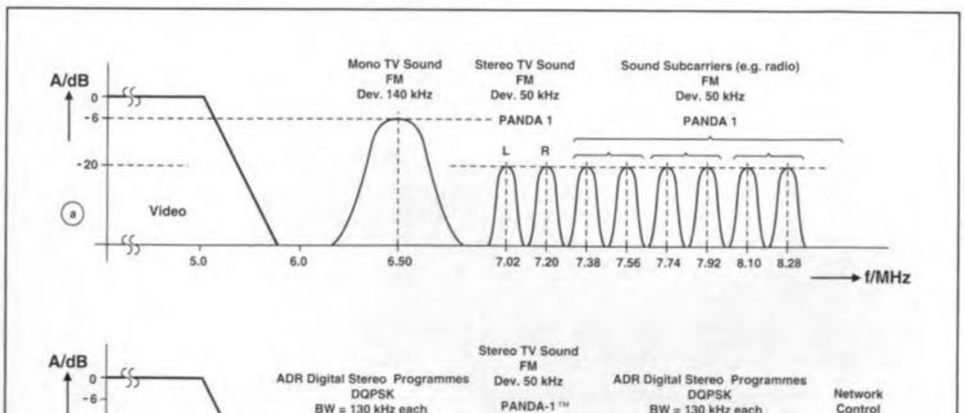


Fig. 1. Baseband signal: (a) with analogue sound subcarriers; (b) with ADR subcarriers.

programme is exclusively transmitted via the 7.02/7.20 MHz subcarrier pair, which retains narrow-band analogue FM, using Panda-1 companding. Above the analogue subcarriers, there is room for five more ADR programmes. This means that each transponder is capable of accommodating 12 ADR programmes. If the TV signal and the analogue FM subcarriers are omitted altogether, a transponder even holds 48 ADR programmes. The subcarrier at 8.595 MHz at the top of the baseband conveys network information, and need not be processed by ADR home receivers. It may be used for data exchange between broadcasters, or for control purposes.

The modulation standard for ADR is quadrature phase shift keying (QPSK). Including a channel encoding system used to prevent bit errors caused by noise in the downlink/uplink path, the ADR signal has a gross data rate of 256 kbit/s. The QPSK signal causes a subcarrier bandwidth of about 130 kHz. The network control signal is modulated in FSK rather than the more complex QPSK.

Figures 1a and 1b also indicate the level ratios between the various types of subcarrier with respect to the video signal.

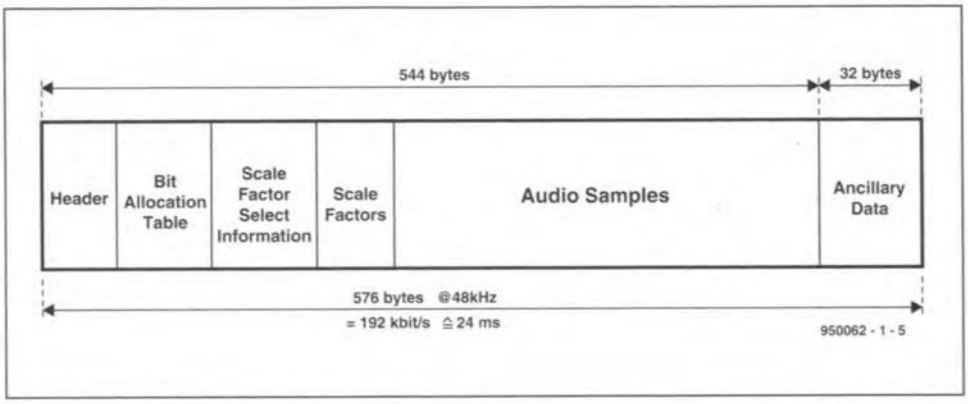


Fig. 2. Contents of the ADR date frame.

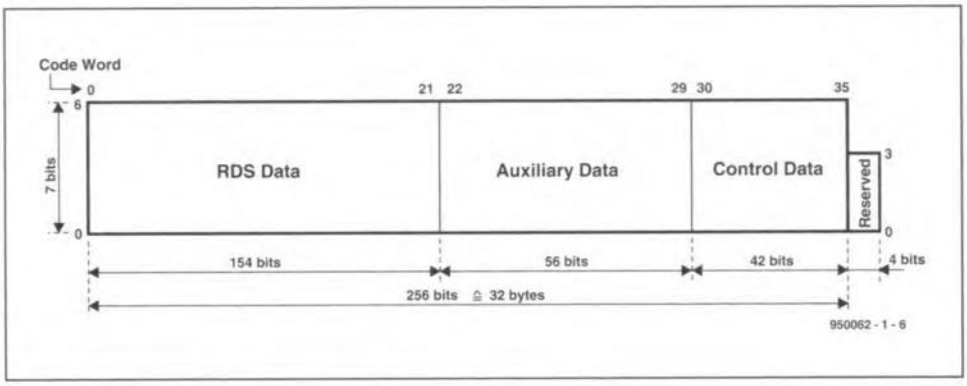


Fig. 3. Structure of the ADR ancillary data block.

The ADR data frame

The actual data frame of the ADR signal has a structure based on ISO/IEC standard 11172-3, which is better known as MPEG-1 Audio, Layer 2. **Figure 2** shows the distribution of the 576 bytes in the frame. The header serves to synchronise the frame. It is followed by the bit allocation table and scale factor select information, which function in a dynamic audio bit allocation system. Then follow the scale factors proper. The largest part of such a data packet is taken by the audio data (samples). Only the last 32 bytes serve to convey ancillary data. The audio sampling frequency used for ADR is 48 kHz.

Inside the ancillary data part (**Fig. 3**), the first block is reserved for **RDS** (radio data system) data. This block allows broadcasters who supply RDS data to their listeners to convey this type of information to local VHF-FM repeaters, via the Astra satellite. The ADR receiver is capable of processing RDS data, and showing them on a display.

The **auxiliary data** block is reserved for internal use by broadcasters, for instance, control information for VHF-FM repeaters.

The border between RDS and auxiliary data is variable, and its exact position is conveyed in the **control data** block. This contains information on the encrypted/non-encrypted status of the relevant programme, in other words, whether it is transmitted in 'free radio mode' or in 'pay radio mode' using DMX encryption. Here, too, the station identification and the programme type (for example, classical music) of the relevant ADR channel are transmitted in hexadecimal code, as usual with RDS. To this is added the ADR-specific programme type according to **Table 1**. This data is processed by the ADR receiver, and may be used to drive a display, or set up search criteria to help the listener find his or her favourite type of music.

The ancillary data are transmitted in the form of 7-bit words. Because channel interleave encoding does not offer sufficient protection for the ancillary data, a further (channel) encoding system is applied. Details of this system, and the measures taken to guarantee the data integrity in the channel, are discussed in next month's final instalment.

Data reduction

The starting point is a full-width, non-compressed, digital audio signal. In accordance with the ADR specifications (Ref. 1), a sampling rate of 48 kHz is used (as with home recordings on DAT tapes). If a different sampling rate is used, for instance, 44.1 kHz with CD, or 32 kHz with DSR) an up-conversion to a sampling rate of 48 kHz is required. In case an analogue stereo signal is available, this is digitized at a sampling rate of 48 kHz. Then follows the data reduction to the Musicam standard.

The **Musicam data reduction method** is based on omitting 'hidden' sound elements which we humans are not capable of perceiving. The frequency range of 20 Hz to 20 kHz is first divided into 32 sub-bands. Within each band, the maximum sound level contained within is measured with the aid of digital signal processing. The Musicam encoder contains a reference set of values representing the hearing thresholds for single frequencies as well as those for co-hearing, i.e., soft sounds obliterated by 'nearby' loud sounds. This reference set

1 - Classical Music
2 - Popular Music
3 - Evergreens
4 - Rock Music
5 - Jazz Music
6 - Country Music
7 - Special Interest Music
8 - Regional
9 - News/Events
0 - General Entertainment

Table 1. ADR programme types according to Ref. 1.

enables the encoder to check if the levels encountered in the different sub-bands are below or above the current hearing threshold. Only those components above the threshold need to be transmitted. The curves in **Fig. 4** give a rough indication of the absolute hearing threshold and the co-hearing threshold for a mixture of three sine waves.

The audio data leaves the Musicam encoder in a format which includes a scale factor. This yields a further data reduction, since leading zeroes need not be transmitted. The total achievable data reduction factor is about 10.

The ADR home receiver

Before the ADR processor will be integrated in satellite TV receivers, stand-alone ADR receivers will be available on the market. To enable the ADR receiver to control the satellite TV receive equipment, it is inserted in the cable between the outdoor unit (LNB) and the indoor unit (satellite TV receiver), as illustrated in **Fig. 5**. In this set-up, the ADR receiver controls the LNB polarization selection (14V/18V) so that the satellite TV receiver can not interfere. When the ADR receiver is switched off, the satellite TV receiver takes control of the polarisation selection again.

An interesting feature is the so-called **scan-and-search mode**. The ADR receiver then scans all transponders and subcarrier frequencies for QPSK modulated ADR programmes, and stores these in a programme memory, together with the relevant program type (**Table 1**). This relieves the user of an awesome programming session, and helps to make sure that all available ADR programs become accessible. The receiver is capable of doing a scan-and-search on its own to ensure that its programme memory is always up to date. A manual command to the same effect is, however, also possible.

Technisat[1] will be among the first manufacturers to release a stand-alone ADR/DMX receiver this year. The cost of this receiver is said to remain below £325.

Philips Semiconductors are working on an integrated ADR receiver in a single very high-integration chip. At the input of this chip, the complete baseband is digitized. The subcarrier detection/filtering and QPSK demodulation operations are based on digital signal processing, which is relatively easy within a spectrum of 'only' 10 MHz considering the speed of today's DSP cores. However, the Philips single-chip solution is not expected to be available before the second generation of ADR receivers.

DMX — Digital Music Express

For the first time in Europe, there will be a satellite pay-radio station which uses

Programme	Subcarrier	Transponder	TV Programme	Frequency / Pol.
HR	7.74 MHz	19	ARD	11.494 GHz H
HR	7.92 MHz			
HR	8.10 MHz			
HR	8.28 MHz			
HR	8.46 MHz			
NDR	7.74 MHz	25	Nord 3	11.582 GHz H
NDR	7.92 MHz			
NDR	8.10 MHz			
NDR	8.28 MHz			
NDR	8.46 MHz			
WDR	6.12 MHz	39	WDR 3	11.053 GHz H
WDR 2	6.30 MHz			
WDR	6.48 MHz			
WDR	6.66 MHz			
WDR	6.84 MHz			
MDR life	6.30 MHz	43	MDR 3	11.112 GHz H
MDR info	6.48 MHz			
MDR culture	6.66 MHz			
MDR sputnik	6.84 MHz			
BR 1	6.12 MHz	45	Bayern 3	11.141 GHz H
BR 2	6.30 MHz			
BR 3	6.48 MHz			
BR 5	6.66 MHz			
BR 4	6.84 MHz			
SWF	6.12 MHz	48	Südwest 3	11.186 GHz V
SWF 3	6.84 MHz			
SES Test 1	8.10 MHz	33	ZDF	10.964 GHz H
SES Test 2	8.28 MHz			
SES Test 3	8.46 MHz			
DMX	7.56 MHz	41	Discovery Ch.	11.082 GHz H
DMX	7.74 MHz			
DMX	7.92 MHz			
DMX	8.10 MHz			
DMX	8.28 MHz			
DMX	8.46 MHz			
DMX	7.74 MHz	42	Bravo/Adult Ch.	11.097 GHz V
DMX	7.92 MHz			
DMX	8.10 MHz			
DMX	8.28 MHz			
DMX	8.46 MHz			
DMX	8.10 MHz	12	Sky News	11.377 GHz V

Table 2. Where to find ADR programmes on the Astra satellites.

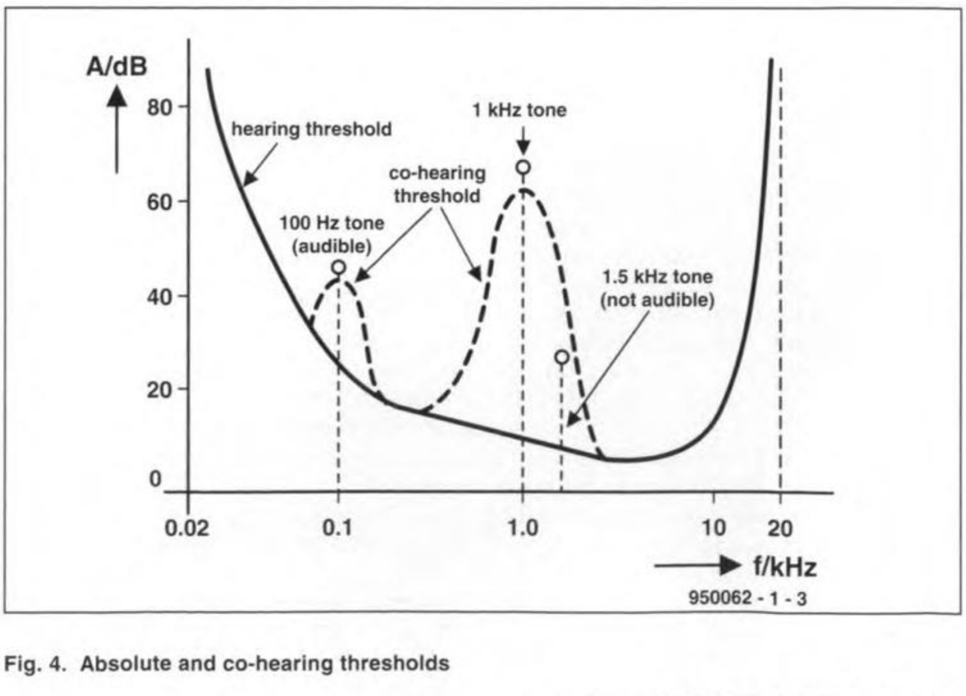


Fig. 4. Absolute and co-hearing thresholds

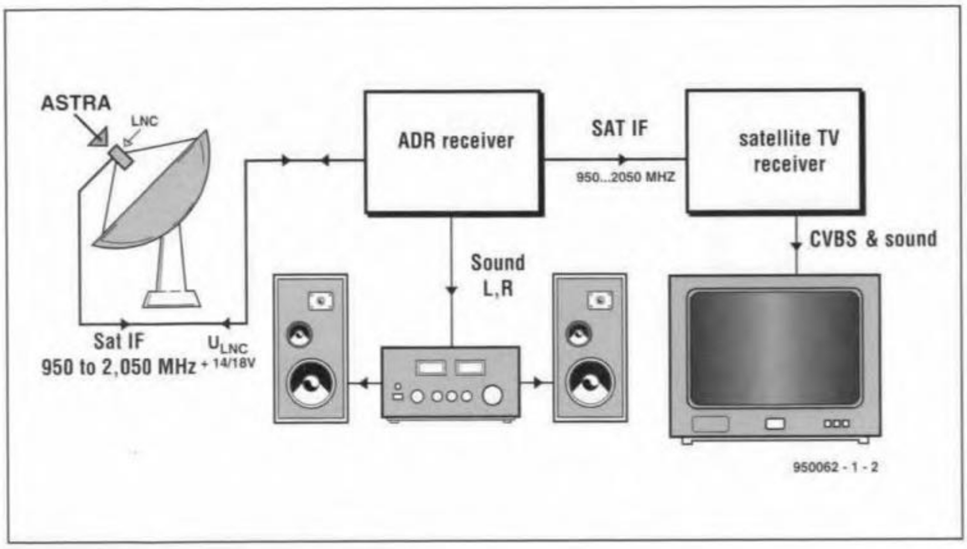


Fig. 5. Inserting the ADR receiver in an existing satellite TV receiver system.

ADR. As with pay-TV (for instance, some of the Sky channels), you, the listener, will have to get hold of a so-called **listening card** for the reception of a program bundle consisting of, initially, 30 channels. The card is inserted into a reader in the ADR/DMX receiver, and enables the DMX programmes to be decrypted. The reception of the 30 channels (see Table 3) with non-stop music (no announcements, no commercials) which covers a wide variety of musical preferences, is rumoured to cost you about £7 per month.

Since October 1994, a DMX acceptance test has been run in a number of villages in South Germany. For the test, receivers from Scientific Atlanta were installed in about 100 homes. Programmes were received from the Intelsat-601 TV satellite at 27.5° West, via a head-end station and then distributed by a privately owned cable network. At the time of writing this article (February 1995), the first DMX subcarriers have appeared on the Astra satellites also. If the acceptance test is successful, more music programmes for the European market will be added. A total of up to 90 ADR channels is expected to be operational in the near future.

