ELECTRONICS WORLD

+ WIRELESS WORLD

Denmark DKr. 70.00
Germany DM 15.00
Greece Dra.760
Holland DFl. 14
Italy L. 7300
IR £3.30
Spain Pts. 780
Singapore S$ 12.60
USA $6.70

SOR DISTRIBUTION

A REED BUSINESS PUBLICATION

FEBRUARY 1994 £1.95

BASS SPECIAL

Bass technology:
Bass deception?
The bass race

Designer bass:
Big bass–small box
Coupled cavities

RF DESIGN
Using broadband RF amplifier blocks

BROADCAST
DAB: sound technology for the next century?

APPLICATIONS
Single chip GPS, inductorless voltage conversion

0 2>
9 770959 833004

20% OFF: Audio and HiFi Handbook
See inside for details

CONTENTS

FEATURES

REGULARS

In next month's issue: Working with realtime computing. There is – or should be – a world of difference between the machines and software which handle business applications and the sort which control a production line or a scientific experiment. OS-9 starts here. Also in this issue: Using transconductance amps.

MARCH ISSUE ON SALE FEBRUARY 24.

Unsound ideas at the European Commission

Nobody doubts for a moment the ingenuity and technical excellence of the BBC's digital audio broadcast system – DAB. Recent demonstrations both here and overseas show it to offer real improvement on standard FM for mobile reception. Although it could be argued that the London venue demonstrations were flawed by the relative siting of the FM and DAB transmitters sending out the test material – the DAB sites were local while the FM site was 20 miles away – it eliminates multipath distortion and co-channel interference. DAB provides an excellent transmission medium for mobile reception. Why then should we think very carefully before committing ourselves to a new sound broadcast receiving system?

It is simply this. The technology, while feasible, takes little account of broadcasting requirements. It has been tailored far too closely to the requirements of a national network with little account of local broadcasting.

The precise details of the technology are dealt with elsewhere in this issue but in essence, each DAB transmitter broadcasts six programmes simultaneously using subcarrier interleave. The frequency spreading reduces the individual data rate/Hz to the point where multipath is no longer a problem but it requires that six programmes are transmitted simultaneously from a single site. Where used for local broadcasting, it would mean that six stations are locked together into an inflexible bundle. Six franchises would have to be offered to serve a local area since DAB only represents efficient use of frequency and financial resource when fully occupied. While this arrangement clearly suits Radios 1 to 5 plus another, it leaves local radio out in the cold.

The European Commission feels compelled to push Eureka DAB to gain advantage over emerging US technology in setting world standards. American digital sound broadcast technology takes as its starting point the elimination of transmission shortcomings from individual radio stations. And most broadcast systems around the world operate like the Americans'. I don't argue with assistance to our home grown DAB on philosophical grounds: let's gain any trading advantage we can. However, the world will surely turn its back on anything which isn't absolutely in tune with a market requirement.

Perhaps I should remind the EU of the MAC TV standards fiasco. And also those at the BBC/ITC with similarly short memories.

No audience or broadcasting organisation can afford to support multiple standards on the airwaves, particularly when the actual cost of hardware for the alternative will be quite high in comparison to the equipment which it is designed to replace. I urge the EU and our broadcast R&D departments to look again at the whole problem, not just selective parts of it. A total solution will receive a wider audience than a simply pragmatic one.

Frank Ogden

EDITOR
Frank Ogden
081-652 3128

DEPUTY EDITOR
Martin Eccles
081-652 8638

CONSULTANT
Derek Rowe

DESIGN & PRODUCTION
Alan Kerr

EDITORIAL ADMINISTRATION
Lorraine Spindler
081-652 3614

ADVERTISEMENT MANAGER
Carol Nobbs
081-652 8327

SALES EXECUTIVE
Pat Bunce
081-652 8339

ADVERTISING PRODUCTION
Paul Burgess
081-652 8355

PUBLISHER
Susan Downey

EDITORIAL FACSIMILE
081-652 8956

CLASSIFIED FACSIMILE
081-652 8931

SUBSCRIPTION HOTLINE
0622721666
Quote ref INJ

SUBSCRIPTION QUERIES
0444 445566

NEWSTRADE DISTRIBUTION ENQUIRIES
Martin Parr
081 652 8171

BACK ISSUES
Available at £2.50
081-652 3614

ISSN 0959-8332

REED BUSINESS PUBLISHING

Electronics World + Wireless World is published monthly. By post, current issue £2.25, back issues (if available) £2.50. Orders, payments and general correspondence to **L333, Electronics World + Wireless World, Quadrant House, The Quadrant, Sutton, Surrey SM2 5AS.** Telex:892984 REED BP G. Cheques should be made payable to Reed Business Publishing Group.
Newstrade: IPC Marketforce, 071 261-5108
Subscriptions: Quadrant Subscription Services, Oakfield House, Perrymount Road, Haywards Heath, Sussex RH16 3DH. Telephone 0444 445566. Please notify a change of address. Subscription rates 1 year (normal rate) £30 UK and £35 outside UK surface mail. Air mail rates are found on card bound into issue.
USA: $78.00 airmail Reed Business Publishing (USA). Subscriptions office, 205 E. 42nd Street, NY 10117.
Overseas advertising agents: France and Belgium: Pierre Mussard, 18-20 Place de la Madeleine,Paris 75008. United States of America: Ray Barnes, Reed Business Publishing Ltd, 205 E. 42nd Street, NY 10117.Telephone (212) 867-2080. Telex 23827.
USA mailing agents: Mercury Airfreight International Ltd Inc, 10(b) Englehard Ave, Avenel NJ 07001. 2nd class postage paid at Rahway NJ Postmaster. Send address changes to above.
Printed by BPCC Magazines (Carlisle) Ltd, Newtown Trading Estate, Carlisle, Cumbria, CA2 7NR
Typeset by Marlin Graphics 2-4 Powerscroft Road, Sidcup, Kent DA14 5DT

Ultra fast SiGe transistors go commercial

The world's first highly integrated silicon-germanium semiconductors will be on the market within nine months following an agreement between Analog Devices and IBM. Analog Devices plans to design a range of ICs for radio frequency and mixed signal applications, which will be made by IBM in New York.

The new production process, which uses ultra high vacuum chemical vapour deposition to create a Ge on Si structure, allows the building of transistors with an f_T of more than 60GHz without extra processing steps. SiGe devices can operate from 3 and 1.5V supplies.

Silicon-germanium semiconductors are seen as the natural successor to cmos for high speed circuits. A 1GHz 12-bit digital to analogue converter will be the first IC.

However, Analog Devices and IBM are likely to concentrate on developing single IC front end chips for wireless communications.

This is a significant blow to the fortunes of gallium-arsenide ICs – previously the only alternative semiconductor process for very high speed circuits. SiGe circuits are significantly cheaper to produce and can be integrated onto an ordinary cmos process.

Richard Siber, director for the wireless communications service at market researcher BIS Strategic Decisions, said: "This process will negate the need for more expensive GaAs for operating frequencies up to 3GHz. This development is truly revolutionary, giving the wireless industry a major boost."

See also Research Notes, p97

NTSC to VGA conversion: ITT's digital TV components perform a TV to computer display conversion for multimedia. The video adaptor uses an NTSC comb filter chip, the DPU2554 deflection processor, progressive scan processor PSP2210 and video codec VCU2134.
Simple multimedia applications do not require data reduction. The picture data is first converted to the square pixel format of computers. These are then interpolated to the required number of pixels for display on the computer screen using a picture format processor chip DTI2250.

Jessi goes commercial

Jessi, the European microelectronics research and development programme, is to produce a series of GSM chips for pocket telephones working on the European digital cellular network. These are an ATM chipset for advanced data communications equipment, a chipset for receiving digital audio broadcasts (DAB), a chipset providing the electronics for digital TVs, and an automotive safety chipset.

If used commercially, the chips could give European systems manufacturers an edge in the market for end equipment. They represent a shift away from the previous objective of Jessi which was to push forward technological capability over various disciplines without the specific intention of producing useful, commercially competitive products.

The ATM project is the most advanced. Project leader, Alcatel, already has the four-chip set in silicon on a 0.8μm process and aims to put them onto the Jessi developed 0.5μm process by 1995.

The other three projects are less advanced. A two-chip GSM chipset has been designed by Alcatel, Bosch, SGS-Thomson and Mietec but is some way from implementation in silicon.The chipset follows the industry pattern of one chip for the radio frequency side and another for the digital signal processing side. The digital TV, DAB and car safety chipsets are still being designed

Bulletin boards face copyright battle

A music publisher in the US is suing CompuServe for copyright infringement on one of its bulletin boards. The case, to be heard in the Federal Court of New York, looks likely to set far-reaching precedents. By suing CompuServe, as provider of the host computer on which the music material is temporarily stored rather than individual system users, the music publishers are creating a precedent for the future. All electronic delivery services could become legally responsible for whatever messages subscribers post through them.

The Harry Fox Agency, part of the National Music Publishers Association, is paying for music publisher Frank Music to pursue the test case on behalf of over 140 other publishers. CompuServe describes itself as "the world's most comprehensive computerised information service". The claim is for $70 million in damages and costs for copyright infringement in just one song. If the publisher wins, the HFA, which represents 12,000 publishers and controls the licensing of 75% of all the music played in the US, will claim from any other electronic distribution system which carries music.

Over a million PCs around the world connect to the CompuServe network to exchange electronic mail messages and access 1700 different bulletin boards. Most of the messages are text, for instance news and views on new technology. But one board, called the MIDI/Music Forum, lets subscribers exchange music.

Because midi code is similar to ascii, subscribers to CompuServe have been converting music into digits and uploading it into the CompuServe computer so that other subscribers can download the code to make a PC or electronic instrument play the tune.

By logging use of the musical bulletin board, the HFA has been able to cite what it describes as 690 "wilful acts of infringement", involving more than 500 songs owned by some of the 12,000 music publishers it represents. **BF**

Co-op venture produces 64Mbit ram

Texas Instruments and Hitachi have unveiled the 64Mbit dram they have been developing for the past two years. The $228mm^2$ chip can store four copies of *Gone with the wind* – some 2000 printed pages.

Dr Tsugio Makimoto from Hitachi said: "We pooled the best of the technologies of each side to realise benefits in terms of technology and development efficiency that greatly exceeded what either would have been able to achieve independently."

He reckons the success of the project "provides a platform for substantially strengthening and expanding our co-operative efforts in this area".

The firms are considering a similar project for the 256Mbit dram. Toshiba, IBM and Siemens have already started joint 256Mbit development.

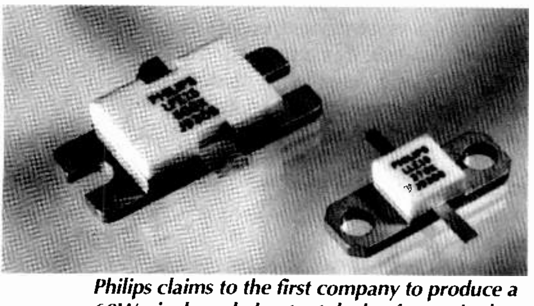
A change from TI's previous drams is that the chip uses stacked rather than trench capacitor techniques.

Beeb in more HDTV tests

The BBC and Thomson-CSF Laboratories recently carried out a second series of HDTV transmissions from Crystal Palace to test reception quality in a wide area around London. Transmitter power was 10kW and channel 34 was used with an 8MHz bandwidth, the same as an ordinary Pal signal.

Bit rate was 30Mbit/s and the compression system used a motion compensated hybrid discrete cosine transform similar but not identical to the MPEG-2 format.

There were 500 carriers using an orthogonal frequency division multiplex and 64-state QAM modulation.



Philips claims to the first company to produce a 60W, single ended output device for use in the band 1.5 to 1.7GHz. The LFE15600X npn transistor achieves a typical power gain of 8.5dB at 1.5GHz and has been designed for class AB linear service: Intermodulation products are 30dB down at 60W pep output.

Cheap packaging threatens CDs

The latest scare story about disintegrating CDs appearing in the UK press may have been hyperbolic in its predictions of millions of CDs doomed to disintegrate but the facts behind the story have substance. The cause is not in the CD standard or manufacturing process, but sulphur content of the packaging.

Although the CD standards set by Philips specified sleeve size they did not specify materials. Analysis of the sleeves supplied by the record companies showed that some were made from high quality processed paper called solid sulphite board, and did not affect any discs, other sleeves were made from untreated wood pulp. This releases sulphur compounds which eat through the protective lacquer on the label surface of the CD to erode the metal reflective layer underneath.

The first proof that CDs would not last forever came in 1988. Pressing plants discovered that the inks they were using to print label information direct onto the disc were eating through the protective lacquer coating and destroying the very thin layer of aluminium which reflects the laser light. So the discs stopped playing. Before that, plants pressing 30cm laser video discs had found that the glues used to stick the two halves of the double-sided discs together were eating away the reflective layer. The factories solved these problems by changing their inks and glues.

Now Philips' plant in Blackburn has found that some CD singles pressed in the late eighties for record companies Polygram and A and M are failing. Investigations have identified the cause as release of sulphur from some of the cardboard sleeves used to pack CD singles. But the concerned organisations do not agree on the basic issues which might place CDs made by other plants at risk.

Philips built the Blackburn PdO plant to press 30cm video discs, but demand was small so it began pressing 12cm CDs. It uses a wet process, similar to that used for making mirrors, to deposit a layer of silver as the reflective surface. Most CD pressing plants sputter aluminium but the necessary equipment was not available when the Blackburn factory was built.

In 1988 the record companies started to sell CD singles, full size 12cm discs carrying only around 20 minutes of music. They cut costs by using cardboard sleeves instead of plastic jewel boxes. In late 1989 the plant found that some discs in board sleeves were refusing to play after a few months use, while others played perfectly. To confuse the issue, some discs would play on some players but not others, and then later fail on all players.

Cheap CD players immediately refused to play the disc while more expensive players with better error correction circuitry played the disc perfectly, but failed after several more months when the holes had grown larger.

The two types of sleeve look the same. "We had a real struggle to distinguish between them, but finally developed a simple test" says Dave Wilson, PdO's Technical Services Director. PdO found that if a drop of methylated spirits is put on the board surface, the material turns clear to reveal a pulp of free fibres if the board is untreated. PdO then worked with Philips to set a standard for CD sleeve chemistry. Although some record companies had started to use plastic jewel boxes for singles, because of the perceived low value of card, the card sleeve is still used in some countries, including France. Record companies in the US use card sleeves for full length CDs.

Polygram believes that the problem was confined to the Blackburn factory because the silver is more susceptible to sulphur than aluminium; the effect is similar to silver cutlery tarnishing. The disc gradually turns from silver to bronze and loses its reflectivity. But Dave Wilson of PdO believes that aluminium will degrade in exactly the same way if the record companies supply any plant with card which has a high sulphur content.

PdO now checks all card for sulphur content. PdO also checks the paper inlay notes which sit inside a jewel box and press against the disc lacquer. Dave Wilson says that all PdO technical information on the sulphur risk was made available to other plants. But the record divisions of EMI, A & M and Polygram appeared unaware of the need to use high quality paper and board for the CD sleeves and inlays which they supply to pressing plants. EMI says it is confident that the problem is confined to discs pressed by PdO. But a spokeswoman for EMI's own CD pressing plant in Swindon was unaware of the need to check paper and card for sulphur content.

It now seem only a question of time before someone, somewhere, reports full length CDs rotting because they have either been packaged in contaminated card boxes or packed with contaminated paper sleeve notes. **Barry Fox**

Major companies in MPEG 2 scramble

Anticipating profit in digital TV, AT&T Microelectronics, LSI Logic and SGS-Thomson Microelectronics have all announced details of MPEG-2 compliant real-time video decoders. The announcements presage the arrival in 1994 of digital TV receivers for satellite broadcasts and cable TV set-top decoders.

AT&T demonstrated its chip, the *AV6101*, late last year at the Western Cable Show in the US. The decoder chip was paired with a real-time encoder system developed by AT&T Transmission Systems.

Anne Schowe, managing director of AT&T's visual solutions business unit, said: "We are claiming victory in the scramble to deliver MPEG-2 video decoder chips to the digital video market. Our *AV6101* chip decodes all-the MPEG-2 video layers in real time without requiring external processor support, which means it's an ideal solution for inexpensive set-top decoder systems."

The IC is expected to cost less than $100 in volume.