Following the introduction of the DMX system in the U.S.A. by Scientific Atlanta in 1991, the number of subscribers has grown to 150,000. The DMX signals are fed into over 800 cable networks in the U.S.A. In the UK, there are currently about 30,000 subscribers in different cities. The DMX programs are also distributed via cable networks in Prague, Denmark, Norway and Ireland. Scientific Atlanta Inc. supplies all the equipment for a cable head-end station, as well as stand-alone DMX receivers.

Further outlook

In stark contrast with the limited success of the German-market oriented DSR system (via the TVSAT-2), ADR is bound to attract a much larger audience right from the start. SES, the owner/operator of the Astra satellite cluster, follows market trends. In the past, Telekom (the TVSAT-2 operator) attempted to set a trend, and failed with the 'forced' introduction of DSR. Because a number of large German broadcasters like ARD and BR are already endorsing the ADR system by running copies of their terrestrial programmes via the Astra satellite, it is reasonable to expect that others will follow suit.

Whether or not DMX will take off is difficult to tell as this service is relatively new. The success, obviously, depends mainly on how listeners will feel about the cost of the listening card.

(950062-1)

Reference:

1. ADR specification, October 1994, SES-Astra, Betzdorf, Luxembourg.

Address reference:

1. Technisat, Postfach 560, D-54541 Daun, Germany.

DIGITAL MUSIC EXPRESS TM

1. Classical
- 1.1 Symphonies
- 1.2 Chamber Music
- 1.3 Operas

2. Hits
- 2.1 English Hits
- 2.2 Dance Music
- 2.3 European Hottest Hits
- 2.4 Rhythm and Blues
- 2.5. US Hottest Hits

3. Rock
- 3.1 Alternative Rock
- 3.2 Heavy Metal
- 3.3. Classic Rock
- 3.4 Album Rock
- 3.5 Folk Rock
- 3.6 Modern Jazz
- 3.7 Classic Jazz
- 3.8 Big Band Music/Swing

4. Light Music
- 4.1 Adult Contemporary
- 4.2 Love Songs
- 4.3 Rhythm and Blues
- 4.4 Country Music

5. Standards
- 5.1 Big Singers
- 5.2 Evergreens

6. Instrumental
- 6.1 Beautiful Instruments

7. World Music
- 7.1 Beat Music
- 7.2 Reggae

8. Music from Europe
- 8.1 French Chansons
- 8.2 Spanish Folk Music
- 8.3 Italo Hits
- 8.4 Dutch Folk Music
- 8.5 German Folk Music and Evergreens

Table 3. European DMX programme allocation.

Readers' Corner

Dear Editor—My husband, a registered computaholic, reads each new edition of your publication without fail. My new motto is "if you can't beat 'em, join 'em", so I too peruse your magazine. I'm sure there must be many thousands more computer wives out there who would welcome a page of their own in your publication. I attach a starter for 10.

Obsession

My husband is obsessed. I keep threatening to have a rubber stamp made so that I can stamp the word 'obsessed' across his forehead. Even on our honeymoon he was pining. Not for the traditional holidaymaker yearnings like home cooking, fish and chips, a good cup of tea or a luxuriously comfortable (by comparison) bed. No, no, he was pining for his computer. We went away on our romantic, once-in-lifetime holiday, armed to the teeth with computer and electronics magazines, and as many other related books as would fit in his suitcase, together with a few incidentals like clothes and suntan lotion. Even at the airport he spent most of the waiting time in W H Smith's checking out the magazine racks in case he had missed an important publication that had only just appeared in the shops that morning. Three quarters of the way into the holiday, he was in a bad way, suffering from Computer Withdrawal Syndrome (a condition that has reached epidemic proportions) and starting to talk about the things he would do when he got home. I thought he was joking.

We arrived home and he just managed to stagger across the threshold before unceremoniously dumping me in the hallway and tearing up the stairs to check that his beloved was still there. I followed him, more to check for myself than take his word for it. And it was true – he was there hugging his computer. If I hadn't seen it, I wouldn't have believed it. That night, whilst I was sitting on our bed surrounded by wedding presents and a gradually mounting pile of discarded wrapping paper, and judiciously listing the names and gifts of our friends, my husband did not join in this glorious amalgamation of at least ten Christmases all rolled into one. Oh no. He was too busy working on his most recent electronics project and whispering sweet nothings into the ear of his computer. But did I care? Good Lord, no. I was having a whale of a time!

Nine months on and the baby is due for delivery any day – before you get the wrong idea, I mean HIS baby. And I've now got used to coming home from work to find my hubby, already ensconced in his study, working on his project. The routine we have imperceptibly fallen into is that I cook dinner, hubby descends from on high to hurriedly eat it and give me a highspeed rundown of his day, quickly washes up (his husband duty) and then rushes back. Time is very precious. He has been known to take a day off work to have the untold luxury of spending the entire day (and night) working on his project. He has managed to wean himself off his original seven nights a week and now finds the time to do some minor socializing. He also used to work late into the night, but has curbed that habit in recent months as he found he couldn't then get up for work.

Sunday mornings you can find us, without fail, at Maplin's where hubby stocks up on his components. Sometimes I go with him for a mooch. You would be surprised what you can buy there. (I've bought myself some small sealable plastic bags for the homemade earrings I make and sell, a wallet for my yuppy mobile phone and those small, round adhesive felt furniture protectors). On the occasions when I don't want to hang around whilst he makes his choice (even though he takes the list he made earlier, it's usually decisions, decisions), I go to haunt the shelves of Waterstone's, my traditional Sunday place of worship.

I have nothing but admiration for my husband and his dedication, determination and belief in the potential success of his project. All we need now is to win the lottery so that he can indulge his obsession full time. In the mean time, I shall buy him an inspirational gift – Calvin Klein's *Obsession*.
Nichola Davies, Lancashire.

A most welcome letter – food for thought for some; inspiration for others? I have never understood why there appear to be so few women engaged professionally in electronics. To me, electronics as a career is ideally suited to them. That is why this magazine has always tried to give coverage of women in electronics whenever possible, but the input has invariably been very small. However, here is some information (which arrived just too late for inclusion in our April issue) on a few women who are successfully engaged in the field of electronics.

[Editor]

Young Woman Engineer of the Year

Hayley Gladstone,28, senior project manager with GPT Strategic Communications Systems in Coventry, was awarded the coveted title of **1994 Young Woman Engineer of the Year**. The award was presented to Hayley by the Rt Hon Gillian Shephard, MP, last January at a special ceremony in London.

The Award, jointly sponsored by The Institution of Electronics and Electrical Incorporated Engineers (IEEIE) and the Caroline Haslett Memorial Trust (CHMT), was inaugurated in 1978 to encourage more young women to pursue a career in electronic and electrical engineering leading to Incorporated Engineer level. In presenting the awards, Mrs Shephard said: "Today's finalists are an excellent testimony to the quality of education and training available for Incorporated Engineers. I have been most impressed by the quality and commitment of the finalists in this year's competition".

Hayley Gladstone was responsible for managing the modernization of communications of the Beijing Metro and the management of the telecommunications element of the London, Tilbury, Southend railway signalling project. Last month, she and her husband, who also works for GPT, moved to Hong Kong for 18 months where she will manage the communications contract for the Hong Kong Mass Transit Corporation extension to the new airport.

Emma Croucher, 26, from Corsham, Wiltshire, was the runner-up for the award. She is an Outage Planner with the National Grid Company, Bristol, and in 1991 was the winner of the Mary George Memorial Prize. With responsibilities including planning system access (outages) and manpower resources involving nearly 400 staff for work on the grid system within the entire South West Area of NGC, Emma is also required to be available 24 hours a day when on standby. Her ability and commitment in a predominantly male area of engineering has won Emma the respect of her colleagues.

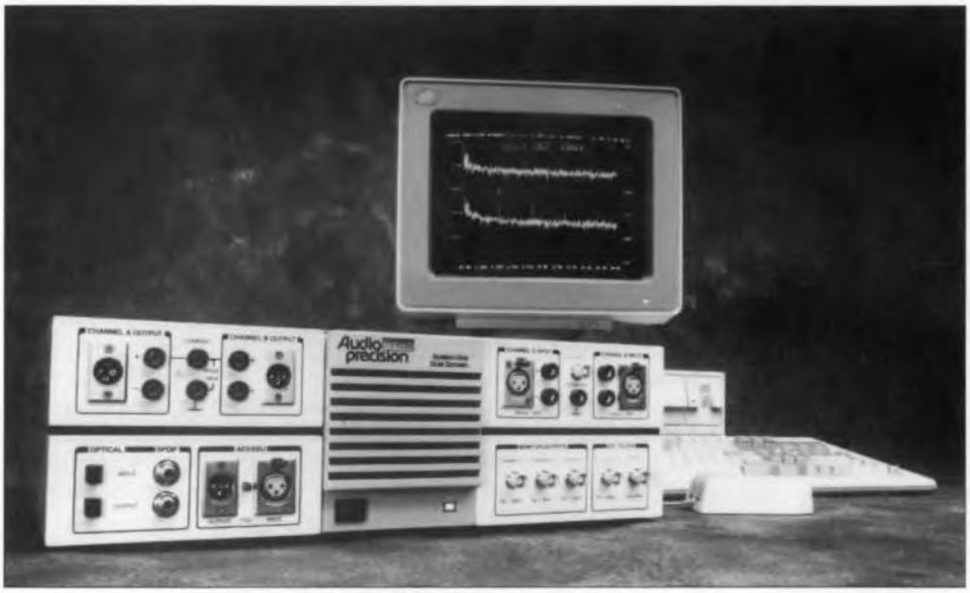
Third-prize winner, Jacqui Baddeley, 29, is a Project Manager and Senior Design Engineer at the the Government Communications Headquarters (GCHQ) in Cheltenham. She is responsible for the project management of 'next generation' secure data communications equipment. She also gives formal lectures in support of GCHQ training commitments. In her spare time, Jacqui is, among other activities, the South West England Regional co-ordinator for the Duke of Edinburgh Award Scheme.

The Mary George Memorial Prize, an additional award given to a young entrant showing particular promise as an Incorporated Engineer, was presented to Ann-Marie Wilkinson, 22. Ann-Marie is an Assistant Scientific Officer with the Defence Research Agency, Malvern, Worcestershire. Her duties range from involvement in analogue and digital circuit design to BS5750 implementation.

Companies who would like to receive information to enable them to nominate their suitably qualified staff for the 1995 Award should contact The Secretary, IEEIE, Savoy Hill House, Savoy Hill, London WC2R 0BS.

FOCUS ON: PC CONTROLLED AUDIO MEASUREMENT SYSTEMS

Today's personal computer, by virtue of its relatively low price and respectable computing power, is a perfect basis for a measurement system. This article looks at the possibilities and the operation of state-of-the-art computer controlled audio technology measurement systems. It also presents the main features of a number of such systems currently on the market. Prices of these systems differ widely from about £40 for hobby applications to £10k and more for high-end systems aimed at professional users.



By our editorial staff

THE rapid evolution of the personal computer over the last, say, ten years has left its marks by the fact that nearly every one these days seems to have or use a computer, be it for personal or business use. The biggest contribution to this phenomenon goes on account of the IBM PC and its compatibles. Today, an investment of less than £1,000 brings you a PC with computing power and storage capacity which was thought impossible or at least 'futuristic' only a few years ago. Coupled with its graphics capacities, that makes the PC eminently suited to more tasks than run-of-the-mill word processing or twiddling with spreadsheets.

Developers of measurement equipment have been long aware of this potential, and a variety of PC-controlled measurement equipment and plug-in cards is currently available. Incidentally, many 'ordinary' measurement systems also contain a full-blown microprocessor core, often with a keyboard to control the relevant measurement equipment, and a display to visualize data. Leafing through the catalogues of companies like Hewlett Packard and Tektronics you will find beautiful high-end test equipment with a plethora of features and control options, all by virtue of built-in computing power. However, this type of equipment is not covered by the present article, which is restricted to audio measurement systems than can be linked to a PC, either via a cable or an insertion card. The bulk of this type of equipment is intended for measurements in the audio field. Consequently, cards functioning as a voltmeter and/or oscilloscope are not covered here. Summarizing, we concentrate on instruments for extensive measurements on audio equipment, including frequency response and distortion measurements.

Classification

From a point of view of their construction, audio measurement systems may be divided into two classes:

- **external measurement systems**, where the function of the PC is restricted to doing the settings on the instrument, and visualizing (as well as, possibly, processing) the measured data;
- **internal measurement systems** which are plugged into the PC and share part of the PC hardware (for instance, an insertion card which copies measured data directly to the PC's memory).

Initially, the differences between these two categories will appear to be marginal. It is even possible for certain measurement systems to come as an internal or an external version. However, the advantage of an external system is that it is easy to relocate to another computer system.

The main difference between an internal and an external PC controlled measurement system lies in the accuracy that can be achieved. Although a high-resolution ADC may be fitted on an insertion card, it will be hard to achieve the specified accuracy in the face of the high noise level that exists inside a PC. This noise is caused by the switch-mode power supply, as well as by switching and data signals with fast edges. The lot severely curtails measurements of small signals. This is where the external measurement system has the edge because it allows sensitive parts like an ADC to be better screened. Only a handful of specialized manufacturers are capable of producing insertion cards which afford enough screening and supply decoupling to ensure high measurement accuracy. Therefore, do not expect too much from an audio measurement system in the form of an insertion card — though it may have a 16-bit A-D converter, this will probably achieve an accuracy of 12 or 13 bits at the most. If you are after really accurate audio measurements, for instance, distortion levels produced by hi-fi amplifiers, there is no alternative but to use an external measurement system. For measurements with 'relaxed' requirements, for instance, sound pressure level (SPL) or frequency response measurements on loudspeaker systems, a plug-in card is good enough, and often much cheaper than an external system.

Evidently, you have to take into account that the performance of any plug-in card depends largely on its internal design. By the same token, do not expect miracles from a measurement system just because it is external. In both cases, the quality strongly depends on the price.

Types of audio measurement systems

There is such a bewildering variety of measurements that can be performed on audio equipment that it is impossible to list them all in this article. Only look at a company like Bruel & Kjaer, which supplies dozens of different instruments for audio measurements only. Fortunately, the diversity is much smaller if we focus on PC controlled systems.

Frequency response plotters

The simplest type of audio measurement

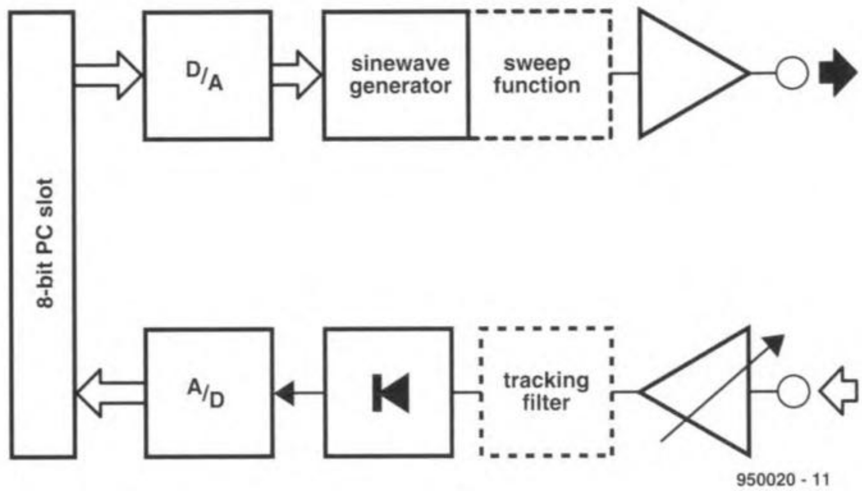
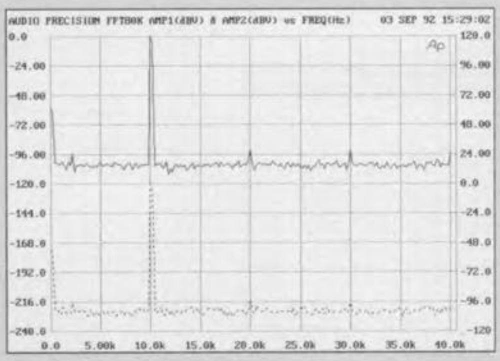
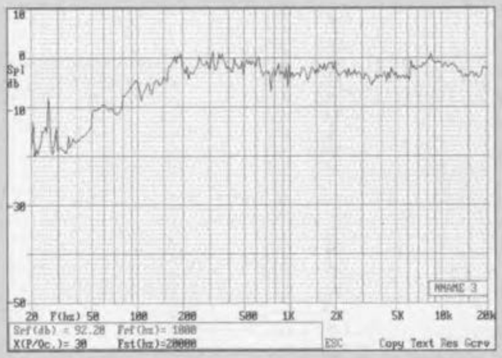


Fig. 1. Block diagrammatical structure of a computer-controlled frequency response plotter.

External instruments





Name:	**Audio Precision System One**	**AES-Mepeg**
Class:	external instrument	external instrument
Computer link:	own interface, PCI card, RS232 or IEEE-488	via printer port
Type:	analogue measurement. system with extra DSP unit and digital unit, MLS measurement. option	sweep generator with voltmeter
Computer requirements:	MS-DOS PC with 8088 or higher, DOS 3.1 or higher, min. 640 KB RAM	MS-DOS PC with 8088 or higher, min. DOS 2.0
Accuracy:	analogue section: THD+N<0.0015% (20 kHz bandwidth) 20 Hz - 20 kHz ±0.05 dB FFT-section: THD+N -120 dB (20 bits) max. fsample 192 kHz	14 bits
Measurement facilities:	frequency, amplitude, phase generator with very low distortion frequency characteristics harmonic distortion intermodulation distortion (also DIM) power measurements DSP section: FFT-analysis Dual domain version: measurements on digital audio signals	frequency characteristic Thiele/Small parameters impedance measurement voltmeter
Price:	from about £7,000	approx. £315
Who/where:	SSE Marketing Ltd. tel. (0171) 387 1262	AE Systeme Schwertstrasse 138 D-47799 Krefeld, Germany tel. (+49) 2151 316071
Options:	miscellaneous filters, IMD board, wow&flutter analyzer, burst/noise generator, DSP unit, dual-domain unit, measurement microphone, switchbox, multi-function module, application software	also available as a kit from about £80 (PCB + software)

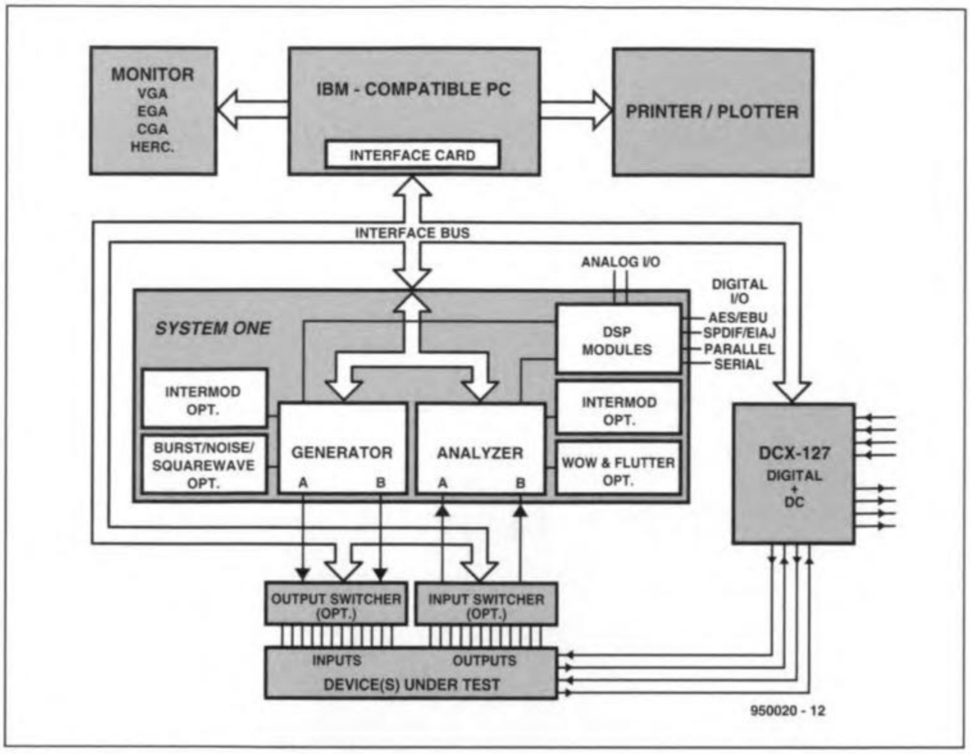


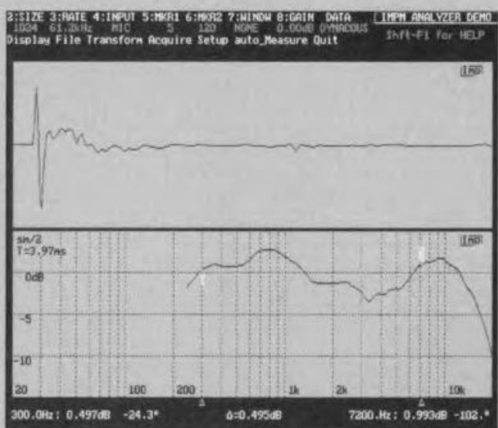
Fig. 2. Block diagram of an extensive analogue measurement system, the Audio Precision System One.

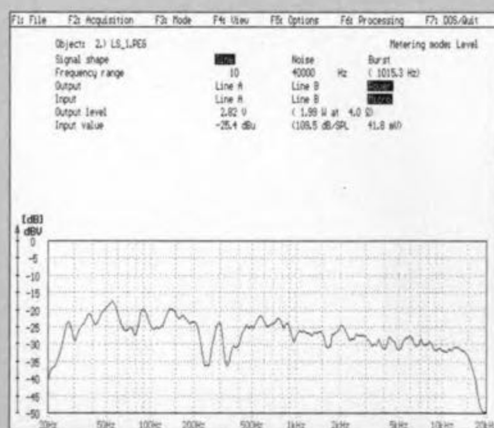
card is a system consisting of a sweep generator amd a voltmeter. A sinewave signal is applied to the equipment under examination (for instance, a loudspeaker or an amplifier), and the voltmeter is used to measure signal level levels at a number of points during the sweep operation. The frequency response is then displayed on the PC screen (see block diagram in **Fig. 1**). Functionally, this corresponds to the well known mechanical frequency response plotter. Unfortunately, a sinewave sweep gives unreliable test results in a 'normal', i.e., non-anechoic, test room because that produces reflections. This can be compensated partly by using narrow-band tracking filters, or by a so-called wobbulator function, which produces rapid variations of the instantaneous frequency within a third or an octave.

Analogue measurement systems

Distortion measurements require special equipment with very steep filters to separate the base frequency from the higher harmonics. The best known test instrument in this field is without doubt the Audio Precision System One (see title

External instruments





Name:	**IMP 2.0**	**Kemsonic AMS PC/ST type 1656**
Class:	external instrument	external instrument
Computer link:	via Centronics port	via RS232 interface
Type:	generator + FFT-analyser, MLS measurement option	sweep generator with voltmeter (and optional tracking filter)
Computer requirements:	MS-DOS-PC with 8088 or higher, min. MS-DOS 3.1, min. 640 KB RAM	MS-DOS-PC, min. MS-DOS 3.1, min. 640 KB RAM (Atari version also available)
Accuracy:	12 bits, f(sample) max. 61.4 kHz	not stated
Measurement facilities:	frequency characteristic phase characteristic impedance measurement Thiele/Small parameters waterfall spectrum	frequency characteristics impedance measurements Thiele/Small parameters room acoustics
Price:	approx. £395	from approx. £425
Who/where:	Clear Sound I. Brouwersteeg 4 NL-8911 BZ Leeuwarden, The Netherlands tel. (+31) 58 159927	Kemsonic Audio Measurement Systems GmbH Teutoburger Strasse 37 D-4800 Bielefeld, Germany tel. (+49) 521 175314, fax (+49) 521 176931
Options:	price incl. measurement microphone	1/3-octave tracking filter, phase measurement card, misc. microphones, vibration transducer, in-/output-module, software for quality checking

photograph), which has been around for more than ten years, and has become a kind of standard for measurements on hi-fi audio equipment. The instrument is, obviously, external, and contains all subcircuits for frequency response plotting, distortion measurements, and more. A version is available with a built-in FFT analyser, as well as an optional extension for measurements on digital audio signals. Prices of the APS One start at about £7,000. That may seem a lot, but considering the impressive performance and possibilities, this sum buys you one of the best measurement systems in its class.

The block diagram of the APS One is reproduced in **Fig. 2**. The ingredients are those of an ordinary measurement system: a low-distortion oscillator, input amplifiers with measurement filters, a frequency meter and a voltmeter. The only non-standard block is the interface between the PC and the measurement system. It allows you, for instance, to set the frequency and output level of the oscillator, while the measurement results are returned to the computer in digitized form. Next, the PC converts the data

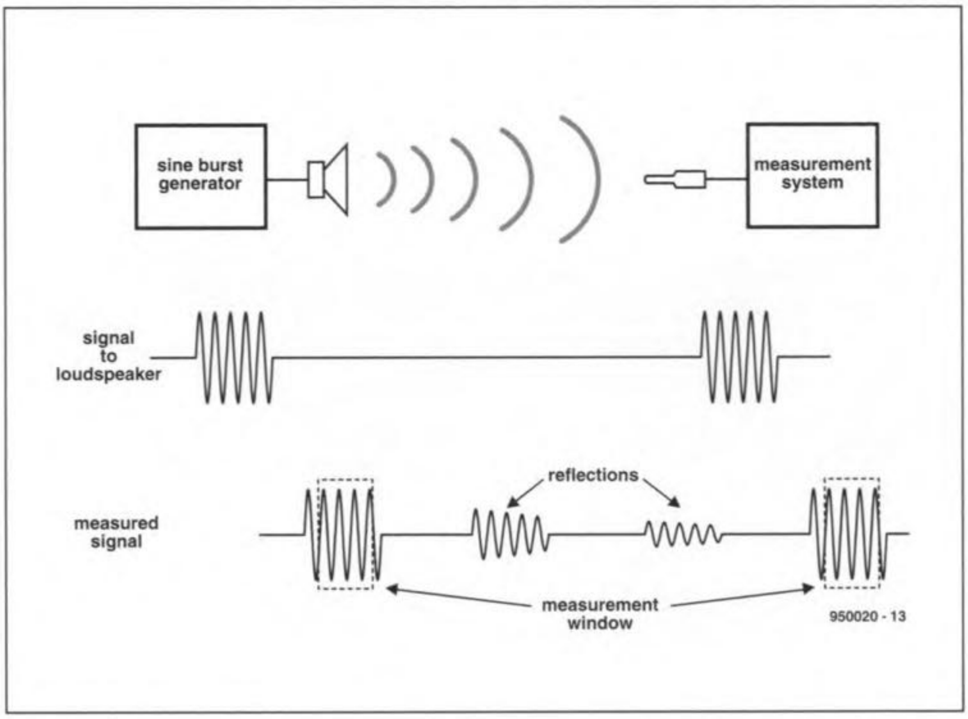
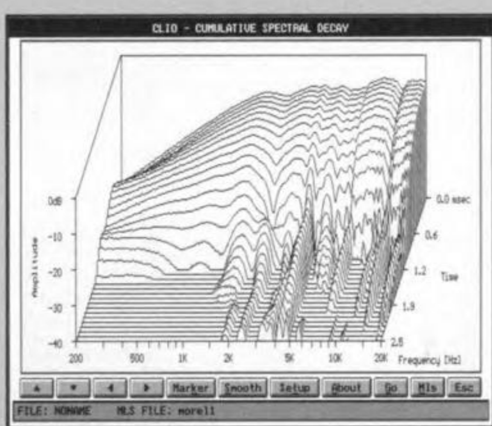
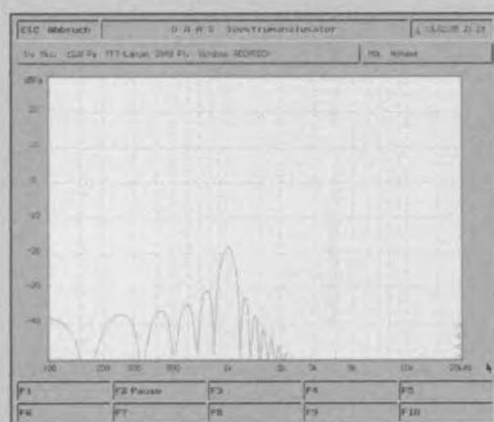


Fig. 3. The gated sinewave measurement is based on sending bursts of sinewaves to the test microphone. The receiver is disabled outside the gate period, so that sound reflections caused by the test room are ignored.

Insertion cards





Name:	**Audiomatica Clio 3.0**	**DAAS 3L**
Class:	8-bit insertion card	16-bit insertion card
Computer link:	internal	internal
Type:	generator + FFT-analyser, MLS measurement feature	generator + FFT-analyser, MLS measurement feature
Computer requirements:	MS-DOS PC with 80286 or higher, min. EGA video card, 640 KB RAM, hard disk	MS-DOS-PC with min. 80386, MS-DOS 5.0, hard disk, min. 2 MB RAM, EGA/VGA card
Accuracy:	2-channel 16-bits, f(sample) max. 51.2 kHz	16-bit 2-channel A/D- and D/A-converter, f(sample) max. 48 kHz
Measurement facilities:	FFT analyser harmonic distortion frequency characteristic phase measurement waterfall response room acoustics Thiele/Small parameters impedance, L and C measurements oscilloscope function	frequency characteristic phase characteristic harmonic distortion intermodulation measurement impedance curve Thiele/Small parameters inductance/capacitance oscilloscope function waterfall spectrum
Price:	approx. £725	to be advised
Who/where:	Acoustical Supply International 100 Cherokee Blvd., Suite 211 Chattanooga, TN37405, U.S.A. tel. (+1) 615 752 1720; fax: (+1) 615 752 1725	adm engineering Steinmaate 24 D-48529 Nordhorn, Germany tel. (+49) 5921 721000
Options:	measurement microphone, microphone preamplifier	

into, for example, a frequency curve which may be viewed on the monitor. Despite (or, perhaps, thanks to) the fairly conventional setup of this system, the accuracy is very high, and sets a standard for all other PC controlled measurement systems. The FFT section has more goodies including breaking down a signal into its frequency components, and performing MLS measurements. The (optional) dual domain unit enables the user to analyse digital audio signals.

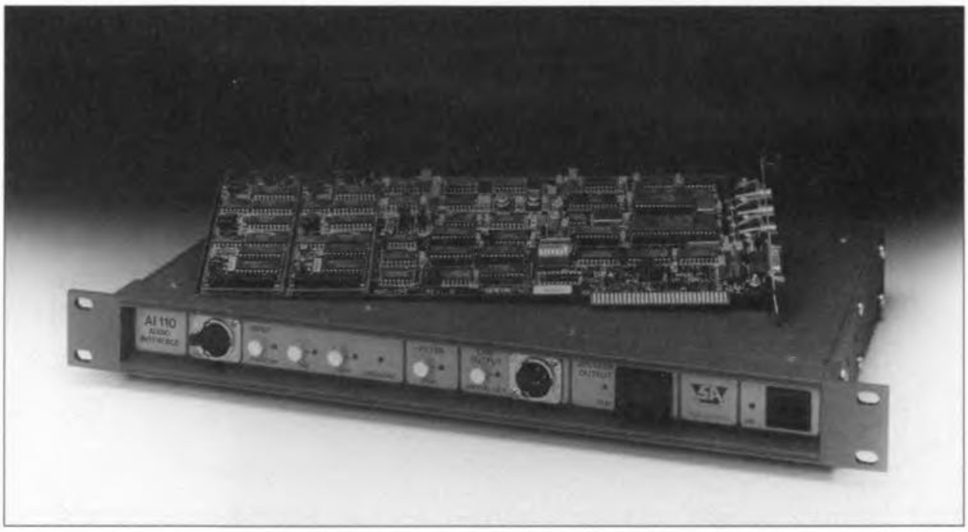


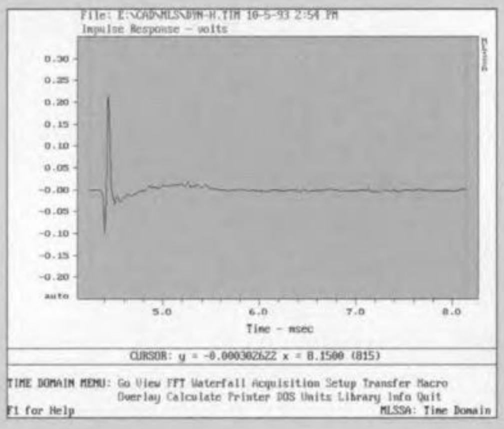
Fig. 4. The industry standard MLSSA measurement card. Below the card is a power amplifier/microphone amplifier extension available as an option from Stage Company.