But AT&T's claim is disputed by SGS-Thomson: "We would question the AT&T claim that its chip is the first real-time MPEG-2 decoder chip to be demonstrated. We have been privately showing ours, the *STi3500*, for a while," said Simon Loe the firm's technical spokesman.

The AT&T chip provides a 4:2:2 raster output and needs 8Mbit of external memory as a frame store. It uses a 27MHz clock and consumes 1.3W.

The chip's i/o operates from a 5V supply whereas the core uses a 3.3V supply. Initial samples are being provided to a few customers but production is not scheduled until later this year.

AT&T says the *AV6101* device is a first generation chip capable of decoding CCIR601 broadcast quality resolution pictures conforming to the main level simple profile format in the MPEG-2 specification. This format excludes bidirectionally predicted, or B frames, in the picture sequence. They are computation intensive to encode and decode.

Main level, main profile format pictures need a minimum of 16Mbit for frame storage.

A second generation chip is planned that can decode a main level, main profile data stream including B frames. However, AT&T believes in the short term system builders will opt for the cheaper main level, simple profile format.

In contrast the *STi3500* chip can accommodate main level, main profile and main level, simple profile picture formats by virtue of a programmable frame store. The chip can directly support between 8 and 32Mbit of dram. An external microcontroller is needed.

Martin Bolton, technical marketing manager for the image compression group, says the external microcontroller is not a disadvantage: "There will usually be a microcontroller somewhere in the system and the load required to control the *STi3500* is small. It is only a matter of setting registers every picture."

Also allowing the decoder to be controlled at the frame and field level externally allows a greater flexibility for handling multiple standards and, importantly, special modes, said Bolton.

The *STi3500* incorporates all the decoder functions, inverse discrete cosine transform, inverse quantise, frame prediction constructor and variable length decoder. The chip is highly pipelined to attain the necessary performance. The output digital video signal is in a YCrCb format multiplexed onto an 8bit bus. The synchronisation signal has to be supplied externally.

The LSI Logic *L64000* video decoder will also work with a main level, main profile picture format and has been developed with Zenith Electronics – one of the companies involved in the US grand alliance to develop an HDTV standard.

Simon Parry, *Electronics Weekly*.

War opens on electronic counterfeit

National Westminster Bank's says it has "high confidence" in the security of its new electronic alternative to cash, the Mondex card. NatWest has been working on the smart card cash system for nearly four years, hopes it will become a global standard. It plans to introduce the system in 1995.

In its basic form Mondex relies on a conventional smart card, made to ISO standard 7816, with inbuilt memory chip and computer processor which store cash credits, and external contacts to connect with a card reader. Anyone can use the card to make a purchase from any shop which has a reader at the till. The user's only security is to use an electronic wallet to lock, and unlock, the card's memory with a personal identification number.

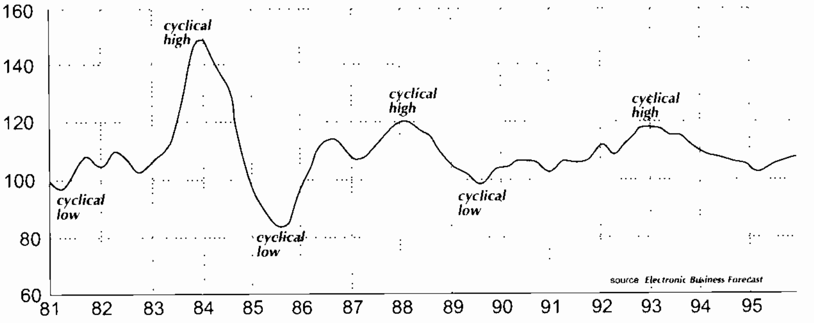
The conceptual breakthrough claimed by NatWest is in the method of proofing the card against counterfeiting, so that criminals cannot make copies of cards or tamper with the memory and credit transfer signals. Tim Jones, NatWest's Head of Information Technology Policy and Strategy, says Mondex is "extraordinarily secure". But court actions brought recently by satellite broadcaster BSkyB reveal that smart cards can be a lot less secure than those who rely on them previously thought.

Credits are loaded into the Mondex card memory or deducted to make payments by a reader at the point of sale. Both National Westminster and Midland Bank will provide hole-in-the-wall readers which let cardholders refresh cards by loading credits into the memory, while debiting their accounts. BT is already designing a domestic "smart" telephone which will let card-holders refresh them from home by calling their bank.

NatWest knows that Mondex is an open invitation to criminals who will try to print money by copying cards or pirating the signals which transfer cash from one memory to another. A spokesman for the bank "Yes we are definitely confident on security. We realise that Mondex will be targeted by criminals. There are many levels of protection against counterfeiting".

The system checks the integrity of the money signal passing from "purse" to "purse", or source and destination, to ensure that a card-holder does not tamper with the digits and so make a transfer of £10 register as £1000. Mondex also checks that each signal only registers once, to stop the same £10 transfer notching up five times to become £50. The system continually checks the validity of each purse, to ensure that the owner of one card cannot suck money from someone else's account. **BF**

Picture of the US electronics industry: normalised, composite graph from Electronic Business Forecast leading indicator demonstrating year on year change in the production, order books, share price, semiconductor bookings and interest rates. Contrary to common perception, the cyclic business swings appear to be smoothing out as industry learns to plan for the perturbations.



AMSTRAD DMP4000 Entire printer assemblies including printhead, platen, cables, stepper motors etc. Everything bar the electronics and case. Good stripper!! Clearance price just £5 REF: MAG5 or 2 for £8 REF: MAG8

VIEWDATA SYSTEMS Brandnew units made by TANDATA complete with 1200/75 built in modem, infra red remote controlled keyboard. BT approved, Prestel compatible, Centronics printer port, RGB colour and composite output (works with any TV) complete with power supply and fully cased. Price is just £20 REF: MAG20 Also some customer returned units available at £10 each REF: MAG10

PPC MODEM CARDS. These are high spec plug in cards made for the Amstrad laptop computers. 2400 baud dial up unit complete with leads. Clearance price is £5 REF: MAG5P1

INFRA RED REMOTE CONTROLLERS Originally made for hi spec satellite equipment but perfect for all sorts of remote control projects. Our clearance price is just £2 REF: MAG2

TOWERS INTERNATIONAL TRANSISTOR GUIDE. A very useful book for finding equivalent transistors, leadouts, specs etc. £20 REF: MAG20P1

SINCLAIR C5 MOTORS We have a few left without gearboxes. These are 12v DC 3,300 rpm 6"x4", 1/4" OP shaft. £25 REF: MAG25

UNIVERSAL SPEED CONTROLLER KIT Designed by us for the above motor but suitable for any 12v motor up to 30A. Complete with PCB etc. A heat sink may be required. £17.00 REF: MAG17

VIDEO SENDER UNIT. Transmits both audio and video signals from either a video camera, video recorder, TV or Computer etc to any standard TV set in a 100' range! (tune TV to a spare channel) 12v DC op. Price is £15 REF: MAG15 12v psu is £5 extra REF: MAG5P2

***FM CORDLESS MICROPHONE** Small hand held unit with a 500' range! 2 transmit power levels. Reqs PP3 9v battery. Tuneable to any FM receiver. Price is £15 REF: MAG15P1

LOW COST WALKIE TALKIES Pair of battery operated units with a range of about 200'. Ideal for garden use or as an educational toy. Price is £8 a pair REF: MAG 8P1 2 x PP3 req'd.

***MINATURE RADIO TRANSCEIVERS** A pair of walkie talkies with a range of up to 2 kilometres in open country. Units measure 22x52x155mm. Complete with cases and earpieces. 2xPP3 req'd. £30.00 pair REF: MAG30.

COMPOSITE VIDEO KIT. Converts composite video into separate H sync, V sync, and video. 12v DC operation. £8.00 REF: MAG8P2.

LQ3500 PRINTER ASSEMBLIES Made by Amstrad they are entire mechanical printer assemblies including printhead, stepper motors etc etc In fact everything bar the case and electronics, a good stripper! £5 REF: MAG5P3 or 2 for £8 REF: MAG8P3

PHILIPS LASER 2MW helium neon tube. Brand new full spec £40 REF: MAG40. Mains power supply kit £20 REF: MAG20P2 Fully built and tested unit £75 REF: MAG 75.

SPEAKER WIRE Brown two core, 100 foot hank £2 REF: MAG2P1

LED PACK of 100 standard red 5mm leds £5 REF: MAG5P4

JUG KETTLE ELEMENTS good general purpose heating element (about 2kw) ideal for allsorts of heating projects etc. 2 for £3 REF: MAG3

UNIVERSAL PC POWER SUPPLY complete with flyleads, switch, fan etc. Two types available 150w at £15 REF: MAG15P2 (23x23x23mm) and 200w at £20 REF: MAG20P3 (23x23x23mm)

OZONE FRIENDLY LATEX 250ml bottle of liquid rubber, sets in 2 hours. Ideal for mounting PCB's, fixing wires etc £2 each REF: MAG2P2

***FM TRANSMITTER** housed in a standard working 13A adapter!! the bug runs directly off the mains so lasts forever! why pay £700? or price is £26 REF: MAG26 Transmits to any FM radio.

***FM BUG KIT** New design with PCB embedded coil for extra stability. Transmits to any FM radio. 9v battery req'd. £5 REF: MAG5P5

***FM BUG BUILT AND TESTED** superior design to kit, as supplied to detective agencies etc. 9v battery req'd. £14 REF: MAG14

TALKING COINBOX STRIPPER originally made to retail at £79 each, these units are designed to convert and ordinary phone into a payphone. The units we have generally have the locks missing and sometimes broken hinges. However they can be adapted for their original pupose or used for something else?? Price is just £3 REF: MAG3P1

100 WATT MOSFET PAIR Same spec as 2SK343 and 2SJ413 (8A, 140v, 100w) 1 N channel and 1 P channel. £3 a pair REF: MAG3P2

VELCRO 1 metre length of each side 20mm wide (quick way of fixing for temporary jobs etc) £2 REF: MAG2P3

MAGNETIC AGITATORS Cosisting of a cased mains motor with lead. The motor has two magnets fixed to a rotor that spin round inside. There are also 2 plastic covered magnets supplied. Made for remotely stirring liquids! you may have a use? £3 each REF: MAG3P3 2 for £5 REF: MAG5P6

TOP QUALITY SPEAKERS Made for HI FI televisions these are 10 watt 4R Jap made 4" round with large shielded magnets. Good quality general purpose speaker. £2 each REF: MAG2P4 or 4 for £6 REF: MAG6P2

TWEETERS 2" diameter good quality tweeter 140R (would be good with the above speaker) 2 for £2 REF: MAG2P5 or 4 for £3 REF: MAG3P4

AT KEYBOARDS Made by Apricot these quality keyboards need just a small modification to run on any AT, they work perfectly but you will have to put up with 1 or 2 foreign keycaps! Price £6 REF: MAG6P3

XT KEYBOARDS Mixed types, some returns, some good, some foreign etc but all good for spares! Price is £2 each REF: MAG2P6 or 4 for £6 REF: MAG6P4

PC CASES Again mixed types so you take a chance next one off the pile £12 REF: MAG12 or two identical ones for £20 REF: MAG20P4 component pack bargain 1,000 resistors +1,000 capacitors (all same value) £2.50 a pack. REF: MAG2P7

1994 CATALOGUE OUT NOW

BULL'S BULLETIN BOARD

MASSIVE

WAREHOUSE CLEARANCE

FANTASTIC £20.00 REDUCTION

REFURBISHED PC BASE UNITS
COMPLETE WITH KEYBOARD
FROM ONLY **£29.00**

AMSTRAD 1512 BASE UNITS
GUARANTEED
PERFECT WORKING ORDER.

A LOW COST INTRODUCTION TO THE HOME COMPUTER MARKET.

AMSTRAD 1512SD

1512 BASE UNIT, 5.25" FLOPPY DRIVE AND KEYBOARD. ALL YOU NEED IS A MONITOR AND POWER SUPPLY. WAS £49.00
NOW ONLY **£29.00**
REF: MAG29

AMSTRAD 1512DD

1512 BASE UNIT AND KEYBOARD AND TWO 5.25" 360K DRIVES. ALL YOU NEED IS A MONITOR AND POWER SUPPLY. WAS £59.00
NOW ONLY **£39.00**
REF: MAG39

SOLAR POWER PANELS

3FT X 1FT 10WATT GLASS PANELS
14.5v/700mA
NOW AVAILABLE BY MAIL ORDER
£33.95
(PLUS £2.00 SPECIAL PACKAGING CHARGE)

TOP QUALITY **AMORPHOUS SILICON CELLS** HAVE ALMOST A TIMELESS LIFESPAN WITH AN INFINITE NUMBER OF POSSIBLE APPLICATIONS, SOME OF WHICH MAY BE CAR BATTERY CHARGING, FOR USE ON BOATS OR CARAVANS, OR ANYWHERE A PORTABLE 12V SUPPLY IS REQUIRED. **REF: MAG34**

ALSO 1FT X 1FT GLASS SOLAR PANELS 12v 200mA ONLY £15.00 **REF: MAG15P3**

FREE SOFTWARE!

Brand new, UNUSED top quality *Famous brand* licensed software discs. Available in 5.25" DSDD or 5.25" HD only. You buy the disk and it comes with free BRAND NEW UNUSED SOFTWARE. We are actually selling you the floppy disc for your own **"MEGA CHEAP" storage facilities**, if you happen to get software that you want/need/like as well........ you get a 'MEGA BARGAIN' to!

DSDD PKT10 £2.99 REF: MAG3P7 PKT100 £16.00 REF: MAG16
HD PKT10 £3.99 REF: MAG4P3 PKT100 £26.00 REF: MAG26P1

LARGER QANTITY PRICES AVAILABLE ON APPLICATION

££££££WE BUY SURPLUS STOCK££££££££

TURN YOUR SURPLUS STOCK INTO CASH
IMMEDIATE SETTLEMENT. WE WILL ALSO QUOTE FOR COMPLETE FACTORY CLEARANCE.
COMING SOON

1994 CATALOGUE.

PLEASE SEND 42P, A4 SIZED SAE FOR YOUR FREE COPY.

MINIMUM GOODS ORDER £5.00 TRADE ORDERS FROM GOVERNMENT, SCHOOLS, UNIVERSITIES, & LOCAL AUTHORITIES WELCOME ALL GOODS SUPPLIED SUBJECT TO OUR CONDITIONS OF SALE AND UNLESS OTHERWISE STATED GUARANTEED FOR 30 DAYS RIGHTS RESERVED TO CHANGE PRICES & SPECIFICATIONS WITHOUT PRIOR NOTICE ORDERS SUBJECT TO STOCK QUOTATIONS WILLINGLY GIVEN FOR QUANTITIES HIGHER THAN THOSE STATED

***SOME OF OUR PRODUCTS MAY BE UNLICENSABLE IN THE UK**

BULL ELECTRICAL

250 PORTLAND ROAD HOVE SUSSEX BN3 5QT

MAIL ORDER TERMS: CASH PO OR CHEQUE WITH ORDER PLUS £3.00 POST PLUS VAT.

PLEASE ALLOW 7 - 10 DAYS FOR DELIVERY
TELEPHONE ORDERS WELCOME
TEL: 0273 203500
FAX: 0273 323077

COMMODORE MICRODRIVE SYSTEM mini storage device for C64's 4 times faster than disc drives, 10 times faster than tapes. Complete unit just £12 REF: MAG12P1

SCHOOL STRIPPERS We have quite a few of the above units which are 'returns' as they are quite comprehensive units they could be used for other projects etc. Let us know now many you need at just 50p a unit (minimum 10).

HEADPHONES 15P These are ex Virgin Atlantic. You can have 8 pairs for £2 **REF: MAG2P8**

PROXIMITY SENSORS These are small PCB's with what look like a source and sensor LED on one end and lots of components on the rest of the PCB. Complete with flyleads. Pack of 5 £3 REF: MAG: 3P5 or 20 for £8 REF: MAG8P4

FIBRE OPTIC CABLE Made for Hewlett Packard so pretty good stuff! you can have any length you want (min5m) first 5m £7 REF: MAG7 thereafter £1 a metre (ie 20m is £22) REF: MAG1 Max length 250m.

SNOOPERS EAR? Originall made to clip over the earpiece of telephone to amplify the sound-it also works quite well on the cable running along the wall! Price is £5 REF: MAG5P7

DOS PACKS Microsoft version 3.3 or higher complete with all manuals or price just £5 REF: MAG5P8 Worth it just for the very comprehensive manual! 5.25" only.