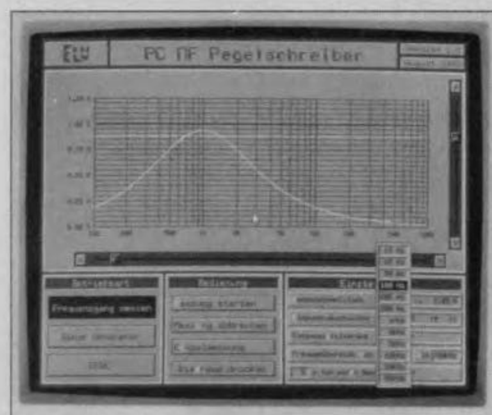
Gated sinewave measurements

Long before computer controlled measurement had reached sufficient power to enable FFT analysis to be performed in a simple way, a method had been discovered (by, among others, KEF), to run accurate measurements on loudspeakers. The big problem with loudspeakers is that the measurement microphone inevitably picks up room reflections also, for instance, while doing a frequency response plot. This can be avoided, but only by doing the test in an anechoic room which unfortunately is very expensive and very large.

The gated sinewave measurement involves driving the loudspeaker under test with a burst-shaped signal with a particular frequency. The system measures the level of the received signal during a short period only, during which the burst reaches the microphone for the first time (**Fig. 3**). All subsequent reflections are thereby ignored and can not cause measurement errors. In this way, you are able to do a fairly accurate frequency response measurement on loudspeakers in an ordinary room. An example of a modern measurement system based on this principle is LMS.

Insertion Cards





Name:	**DRA MLSSA V9.0**	**ELV PC frequency curve writer**
Class:	8-bit insertion card	8-bit insertion card
Computer link:	internal	internal
Type:	generator + FFT-analyser, MLS measurement feature	sweep generator with voltmeter
Computer requirements:	MS-DOS PC, min. 8086, 640 KB RAM, coprocessor, min. MS-DOS 2.1	MS-DOS PC
Accuracy:	12-bit A/D converter, f(sample) max. 160 kHz	not stated
Measurement facilities:	frequency characteristic phase characteristic time-delay curve impedance curve Thiele/Small parameters radiation pattern of loudspeakers waterfall spectrum room acoustics	frequency response measurement with the aid of a sinewave sweep
Price:	approx. £4200	approx. £90
Who/where:	DRA Laboratories 4587 Cherrybark Court Sarasota, FL 34241, U.S.A. tel. (+1) 813 927-2617, fax (+1) 813 925-0964	ELV GmbH D-26787 Leer Germany tel. (+49) 491 6008-0 fax (+49) 491 7016
Options:	audio interface SA AI 110	

Pulse measurements

Another method which was frequently used in the past (in particular, by KEF), involved measuring the pulse response of a system. A very short pulse (called dirac) in principle contains an infinitely wide frequency spectrum. With the aid of FFT analysis, the shape of the measured response to the pulse allows the transfer function of the system to be determined. In principle, this method allows the effects of the room to be virtually eliminated by doing an FFT analysis on the first pulse only which is received by the test microphone. A number of measurement systems including Clio and PC Audiolab are capable of running pulse response tests alongside other measurement methods.

MLS systems

Although the acronyms LMS and MLS are easily mixed up, they stand for completely different things. MLS stands for Maximum Length Sequence, and refers to a smart FFT method introduced by Douglas D. Rife about ten years ago. The basis is a short, accurately defined noise signal. This is picked up by the measurement microphone and fed to the FFT

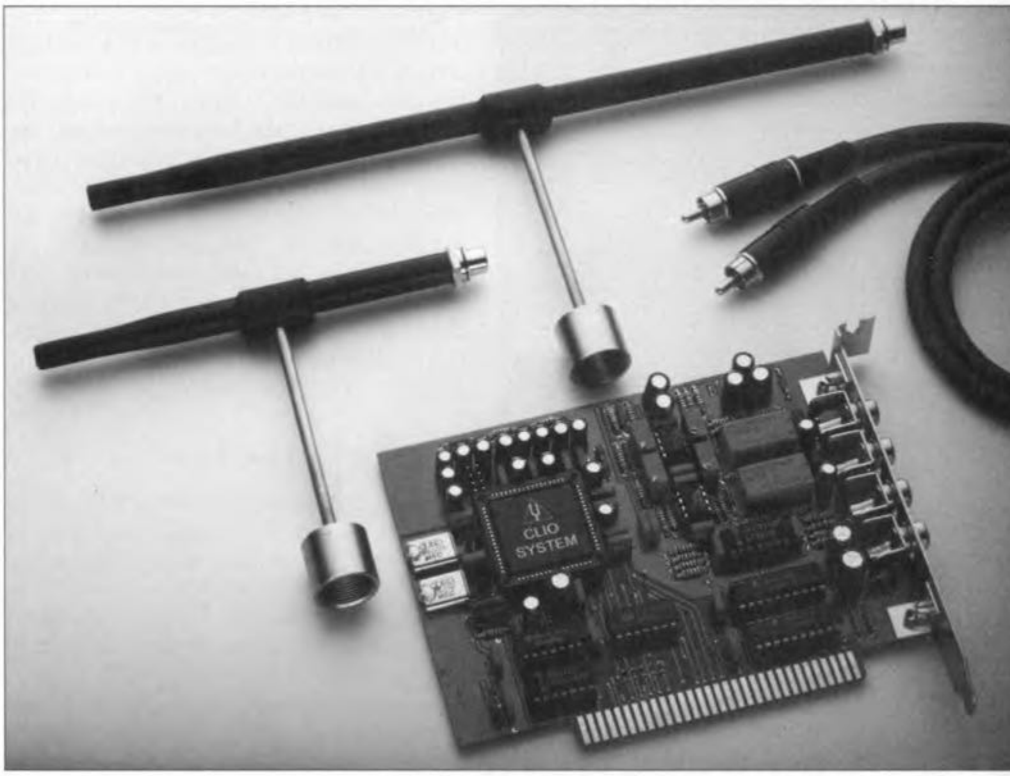
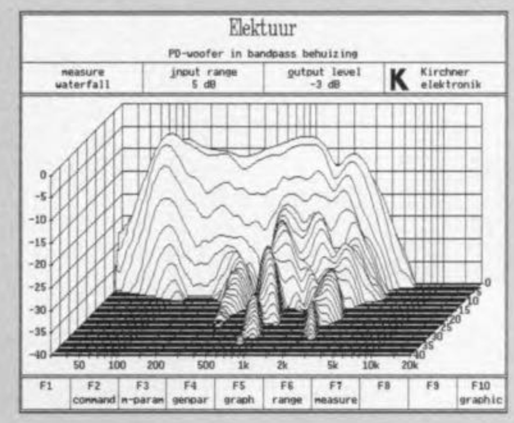
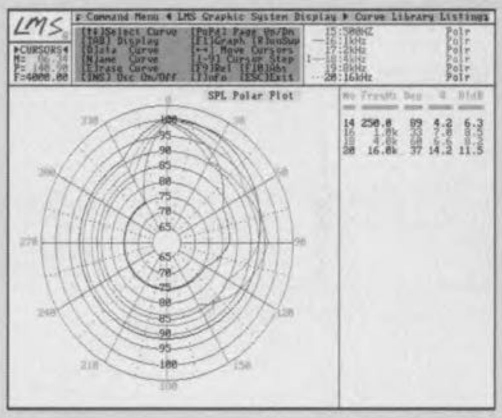


Fig. 5. A modern measurement system called Clio, together with a couple of test microphones. Note the small size of this card compared to the MLSSA card.

Insertion Cards





Name:	**Kirchner ATB 2.4**	**LMS**
Class:	8-bit insertion card	8-bit insertion card
Computer link:	internal	internal
Type:	generator + FFT-analyser	sweep(burst) generator and voltmeter with "gating"-technique
Computer requirements:	MS-DOS PC	MS-DOS PC
Accuracy:	12-bit converter, f(sample) max. 100 kHz	not stated
Measurement facilities:	frequenct characteristic impedance measurement phase measurement radiation patter FFT analysis harmonic distortion waterfall spectrum oscilloscope function	"gated" sound presure level measurements frequency characteristics impedance curves transient response (via inverse FFT) Nyquist diagram radiation pattern of loudpeakers
Price:	approx. £1280	approx. £1000
Who/where:	Kirchner Elektronik Wendenstrasse 53 D-3300 Braunschweig, Germany tel/fax (+49) 531 46412	Linear X Systems Inc. In the UK: Munro Associates tel. 0171 379 7600
Options:	software for quality checking in-/output-module	measurement microphone included

Fig. 6. These days it is possible to do pulse response, frequency response plotting and distortion measurements 'on a shoestring' with the aid of programs like AIRR and PC Audiolabs, which use the PC's internal sound card.

analyser, which 'dissects' it, and computes the pulse response of the system. The short length of the noise 'burst' and the fact that the FFT window can be set for the calculation of, for example, the frequency response, it is possible to virtually eliminate the effect of the room. The *de facto* standard in this field is the MLSSA system. This plug-in card, although now a bit rusty, has a resolution of 12 bits which is really quite sufficient for measurements on loudspeakers. Loudspeakers manufacturers meanwhile consider MLSSA to be a standard.

Product overview

The four test methods described above cover most of the currently available audio measurement systems. Depending on your wishes, you have a choice from 12 systems which are briefly introduced below. It should be noted that this survey can not possibly cover all available products in the field. Undoubtedly you may come across competitive products which are not discussed here.

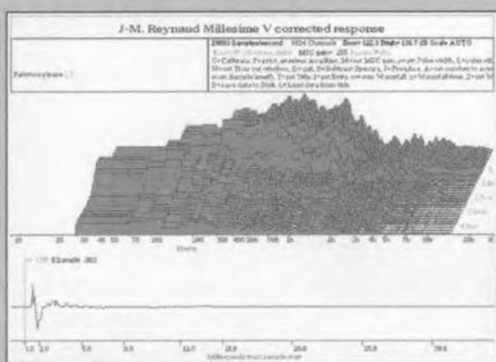
Prices start from about £85 for a simple frequency response plotter, and then rise rapidly. Prices of systems capable of doing serious measurements, including FFT analysis and MLS-based testing, start at about £350.

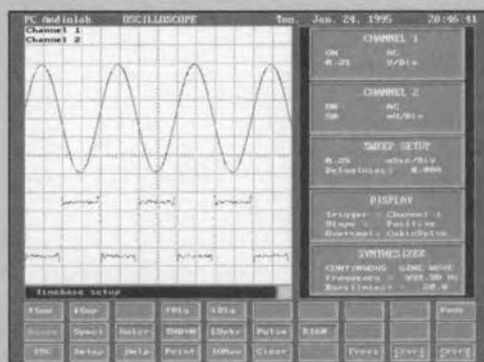
Two items from the overview, AIRR and PC Audiolab deserve special attention. Normally, a measurement system consists of a card, or a box with a connection cable to the computer. That, by definition, makes the system fairly expensive. To reduce cost, AIRR and PC Audiolab make use of the sound card in the PC. Today's generation of Soundblaster and Adlib compatible cards offer 16-bit sound recording and playback facilities at prices just under £100. Backed up by the appropriate software, such cards offer measurement options comparable to those of a system like MLSSA, at much lower cost. However, these developments are in the initial stages as yet, and we may have to wait for some time for affordable software to become available which offers the same possibilities as MLSSA. To those of you keen on having a first go at this challenging field, the dirt-cheap AIRR software from Old Colony Sound Lab in the U.S.A. comes highly recommended.

(950020)

Software

Reviews of these programs in *Speaker Builder*, 8/94. For address, see *SB* advert elsewhere in this issue





Name:	**AIRR (Anechoic and In-Room Response)**	**PC AudioLab 2.0**
Class:	software	software
Computer link:	—	—
Type:	program which uses PC sound card for generator function and FFT-analysis	program which uses PC sound card for generator function and FFT-analysis
Computer requirements:	MS-DOS PC	MS-DOS PC witht 80386 or higher, min. MS-DOS 3.3, min. 4 MB RAM, VGA card
Accuracy:	depending on sound card	depending on sound card
Measurement facilities:	pulse measurements FFT analysis frequency characteristic waterfall spectrum	FFT-analyser pulse measurements harmonic distortion frequency characteristic (with sweep) phase characteristic impedance curve room acoustics oscilloscope function network analyser waterfall spectrum
Price:	$49.95 (+ $10 P&P)	from about $300
Who/where:	Old Colony Sound Lab PO Box 243, Department B94 Peterborough, NH 03458-0243 USA tel. (+1) 603.924.6371 fax (+1) 603.924.9467	Microacoustics, Audio Software Products 2553 Carpenter St Thousand Oaks, CA 91362 tel/fax (+1) 805 495-8945

Measurement Products

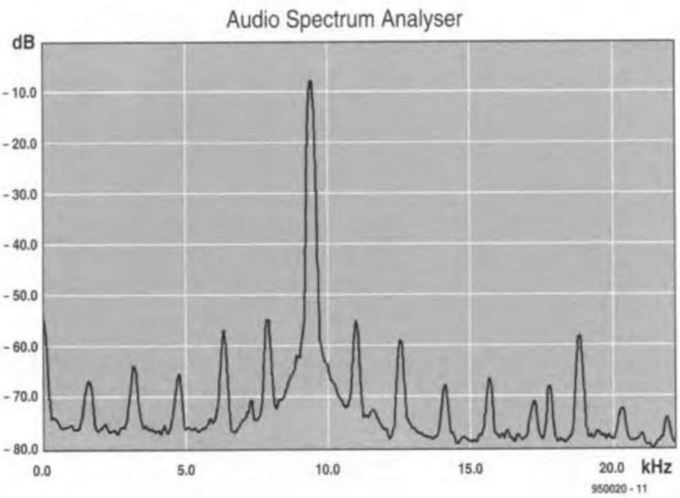
Pico Technology's Range of Virtual Instruments

Pico's ADC range of devices are ideal for making audio measurements on a PC. For example, the ADC-100, together with PicoScope software, provides nearly all the test equipment an audio engineer requires: a dual channel oscilloscope, spectrum analyser, frequency meter, decibel meter and voltmeter.

The ADC-100 is a 12-bit, 100 kHz unit that plugs into the parallel port of any PC and requires no power supply. Its flexible input ranges (from ±20 V to ±50 mV) allow line or microphone signals to be directly connected.

One typical application is the monitoring of a mixing desk. During the sound check, the spectrum analyser is used to help set EQ levels. Once the performance has started, it has proved invaluable in spotting early signs of feedback and identifying its frequency.

Other interesting applications have been the automatic testing and calibration of loudspeakers and even the selection of wood with good resonant qualities for a violin maker. With the optional PicoLog software, it can also perform data logging, for example, monitoring sound levels with an alarm if levels go above legal limits.



The ADC-100 costs £199 with PicoScope software and £209 with PicoScope and PicoLog.

Full details from **Pico Technology Ltd, 149–151 St Neots Road, Hardwick, Cambs CB3 7QJ.** Telephone 01954 211 717; Fax 01954 211 880.

Measurement microphone

Microphone Type M31 from Linear X Systems is a low-voltage electret capacitor microphone specifically designed for the measurement of transducers, loudspeaker systems, room response, and other similar acoustic characteristics. Also, it provides a cost-effective solution to many general-purpose acoustic measurements. It is small, has a wide, smooth frequency response and its interface/power requirements are straightforward.

The small diameter (0.3 in. = 8 mm) and depth make the M32 an excellent choice for near-field measurements and in other locations where space is at a premium. The small diameter also produces less directional sensitivity for high-frequency measurements. The low voltage and current requirements of the microphone/preamp allow for easy setup and operation from a wide variety of power supplies and can even be powered from a standard 9 V battery.

Full details from **Linear X Systems Inc., 7556 SW Bridgeport Road, Portland, OR 97224, USA.** Telephone +1 (503) 620-3044; Fax +1 (503) 598 9258.

High performance audio/hi-fi/loudspeaker components for music lovers and audiophiles are available from **North Creek Music Systems, Main Street, P.O. Box 1120, Old Forge, NY 13420, USA.** Telephone/fax +1 (315) 369 2500.

Useful addresses for information on noise measurement:

Lucas CEL Instruments Ltd, 35-37 Bury Mead Road, Hitchin, Herts England. Telephone +44 (0)1462 422411; Fax +44 (0)1462 422511.

CEL Akustik GmbH, Erzbergerstrasse 115, 4050 Mönchen Gladbach, Germany. Telephone +49 (0)2161 41071/2; Fax +49 (0)2161 41073.

Lucas Industrial Instruments, 760 Ritchie Highway, Suite N6, Severna Park, MD, 21146, USA. Telephone +1 (301) 544 8773; Fax +1 (301) 544 9054.

Other Products

Jackson Brothers Limited

From time to time, Jackson's products – variable capacitors, slow-motion drives and panel furniture – are specified in published constructional projects. Because of the very large product range, it is inevitable that items specified or suitable substitutes are not always available from stock.

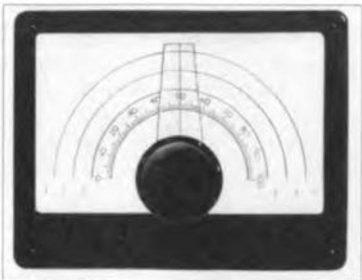
Arrangements have been made for Isoplethics – see next page – to hold stocks of Jackson's products for published projects. Where necessary, Isoplethics will also arrange supplies of any special mounting brackets or related hardware, and can provide technical support.

Jackson Brothers Ltd, 58–72 Dalmain Road, London SE23 1AX. Telephone 0181 681 2754/7.

New Products

Return of an old favourite

To mark the inception of their distributor service for Jackson Bros variable capacitors, drive components and related hardware, Isoplethics announces the renewed availability of the 6/36 slow motion drive and dial assembly (Cat. No. 4103/A).



Measuring 123×95 mm overall, the assembly comprises a dual ratio 6:1 and 36:1 ball drive, black, plastic-coated escutcheon, printed card scale, hair-line pointer and clear plastic window. The scale itself has an inner 0–100 logging scale and three blank 180° scale arcs for calibration by the user.

Ideal for home-construction projects, including direct-conversion receivers, grid-dip oscillators, wavemeters and VFOs, the 6/36 drive assembly is available direct from Isoplethics in the UK and the rest of the European Union for £19·50. In common with all Isoplethics prices, this includes VAT and post & packing. **Isoplethics, 13 Greenway Close, North Walsham, Norfolk NR28 0DE**. Phone 01692 403 230.

Constructors' hardware for the 90s

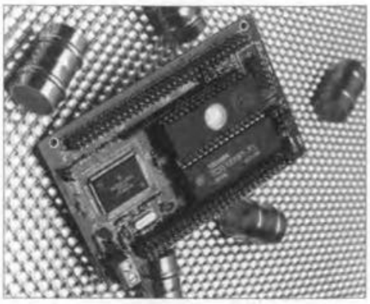
A 48-page 1995 catalogue containing a range of hardware (boxes, meters, brackets, tools, etc) for the electronics constructor is available from
Sescom, Inc., 2100 Ward Drive, Henderson, NV 89015-4249, USA.
Telephone +1 (702) 565 3400
Fax +1 (702) 565 4828

First low-cost in-circuit emulator for the Siemens 80C166 family

In response to the demand for a lower cost alternative to its high specification T32 in-circuit emulation system, Hitex has announced the release of the new AX166, PC-hosted 80C166 family emulator.
Hitex (UK) Ltd, Warwick University Science Park, Coventry CV4 7EZ, England. Telephone +44 (0)1203 692 066
Fax +44 (0)1203 692 131

Complete CPU with PC starter pack

The new Micro-Midget from CMS is a small (3.8"×2" approx), powerful 16/32 bit controller. It is ideal as a component in intelligent control systems, with an



advanced royalty-free, real-time operating system and full support for high-level languages, including C. The controller has up to 22 digital I/O lines which can be configured for input or output as required, a single serial port operating at up to 38400 baud with RS232 or RS485 driver options and two 16-bit timer/counters. The peripheral expansion bus can be used with 68000 type devices, 8051 devices, or I2C bus peripherals. Full details from
Cambridge Microprocessor Systems Limited, Units 17-18, Zone 'D', Chelmsford Road Ind. Est., Great Dunmow, England M6 1XG. Telephone +44 (0)1371 875 644; Fax +44 (0)1371 876 077.

Major breakthrough in Electromagnetic circuit simulation

Number One Systems have announced the world's first affordable Electromagnetic Simulation program, called **LAYAN**. This program, which runs on a standard 486, breaks through the barriers which have until now prevented most design engineers from fully understanding the behaviour of their PCB-based designs. Full details from

Number One Systems, Harding Way, Somersham Road, St Ives, Huntingdon Cambs, England PE17 4WR. Telephone +44 (0) 1480 461 778; Fax +44 (0) 1480 494 042.

DIL-packaged DC-DC converters

When printed-circuit board space is at a premium, and a well-regulated d.c. power supply is required close to the devices being driven, one of Gresham Power Electronics' EL5 Series of high-isolation 5 W d.c. to d.c. converters may well fit the bill. Full details from
Gresham Power Electronics, Telford Road, Salisbury, Wiltshire, England SP2 7PH. Phone +44 (0)1722 413 060.

Compact LED DPM

The DPM959 is an LED Digital Panel Meter from the new Lascar 900 Series of digital panel instruments, which can be scaled easily by the user to indicate voltage, current, or other engineering units.

Although housed in a compact moulding with only 15 mm depth behind the panel, the DPM959 has high-efficiency 14.2 mm LED displays. Full details from
Lascar Electronics, Module House, Whiteparish, Salisbury, Wiltshire, England SP5 2SJ. Phone +44 (0)1794 884 567; Fax +44 (0)1794 884 616.

'Green' controller from Microchip

Microchip's new Energy Management Controller can reduce consumption by up to 30%. The MTE1122 is ideal for all residential, commercial and industrial equipment that uses a.c. motors, including refrigerators, freezers, washing machines, dryers, swimming pool pumps, and heating, ventilation and air conditioning equipment. Details from
Arizona Microchip Technology, Unit 6, The Courtyard, Meadowbank, Furlong Road, Bourne End, Bucks, England SL8 5AJ. Phone +44 (0)1

Micro PLC with 16 I/O ports and analogue capability for under £100

The Dianamic MPLC1000 is a new low-power Micro PLC with a standard configuration of 16 I/O ports. The ports can be configured as analogue ports with an 8-bit resolution, digital ports or high-speed counters with a counting frequency of 2.5 kHz. The power supply is d.c.-d.c. with an input voltage tolerance of between 6.5 V and 30 V. Standard consumption is 3 mA when running with all ports configured as inputs.

Measuring 72×67×17 mm, the unit features full BASIC language implementation with full mathematical manipulation, including Boolean logic. Further details from
NMB Marketing, London House, 100 New Kings Road, London, England SW6 4LX. Phone +44 (0)171 731 8199; Fax +44 (0)171 731 8312.

Stepping motor kits

JPG Electronics has introduced a new range of kits for controlling stepping motors using a computer via the parallel port or manually with switches for automating machines, making robots, and learning about stepping motors. The kits come complete with stepping motors with models suitable for bipolar (4-wire types) and unipolar (6-wire types).

Top of the range is the Comstep for computer control of two stepping motors simultaneously (motors with 9–24 V coils and a coil current of up to 4 A with an optional power interface). The kit comes complete with control software for a PC (XT to 486).

For further information, contact **JPG Electronics, 276–278 Chatsworth Road, Chesterfield, England S40 2BH**. Telephone +44 (0)1246 211 202; Fax +44 (0)1246 550 959.

Video enhancer from Maplin

The Video Enhancer with sharpness control is designed to improve the TV picture quality when dubbing or playing back home video, and is instrumental in improving poor recordings. A recording can be made on two VCRs at the same time, while audio can be recorded in stereo, or lefthand and righthand channels may be combined for recording in a mono system VCR.

The Video Colour Processor allows the user to correct colour saturation and contrast from a camcorder or VCR. The unit features a gain control for improving the image when shooting under low-light conditions, a colour control to provide compensation for colour saturation, and editing facilities in the form of video and audio fade controls.

The Video Enhancer/Editor is a high-quality unit, with stereo sound, and is ideal for editing, processing and enhancing home videos quickly and easily whilst adding an extra sound track and narration.

Details from **Maplin Electronics, P O Box 3, Rayleigh, Essex, England SS6 8LR**. Phone +44 (0)1702 552 911. Fax +44 (0)1702 553 935.

Low-profile sealed keypads

The Grayhill Series 84LS low-profile range of environmentally sealed keypads is now available from EAO Highland. The new range has a thickness of only 5.26 mm (9.07 mm incl. keys) and is available in 12- and 16-key versions. The keys feature audible, tactile contacts with a low contact resistance, which is compatible with MOS, TTL and DTL circuitry. Details from
EAO-Highland Electronics, Albert Drive, Burgess Hill, West Sussex, England RH15 9TN. Phone +44 (0)1444 236 000; Fax +44 (0)1444 236 641.

APPLICATION NOTE

The content of this note is based on information received from manufacturers in the electrical and electronics industries or their representatives and does not imply practical experience by *Elektor Electronics* or its consultants.

LM3886 150 W audio power amplifier with mute

A National Semiconductor Application

The LM3886 is a member of National Semiconductor's *Overture™Audio Power Amplifier Series.* It is a high-performance amplifier capable of delivering 60 W of continuous average power to a 4 Ω load or 30 W into 8 Ω with 0.03% (THD + N) from 20 Hz to 20 kHz.

The performance of the LM3886, because of its *Self Peak Instantaneous Temperature* (K) (**SPiKe**) protection circuitry, puts it in a class above discrete and hybrid amplifiers by providing an inherently, dynamically protected, *Safe Operating Area* (SOA). SPiKe protection means that these parts are completely safeguarded at the output against overvoltage, undervoltage, overloads, including shorts to the supplies, thermal runaway, and instantaneous temperature peaks. See box further on in this article.

The LM3886 maintains an excellent signal-to-noise ratio of >92 dB with a typical low noise floor of 2.0 μV. It provides very low distortion (THD + N of 0.03%) at the rated output into the rated load over the audio spectrum and provides excellent linearity with an IMD (SMPTE) rating (typical) of 0.004%.

Typical applications of the LM3886 include compact stereo, self-powered speakers, surround sound amplifiers, and high-end stereo TV receivers.

Single-supply application

In the typical single-supply audio amplifier application in **Fig. 1**, R_{IN} acts as a volume control by setting the voltage level allowed to the amplifier's input. Resistors R_A provide d.c. bias for the single-supply operation and bias current for the positive input terminal. Capacitor C_A provides bias filtering, while C provides a.c. coupling at the input and output of the amplifier for single-supply operation.

Capacitor C_C reduces the gain (bandwidth of the amplifier) at high frequencies to avoid quasi-saturation oscillations of the output transistor. It also suppresses external electromagnetic switching noise created by fluorescent lamps.

Resistor R_i provides a.c. gain in conjunction with feedback resistor R_{f1}. C_i is a feedback capacitor that ensures unity gain at d.c. It also provides a low-frequency pole (highpass roll-off) at $f_c = 1/2\pi R_i C_i$.

Feedback resistor R_{f2}, in conjunction with C_f, R_{f1} and R_i, provides lower a.c. gain at higher frequencies. Also, a high-frequency pole (lowpass roll-off) exists at

$$f_c = [R_{f1}R_{f2}(s+1/R_{f2}C_f)]/[(R_{f1}+R_{f2}) \times \{s+1/C_f(R_{f1}+R_{f2})\}].$$

C_f is a compensation element which, in conjunction with R_{f1} and R_{f2}, reduces the a.c. gain at higher frequencies.

R_M is the mute resistance set up to allow 0.5 mA to be drawn from pin 8 to turn the muting function off. Its value is calculated from $R_M \leq (|V_{EE}|-2.6\ V/I_8$, where $I_8 \geq 0.5$ mA.

C_M is the mute capacitance set up to create a large time constant for turn-off and turn-off muting.

Features

- 60 W continuous average output power into 4 Ω at $V_{cc} = \pm 28$ V
- 30 W continuous average output power into 8 Ω at $V_{cc} = \pm 28$ V
- 50 W continuous average output power into 8 Ω at $V_{cc} = \pm 35$ V
- 150 W instantaneous peak output power capability
- Signal-to-noise ratio ≥92 dB
- Input mute function
- Output protection from a short to ground or to the supplies via internal current limiting
- Output overvoltage protection against transients from inductive loads
- Supply undervoltage protection, not allowing internal biasing to occur when $|V_{EE}| + |V_{cc}| \leq 12$ V, thus eliminating turn-on and turn-off transients
- 11-lead TO-220 package

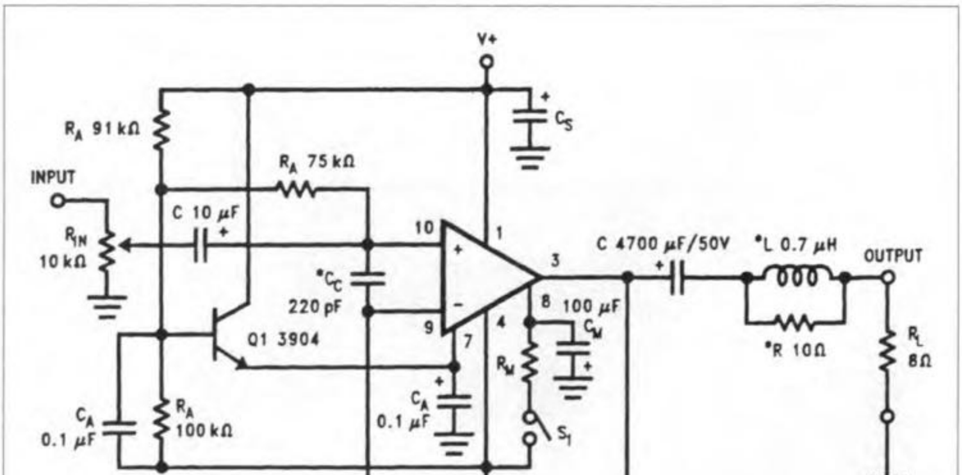


Fig. 1. Typical amplifier circuit based on LM3886.

R_{SN} and C_{SN} stabilize the output stage by creating a pole that eliminates high frequency oscillations

($f_c = 1/2\pi R_{SN}C_{SN}$).

Inductance L provides a high impedance at high frequencies so that R may decouple a highly capacitive load and reduce the *Q* of the series resonant circuit caused by the capacitive load. It also provides a low impedance at low frequencies to short out R and pass audio signals to the load.

Capacitor C_S provides power supply filtering and bypassing.

S_1 is the mute switch that mutes the music going into the amplifier when open.

Although the components have specific desired functions that are designed to reduce the bandwidth and eliminate unwanted high-frequency oscillations, they may cause certain undesirable effects when they interact. Interaction may occur where the reactances of components are close to one another. An example is the coupling capacitor, C_C, and the compensation capacitor, C_f. These two components act as low impedances to certain frequencies which will couple signals from the input to the output. Careful consideration needs to be given to the function of these components when designing them into the circuit.

Maximum power dissipation

The following equations can be used to accurately calculate the maximum and average IC power dissipation for an amplifier design, given the supply voltage, rated load and output power.

$$P_{D(max)} = V_{cc}^2/2\pi^2R_L \qquad \text{[Eq. 1]}$$

$$P_{D(ave)} = (V_{o(pk)}/R_L)\,(V_{cc}/\pi - V_{o(pk)}/2) \qquad \text{[Eq. 2]}$$

$$P_{D(ave)} = V_{cc}V_{o(pk)}/\pi R_L - V_{o(pk)}^2/2R_L \qquad \text{[Eq. 3]}$$

where P_D is the power dissipation, (ave) is average, V_{cc} is the total supply voltage, $V_{o(pk)}$ is the peak output voltage = V_{cc}/π, and R_L is the rated load.

Heat sinks

The choice of a heat sink for a high-power audio amplifier is made entirely to keep the die temperature at a level such that the thermal protection circuitry does not operate in normal circumstances. The heat sink should be chosen to dissipate the maximum IC power for a given supply and rated load. With high-power pulses of longer duration than 100 ms, the case temperature will rise drastically when a heat sink is not used. Therefore, the case temperature, as measured at the centre of the package bottom, is entirely dependent on heat sink design and the mounting of the IC to the heat sink.

Proper mounting of the IC is necessary to minimize the thermal drop between the package and the heat sink. The heat sink must also have enough metal under the package to conduct heat from the centre of the package bottom to the fins without excessive temperature drop.

A heat conducting paste (thermal grease) should be used when mounting the package to the heat sink. Without this compound, thermal resistance will be no better than 0.5 K W^{-1}, and probably much worse. With the compound, thermal resistance will be 0.2 K W^{-1} or smaller, assuming under 0.005 in. (0.13 mm) combined flatness runout for the package and heat sink.

Should it be necessary to isolate V^- from the heat sink, an insulating washer (beryllium oxide, anodized aluminium, mica, silicone-rubber) is required. Experience has shown that rubber washers deteriorate and must be replaced when the IC is dismounted. Hard washers require the use of thermal compound on both faces.

When the maximum IC power dissipation is known for a given supply voltage, rated load and the desired rated output power, the maximum thermal resistance (in K W^{-1}) of a heat sink can be calculated. This calculation is made with Eq. (4), which is based on the fact that thermal heat flow parameters are analogous to electrical current flow properties.

It is also known that typically the thermal resistance, θ_{JC} (junction to case), of the LM3886 is 1 K W^{-1} and that using a good thermal grease provides a thermal resistance, θ_{CS} (case to heat sink), of about 0.2 K W^{-1}.