DOS PACK Microsoft version 5 Original software but no manuals hence only £3 REF: MAG3P6 5.25" only.

FOREIGN DOS 3.3-German, French, italian etc £2 a pack with manual. 5.25" only. REF: MAG2P9

MONO VGA MONITOR Made by Amstrad, refurbished £49 REF: MAG49

CTM644 COLOUR MONITOR Made to work with the CPC464 home computer. Standard RGB input so will work with other machines. Refurbished £59.00 REF: MAG59

JUST A SMALL SELCTION of what we have- to see more get our 1994 catalogue (42p stamp) or call in Mon-Sat 9-5.30

HAND HELD TONE DIALLERS Ideal for the control of the Response 200 and 400 machines. £5 REF: MAG5P9

PIR DETECTOR Made by famous UK alarm manufacturer these are hi spec, long range internal units. 12v operation. Slight marks on case and unboxed (although brand new) £8 REF: MAG8P5

WINDUP SOLAR POWERED RADIO AM/FM radio complete with hand charger and solar panel! £14 REF: MAG14P1

COMMODORE 64 Customer returns but ok for spares etc £12 REF: MAG12P2 Tested and working units are £69.00 REF: MAG69

COMMODORE 64 TAPE DRIVES Customer returns at £4 REF: MAG4P9 Fully tested and working units are £12 REF: MAG12P5

COMPUTER TERMINALS complete with screen, keyboard and RS232 input/output. Ex equipment. Price is £27 REF: MAG27

MAINS CABLES These are 2 core standard black 2 metre mains cables fitted with a 13A plug on one end, cable the other. Ideal for projects, low cost manufacturing etc. Pack of 10 for £3 REF: MAG3P8 Pack of 100 £20 REF: MAG20P5

SURFACE MOUNT STRIPPER Originally made as some form of high frequency amplifier (main chip is a TSA5511T 1.3GHz synthasiser) but good stripper value, an excellent way to play with surface mount components £1.00 REF: MAG1P1.

MICROWAVE TIMER Electronic timer with relay output suitable to make enlarger timer etc £4 REF: MAG4P4

PLUG 420? showing your age? pack of 10 with leads for £2 REF: MAG2P11

MOBILE CAR PHONE £5.99 Well almost! complete in car phone excluding the box of electronics normally hidden under seat. Can be made to illuminate with 12v also has built in light sensor so display only illuminates when dark. Totally convincing! REF: MAG6P6

ALARM BEACONS Zenon strobe made to mount on an external bell box but could be used for caravans etc. 12v operation. Just connect up and it flashes regularly) £5 REF: MAG5P11

FIRE ALARM CONTROL PANEL High quality metal cased alarm panel 350x165x80mm. Comes with electronics but no information. £15 REF: MAG15P4

SUPER SIZE HEATSINK Superb quality aluminium heatsink. 365 x 183 x 61mm, 15 fins enamble high heat dissipation. No holes! £9.99 REF: MAG10P1P

REMOTE CONTROL PCB These are receiver boards for garage door opening systems. You may have another use? £4 ea REF: MAG4P5

LOPTX Line output transformers believed to be for hi res colour monitors but useful for getting high voltages from low ones! £2 each REF: MAG2P12 bumper pack of 10 for £12 REF: MAG12P3.

PORTABLE RADIATION DETECTOR

£49.99

A Hand held personal Gamma and X Ray detector. This unit contains two Geiger Tubes, has a 4 digit LCD display with a Piezo speaker, giving an audio visual indication. The unit detects high energy electromagnetic quanta with an energy from 30K eV to over 1.2M eV and a measuring range of 5-9999 UR/h or 10-99990 Nr/h. Supplied complete with handbook.
REF: MAG50

CIRCLE NO. 104 ON REPLY CARD

RESEARCH NOTES

High-density storage gets chrome finish

Scientists working at the US National Institute of Standards and Technology (Nist) have glimpsed a future where very high density storage devices could be built around individual atoms of chromium.

The Nist team has used a laser to deposit neat rows of individual chromium atoms on a silicon substrate. But the significant achievement is that the rows are a mere 65nm wide – considerably smaller than the smallest structure that can be created by conventional lithographic techniques.

Atom optics, as the technology is called, involves using a split laser beam to create a stable interference pattern – a standing wave – just above a silicon substrate. The pattern of standing waves consists essentially of alternating light and dark bands, and into it is fired a beam of chromium atoms, all inside an evacuated chamber.

The light and dark bands behave like an array of atomic lenses, focusing the chromium atoms and depositing an identical pattern on the surface of the silicon. Chromium atoms are deposited in the areas where the light intensity is lowest.

According to the Nist researchers (Science, Vol 262, 877) the laser fields influence atomic trajectories by causing them to absorb and re-radiate photons. They also create a dipole force proportional to the intensity gradient in the oscillating electric field of the laser. Both effects are at their strongest when the laser frequency coincides with the atom's natural resonance.

AT&T Bell Laboratories first used the technique to deposit patterns of sodium atoms. But sodium is not stable in air and the resulting structure had poor life.

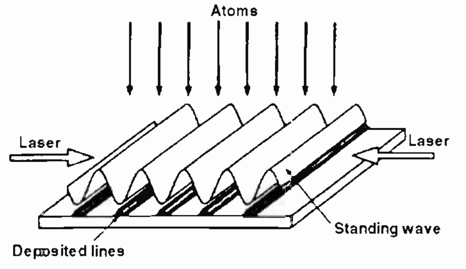
The Nist creation, being made of chromium, is more permanent. Next stage, according to Nist's Bob Celotta, is to create a two-dimensional optical standing wave that would allow deposition, not just of lines, but of dots. Careful movement of the substrate would then permit successive dots no bigger than a few atoms in size to be deposited next to each other in arbitrary patterns.

The ability to create such structures with details as small as 5nm (about 15 atoms) would clearly open up a whole new area of nanotechnology.

Practical problems are still immense. Quantum effects, for example, play a big part and the team is currently investigating how thin a piece of chromium 'wire' can be before it ceases to behave electrically like a piece of wire.

But the choice of chromium will allow the researchers to investigate more than mere electrical conductivity. Celotta points out that chromium dioxide has magnetic properties that are already widely exploited in magnetic data storage. Given the ability to deposit the material in well-ordered arrays at the atomic level, the prospects of being able to develop advanced new storage technologies are considerable.

Celotta says, "We are going to try to make magnetic media with this material and that should lead to high density magnetic storage... Eventually we will also be pursuing other materials, such as silicon and gallium arsenide semiconductors".



Nanometre-wide lines of chromium are deposited by a standing-wave laser field that forms cylindrical lenses. These focus the chromium atoms into strips that could be the basis of very high density storage devices.

Hole story behind superfast p-mosfet

Cornell University electrical engineers have fabricated a p-channel mosfet that overcomes one of the seemingly-inherent disadvantages of such devices – the poor mobility of holes compared to electrons.

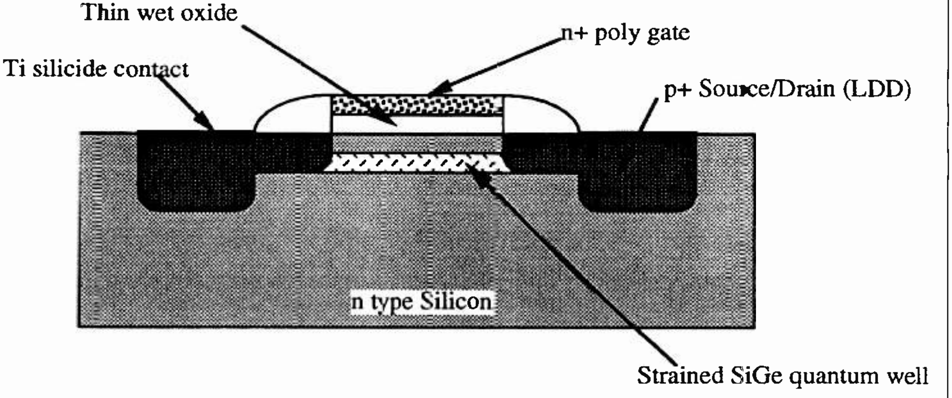
In conventional silicon microelectronics, holes travel three times slower than electrons, leading to the disparity in performance between complementary n- and p-channel devices. But Cornell Assistant Professor Yosef Shacham-Diamand and PhD student Kaushik Bhaumik have developed a new type of p-channel device with a 10nm (50 atom diameters) layer of silicon-germanium. This layer cracks the speed problem by creating a quantum well that captures holes, providing them with a sort of conduction 'fast lane'. Shacham-Diamand says that, for an equivalent terminal voltage, the new device passes 40% more current (about 5mA) and switches 40% faster than a typical p-channel device.

"Now we have a p-channel device that's just as fast as an n-channel device," he says.

Electron-beam-lithography capabilities of the National Nanofabrication Facility at Cornell were used to define the gate length of the transistor to less than 0.2μm. At this sort of gate dimension, the holes in the SiGe layer travel at speeds exceeding that of holes in silicon, an enhancement reflected in the overall performance of the device.

Another benefit is a possible reduction in the operating voltage, in this case to 1.5V rather than the more common 3.3V. The resultant increased chip density makes greater complexity possible as well as faster speed. Despite the improvements, no special fabrication techniques are needed and there are no extra processing steps.

On test, the new p-channel mosfet was clocked at 35GHz, substantially faster than the best that can be achieved with standard p-channel devices (about 10GHz), and even marginally faster than the equivalent n-channel mosfet (32GHz).



In complementary mosfet ICs, p-channel devices perform less well than their n-channel counterparts due to low hole mobility. This new p-channel structure could lead to faster complementary mosfet ICs with the bonus of lower operating voltage and higher density.

Jupiter's big bang could shake scientific world

The scientific community is expectantly grabbing its seats to observe – albeit indirectly – one of the biggest collisions in the Solar System in recent times. The occasion will be when over twenty fragments of a broken comet, on collision course for Jupiter, hit the giant planet at a speed of 60km/s in July. Unfortunately, Comet Shoemaker-Levy 9 will hit the side of Jupiter facing away from the Earth. Even so, the International Astronomical Union says it expects the comet to undertake a final dramatic act of suicide in which the fragments will hit the surface of Jupiter with a force equivalent to a 100Gt nuclear weapon. The impact would be of the same order as that thought to have occurred 65 million years ago on Earth and which may have led to the extinction of the dinosaurs.

Shoemaker-Levy 9, named after its discoverers, is one of the strangest objects in the solar system. Instead of being a single body, it looks more like a string of pearls in the sky. Pictures taken by the Hubble space telescope show fragments of up to 5km in diameter, chasing each other in a procession tens of thousands of kilometres long.

The orbits of Jupiter and Shoemaker-Levy had been getting progressively closer for some time. But the history of the bizarre multi-comet actually goes back no further than July 1992 when, as a normal comet, it almost crashed into Jupiter. Instead the powerful gravity of Jupiter simply wrenched the comet into tiny pieces.

The now-awaited collision is so important that, in spite of the fact that it is happening on the dark side of the planet, astronomers are thinking up ingenious ways to observe what happens.

One idea is to look at Jupiter's various moons to see if they reflect any light from what is bound to be a firework display on the grand scale. Another idea is to look for refracted light around the rim of the planet.

Jupiter rotates very fast on its axis so astronomers will not have to wait more than an hour or two for the scars on the planet's surface to swing round into view. But observers hope to be able to see the impact directly – using Voyager 2. The spacecraft is too far beyond Jupiter to resolve a clear image, but it should still be able to measure the intensity of the flashes of light.

The other hope is that Galileo, still on its way to Jupiter, might able to catch an oblique view of what's going on behind.

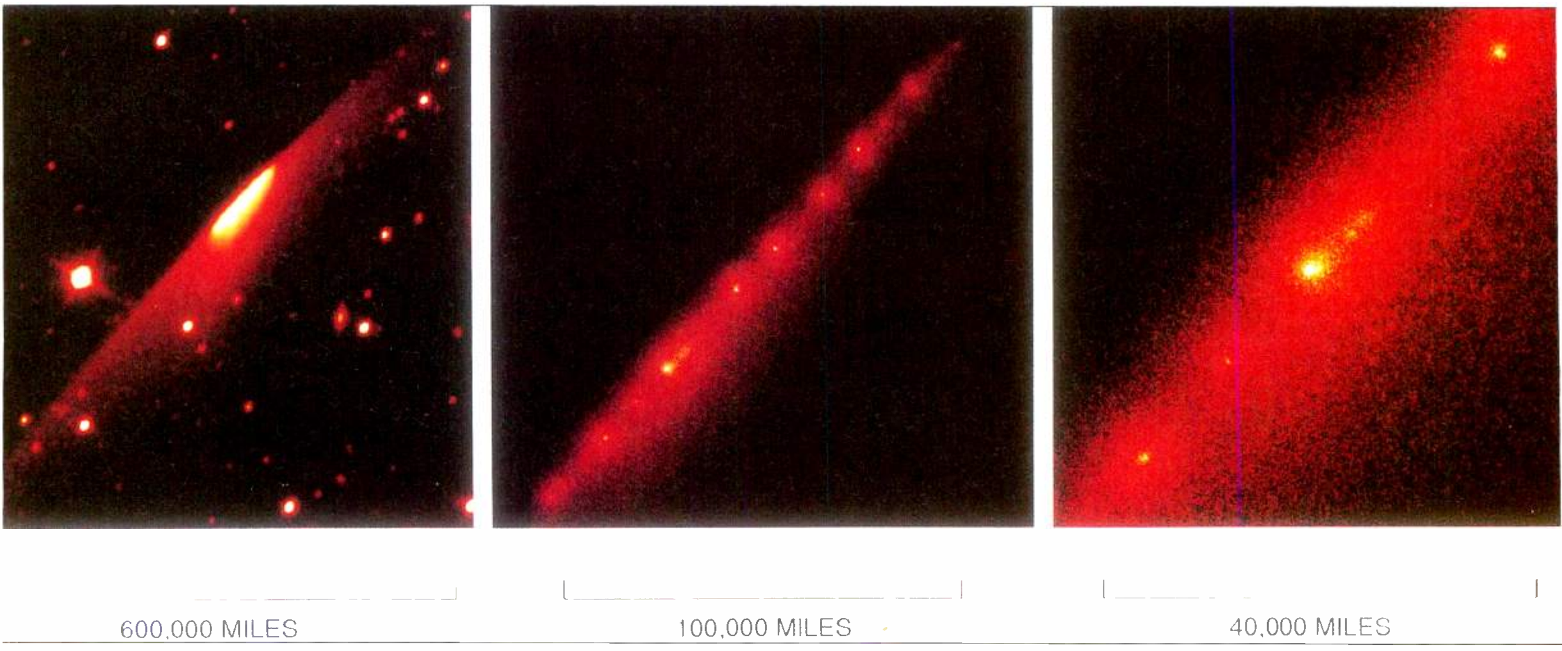
Precisely what will happen when all the fragments of Shoemaker-Levy hit the giant red planet remains a matter of some debate. It is expected to be a bright and spectacular event, with huge holes punched in Jupiter's atmosphere and with gigantic shock waves reverberating around the planet. Researchers at Sandia National Laboratories in Albuquerque are currently attempting to gain a better understanding of this cataclysmic event by using a supercomputer system originally developed to model nuclear weapon blasts.

Sandia's Mark Boslough says that the Jovian collision will have certain differences relative to what happened here on earth 65 million years ago. On earth, the impact would have caused an instant atmospheric pressure rise, with all the force of a massive nuclear weapon. But because Jupiter does not have a solid surface, the impact may have different characteristics. Sandia's supercomputer simulation predicts that when Shoemaker-Levy 9 enters the atmosphere of Jupiter, it will at first slice through unhindered. After that, the pressure will build up gradually until the comet pieces break up further.

At this point, each piece will have lost 2% of its kinetic energy. The remaining 98% will be carried beneath Jupiter's clouds where it will be explosively released.

Still to be worked out on this model is whether the final big bang will result in a giant mushroom cloud, another Great Red Spot or nothing at all.

The answer, for which we may need to wait until July, is more than just a matter of curiosity. Astronomers believe that impacts of this sort are the means by which the planets were created in the first place. So the way in which Shoemaker-Levy 9 commits suicide could answer some fundamental questions about our own history.



'String of Pearls' comet on collision course with Jupiter. Left is taken from Earth, centre and right from Hubble.