The thermal resistance from the die (junction) to the outside air (ambient) is a combination of three thermal resistances, two of which, θ_{JC} and θ_{CS}, are known. Since convection heat flow (power dissipation) is analogous to current flow, thermal resistance is analogous to electrical resistance, and temperature drops are analogous to voltage drops, the power dissipation from the LM3886 is equal to

$$P_{D(max)} = (T_{J(max)} - T_a)/\theta_{JA},$$

where $\theta_{JA} = \theta_{JC} + \theta_{CS} + \theta_{SA}$. Since $P_{D(max)}$, θ_{JC} and θ_{SC} are known,

$$\theta_{SA} = [T_{J(max)} - T_a) - P_{D(max)}(T_{JC} + \theta_{CS})]/P_{D(max)} \qquad \text{[Eq. 4]}$$

Again, it must be noted that the value of θ_{SA} is dependent on the system designer's amplifier application and its corresponding parameters. If the ambient temperature that the audio amplifier is to be working in is higher than the normal 25 °C, the thermal resistance for the heat sink, all other things being equal, will need to be smaller.

Equations (1) to (4) are the only ones needed to determine the maximum heat sink thermal resistance. This is, of course, given that the system designer knows the required supply voltages to drive his rated load at a particular power output level and the parameters provided by the semiconductor manufacturer. These parameters are the case thermal resistance, θ_C, $T_{J(max)}$ = 150 °C, and the compound resistance, θ_{CS}.

Supply bypassing

The LM3886 has excellent power supply rejection and does not require a regulated supply. However, to eliminate possible oscillations all op amps and power op amps should have their supply leads bypassed with low-inductance capacitors with short leads and located close to the package terminals. Inadequate power supply bypassing will manifest itself by a low-frequency oscillation known as motorboating or by high-frequency instabilities. These instabilities can be eliminated through multiple bypassing with a large tantalum or electrolytic capacitor (≥10 µF) which is used to absorb low-frequency variations and a small ceramic capacitor (0.1 µF) to prevent any high-frequency feedback through the power supply lines.

If adequate bypassing is not provided, the current in the supply leads, which is a rectified component of the load current, may be fed back into internal circuitry. This signal causes low distortion at high frequencies, requiring that the supplies be bypassed at the package terminals with an electrolytic capacitor of ≥470 µF.

Lead inductance

Power op amps are sensitive to inductance in the output lead, particularly with heavy capacitive loading. Feedback to the input should be taken directly from the output terminal, minimizing common inductance with the load.

Lead inductance can also cause voltage surges on the supplies. With long leads to the power supply, energy is stored in the lead inductance when the output is shorted. This energy can be dumped back into the supply bypass capacitors when the short is removed. The magnitude of this transient is reduced by increasing the value of the bypass capacitor near the IC. With not less than 20 µF local bypass, these voltage surges are important only if the lead length exceeds a couple of feet (about 1 metre) (≥1 µH lead induc-

SPIKe protection

The **Overture**™ Audio Power Amplifier Series possesses a unique protection system that saves audio designers components, size and cost of their systems. This translates into higher-power, more functional, more reliable, compact audio amplification systems.

These advantages, generally provided only in high-end discrete amplifiers, are accomplished by providing a protection mechanism within a monolithic power package. Since audio amplifier designers generally need to provide some sort of protection to the output transistors in order to keep product failures to a minimum, National Semiconductor's Audio Group has designed SPiKe (Self Peak Instantaneous Temperature [K]) protection. This is a mechanism designed to safeguard the amplifier's output from overvoltages, undervoltages, shorts to ground or to the supplies, thermal runaway, and instantaneous temperature peaks.

Temperature limiting

When the output transistor's temperature reaches about 250 °C, the amplifier, depending on its present operating conditions, will reduce the output drive transistor's base current, keeping the transistor within its safe operating area (SOA).

The uniqueness of SPiKe protected audio amplifiers is their ability to monitor the output drive transistor's SOA dynamically, regardless of an output to ground short, an output to supply short, or the reaching of its power limit by any pulse within the audio spectrum.

Overvoltage protection

In general, the amplifier should not be stressed beyond its *Absolute Maximum* (no signal) *Voltage Supply Rating* and should be protected against any condition that may lead to this type of voltage stress level. This type of protection generally requires the use of costly zener or fast-recovery Schottky diodes from the output of the amplifier to each supply rail.

However, SPiKe-protected audio amplifiers possess a unique overvoltage protection scheme that allows the device to sustain overvoltages for nominally rated speaker loads. In **Fig. 2**, the protection mechanism functions by first sensing that the output has exceeded the supply rail, then immediately turning the driving output transistor off, so that its breakdown voltage is not exceeded. The circuitry continues to monitor the output, waiting to turn the output drive transistor back on when the overvoltage fault has ceased.

Moreover, SPiKe-protected amplifier ICs possess an internal supply-clamping mechanism; a zener plus a diode drop from the output to the positive supply rail and an intrinsic diode drop from the output to the negative rail. This equates to clamping of about 8 V on the positive rail and 0.8 V on the negative rail.

Figures 3a and **3b** model the output stage for each overvoltage condition, showing how the voltage waveforms are clamped to their respective values for high-frequency waveforms.

Popless power on/off

Undervoltage protection is provided because all d.c. shifts or 'pops' at the output should be avoided in any amplifier design because of their destructive capability on a speaker. These pops are generally a result of the unstable nature of the output as internal biasing is established while the power supplies are coming up.

In SPiKe-protected ICs, this is accomplished by disabling the output, placing it in a high-impedance state, while its biasing is established. This function is achieved through the disabling of all current sources within the IC as denoted by control signal V_c in Fig. 2. In the LM388, the control signal will not allow the current sources to function until (a) the total supply voltage, from the positive rail to the negative rail, is >14 V, and (b) the negative voltage rail exceeds –9 V. It is this –9 V protection that causes the undervoltage protection scheme to disable the output up to 18 V between the positive and negative rail, assuming that both rails come up simultaneously.

The –9 V undervoltage protection is ground referenced to eliminate the possibility of large voltage spikes that occur on the supplies from momentarily enabling the relative 14 V undervoltage.

It should be noted that the isolation from the input to the output, when the output is in the high-impedance state, is dependent on the interaction of external components and traces on the circuit board.

Current limiting

The likelihood of an amplifier output short to ground always exists. If current limiting is not provided within the IC, the output drive transistor may be damaged. All SPiKe-protected ICs provide current limiting internally. The value of the current limit will vary for each particular audio amplifier and its output drive capability; in the LM3886 it is 7 A ($|V^+| = |V^-| = 20$ V, $T_{ON} = 10$ ms, $V_0 = 0$ V).

The internal current limiting circuitry functions by monitoring the output drive transistor current. The sensing of an increase in this current signals the circuitry to pull away drive current from the base of the output drive transistor as shown in Fig. 2. The harder the input tries to drive the output, the more current is pulled away from the output drive transistor, thus internally limiting the output current.

SPike-protected ICs are protected against momentary shorts from the output to either supply rail by limiting the current flow through the output transistors. Although accidents like these occur infrequently, they do happen. Normally, when such an accident occurs in a discrete design with no current-limiting protection, the output transistor would be subjected to the full output swing plus a large current drawn from the supply. This type of stress would destroy an output stage discrete transistor, whereas with SPiKe-protected amplifiers the current is internally limited. It should be noted, however, that this protection is not sustained indefinitely. In essence, the output shorts to either supply rail should not be sustained for any period greater than a few seconds, otherwise the long-term reliability of the IC is not guaranteed.

Thermal shutdown

SPiKe-protected audio amplifiers are safeguarded from thermal runaway, an area of concern for any complementary-symmetry amplifier. Thermal runaway is an excessive amount of heating and power dissipation of the output transistor from an increased collector current caused by the two complementary transistors not having the same characteristics or from an uncompensated V_{BE} being reduced by high temperatures.

If proper heat sinks are not used, the die will heat up owing to the poor dissipation of power when the amplifier is being driven hard for a long period of time. Once the die reaches its upper temperature limit of about 165 °C, the thermal shutdown protection circuitry is enabled, driving the output to ground. A pseudo pop at the output may occur when this point is reached, owing to the sudden interruption of the flow of music to the speaker. The device will remain off until the temperature of the die decreases about 10 °C to its lower temperature limit of 155 °C. It is at this point that the device will turn itself on, again amplifying the input signal.

Reference: National Semiconductor: Application Note 898.

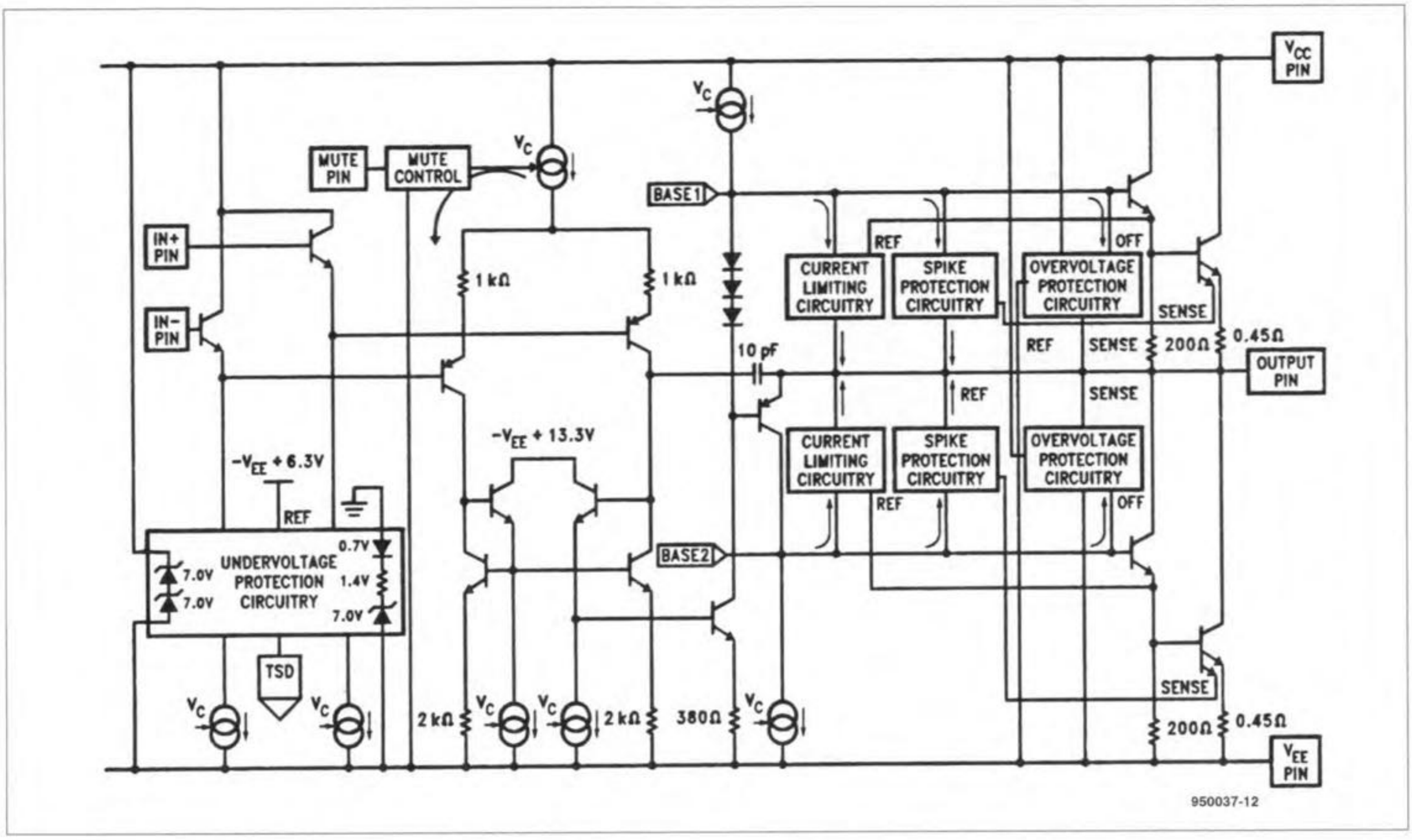


Fig. 2. Amplifier equivalent circuit with simplified SPiKe protection circuitry.

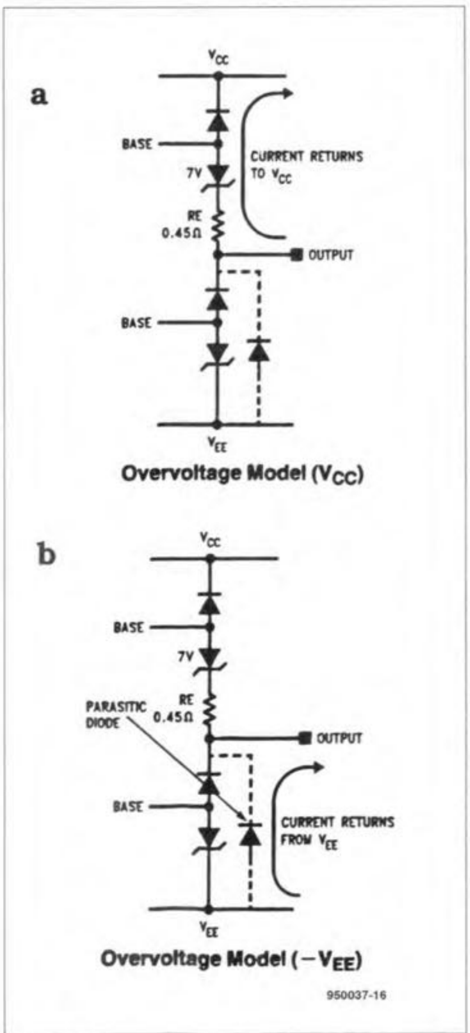


Fig. 3. Output stage overvoltage models

tance). Twisting together the supply and ground leads minimizes the effect.

Layout, earth loops, stability

The LM3886 is designed to be stable when operated at a closed-loop gain of 20 dB (×10) or greater, but as any other high-current amplifier, the LM3886 can be made to oscillate in certain circumstances. These usually involve PCB layout or output/input coupling.

In general, in fast, high-current circuits all sorts of problem can arise from improper earthing, which can be avoided by returning all grounds separately to a common point. Without isolating the ground signals and returning the grounds to a common point, earth loops may occur.

Earth loop is the term used to describe situations that occur in ground systems where a difference in potential exists between two ground points. Ideally, a ground is a ground, but unfortunately, in order for this to be true, ground conductors with zero resistance are necessary. Since real-world ground leads possess finite resistance, currents running through them will cause finite voltage drops. If two earth return lines tie into the same path at different points, there will be a voltage drop between them.

The solution to most ground loop problems is always to use a single-point ground system, although this is sometimes impractical. The single-point earth concept should be applied rigorously to all components and all circuits when possible. Violations of single-point grounding are most common among PCB designs, since the circuit is surrounded by large earth areas which invite the temptation to run a device to the closest earth spot.

Occasionally, current in the output leads (which function as antennas) can be coupled through the air to the amplifier input, resulting in high-frequency oscillation. This normally happens when the source impedance is high or the input leads are long. The problem can be eliminated by placing a small capacitor, C_c (50–500 pF) across the LM3886 input terminals.

Reactive loading

It is hard for most power amplifiers to drive highly capacitive loads very effectively and normally results in oscillations or ringing on the square-wave response. If the output of the LM3886 is connected directly to a capacitor with no series resistance, the square-wave response will exhibit ringing if the capacitance is >0.2 μF. If highly capacitive loads are expected caused by long speaker cables, a method commonly employed to protect amplifiers from low impedances at high frequencies is to couple the load through a 10 Ω resistor in parallel with a 0.7 μH inductor. The inductor-resistor combination shown in Fig. 2 isolates the feedback amplifier from the load by providing high output impedance at high frequencies, thus allowing the 10 Ω resistor to decouple the capacitive load and reduce the *Q* of the series resonant circuit. The *LR* combination also provides low output impedance at low frequencies, thus shorting out the 10 Ω resistor and allowing the amplifier to drive the series *RC* load (large capacitive load caused by long speaker cables) directly.

Audio power amplifier design

General

Some of the following parameters should be known when an audio amplifier design is started:

* desired power output;
* input impedance;
* maximum supply voltage;
* input level;
* load impedance;
* bandwidth.