SMALL SELECTION ONLY LISTED – EXPORT TRADE AND QUANTITY DISCOUNTS – RING US FOR YOUR REQUIREMENTS WHICH MAY BE IN STOCK

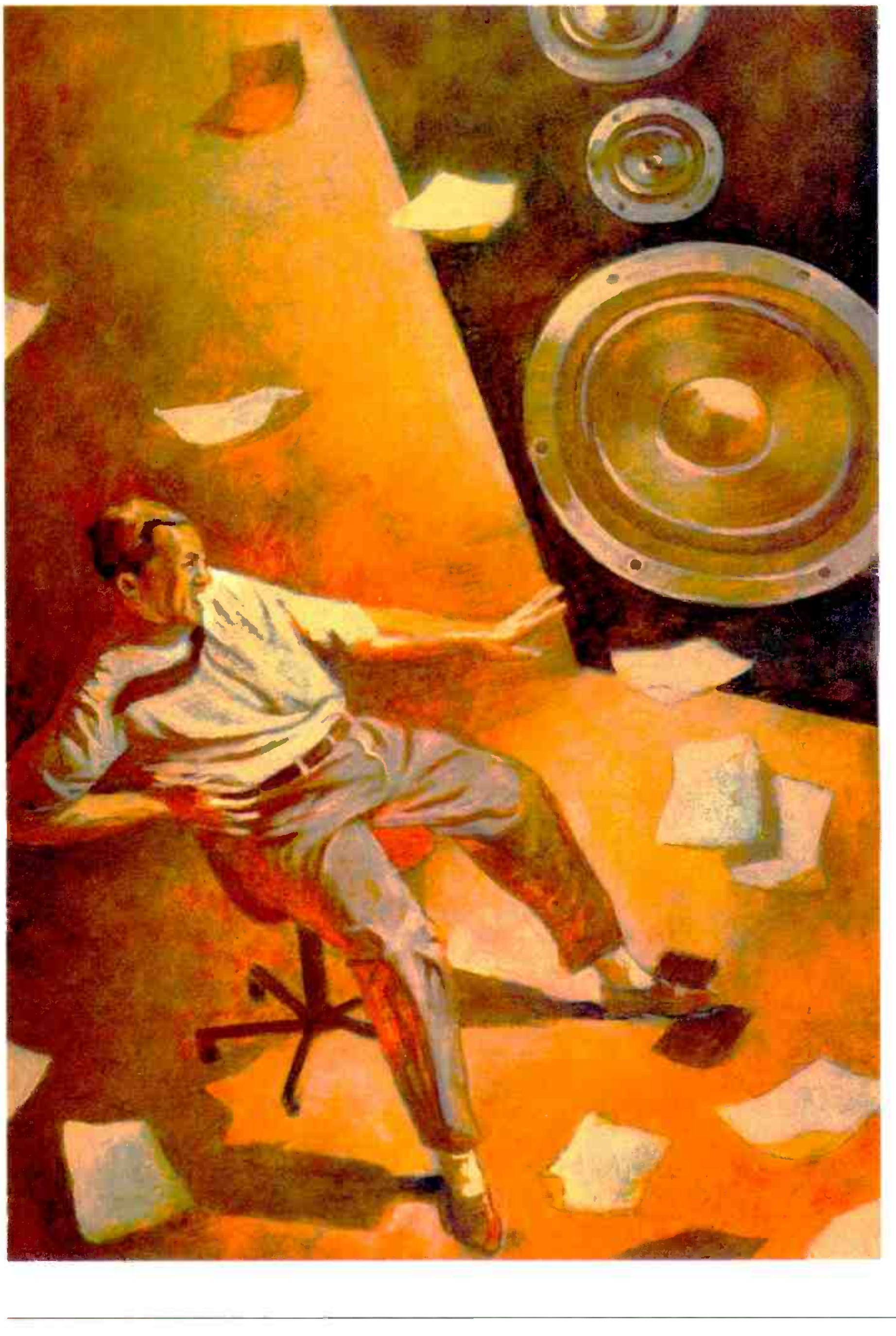
Marconi TF2008 – AM-FM signal generator – Also sweeper – 10Kc/s – 510Mc/s – from £350 tested to £500 as new with manual – probe kit in wooden carrying box.
HP Frequency comb generator type 8406A – £400.
HP Sampling Voltmeter (Broadband) type 3406A – £200.
HP Vector Voltmeter type 8405A – £400 to £600 – old or new colour.
HP Synthesiser/signal generator type 8672A – 2 to 18GHz £4000.
HP Oscillographic recorder type 7404A – 4 track – £350.
HP Plotter type 9872B – 4 pen – £300.
HP Sweep Oscillators type 8690 A & B + plug-ins from 10Mc/s to 18GHz also 18-40GHz. P.O.R.
HP Network Analyser type 8407A + 8412A + 8601A – 100Kc/s – 110Mc/s – £500–£1000.
HP Down Converter type 11710B – .01-11Mc/s – £450.
HP Pulse Modulator type 11720A – 2-18GHz – £1000.
HP Modulator type 8403A – £100-£200.
HP Pin Modulators for above-many different frequencies – £150.
HP Counter type 5342A – 18GHz – LED readout – £1500.
HP Signal Generator type 8640B – Opt001 + 003 – .5-512Mc/s AM/FM – £1000.
HP Amplifier type 8447A – 1-400Mc/s £200 – HP8447F .1-1300Mc/s £400.
HP Frequency Counter type 5340A – 18GHz £1000 – rear output £800.
HP 8410 – A – B – C Network Analyser 110Mc/s to 12GHz or 18GHz – plus most other units and displays used in this set-up – 8411A – 8412 – 8413 – 8414 – 8418 – 8740 – 8741 – 8742 – 8743 – 8746 – 8650. From £1000.
HP Signal Generator type 8660C – .1-2600Mc/s AM/FM – £3000 1300Mc/s £2000.
HP Signal Generator type 8656A – 0.1-990Mc/s. AM/FM – £2000.
HP 8699B Sweep PI – 0.1-4GHz £750 – HP8690B Mainframe £250.
Racal/Dana 9301A-9302 RF Millivoltmeter – 1.5-2GHz – £250-£400.
Racal/Dana Counters 9915M – 9916 – 9917 – 9921 – £150 to £450. Fitted FX standards.
Racal/Dana Modulation Meter type 9009 – 8Mc/s – 1.5GHz – £250.
Racal – SG Brown Comprehensive Headset Tester (with artificial head) Z1A200/1 – £350.
Marconi AF Power Meter type 893B – £200.
Marconi RCL Bridge type TF2700 – £150.
Marconi/Saunders Signal Sources type – 6058B – 6070A – 6055B – 6059A – 6057B – 6056 – £250–£350. 400Mc/s to 18GHz.
Marconi TF1245 Circuit magnification meter + 1246 & 1247 Oscillators – £100-£300.
Marconi microwave 6600A sweep osc., mainframe with 6650 PI – 18-26.5GHz or 6651 PI – 26.5-40GHz – £1000 or PI only £600.
Marconi distortion meter type TF2331 – £150. TF2331A – £200.
Microwave Systems MOS/3600 Microwave frequency stabilizer – 1GHz to 40GHz £1k.
Tektronix Plug-Ins 7A13 – 7A14 – 7A18 – 7A24 – 7A26 – 7A11 – 7M11 – 7S11 – 7D10 – 7S12 – S1 – S2 – S6 – S52 – PG506 – SC504 – SG502 – SG503 – SG504 – DC503 – DC508 – DD501 – WR501 – DM501A – FG501A – TG501 – PG502 – DC505A – FG504 – P.O.R.
Alltech Stoddart receiver type 17/27A – .01–32Mc/s – £2500.
Alltech Stoddart receiver type 37/57 – 30-1000Mc/s – £2500.
Alltech Stoddart receiver type NM65T – 1 to 10GHz – £1500.
Gould J3B Test oscillator + manual – £200.
Infra-red Binoculars in fibre-glass carrying case – tested – £100. Infra-red AFV sights £100.
ACL Field intensity meter receiver type SR – 209 – 6. Plugs-ins from 5Mc/s to 4GHz – P.O.R.
Tektronix 491 spectrum analyser – 1.5GHz-40GHz – as new – £1000 or 10Mc/s 40GHz.
Tektronix Mainframes – 7603 – 7623A – 7633 – 7704A – 7844 – 7904 – TM501 – TM503 – TM506 – 7904 – 7834 – 7104.
Knott Polyskanner WM1001 + WM5001 + WM3002 + WM4001 – £500.
Alltech 136 Precision test RX + 13505 head 2 – 4GHz – £350.
SE Lab Eight Four – FM 4 Channel recorder – £200.
Alltech 757 Spectrum Analyser – 001 22GHz – Digital Storage + Readout – £3000.
Dranetz 606 Power line disturbance analyser – £250.
Precision Anerold barometers– 900-1050Mb – mechanical digit readout with electronic indicator – battery powered. Housed in polished wood carrying box – tested – £100-£200-£250. 1, 2 or 3.
HP141T SPECTRUM ANALYSERS – ALL NEW COLOURS
TESTED WITH OPERATING MANUAL
HP141T + 8552A or B IF-8553B RF – 1kHz–110Mc/s-A IF – £1300 or B IF – £1400.
HP141T + 8552A or B IF-8554B RF – 100kHz–1250Mc/s-A IF – £1400 or B IF – £1500.
HP141T + 8552A or B IF-8555A RF – 10Mc/s–18GHz-A IF – £2400 or B IF – £2500.
HP141T + 8552A or B IF-8556A RF – 20Hz–300kHz-A IF-A IF – £1200 or B IF – £1300.
HP8443A tracking generator/counter – 100kHz–110Mc/s – £500.
HP8445B tracking pre-selector DC–18GHz – £750.
HP ANZ UNITS AVAILABLE SEPARATELY – NEW COLOURS – TESTED.
HP141T mainframe – £550 – 8552A IF – £450 – 8552B IF – £550 – 8553B RF – 1kHz–110Mc/s – £550 – 8554B-RF – 100kHz–1250Mc/s – £650 – 8555A-RF – 10Mc/s–18GHz – £1550.
HP 3580A LF-spectrum analyser – 5kHz to 50kHz – LED readout – digital storage – £1600 with instruction manual – internal rechargeable battery.
Tektronix 7D20 plug-in 2-channel programmable digitizer – 70 Mc/s – for 7000 mainframes – £500 – manual – £50.
Datron 1065 Auto Cal digital multimeter with instruction manual – £500.
Racal MA 259 FX standard. Output 100kc/s–1Mc/s–5Mc/s – internal NiCad battery – £150.
Aerial array on metal plate 9"×9" containing 4 aerials plus Narda detector – .100–11GHz. Using N type and SMA plugs & sockets – ex eqpt – £100.
EIP 451 microwave pulse counter 18GHz – £1000.
Marconi RF Power Amplifier TF2175 – 1.5Mc/s to 520Mc/s with book – £100.
Marconi 6155A Signal Source – 1 to 2 GHz – LED readout – £600.
Schlumberger 2741 Programmable Microwave Counter – 10Hz to 7.1GHz – £750.
Schlumberger 2720 Programmable Universal Counter 0 to 1250Mc/s – £600.
HP 2225CR Thinkjet Printer – £100.
TEK 576 Calibration Fixture – 067-0597-99 – £250.
HP 8006A Word Generator – £150.
HP 1645A Data Error Analyser – £150.
Texscan Rotary Attenuators – BNC/SMA 0-10-60-100DBS – £50-£150.
HP 809C Slotted Line Carriages – various frequencies to 18GHZ – £100 to £300.
HP 532-536-537 Frequency Meters – various frequencies – £150-£250.
Barr & Stroud variable filter EF3 0.1Hz–100kc/s + high pass + low pass – £150.
S.E. Lab SM215 Mk11 transfer standard voltmeter – 1000 volts.
Alltech Stoddart P7 programmer – £200.
H.P. 6941B multiprogrammer extender. £100.
Fluke Y2000 RTD selector + Fluke 1120A IEEE-488-translator – Fluke 2180 RTD digital thermometer + 9 probes. £350 all three items.
H.P. 6181 DC current source. £150.
H.P. 59501A – HP-IB isolated D/A/power supply programmer.
H.P. 3438A digital multimeter.
H.P. 6177C DC current source. £150.
H.P. 6207B DC power supply.
H.P. 741B AC/DC differential voltmeter standard (old colour) £100.
H.P. 6209B DC power unit.
Fluke 80 high voltage divider.
Fluke 431C high voltage DC supply.
Tektronix M2 gated delay calibration fixture. 067-0712-00.
Tektronix precision DC divider calibration fixture. 067-0503-00.
Tektronix overdrive recovery calibration fixture. 067-0608-00.
Avo VCM163 valve tester + book £300.
H.P. 5011T logic trouble shooting kit. £150.
Marconi TF2163S attenuator – 1GHz. £200.
PPM 8000 programmable scanner.
Fluke 730A DC transfer standard.
B&K 4815 calibrator head.
B&K 4812 calibrator head.
Farnell power unit H60/50 – £400 tested.
H.P. FX doubler 938A or 940A – £300.
Racal/Dana 9300 RMS voltmeter – £250.
H.P. sweeper plug-ins – 86240A – 2–8.4GHz – 86260A – 12.4–18GHz – 86260AH03 – 10–15GHz – 86290B – 2–18.6GHz. 86245A 5.9–12.4GHz.
Telequipment CT71 curve tracer – £200.
H.P. 461A amplifier – 1kc–150Mc/s – old colour – £100.
H.P. 8750A storage normalizer.
Tektronix oscilloscopes type 2215A – 60Mc/s – c/w book & probe – £400.
Tektronix monitor type 604 – £100.
Marconi TF2330 or TF2330A wave analysers – £100–£150.
HP5006A Signature Analyser £250 + book.
HP10783A numeric display. £150.
HP 3763A error detector. £250.
Racal/Dana signal generator 9082 – 1.5–520Mc/s – £800.
Racal/Dana signal generator 9082H – 1.5–520Mc/s – £900.
Claude Lyons Compuline – line condition monitor – in case – LMP1 + LCM1 £500.
Efratom Atomic FX standard FRT – FRK – 1–1-5–10Mc/s. £3K tested.
Racal 4D recorder – £350 – £450 in carrying bag as new.
HP8350A sweep oscillator mainframe + HP11869A RF PI adaptor – £1500.
Alltech – precision automatic noise figure indicator type 75 – £250.
Adret FX synthesizer 2230A – 1Mc/s. £250.
Tektronix – 7S12–7S14–7T11–7S11–S1–S52–S53.
Rotek 610 AC/DC calibrator. £2K + book.
Marconi TF2512 RF power meter – 10 or 30 watts – 50 ohms – £80.
Marconi multiplex tester type 2830.
Marconi digital simulator type 2828A.
Marconi channel access switch type 2831.
Marconi automatic distortion meter type TF2337A – £150.
Marconi mod meters type TF2304 – £250.
HP 5240A counter – 10Hz to 12.4GHz – £400.
HP 3763A error detector.
HP 8016A word generator.
HP 489A micro-wave amp – 1–2GHz.
HP 8565A spectrum analyser – .01–22GHz – £4k.
HP 5065A rubidium vapour FX standard – £5k.
Fluke 893A differential meters – £100 ea.
Systron Donner counter type 6054B – 20Mc/s–24GHz – LED readout – £1k.
Takeda Riken TR4120 tracking scope + TR1604P digital memory.
EG&G Parc model 4001 indicator + 4203 signal averager PI.
Systron Donner 6120 counter/timer A+B+C inputs – 18GHz – £1k.
Racal/Dana 9083 signal source – two tone – £250.
Systron Donner signal generator 1702 – synthesized to 1GHz – AM/FM.
Systron Donner microwave counter 6057 – 18GHz – Nixey tube – £600.
Racal/Dana synthesized signal generator 9081 – 520Mc/s – AM–FM. £600.
Farnell SSG520 synthesized signal generator – 520Mc/s – £500.
Farnell TTS520 test set – £500 – both £900.
Tektronix plug-ins – AM503 – PG501 – PG508 – PS503A.
Tektronix TM515 mainframe + TM5006 mainframe.
Cole power line monitor T1085 – £250.
Claude Lyons LCM1P line condition monitor – £250.
Rhodes & Schwarz power signal generator SLRD-280 – 2750Mc/s. £250–£600.
Rhodes & Schwarz vector analyser – ZPV + E1 + E3 tuners – .3–2000Mc/s.
Bell & Howell TMA3000 tape motion analyser – £250.
Ball Efratom PTB-100 rubidium standard mounted in Tek PI.
Ball Efratom rubidium standard PT2568-FRKL.
Trend Data tester type 100 – £150.
Farnell electronic load type RB1030-35.
Fairchild interference analyser model EMC 25 – 14kc/s–1GHz.
Fluke 1720A instrument controller + keyboard.
Marconi 2442 – microwave counter – 26.5GHz – £1500.
Racal/Dana counters – 9904 – 9905 – 9906 – 9915 – 9916 – 9917 – 9921 – 50Mc/s – 3GHz – £100–£450 – all fitted with FX standards.
B&K 7003 tape recorder – £300.
B&K 2425 voltmeter – £150.
B&K 4921 + 4149 outdoor microphone.
Wiltron sweeper mainframe 610D – £500.
HP3200B VHF oscillator – 10–500Mc/s – £200.
HP3747A selective level measuring set.
HP3586A selective level meter.
HP5345A electronic counter.
HP4815A RF vector impedance meter c/w probe. £500–£600.
Marconi TF2092 noise receiver. A, B or C plus filters.
Marconi TF2091 noise generator. A, B or C plus filters.
Tektronix oscilloscope 485 – 350Mc/s – £500.
HP180TR, HP182T mainframes £300–£500.
Bell & Howell CSM2000B recorders.
HP5345A automatic frequency convertor – .015–4GHz.
Fluke 8506A thermal RMS digital multimeter.
HP3581A wave analyser.
Philips panoramic receiver type PM7800 – 1 to 20GHz.
Marconi 6700A sweep oscillator + 6730A – 1 to 2GHz.
Wiltron scaler network analyser 560 + 3 heads. £1k.
R&S signal generator SMS – 0.4–1040Mc/s – £1500.
HP8558B spectrum ANZ PI – .1–1500Mc/s – o/c – £1000. N/C – £1500 – To fit HP180 series mainframe available – £100 to £500.
HP8505A network ANZ + 8503A S parameter test set + 8501A normalizer – £4k.
HP8505A network ANZ + 8502A test set – £3k.
Racal/Dana 9087 signal generator – 1300Mc/s – £2k.
Racal/Dana VLF frequency standard equipment. Tracor receiver type 900A + difference meter type 527E + rubidium standard type 9475 – £2750.
Marconi 6960–6960A power meters with 6910 heads – 10Mc/s – 20GHz or 6912 – 30kHz–4.2GHz – £800–£1000.
HP8444A–HP8444A opt 59 tracking generator £1k–£2k.
B&K dual recorder type 2308.
HP8755A scaler ANZ with heads £1k.
Tektronix 475 – 200Mc/s oscilloscopes – £350 less attachments to £500 c/w manual, probes etc.
HP signal generators type 626 – 628 – frequency 10GHz–21GHz.
HP 432A–435A or B–436A – power meters + powerheads – 10Mc/s–40GHz – £200–£280.
HP3730B down convertor – £200.
Bradley oscilloscope calibrator type 192 – £600.
Spectrascope SD330A LF realtime ANZ – 20Hz–50kHz – LED readout – tested – £500.
HP8620A or **8620C** sweep generators – £250 to £1k with IEEE.
Barr & Stroud variable filter EF3 0.1Hz–100kc/s + high pass + low pass – £150.
Tektronix 7L12 analyser – .1Mc/s–1.8GHz – £1500 – **7L14 ANZ** – £2k.
Marconi TF2370 spectrum ANZ – 110Mc/s – £1200–£2k.
Marconi TF2370 spectrum ANZ + TK2373 FX extender 1250Mc/s + trk gen – £2.5k–£3k.
Racal receivers – RA17L–RA1217–RA1218–RA1772–RA1792 – P.O.R.
Systron Donner microwave counter 6057 – 18GHz – nixey tube – £600.
HP8614A signal gen 800Mc/s–2.4GHz old colour £200, new colour £400.
HP8616A signal gen 1.8GHz–4.5GHz old colour £200, new colour £400.