The power output and load impedance determine the power supply requirements. However, depending on the application, some system designers may be limited to certain maximum supply voltages. If such a limitation exists, a practical load impedance should be chosen, which allows the amplifier to provide the desired output power, keeping in mind the current limiting capabilities of the device. In any case, the output signal swing and current are found from

$$V_{o(peak)} = \sqrt{(2R_LP_o)} \quad \text{[Eq. 5]}$$

$$I_{o(peak)} = \sqrt{(2P_o/R_L)} \quad \text{[Eq. 6]}$$

To determine the maximum supply voltage, the following parameters must be considered. Add the dropout voltage, $V_{o(d)}$ (4 V for the LM3886) to the peak output swing, $V_{o(peak)}$, to get the supply rail value (i.e., $\pm[V_{o(peak)} + V_{o(d)}]$ at a current of $I_{o(peak)}$). The regulation of the supply determines the unloaded voltage, usually about 15% higher. Supply voltage will also rise 10% during high line conditions. Therefore, the maximum supply voltage, $V_{cc(max)}$ is

$$\pm(V_{o(peak)} + V_{od})(1 + \text{regulation})(1.1) \quad \text{[Eq. 7]}$$

The input sensitivity and the output power specs determine the minimum required amplification, A_v:

$$A_v \geq (\sqrt{P_oR_L})/V_{in} = V_{o(rms)}/V_{in(rms)}. \quad \text{[Eq. 8]}$$

Normally, the amplification is set between 20 and 200; for a 40 W, 8 Ω audio amplifier, this results in a sensitivity of 894 mV and 89 mV respectively. Although higher gain amplifiers provide greater output power and dynamic headroom capabilities, there are certain shortcomings that go along with the so-called 'gain'. The input referred noise floor is increased and hence the signal-to-noise ratio is worse. With the increase in amplification, there is also a reduction of the power bandwidth which results in a decrease in feedback, thus not allowing the amplifier to respond quickly enough to nonlinearities. This decreased ability to respond to nonlinearities increases the THD + N specification.

The desired input impedance is set by R_{in}. Very high values can cause board layout problems and d.c. offsets at the output. The value of the feedback resistance, R_{f1}, should be chosen to be a relatively large value (10–100 kΩ), and the other feedback resistance, R_i, is calculated with standard op amp configuration gain equations. Most audio amplifiers are designed from the non-inverting amplifier configuration.

40 W/4 Ω audio amplifier

Given:

Power output	40 W
Load impedance	4 Ω
Input level	1 V (max)
Input impedance	100 kΩ
Bandwidth:	20 Hz – 20 kHz ±0.25 dB

Equations [5] and [6] give

$$V_{o(peak)} = 17.9 \text{ V}$$

and

$$I_{o(peak)} = 4.5 \text{ A}.$$

Thus, the supply required is ±21.0 V at 4.5 A.

With 15% regulation and high line, the final supply voltage is ±26.6 V (from Eq. 7). At this point it is a good idea to check the power output vs supply voltage to ensure that the required output power is obtainable from the device while maintaining low THD + N. It is also advisable to check the power dissipation vs supply voltage to ensure that the device can handle the internal power dissipation. At the same time, designing in a practical, relatively sized heat sink with a low thermal resistance is also important (see section on heat sinks).

The minimum amplification from Eq. 8 is $A_v = 12.6$. Select an amplification of 13 (non-inverting amplifier), which results in a sensitivity of 973 mV.

Letting R_{in} equal 100 kΩ gives the required input impedance, but this would eliminate the 'volume control' unless an additional input impedance were placed in series with the wiper of R_{in} in Fig. 1. Adding the additional 100 kΩ resistor would ensure the minimum required input impedance.

For low d.c. offsets at the output, let R_{f1} = 100 kΩ. Solving for R_i (non-inverting amplifier) gives the following: $R_i = R_{f1}/(A_v - 1) = 100/(13 - 1) = 8.5$ kΩ (use 8.2 kΩ).

The bandwidth requirement must be stated as a pole, i.e., the 3 dB frequency. Five times away from a pole gives 0.17 dB down, which is better than the required 0.25 dB. Therefore:

$$f_L = 20/5 = 4 \text{ Hz};$$

$$f_H = 20 \times 10^3 \times 5 = 100 \text{ kHz}.$$

At this point, it is a good idea to ensure that the gain-bandwidth product (GBWP) for the part will provide the designed amplification out to the upper 3 dB point of 100 kHz. This is why the minimum gain-bandwidth of the LM3886 is important:

$$\text{GBWP} \geq A_v \times f_{3dB} = 13 \times 100 \text{ kHz} = 1.3 \text{ MHz}.$$

The GBWP for the LM3886 is 2.0 MHz (min).

Solving for the low-frequency roll-off capacitor C_i:

$$C_i \geq 1/2\pi R_i f_L = 4.85\ \mu\text{F (use 4.7 }\mu\text{F)}.$$

Reference: National Semiconductor Lit#108048-001.

RECHARGEABLE ALKALINE BATTERIES

By our Editorial Staff

The alkaline-manganese battery is an example of the type of battery that until some years ago was available only in non-rechargeable (primary) form, but has since become available in rechargeable (secondary) form. It must be said, however, that this is primarily so in the USA and Canada; in Europe they are still very scarce, but demand (and supply) is growing. Manufacturers that supply rechargeable alkaline manganese cells and batteries include Union Carbide (Eveready), Ray-O-Vac (both USA) and Pure Energy Corporation (Canada).

These batteries use a unique electrochemical system, are maintenance free, hermetically sealed, and will operate in any position. Their discharge characteristics (voltage decrease, internal resistance and self-discharge) are very similar to those of a primary alkaline-manganese battery. Their capacity is comparable to that of a nickel-metal-hydride (NiMH) battery, that is, somewhat higher than that of a NiCd battery, but rather lower than that of a primary alkaline-manganese battery. Like NiMH batteries (but in contrast to NiCd batteries), rechargeable alkaline-manganese batteries do not contain heavy metals. They are somewhat dearer than NiCd batteries, but cheaper than NiMH batteries.

Charging (pulsed charging at a constant voltage of 1.8 V) is rather different from that of nickel-based batteries. The charging time for AA/R6/HP7 size batteries is 16–18 hours: fast charging is not (yet) possible. On the other hand, the battery can be charged at any time, irrespective of the state of residual charge: discharging it beforehand is not necessary.

The charge retention properties of secondary alkaline manganese batteries are as good as those of primary batteries.

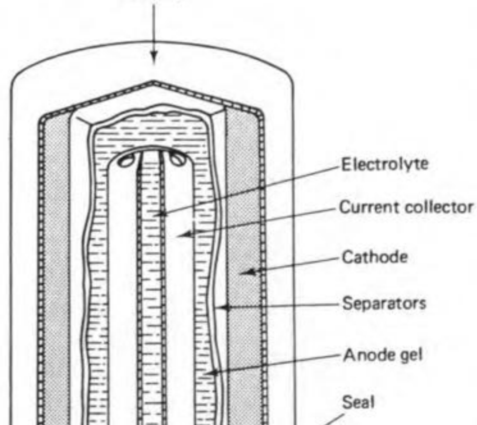


Fig. 1. Basic construction of a secondary alkaline manganese cell.

Technical data

The basic construction of an alkaline manganese cell is shown in **Fig. 1**. The cell uses electrodes of powdered zinc and manganese dioxide with an electrolyte of potassium hydroxide. These are put together as shown and the cell is then hermetically sealed. The voltage per cell is 1.5 V. Batteries of higher voltage are made by connecting the requisite number of (similar) cells in series and sealing them in a metal case.

When the cell is being discharged, the manganese dioxide gives off hydrogen, which reduces its mass, while the zinc reacts with the hydrogen to form zinc oxide. When the cell is being charged, the zinc oxide is reduced to zinc again which is made possible, among others, by the separators (made of non-woven fabric) that are much stronger than those in a primary cell.

When energy is withdrawn from the cell, the terminal voltage drops slowly. The total voltage drop for a given energy withdrawal increases as the number of discharge/charge cycles rises. Furthermore, the available energy per cell diminishes with each discharge/charge cycle although the e.m.f. remains constant. If the power demands exceed the rated battery capacity, the cell's cycle life will decrease more quickly. Nevertheless, when the cell is discharged at the maximum rate for a period of time and then recharged as recommended by the manufacturers, the discharge/charge cycle can be repeated many times before the cell e.m.f. will drop below 0.9 V.

Cycling

In a practical test, four sample batteries were discharged with the battery tester described elsewhere in this issue and then pulse-charged with a proprietary charger. Each of the four batteries was discharged through a 4.7 Ω resistor until its terminal voltage had dropped to 0.9 V. The batteries were then allowed to recover for two hours. Subsequently, they were charged for exactly 16 hours and 30 minutes, allowed to stabilize for two hours, and then discharged again as before. It was found that the time for the terminal cell voltage to fall to 0.9 V during discharge levelled out after 10 discharge/charge cycles—see **Fig. 2**.

It is seen that the batteries released most energy during the first discharge: at an average discharge current of 255 mA, the capacity of the four batteries varied between 1.120 Ah and 1.230 Ah, which is rather lower than obtainable from primary batteries (1.8 Ah). Moreover, this initial high capacity could not be recovered by charging, but kept dropping, until after 10 cycles it levelled out as stated earlier. However, the average capacity had dropped to 510–638 mAh after 12 discharge/charge cycles. Noteworthy is that the difference in capacities of the best and the worst battery was over 20%.

Compared with NiCd batteries, secondary alkaline manganese cells have the advantages of not containing heavy met-

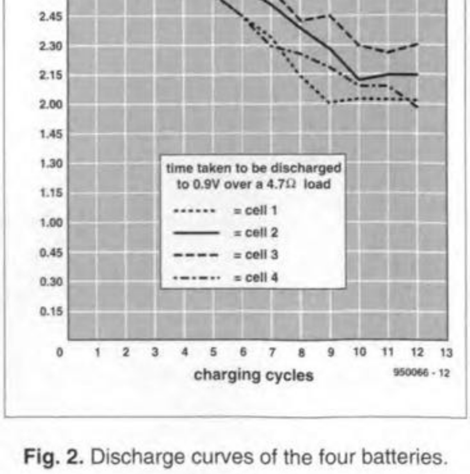


Fig. 2. Discharge curves of the four batteries.

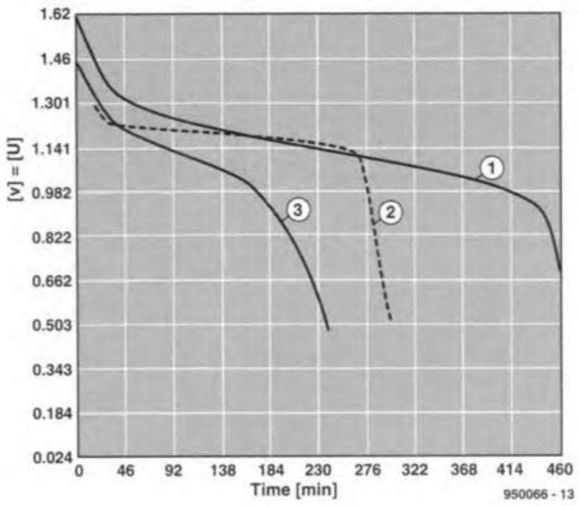


Fig. 3. Discharge curves of three HP7 batteries: (1) new secondary alkaline manganese; (2) NiCd; (3) well-used secondary alkaline manganese.

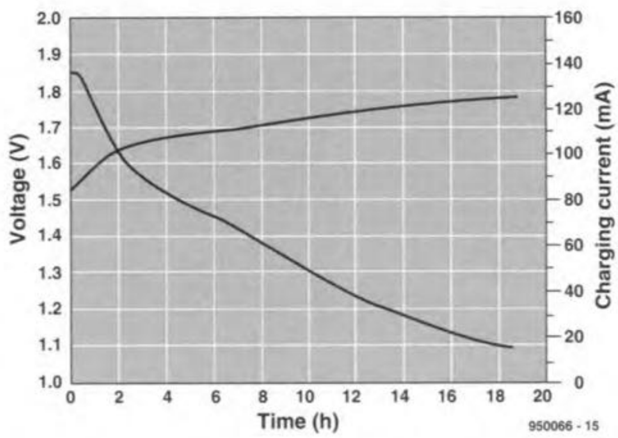


Fig. 4. Typical curves of the charging voltage and charging current measured during the charging of an HP7 secondary alkaline manganese battery.

als, unproblematic charging and very small self-discharge. However, their higher price is not compensated by higher capacity.

Compared with the relatively new and expensive NiMH batteries, secondary alkaline manganese cells have the advantages of better charge retention (because of their low self-discharge) and lower price. NiMH batteries have a higher capacity and shorter charging time.

The discharge curves of a new alkaline manganese battery (1), a 700 mAh NiCd battery (2) and a well-used alkaline manganese battery (3) are shown in **Fig. 3**. All of them were discharged via a 6.8 Ω resistor (discharge current about 150 mA). It is seen that even in the case of a new alkaline cell the terminal voltage is higher than that of a NiCd cell only during the first 30% or so of the discharge time (at average discharge currents).

It should, however, be borne in mind that in most small electronic equipment the discharge currents do not normally exceed 50 mA, so that the battery voltage drops only very gradually. This means that the average supply voltage to the equipment from alkaline manganese batteries is higher than that provided by NiCd batteries.

It should also be borne in mind that the curves in Fig. 2 can not really be taken as the basis for a final judgment as to the practicability of secondary alkaline manganese batteries since they were obtained from early samples charged with a pre-production charger. All that can be assumed at this stage is that the reduction in capacity with small discharge currents and intermittent discharging is not as severe as with average to high discharge currents.

Conclusion

The introduction of rechargeable alkaline manganese batteries is an interesting development. These batteries compare very favourably with NiCd and NiMH batteries as far as self-discharge is concerned. In fact, in this respect they are the best on the market as long as 1.5 V lithium-ion cells are not available in the usual battery sizes.

There remains a question mark over the practical usable capacity and the number of discharge/charge cycles of secondary alkaline batteries. This makes their price/power ratio high compared with other secondary batteries.

Furthermore, they can not be used where high currents are required; they can not be charged rapidly; and they are not proof against deep discharging. Avoiding deep discharging is a well-known problem with sealed lead-acid batteries, which leads to complicated (not user-friendly) usage and is likely to affect the battery life in the long term

As far as environment-friendliness is concerned, rechargeable alkaline batteries are much to be preferred over NiCd batteries, but, at present, there is not much to choose between them and NiMH batteries

Charging methods

The pulse-charging technique specially developed for secondary alkaline manganese batteries is protected by patent registration. However, for private use in home-built chargers no permission needs to be sought from the patentee.

In the interest of battery life and capacity, deep discharging of the battery (below 0.9 V per cell) should be avoided. It is also beneficial to leave the batteries at rest for a period of 2 hours before and after charging. Newly bought batteries MUST NOT be charged before they have been used.

Charging takes place with a 100 Hz pulsating direct voltage with a duty factor (pulse/pause ratio) of 5:3. Charging must be discontinued when the cell voltage has risen to 1.8 V. Charging of flat batteries, that is, discharged down to

- Nominal voltage 1.5 V
- Rechargeable over 100 times
- High capacity (HP7: 1 Ah)
- Can be recharged at any time
- Very low self-discharge
- New battery immediately usable
- Contains no mercury, cadmium,lead or nickel
- Discharge characteristics similar to those of primary batteries, so that:
- Internal resistance higher than that of NiCd batteries
- Not suitable for high discharge currents

Table 1. Concise characteristics of secondary alkaline manmganese batteries.

Nominal voltage: 1.5 V
Voltage at end of charging:
1.75 V typical; 1.8 V maximum
Continuous discharge current:
0.3–0.5 A
Pulsed discharge current:
0.5–1.0 A maximum
Operating temperature range:
–30 °C to +60 °C
Voltage at end of discharging:
1.0 V typical; 0.9 V minimum
Self-discharge loss (at 21 °C):
4% annually (typical)
0.2% per month (typical)

Table 2. Brief technical data of secondary alkaline manganese batteries.

0.9 V, should take 18–20 hours.

When the peak value of the charging voltage pulses is limited to 1.8 V, series resistors with values as shown for the various size batteries are recommended as follows:

HP16, AAA,UM4 (Micro): 1.5 Ω;
HP7, AA, UM3 (Mignon): 1.0 Ω;
HP11, C, UM2 (Baby): 0.5 Ω;
HP2, D, UM1 (Mono): 0.3 Ω.

If the charging voltage is switched via transistors, their ON resistance must be deducted from these values.

When charging is carried out with a pulsed direct voltage of 1.8 V, the r.m.s. value of the charging current through the series resistors or current sources must not exceed

HP16, AAA,UM4 (Micro): 60 mA;
HP7, AA, UM3 (Mignon): 120 mA;
HP11, C, UM2 (Baby): 200 mA;
HP2, D, UM1 (Mono): 300 mA.

For back-up applications, charging may be carried out with a non-pulsed direct voltage of 1.8 V. Series resistors must then be used to keep the charging current within the limits shown for the various battery sizes.

[950066]

RODENT DETERRENT

Though it may sound hard to believe, one day you may be unable to start your car in the morning not because of a flat battery or engine trouble, but because the ignition cables have been chewn away. Certain rodents, in particular, stone martens, appear to be very fond of all kinds of plastic and rubber objects which they happily destroy without actually eating these materials. The circuit discussed here keeps these beautiful, nimble animals at a safe distance from your car.

Design by G. Geissler

ALTHOUGH the problem of martens damaging car electrical systems is by no means massive, the German Automobile Association, ADAC, claims to have gathered some 350 damage reports from its members in one year. Following a publication on this subject, a research programme was launched to find ways of preventing damage to cars by stone martens.

Fortunately, it is possible to keep stone martens at bay with relatively simple means. The electronic deterrent discussed here is sure to be successful in this respect. By virtue of its design, the deterrent can be adjusted to scare other rodents, pets and vermin as well.

Towards the cities

Stone martens are small predators which until recently stayed away from towns and cities. These days, however, they appear to change their habitat from woods and open fields to the city. In some European countries, particularly Switzerland, Austria and the South of Germany, stone martens can be a real nuisance to motorists, being the cause of huge garage bills.

Research has shown that these little animals are initially attracted by the heat of the engine under the bonnet, where a comfortable sleeping place is found, particularly during the winter season. Next, the question arises why stone martens put their (very sharp) teeth in soft plastic parts, rubber cables, sleeves and covers. To find the answer to this question, researchers analysed the animal's behaviour in great detail. The idea that the marten uses the plastic and rubber as nutriments was soon dispelled because all the tiny bits of cable insulation, sleeving, etc., which had been chewn away could be found on the ground, under the engine, and pieced together to give exactly what was missing. So, the marten does not actually eat these materials. In fact, its chewing habit is merely playful, and in this respect the marten behaves just like a dog or a cat.

What to do?

Although the stone marten does not form a real problem in most European countries, including the UK, those of you who intend to go on a motoring holiday to Austria, Switzerland or Southern Germany are well advised to take the risk posed by this nimble, playful animal into account.

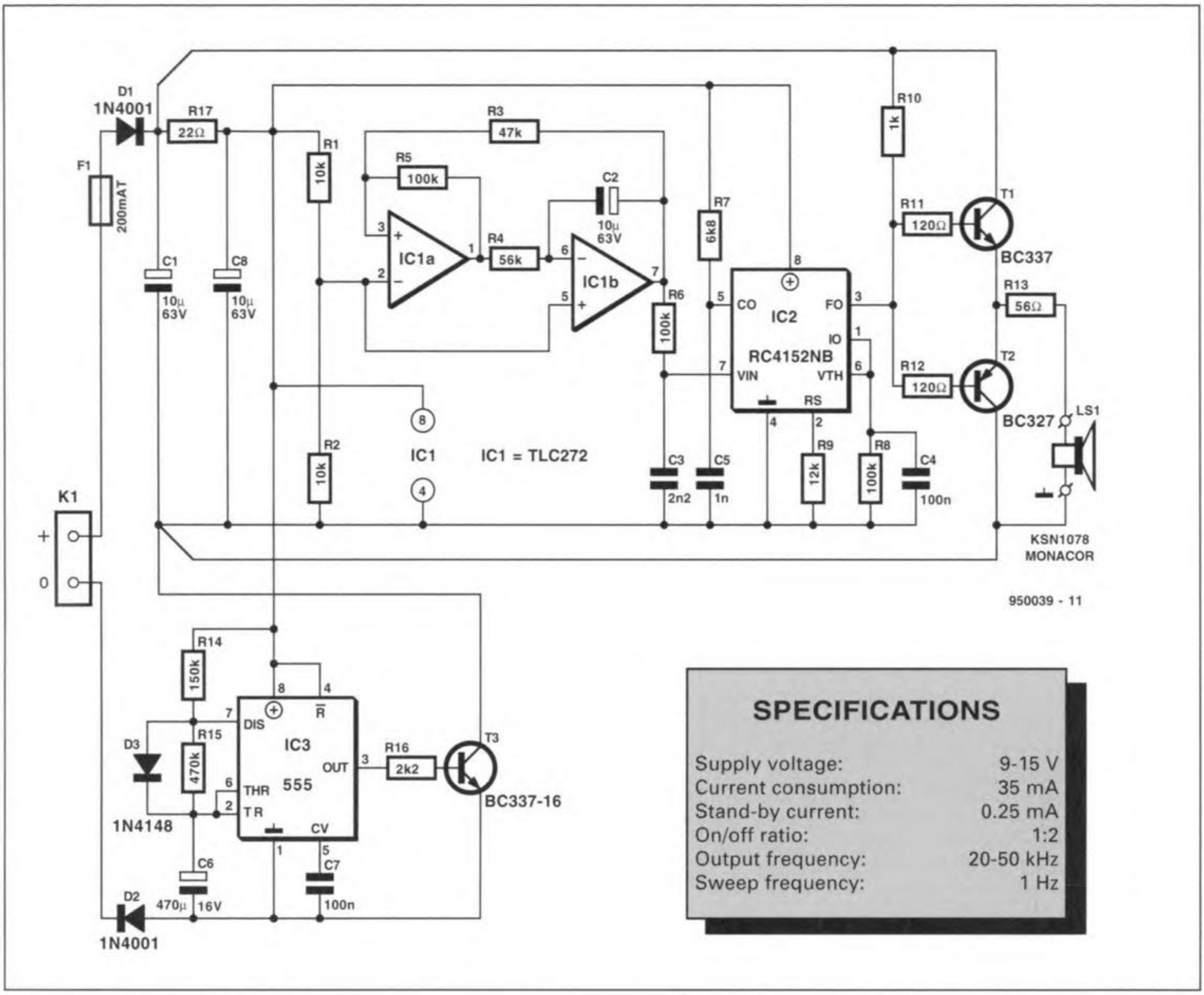
The ADAC has tested a number of marten deterrent systems on the market. Their conclusion was that the best protection is afforded by a system which produces a lot of ultrasonic noise. The best marten deterrent was a circuit which produced a signal with a frequency between 17 kHz and 19.5 kHz with a sound pressure of between 90 dB and 110 dB at a distance of about 30 cm. Although such a signal causes some discomfort to human beings, too, it is generally only noticed by young people who are close enough to the vehicle.

Further evidence is available which indicates that the above system also deters mice, rats, cockroaches, ants and fleas. Dogs and cats too, run off when confronted with such an amount of ultrasonic sound.

Circuit description

The circuit diagram of the rodent deterrent is shown in **Fig. 1**. The design is such that the application is not limited to scaring stone martens alone.

A square/triangle wave oscillator is constructed around opamps IC_{1a} and IC_{1b}. The triangular waveform is available on pin 7 of IC_{1b}. This signal has a frequency of about 1 Hz, and a level which varies between 3 V and 7 V. It is used to drive a voltage-to-frequency converter (VCO) built around an RC4152 (from Raytheon) or its pin-compatible equivalent, the LM331 (from National Semiconductor). These ICs were chosen because they only require a handful of passive parts to implement the VCO. The integrated VCO is capable of generating square-wave signals with a frequency between 1 Hz and 100 kHz, which is more than adequate for the present application.The drive signal for IC_2 is applied to pin 7



SPECIFICATIONS

Supply voltage:	9-15 V
Current consumption:	35 mA
Stand-by current:	0.25 mA
On/off ratio:	1:2
Output frequency:	20-50 kHz
Sweep frequency:	1 Hz

Fig. 1. Circuit diagram of the rodent deterrent. Here, the application is geared to scaring stone martens from the engine compartment of cars parked in the street overnight.

via an *RC* network consisting of R_6 and C_3. The frequency-determining components in the VCO are R_7, R_8, R_9 and C_4. The output signal frequency, f_o, is calculated from

$$f_o = \frac{U_{in}}{2.09\text{V}} \times \frac{R_9}{R_8} \times \frac{1}{R_7 C_5}$$

Entering the drive voltage level and the component values used in the present circuit into this equation, you get a frequency range of 22 kHz to 50 KHz. In practice, most of the above mentioned animals and vermin respond favourably (for us, that is) to this signal.

The VCO output signal is buffered by a simple transistor amplifier built around T_1 and T_2, which drive a high-impedance ceramic tweeter. The prototype was tested with a type KSN1078 tweeter from Monacor (Monarch), which has a frequency range of 5 kHz to 20 kHz, and an efficiency of 98 dB (2.83 V/1 m). Thanks to the high impedance of this tweeter, the 'power driver' can be built with two small signal transistors from the BC series. A 56-Ω series resistor, R_{13}, is added to make the output driver short-circuit resistant.

Because the circuit is to be fitted in the engine compartment, it is convenient to power it from the car battery. Low energy consumption is, therefore, a major design consideration. To keep the load on the battery as small as possible, the deterrent automatically switches itself on and off. The 'on' period lasts 1 minute, the 'off' period, two minutes. This pulse/pause ratio is ensured by IC_3, a 555 timer, which is wired as an astable multivibrator. Unlike the rest of the circuit, it is permanently connected to the car battery. As long as the output of IC_3 is high, transistor T_3 conducts. Consequently, IC_1, IC_2 and the output driver are powered via the collector-emitter junction of T_3. During the 'off' period, T_3 is off, and effectively breaks the ground connection of the rest of the circuit. The rodent deterrent is then switched off for about 2 minutes, until T_3 conducts again.

A few components are included in the circuit for electrical protection. F_1 is a 200-mA fuse which protects the car battery in the event of a short-circuit in the deterrent. Diodes D_1 and D_2 protect the circuit against polarity reversal of the battery voltage. Finally, C_1, R_{17} and C_8 suppress noise and voltage surges on the battery voltage.

Construction

The design of a printed circuit board for the rodent deterrent is given in **Fig. 2**. Unfortunately, this board is not

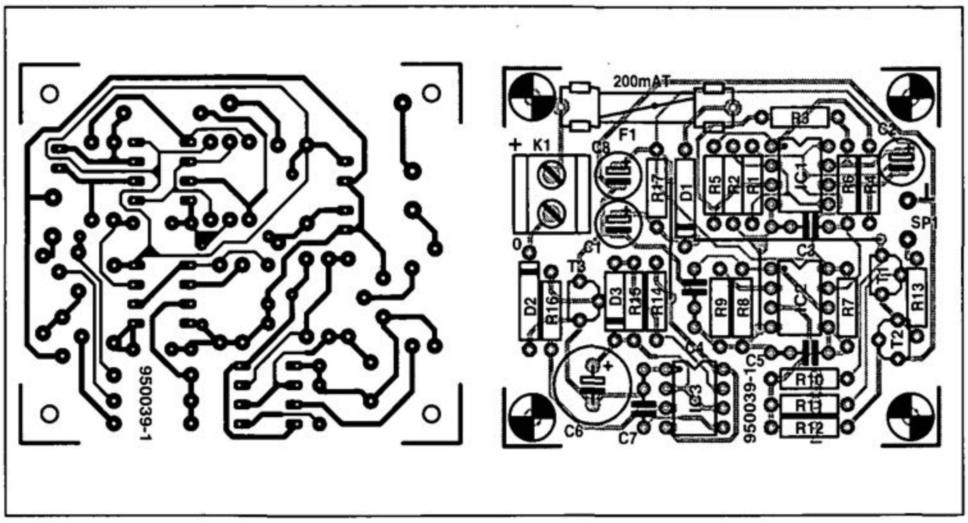


Fig. 2. Copper track layout and component mounting plan of the printed circuit board designed for the rodent deterrent (PCB not available ready-made through the Readers Services).

available ready-made through the Readers Services, so you have to make it yourself. The circuit is housed in a compact enclosure with a size of about 80×80×55 mm, which should fit in the engine compartment of almost any car.

Start the construction by fitting the wire links on the board. Next, mount the passive parts and the PCB terminal block. The last parts to be fitted are the integrated circuits. IC sockets are not strictly necessary.

Take the measurements of the tweeter you intend to use, and cut a clearance in the case to secure it. Drill a hole for the rubber grommet which passes the wiring. Next, mount the completed PCB into the case, and connect it to the loudspeaker wires. To prevent moisture entering the case, the supply wires should pass through a grommet. The cover of the case, too, should be securely fitted and made water-resistant by inserting a rubber gasket.

The circuit should be connected to the part of the electrical system which always carries the battery voltage (unswitched 12 V). Suitable connection points may be found on the cigarette lighter or the connection block for the car radio.

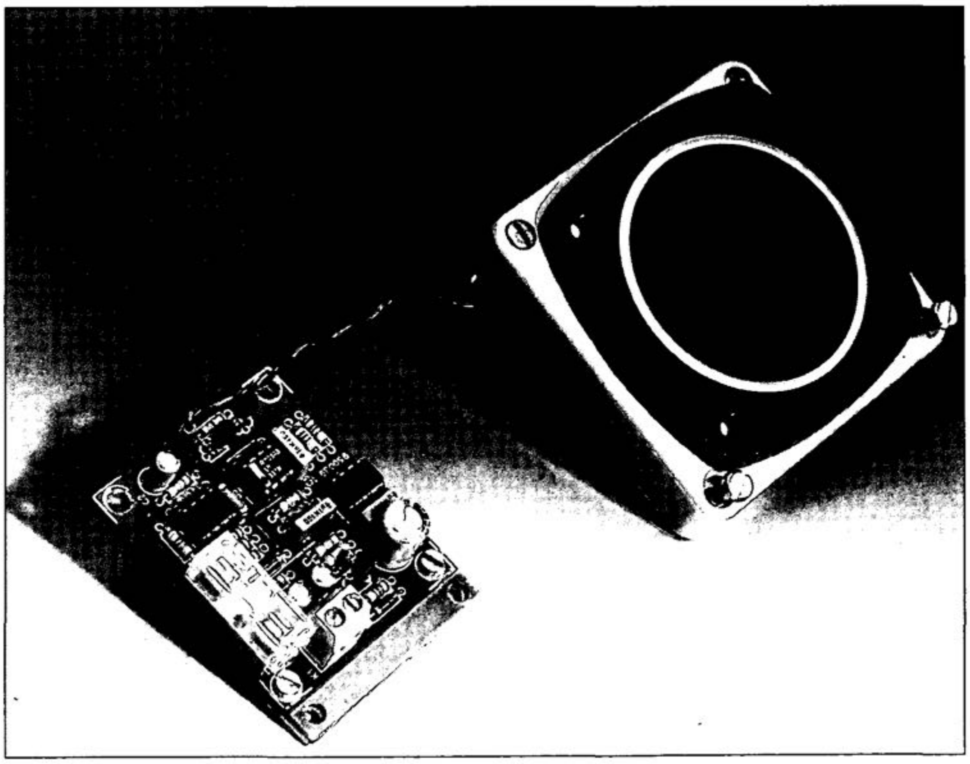


Fig. 3. Finished prototype of the circuit. It is essential that the enclosure is waterproof.

COMPONENTS LIST

Resistors:
R_1,R_2 = 10kΩ
R_3 = 47kΩ
R_4 = 56kΩ
R_5,R_6,R_8 = 100kΩ
R_7 = 6kΩ8
R_9 = 12kΩ
R_{10} = 1kΩ
R_{11},R_{12} = 120Ω
R_{13} = 56Ω
R_{14} = 150kΩ
R_{15} = 470kΩ
R_{16} = 2kΩ2
R_{17} = 22Ω

Capacitors:
C_1,C_2,C_8 = 10μF 63V
C_3 = 2nF2
C_4,C_7 = 100nF
C_5 = 1nF
C_6 = 470μF 16V

Semiconductors:
D_1,D_2 = 1N4001
D_3 = 1N4148
T_1 = BC337
T_2 = BC327
T_3 = BC337-16
IC_1 = TLC272
IC_2 = LM331 or RC4152NB
IC_3 = 555

Miscellaneous:
K_1 = 2-way PCB terminal block, pitch 5 mm.
SP_1 = ceramic tweeter (e.g. Monacor* KSN1078).
F_1 = fuse 0.2 AT, with PCB mount holder.
1 case, dim. 82 x 80x55 mm (e.g., Bopla** Euromas T210).

* Monacor Netherlands, tel. (+31) 80 585555, fax (+31) 80 584790.
** Phoenix Mecano Ltd. tel. (01296) 398855.

Other applications

As already mentioned, the circuit is primarily designed to keep stone martens out of the engine compartment. In practice, the circuit is also suitable for other animals. Owners of a sandpit will be pleased to know that the present deterrent is a perfect way to prevent cats and dogs from fouling the sand. Mice, rats, ants, cockroaches and fleas also keep out of areas guarded by the present circuit, which goes to show that there are environmentally safe alternatives to chemical and/or bloody warfare on these animals. (950039)