ITEMS BOUGHT FROM HM GOVERNMENT BEING SURPLUS. PRICE IS EX WORKS. S.A.E. FOR ENQUIRIES. PHONE FOR APPOINTMENT OR FOR DEMONSTRATION OF ANY ITEMS. AVAILABILITY OR PRICE CHANGE. VAT AND CARR., EXTRA.
ITEMS MARKED TESTED HAVE 30-DAY WARRANTY. WANTED: TEST EQPT – VALVES – PLUGS & SOCKETS – SYNCROS – TRANSMITTING & RECEIVING EQPT. ETC.

Johns Radio, Whitehall Works, 84 Whitehall Road East, Birkenshaw, Bradford BD11 2ER. Tel. No. (0274) 684007. Fax 651160.

CIRCLE NO. 107 ON REPLY CARD

Extending BASS



Coupled-cavity loudspeakers have the combined advantages of extended bass response and shallow lf roll-off. Ian Gosling highlights the design process with a worked sub-woofer example.

In recent years manufacturers have followed a trend towards smaller speaker enclosures. Sparsely furnished rooms exhibit undesirable room resonance at around 30Hz, and small speakers are more popular for aesthetic reasons. As a result, few reasonably priced systems now offer a –3dB point much lower than 70Hz.

Bass response has suffered. This is a pity, since the bottom octaves of piano and bass guitar both extend down to 40Hz, and percussion much lower. Many commercial systems use vented boxes to reduce footprint, but with the penalty of steep response roll-off. Venting also degrades transient response – an important factor in perceived sound quality.

Compact discs can reproduce the full audio spectrum from 2Hz to 20kHz. I therefore looked for a suitable sub-woofer design to extend my existing system. Its specification was a frequency response flat down to 20Hz, with a gradual roll-off of 12dB/octave or better.

I chose 80Hz as the subwoofer crossover frequency. There are a few choices of enclosure, each with their own merits. I chose the coupled-cavity enclosure. It is smaller than a transmission line and much smaller than a horn. In addition, radiation is only emitted from the port, which is quite small. This reduces the area of grille cloth needed, making the unit surprisingly inconspicuous in a domestic environment.

Since the high frequency response of the cavity-based subwoofer rolls off naturally, a crossover network is not needed. If you wish not to enter into filter theory, a rough design for a simple closed box can be completed in a few minutes using a calculator. For more complex enclosure types however, a first cut design procedure can give disappointing results. It is a good idea to model the frequency response on a computer before starting to cut timber.

The enclosure can be modelled as an acoustic circuit on a Spice type circuit simulator. Prices of such software nowadays can be much less than the cost of speaker materials.

Acoustic circuits

Just as voltage V and current I describe an electrical signal, so a sound wave can be described by air pressure p in N/m^2 and the rate of movement of air or volume velocity v in m/s, **Fig. 1**. Equivalent electrical components can represent acoustical loudspeaker components.

Consider first a sealed box with an air inlet, **Fig. 2**. When air is pumped in, the pressure rises. This is analogous to an electrical capacitor, where injecting charge causes the potential difference to rise. The acoustic equivalent of capacitance is the acoustic compliance C_A measured in m/N.

A similar argument applies to air being pumped into the space near an elastic mechanical part such as the cone suspension. When deflected from it rest position, it exerts a pressure on the air through the attached cone. The acoustic compliance is proportional to the mechanical compliance or spring constant.

Mass of the drive unit cone and the mass of air trapped in the port tube also have acoustic equivalents. A mass moving at the same velocity as the air behind it is in equilibrium and no forces act. If the air velocity changes suddenly, the mass cannot react instantaneously to move with it, so the air starts to press on the mass. This is analogous to change of current through an inductor, which causes a back emf.

The equivalent of inductance is acoustic mass M_A measured in kg/m^4. Radiation of sound energy is analogous to electrical power dissipation. As a result, the circuit component is an acoustic radiation resistance. There is also a reactance due to the mass of air trapped just in front of the radiating surface.

Finally, the electrical part of the circuit must also be modelled. An electrical resistance R_E appears through the transformer action of the voice coil motor as an acoustic resistance

$$R_A = \frac{(Bl)^2}{S_D^2 R_E},$$

where B_l is magnetic field in the voice coil multiplied by the coiled length of wire and R_E is the sum of amplifier output impedance, speaker cable resistance and voice coil resistance. Typically, R_E is around 8Ω. Sensibly chosen speaker cable has a negligible contribution to this figure – another nail in the coffin for fancy cables.

Enclosure design process **Fig. 3** shows the enclosure configuration. Two drive units are used to reduce the large cone excursion nec-

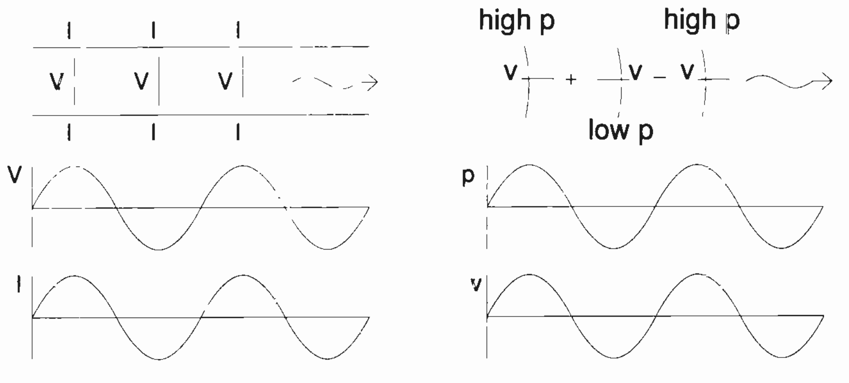


Fig. 1. Voltage and current in a travelling wave on an electrical two-wire transmission line compared to pressure and velocity in a sound wave. The air has maximum and minimum density at the places marked + and – respectively.

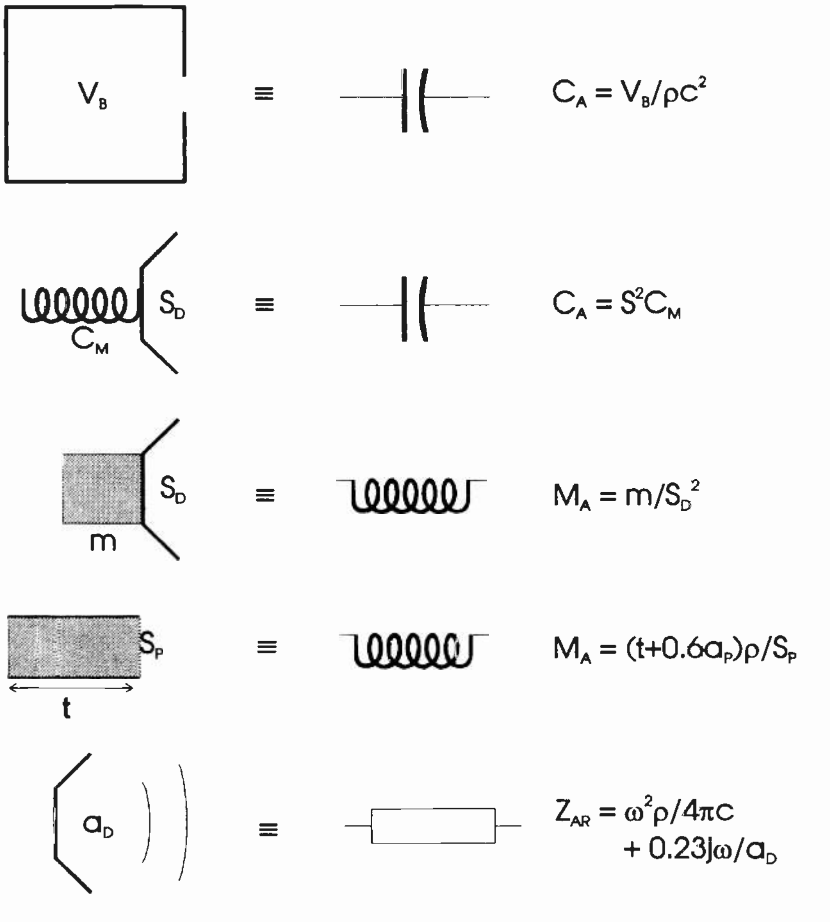


Fig. 2. Top to bottom, electrical components analogous to a sealed box of volume V_B, a cone suspension of spring constant C_M, a cone of mass m, a port tube of volume SP×t and a radiating surface of radius a_D. Parameter ρ is the density of air, c the speed of sound and S_D cone surface area.

essary at low frequencies.

Behind each is a sealed box of volume V_B. One drive unit may be mounted back to front to reduce second harmonic distortion. In this case its electrical connections must also be reversed.

In acoustic circuit **Fig. 4**, $R_{1,2}$ are the voice coil resistances, $R_{3,4}$ the energy absorbed in the cone suspension, $L_{1,2}$ the cone masses and $C_{1,2}$ the suspension compliances. $C_{3,4}$ represent the compliance of air in boxes V_B. Since all these components move together, they displace the same amount of air, so must share the same volume velocity v. This is equivalent to electrical components that all carry the same current. They are therefore connected in series.

The cavity of volume V_C (C_5) can be regarded as a sealed box into which air can flow from the drive units and from the port tube L_3. Since these volumes of air add, there is a volume summing junction at one end of C_5, equivalent to a current summing junction.

For a closed box, the acoustic circuit has a high-pass resonance below which there is no useful sound output. Inspection of this circuit shows that there is a band-pass resonance. Since this lies at the centre of the operating frequency band, the lower 3dB point can be at least an octave below that obtainable with simpler enclosure types.

The schematic can be entered on a PC-based circuit simulator and output sound pressure level plotted as the voltage across the resistive part of the radiation impedance Z_1. Alternatively, a short program can be written to perform the circuit calculations and display the results. Enclosure and port dimensions are then adjusted to obtain the required response.

For a speaker response extending down to 20Hz, the enclosure resonant frequency should be about 40Hz and the free air resonance of the drive units, f_s, should be less than 30Hz. Manufacturers provide drive unit parameters, but they also can be measured as shown in the panel.

I used the KEF *B139B*, which has low distortion and a free air resonance of 29.5Hz. For a design procedure not involving filter theory, initially the box volume V_B can be made infinite. Parameter V_C and the acoustic mass of the port are chosen to give the required centre frequency. Then V_B is reduced to a sensible size and final adjustments made to the bandwidth and response shape adjusted.

To meet my specification, V_B should be $0.08m^3$, V_C is $0.056m^3$ and port cross-section

Bass speaker enclosures

Loudspeaker drive units were first designed for use with an infinite baffle. A closed back to the enclosure degraded the performance rather than enhancing it.

More recent air suspension drivers are designed to work with an enclosure. Compressing the air in the sealed box creates pressure on the cone, and this can be employed to provide the restoring force for the cone as opposed to the traditional stiff cone suspension. Sealed enclosures have a high-pass response with a gradual low frequency roll-off of 12dB/octave.

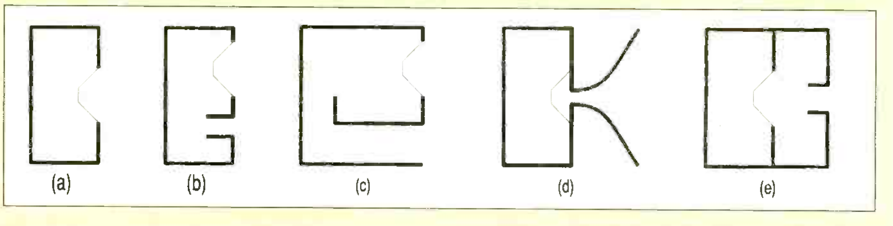
Vented or reflex enclosures have an additional tube terminated by a port which can radiate in addition to the drive unit. This lowers the cut-off frequency of the response somewhat, but increases the roll-off slope to 18dB/octave.

In the transmission line enclosure, the port tube is lengthened to be a half wavelength at the cut-off frequency. Sound then emerging from the port is in phase with that directly radiated from the drive unit. This increases the sound output, extending the bass response. But the tube now becomes 8.7m long at 20Hz. In addition, at frequencies where the port is a whole number of wavelengths long, destructive interference occurs, causing dips in the response.

Fibre packing in the port tube can help by attenuating the signal at mid bass frequencies and by reducing the wavelength by up to three times, as in the Bailey design[1]. Packing however causes pressure build-up behind the cone which can spoil the frequency response.

Sound output at low frequencies can be improved by making the cone effectively larger by attaching it to a horn. Its mouth should ideally be at least one wavelength in circumference, i.e. 4.3m across for a flat response down to 20Hz. The horn enclosure has a band-pass response with a very shallow lf roll-off of 6dB/octave.

Coupled-cavity enclosures[2] have a cavity and a port in front of the drive unit. This also results in a band-pass response. The lower 3dB point is well below that of a comparable closed box while roll-off is unchanged.



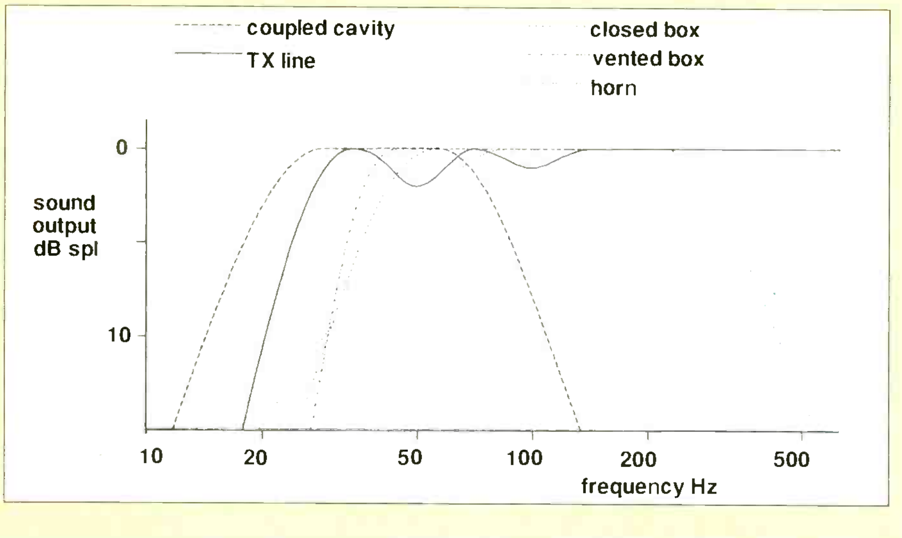


Fig. 1. Speaker enclosure types. Diagram (a) is basic closed box, (b) is a vented box, (c) represents a transmission line, (d) a horn and (e) a coupled cavity.

Fig. 2. Comparison of frequency responses of different enclosures of broadly similar floor footprint. These curves can vary considerably with the enclosure design and drive unit parameters.

Measuring drive unit parameters

If the drive unit parameters are unknown, the following procedure can be used to obtain reasonably accurate values using readily available instruments.

First measure area S_D as the area of the cone itself plus half the surround. Measure across the front of the chassis, not along the slope of the cone. Next apply a small weight to the cone. Suspension compliance C_{AS} is then S_D^2 multiplied by the ratio of cone deflection to the mass applied.

Now measure the DC voice coil resistance R_E. Connect to a signal generator via a voltmeter and ammeter and find the free air resonant frequency, f_s, where current is at its minimum. Divide the AC voltage by the current to find the total resistance of the drive unit, calling this R_E+R_{EC}. Subtract R_E to obtain R_{EC}.

If B_l is not known, measure the frequencies f_1 and f_2 at which AC resistance is,

$$\sqrt{R_E(R_E+R_{EC})}.$$

Calculate the electrical

$$Q_{EC}=f_s/(f_2-f_1),$$

and hence the component value

$$R_1=\frac{(Bl)^2}{S_D^2 R_E},$$

using $R_1=1/(2\pi f_s Q_{EC} C_{AS})$. Calculate $R_{AS}=R_1 R_E/R_{EC}$. Then finally,

$$M_{AD}=\frac{1}{4\pi^2 f_s^2 C_{AS}}-0.46\sqrt{\frac{S_D}{\pi}}$$

The second term allows for the mass of air trapped next to the cone.

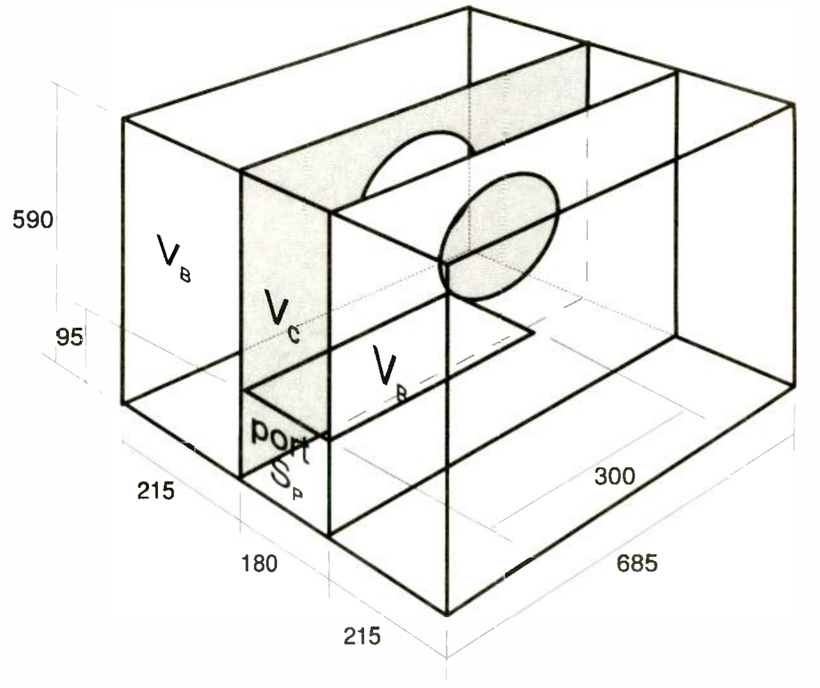


Fig. 3. Twin-unit coupled cavity enclosure. Circular holes are for the drive units. Sound output comes from the open end of the port S_P. A smaller model using only one drive unit may be constructed by dividing the drawing in half along the centre line of the port tube. The speakers are mounted nose to tail.

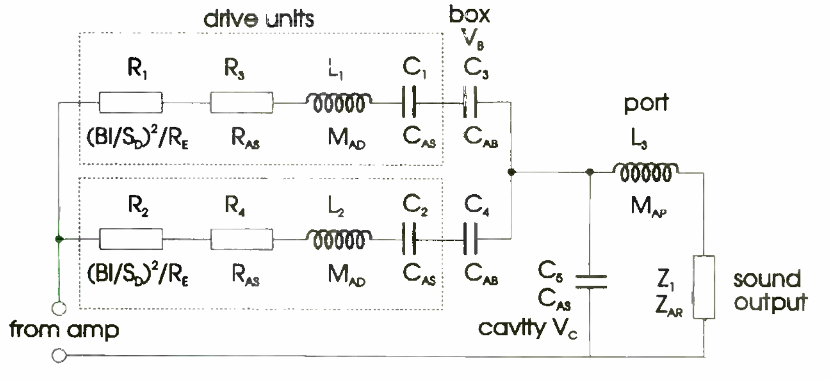


Fig. 4. Acoustic circuit corresponding to Fig. 3. Since all the components move together, they displace the same amount of air and have the same velocity.

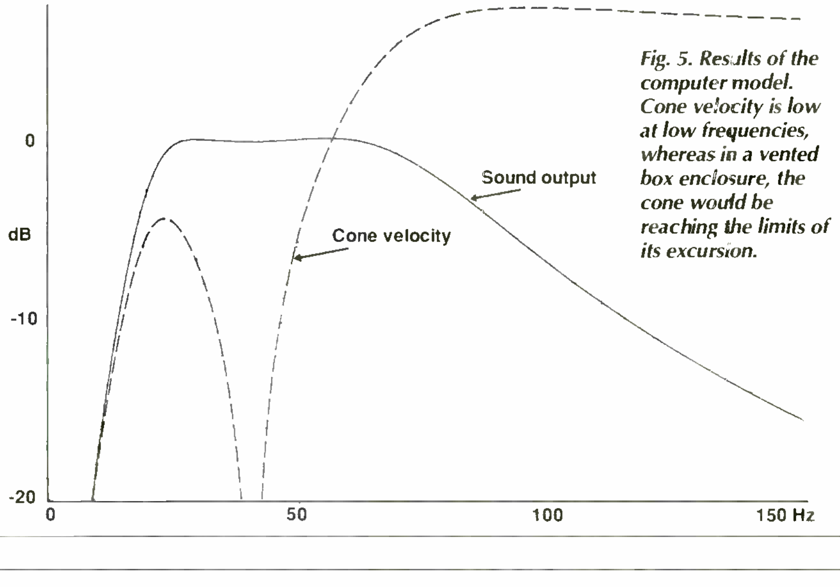


Fig. 5. Results of the computer model. Cone velocity is low at low frequencies, whereas in a vented box enclosure, the cone would be reaching the limits of its excursion.

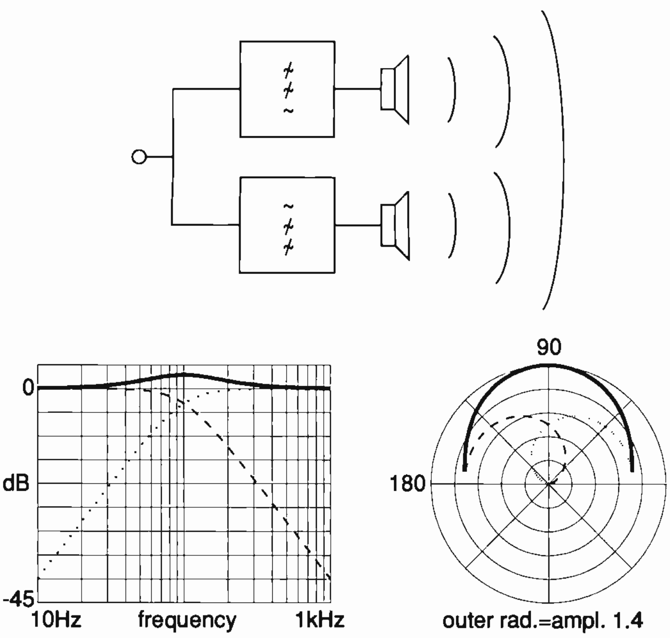


Fig. 6 In a normal crossover network for loudspeakers, signal is split into low-frequency and high-frequency components using standard filters, and the components are then recombined (solid line). The recombined signal has a peak at the crossover frequency, and there is a 180° phase shift across the band.

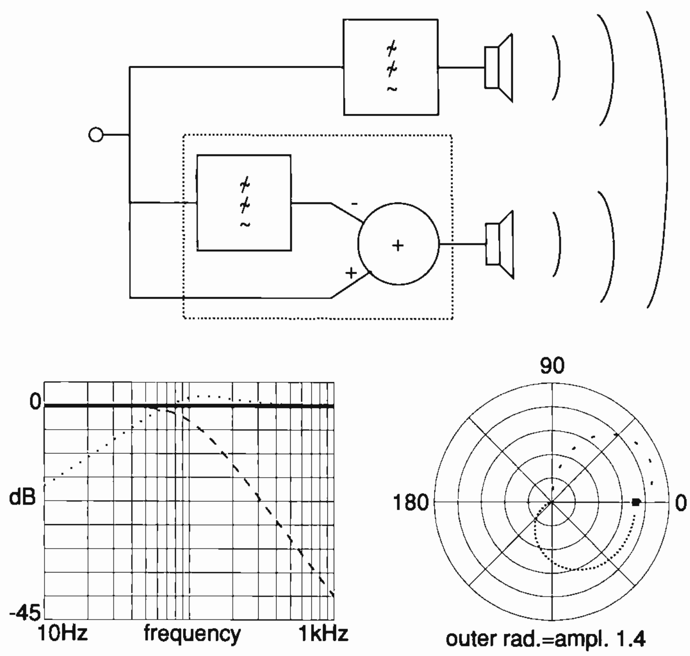


Fig. 7. Using the high-pass filter described in the text, when low and high-frequency components are recombined, the resulting signal is flat across the band and has no phase shift.

S_P is 0.017m^2 by length of 0.5m. This gives an almost ideal response, **Fig. 5**, with slightly under-damped low and high frequency roll-off.

Electronic filter design

Electronic circuitry is required before the power amplifiers to separate the electrical drive for the sub-woofer from that for the midrange and treble. However, there can be pitfalls if this is done without care. In a loudspeaker crossover network, the electrical signal is separated into low- and high-frequency components using a low-pass and a high-pass filter. The two components are recombined by summation in the listener's ear.

A simple circuit simulation using second-order filters with Q=0.7, **Fig. 6**, shows that, surprisingly, the recombined signal does not have a flat frequency response. The low frequency component is also in antiphase to the high frequency one. This is usually corrected by reversing the connections to one speaker drive unit – in which case the frequency response of the recombined signal has a dip.

If the circuit is modified so that the high-pass filter is made from a duplicate low-pass filter plus a differential amplifier, **Fig. 7** then the effect of the identical low-pass filters cancels out exactly when the signals are recombined at the ear. The resulting frequency response is perfectly flat and the phase of the response is 0° at all frequencies.

Figure 8 shows the filter circuitry for the coupled cavity subwoofer. Low-pass filter action is accomplished by the speaker acoustic circuit, so no high-power crossover network is required. The duplicate low-pass filter is a Sallen and Key op-amp filter, U_2. Subtraction is performed by U_3: the output from U_3 is then taken to the midrange power amplifier.

Drive for the sub-woofer drive is obtained by summing the stereo channels in U_{4A}. A phase shifter, comprising U_{4B} and U_5, may be included to correct for the room position of the sub-woofer. It comprises a very low-Q LCR all-pass filter using the gyrator U_{5B} as the inductor, C_6 and R_{28}. It gives 180° adjustment range; the other 180° is covered by swapping over the speaker cables.

Construction

Chipboard or medium density fibreboard are suitable materials for the enclosure. The two large unbraced panels should be 25mm thick, while the remainder is 18mm. KEF suggests connecting the drive units with a metal rod to give extra stiffness, but this requires custom-designed drive units. Here the port tube provides the bracing.

Length t of the port tube (see Fig. 3) includes the length of the bend where it joins the cavity V_C – a component not included in the acoustic circuit. Damping material should be fixed to the inside of the panels to reduce cabinet vibration and to eliminate high frequency cavity resonances, since there is no crossover network. I used fibreboard pressure treated with bitumen, generously glued all over. This was followed by a 50mm layer of open cell polyester foam cushion stuffing. In practice, this also favourably reduces the Q of the response shape.

The port tube should not be lined. The bottom face should not be glued, but screwed on to allow interior access, using a foam gasket. All other joints should be glued and screwed using 25mm square battens, ensuring airtight joints. A grille cloth may be added, but the material should offer minimum air resistance as the air volume velocity in the port is high.

References

1. Bailey, A.R., *Non-Resonant Loudspeaker Enclosure Design, High Fidelity Designs, Wireless World*, 1974, pp 87-90.
2. Berriman, D., *Sound Concepts* , *Electronics World and Wireless World*, Sep. 1990, pp 774-779.

Further reading

Beranek, L.L., *Acoustics* , American Institute of Physics, 1986.
Small, R.H., *Closed-Box Loudspeaker Systems, Journal of Audio Engineering Soc.*, 2 parts, Dec. 1972, Jan/Feb. 1973.

Coupled-cavity design software

Turbo Pascal listings for plotting frequency response and time-domain square-wave response of a twin-driver coupled-cavity speaker is available on disk. It can be obtained by sending £10 plus vat to *EW&WW*'s editorial offices at the address in the front of the magazine. This PC format software is not a fully-worked speaker design package but rather intended as a template for the computer-literate experimenter. It comprises three pascal files, each over 5K. Documentation is limited to REM statements in the listings.

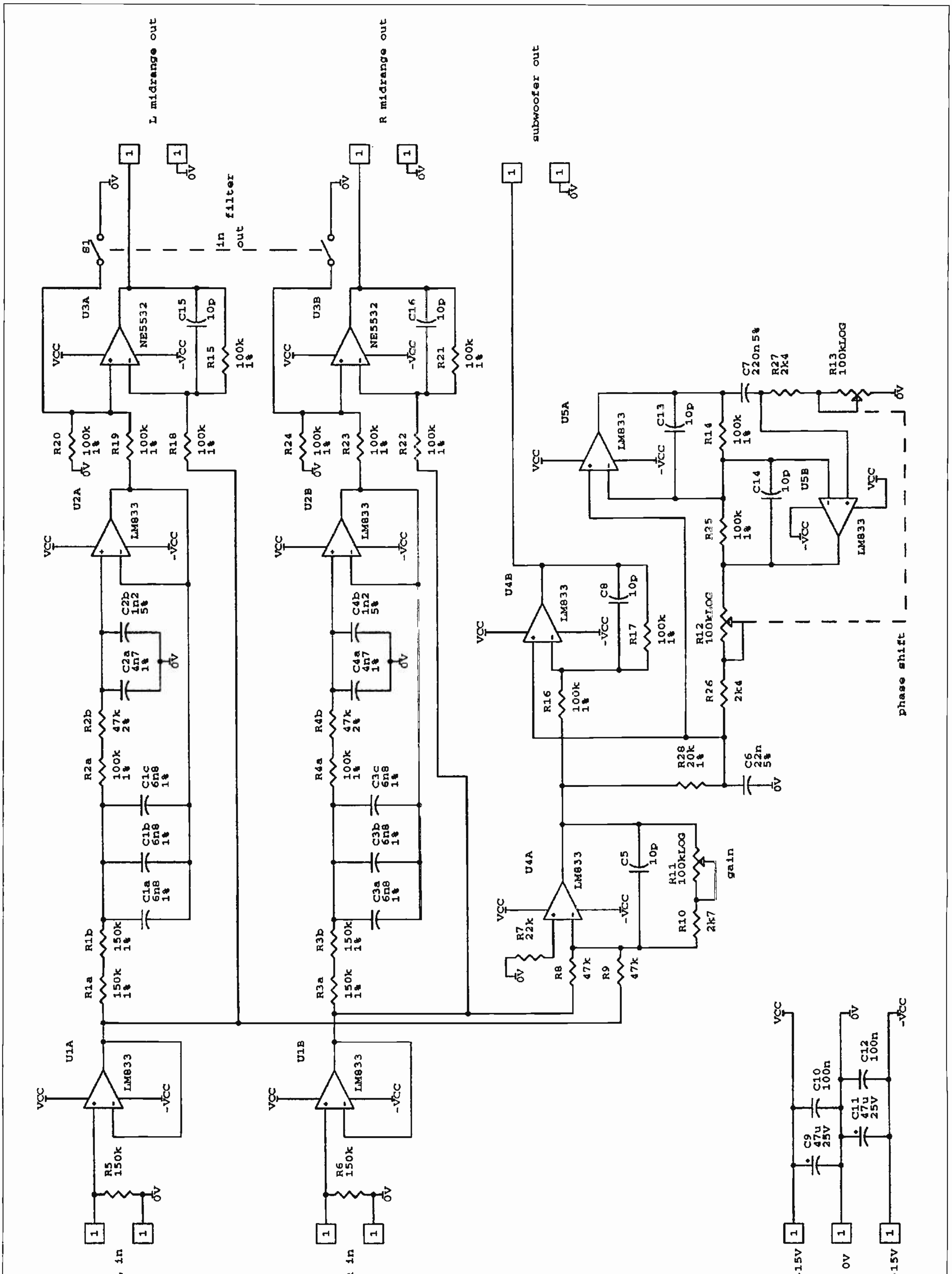
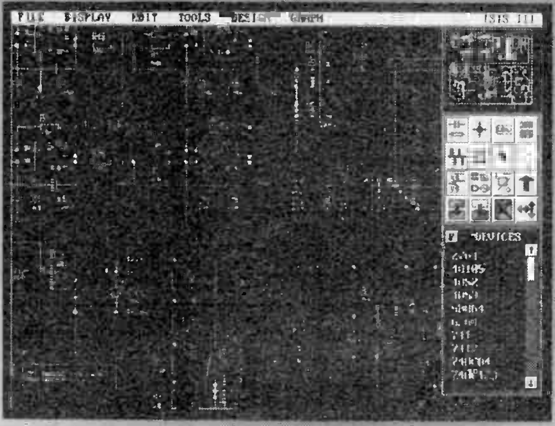


Fig. 8. Phase control is added to this electronic crossover to compensate for the position of the subwoofer relative to the main units.

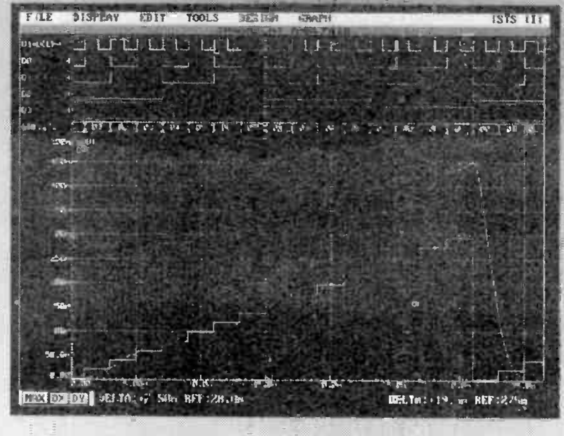
PROTEUS

The Complete Electronics Design System



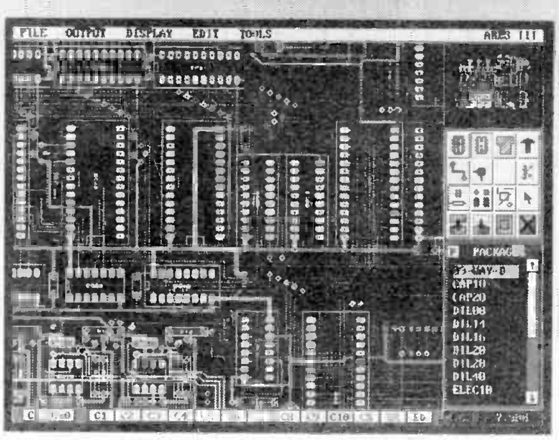
Schematic Capture

- Easy to Use Graphical Interface.
- Netlist, Parts List & ERC reports.
- Hierarchical Design.
- Extensive component/model libraries.
- Advanced Property Management.
- Seamless integration with simulation and PCB design.



Simulation

- Non-Linear & Linear Analogue Simulation.
- Event driven Digital Simulation with modelling language.
- Partitioned simulation of large designs with multiple analogue & digital sections.
- Graphs displayed directly on the schematic.



PCB Design

- Multi-Layer and SMT support.
- Unlimited Design Capacity.
- Full DRC and Connectivity Checking.
- Advanced Multi-Strategy Autorouting.
- Output to printers, plotters, Postscript, Gerber, DXF and DTP bitmaps.
- Gerber View and Import capability.

labcenter
Electronics

Write, phone or fax for your free demo disk, or ask about our full evaluation kit.
Tel: 0274 542868. Fax: 0274 481078.
14 Marriner's Drive, Bradford BD9 4JT.

Proteus software is for PC 386 compatibles and runs under MS-DOS. Prices start from £475 ex VAT; full system costs £1495.
Call for information about our budget, educational & Windows products. All manufacturers' trademarks acknowledged.

CIRCLE NO. 108 ON REPLY CARD

J P Macaulay has thrown his subwoofer away, arguing that the best route to good low-frequency response is to combine the advantages of a small reflex enclosure with electronic compensation.

BIG BASS Small box

It is now nearly seventy years since the introduction of the moving coil speaker and small speaker systems with extended bass response are still rare. In a way this is not surprising. A thorough understanding of how speaker systems operate in the bass region has only been available since the 1970s. In addition, the work of Thiele[1], extended and enhanced by Small[2] and others, has only recently been widely disseminated. This work forms the backbone of modern lf speaker design.

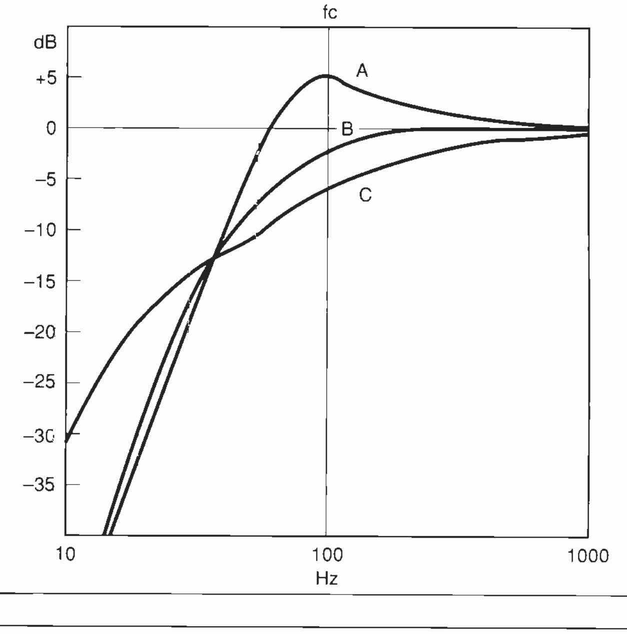
An unmounted loudspeaker can be modelled by an electrical second order high pass filter. Typical response curves are shown in **Fig. 1** Note that the response shape depends on the Q of the filter. Usually, the maximally flat Butterworth filter is assumed to be the best since it combines maximum pass band response flatness without peaking. However the best transient response is obtained with a Q of 0.5, although with the penalty of a drooping lf response.

Filter action is a byproduct of the speaker's fundamental resonance. This resonance occurs between the mass of the cone and the compliance of the cone surround. Response curve Q is mainly determined by electrical damping, or lack of it, imposed by the magnet assembly.

Antiphase bass radiated from the rear of the cone cancels radiation from the front. Sealing the rear of the speaker in an enclosure solves the problem but at the expense of increasing the driver's resonant frequency. This is due to of compliance of the enclosed air effectively stiffening the surround.

Using Thiele/Small theory, the response of

Fig. 1. Typical curves for a sealed loudspeaker enclosure. Designers usually strive for a Q of 0.7 as in curve B. Curve A is for a Q of 2 while curve C is 0.5.



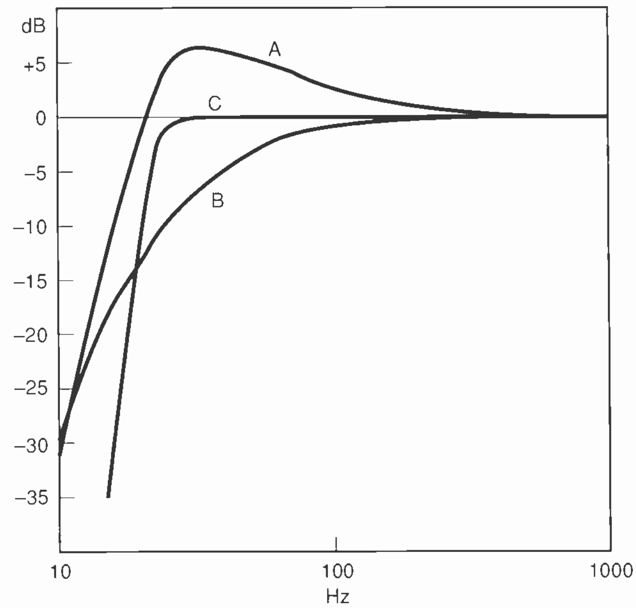


Fig. 2. Sixth-order curve C is system response produced by combining fourth-order roll off of a reflex speaker with second-order electronic low pass filtering.

such a system can be easily be calculated – at least for the bass region. Three basic parameters of the driver are needed. First is the bass resonant frequency, f_s, measured in free air and second V_{as}, the volume of air whose compliance is equivalent to the compliance of the speaker. Finally the parallel equivalent of the mechanical and electrical bass resonant Q_s, or Q_{ts}, needs to be known. Nowadays any reputable driver manufacturer should be able to supply these details on their products.

The other widely used method of mounting a speaker is to introduce a duct or port into the otherwise sealed enclosure – the bass reflex system – as detailed in the panel. By juggling the speaker parameters, enclosure volume and port tuning it is possible to devise a good reflex system. But due to physical restraints, most affordable, reasonably-sized systems end up with a –3dB point somewhere between 50 and 70Hz. Considering that most program

Reflex enclosure design

Rather than being airtight, the reflex enclosure has a duct or vent. Air mass in the duct forms a resonant circuit with the compliance of the air in the cabinet – a form of the Helmholtz resonator.

At resonance, the reflex port inverts the phase of the cone's rear radiation so the duct radiates sound in phase with the front of the speaker. This action increases efficiency at low frequencies. As an added bonus the speaker sees a high impedance at the resonant frequency of the cabinet and cone excursion is greatly reduced for a given acoustic output. Furthermore the resonant frequency of the speaker mounted in the cabinet is hardly raised at all from its free air value.

There is a trade off with the reflex design. The speaker system now responds like a fourth-order high-pass filter with potentially worse transient response.

Response of a reflex enclosure can be determined from the following relationships.

$A=(f_b/f_s)_2$
$B=AQ_{ts}^{-1}+f_b(Q_i f_s)$
$C=1+A+V_{as}/v_b)+f_b(Q_i f_s Q_{ts})$
$D=1/Q_{ts}+f_b/(Q_i f_s)$

For any given frequency f, the relative response in dB (dB=20log$R$) can be found from,

$$R=\frac{f_n^4}{\sqrt{(f_n^4-Cf_n^2+A)^2+f_n^2(Df_n^2-B)^2}}$$

where f_s is free air resonant frequency of the driver, f_b is resonant frequency of the enclosure and port, V_{as} is the volume of air whose compliance is equal to that of the driver, Q_{ts} is Q of the driver's bass resonance, Q_i is a measure of box losses and can be taken as 7 for normal enclosures, F_n is f/f_s and v_b is enclosure volume.

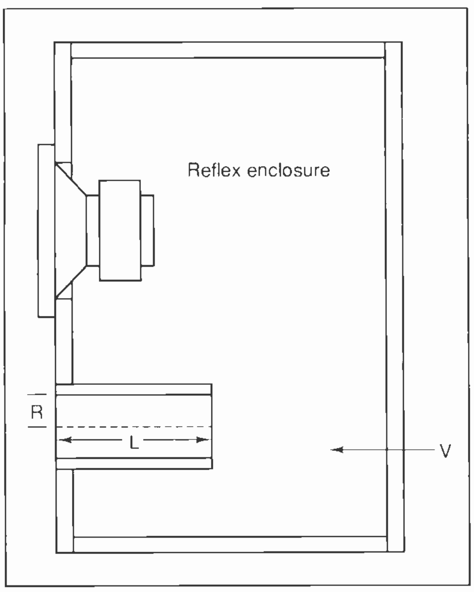
The resulting curve resembles that of a fourth-order filter in the case of the optimum enclosure but closely follows that of a second-order filter for very small enclosures. Small in this case refers to the V_b/V_{as} ratio.

A closed box system has a straightforward second-order frequency response in the bass region. This can be calculated from the following.

$A=(f/f_c)^2$
$B=(A-1)^2$
$C=f/(Q_o f_c)$
$D=C^2$
$E=\sqrt{(B+D)}$
$f\omega=A/E$
$N(\text{dB})=20\log(f_\omega)$

where f_c is resonant frequency of the driver in its enclosure, Q_o is the resonance and f is the frequency of interest.

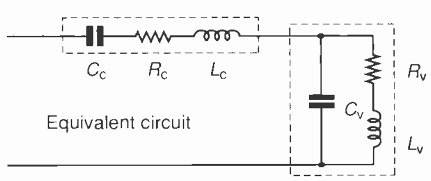
When the enclosure is sufficiently small to produce a second-order response, the curve is analysed and the equivalent f_c and Q_o determined. This information is applied to check the accuracy of the deduction by calculating the equivalent infinite-baffle response, inverting it and adding it to the enclosure response. Enclosure size is determined mainly by the required cut-off frequency.



Tuning the reflex. The reflex enclosure resonant frequency is independent of the speaker fittted. Knowing the vent dimensions it can be shown that the resonant frequency f_c is

$$f_c=\frac{344.8\,R}{2\sqrt{\pi V_b}\,(L+1.7\,R)}$$

R = vent radius
L = vent length
V = cabinet volume



where
C_c =speaker compliance
L_c =cone mass+air load
R_c =radiation resistance
C_v=boxair compliance
L_v=vent air mass
R_v=vent radiation resistance

sources have a bass response that extends to 20Hz or lower, at least an octave of musical information is lost.

To make up the shortfall in response, subwoofers are becoming popular. Nevertheless in order to respond down to the lowest octave such systems need to be large. The only sane alternative way of extending bass response is to use equalisation. I hasten to add that I am not advocating the use of graphic equalisers, but rather a precise inverse frequency response tailored to the system being used[3].

Probably the best known example of this kind of system is the sixth-order alignment proposed by Keele[4]. The label 'sixth order' refers to the fourth-order roll off of a reflex speaker whose lf output is augmented by a second-order low pass filter.

Resulting roll off is that of a sixth-order filter, as in **Fig. 2**. Here a reflex cabinet is tuned low to produce an over-damped lf response which is then boosted flat by the auxiliary high-pass filter.

Bandpass subwoofer

While experimenting with a similar system I discovered an interesting approach to the problem. I had a Kef *B200* mounted in a 19 litres sealed cabinet. This 200mm unit has a free-air resonance at 25Hz, a Q_{ts} of 0.51 and a V_{as} of 161 litres. With the speaker mounted in the cabinet, response of the system is that of a second order filter with an f_c of 90Hz and a Q of 1.8.

The system was equalised for a band-pass response flat between 30Hz and 100Hz to –3dB. This was made possible by feeding the input signal through a low-pass second order filter, **Fig. 3**. Inspiration for this system was the subwoofer unit presented by Harcourt[5].

Although response was well extended, power handling was inadequate. Of course a reflex speaker in a small cabinet has a second-order response similar to the closed box. I reasoned that I could possibly increase power handling by fitting a suitable vent and adjusting the equalisation.

The result exceeded my best expectations. Power handling was vastly improved. Moreover the amount of deep bass output was amazing and without wind noises from the decidedly small 50mm diameter vent.

Initially I could not account for the performance of the unit until I read an article by Plach and Williams from 1951. It transpires that when a reflex cabinet is made sufficiently small compared to the driver's V_{as} the phase angle between the vent and driver output changes slowly through the vent resonant frequency. The result is that vent radiation is still in phase with the driver's above and below the resonant frequency.

This phenomenon accounts for the unequalised response curve which rolls off slowly down to 10Hz and then rapidly plummets due to the vent and driver radiation being in antiphase. Just as importantly though the vent radiation, in a 'normal' reflex design peaks strongly at the vent's resonant

Linkwitz-Riley filters

The Linkwitz-Riley filter is an ingenious and versatile network originally designed for use with closed box speaker systems. Normally, when designing such a system the final response curve in the bass region is determined by the driver parameters. Unfortunately, with the majority of drivers a small enclosure means a high cut-off frequency combined with an undesirable peak in response.

A well designed Linkwitz-Riley filter provides bass boost and correction for the peak. The filter is placed before the power amplifier driving the speaker system. The net result is a system that will respond deeper into the bass region, with a better transient response than that of the original speaker driven by the power amplifier alone.

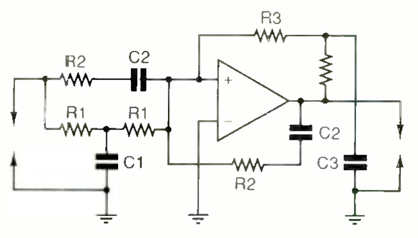
In a closed box system the available acoustic output at low frequencies is limited to the amount of air that can be shifted. This in turn is limited by the maximum displacement of the cone. Most modern long-throw drivers are limited to a linear excursion of 6mm peak to peak. Typically, damage will occur at twice this limit so it is vital to ensure that this cannot occur. Contrary to popular belief the amount of power required at low frequencies is substantially less than the driver's normal rating. For every halving of input frequency the cone excursion will double for a constant drive voltage.

Luckily the low frequency content of musical signals diminishes rapidly below 100Hz. Another objection to bass boosting is that, below resonance, the output is controlled by the linearity of the cone suspension. However research has shown that 40% THD is inaudible on program at 40Hz*.

With the design proposed – a severely overdamped reflex system – efficiency is substantially higher because of the contribution of the vent. A feature of reflex systems is that the resonant frequency is only slightly higher than the free air value so distortion due to sub-resonant operation is minimal. What distortion there is substantially reduced by the high impedance air load provided by the vent.

**This will apply to first harmonic content only – Ed.*

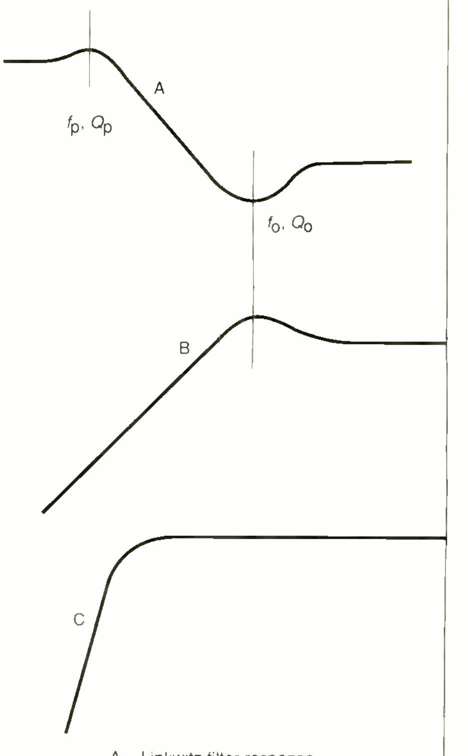
Linkwitz filter



Design equations

- Choose f_o, Q_o, f_p, Q_p
- $k = (f_o/f_p) - (Q_o/Q_p)/(Q_o/Q_p - f_p/f_o)$ $k < 1$
- Choose C_2: $R = 1/6.28\ f_o\ C_2\ [2\ Q_o\ (1 + k)]$
- $R_2 = 2\ k\ R_1$
- $C_1 = C_2\ [2\ Q_o\ (1 + k)]^2$
- $C_3 = C_1\ (f_p/f_o)^2$
- $R_3 = R_1\ (f_o/f_p)^2$

Max network gain = 40 log (f_o/f_p)
k>1 required



A = Linkwitz filter response
B = Speaker response
C = System response

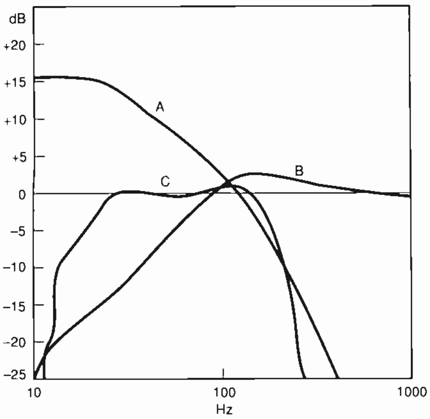


Fig. 3. Reducing the size of the reflex enclosure produced second-order characteristic B. Adding electronic second-order low-pass transfer function A produced a bandpass system exhibiting curve C.

frequency. With the system described here the Q of this peak is much reduced, **Fig. 4**. It follows that the vent can be made smaller because of the lower sound velocity at resonance without generating wind noise. I have found that a vent area $^1/_{16}$ of the driver radiating area suffices.

Just as important is the inherently low rate of roll off in the system's stop band. We have a reflex system with the roll off characteristics of a sealed box and with it the better transient response of a sealed enclosure system.

Building a full range system based on the principle proved to be quite simple. All that is needed is a Linkwitz-Riley filter[3]. This network is designed to compensate for a given second-order response and substitute it with another of the designer's choosing.

Originally the Linkwitz-Riley network was designed for use with closed box systems. However its second-order response curve is easily calculated. All that remains before designing the network is to determine the equivalent f_0 and Q of the unequalised system response.

The system has the most desirable characteristic of a sealed speaker cabinet – superior transient response. Since the vent and enclosure need to be small to obtain the correct response, the cabinet can be made unobtrusive. It has an extended low frequency response combined with low cone excursions due to the favourable phase shift between driver and vent. Because the final response curve is equalised by an external filter, drivers with relatively high f_s and Q_{ts} – i.e. cheap ones – will suffice.

Full-range system details

In my design, called the Microflex, I used a pair of Kef *B200*s for bass and mid range together with a pair of Audax *DTW 100 T1 25 F FFG* tweeters.

My enclosure is made from 15mm medium density chipboard panels with overall dimensions of 355mm high by 258mm wide by 303mm. This yields an internal volume of 19.96l. The cabinet is unlagged and tuned to 34Hz by a 215mm long 51mm diameter vent.

The Kef speakers have a high frequency roll off above 3.5kHz accompanied by a rather large response peak. Roll off is that of a second order filter with a Q of 3.5. I had decided from the outset that the speaker system was to be active so I turned this roll off into an advantage. I fed the bass signal through a 3.5kHz low pass second order filter with a Q of 0.143. This combination gives an acoustic response from the *B200* of a fourth order low pass with a Q of 0.5.

This is ideal since the aim was to implement a Linkwitz-Riley-Riley crossover. This type of crossover uses fourth order filtering for both high and low pass sections combined with a Q of 0.5 for the best possible transient response. My tweeters are titanium dome types with a natural resonance frequency of 1.2kHz which is over an octave below the crossover frequency. Filtering for the tweeter is provided by a cascaded pair of Sallen and Key second-order high pass filters built around a dual op-amp, **Fig. 5**.

At the bass end, the input signal is first buffered before being processed. Active filters only operate as intended when driven from a low impedance source. The signal is then fed into a second-order high-pass filter with a turnover frequency of 30Hz and a Q of 1. This filter is an optional extra and is most useful when using vinyl discs.

Bass information is now fed to the Linkwitz-Riley filter. Values shown are determined empirically. From here the signal is fed into the low pass filter previously described, which forms part of the low pass crossover network. Also fed from the input buffer are the cascaded high-pass crossover filter sections. From here, output is fed to the power amplifier via a preset potentiometer to equalise signal levels to the speakers.

Bass response is flat down to 30Hz and the system can produce a good acoustic sine wave at this frequency. Transient response is good and system performance is good on all material – especially speech. I have dismantled my subwoofer as it is no longer necessary. I now feel that the real historical significance of Thiele/Small theory will be showing us where to set our equalisers. ■

References

1. Thiele, A. N., *Loudspeakers in vented boxes, J. Aud. Eng. Soc.*,Vol 19. May/Jun 1971.
2. Small, R.H., *Closed box loudspeaker systems, J. Aud. Eng. Soc.* Vol 21, Jun/Sep 1973
3. Linkwitz-Riley, S., A *three enclosure loudspeaker system*, parts 1-3, *Loudspeaker Builder* 2-4, 1980.
4. Keele, D. B., A *new set of sixth-order vented box loudspeaker system alignments, J. Aud. Eng. Soc.*, vol 23, pp354-360 June 1975.
5. Harcourt, R. I., *An acoustically small loudspeaker, Wireless World,* Oct. 1980.

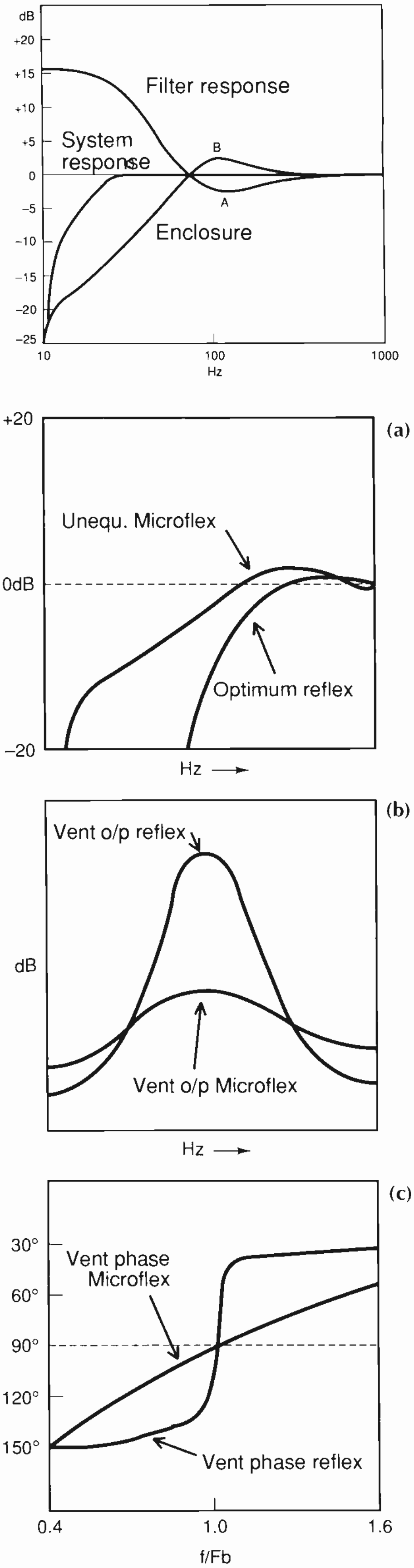
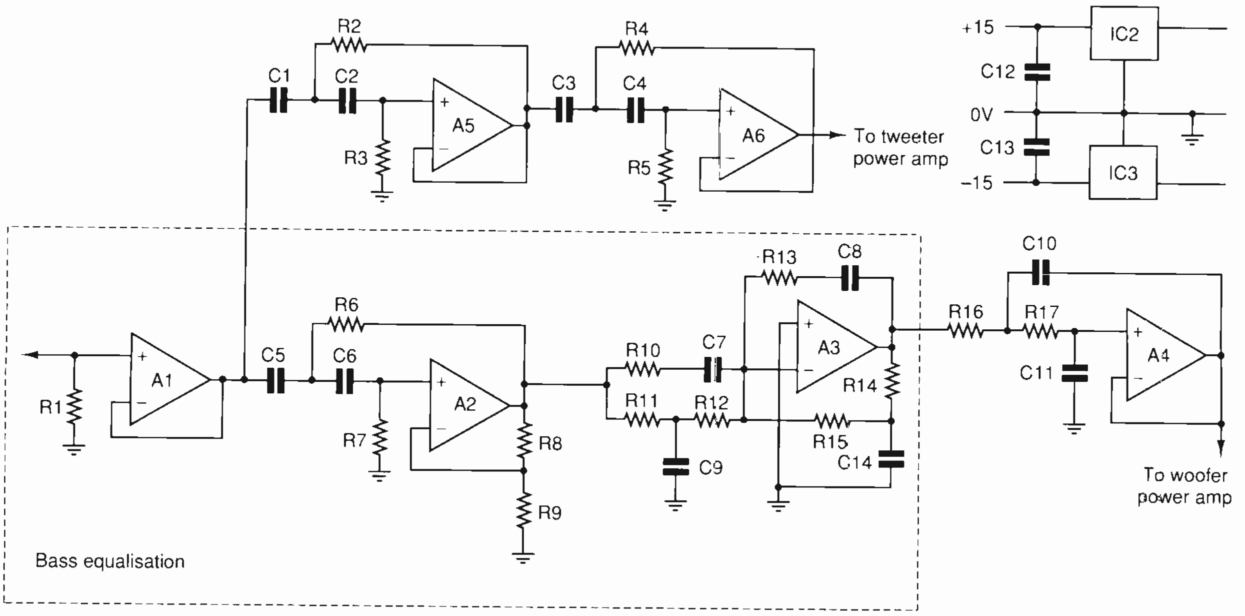


Fig. 4. Curves for the electronically augmented reflex enclosure show that response is flat down to the cut-off point.



Values

Resistors, 1%

$R_{1,6,7,10,13}$	56k
$R_{2,4}$	12k
$R_{3,5}$	22k
$R_{8,9}$	15k
$R_{11,12}$	100k
$R_{14,15}$	820k
$R_{16,17}$	4k7

Capacitors

$C_{1\text{-}4}$	2n7
$C_{5,6}$	100n
$C_{7,8}$	4n7
C_9	10n
C_{10}	3n3
C_{11}	33n
$C_{12,13}$	100n cer
C_{14}	68n

Semconductors

$A_{1\text{-}6}$	*TL072* (3off)
IC_2	*7815*
IC_3	*7915*

Fig. 5. Linkwitz-Riley crossover with electronic woofer equalisation. This circuit compensates for low-frequency response deviation in a small reflex enclosure – Microflex. Further optional filtering removes sub-sonics for disc replay.

The spirit of BASS

Unlike traditional Western music producers, many modern artists rely heavily deep bass for exciting their audiences. Ben Duncan discusses why U2 never quite sounds the same in your living room.

Hard and fast lines cannot be drawn, but bass energy in music starts somewhere around 300Hz, contains over half the octaves that music spans, and turns into sub-bass – something you feel more than hear – below about 30Hz.

Bass energies have been important to mankind for more than 10,000 years. The rock festival that can be heard miles away uses kilowatts of electricity to parallel tribal drumming ceremonies, audible at over ten miles. But the lowermost three octaves, 10-80Hz, are rarely appreciated in the majority of Western listening, which takes place on compromised, so called high-fidelity systems.

While it is superficially possible to appreciate much 'classical' Western music largely without these lower registers, the fulsome reproduction of bass profoundly enhances and expands the experience and consciousness of the serious music lover. With an 18th century oratorio or a pipe organ, the low content may only be the grandiose. But most of rhythmic music's therapeutic power resides in the low bass.

There is a widespread misconception among those without a global appreciation of music that very low bass is somehow dangerous*. I have repeatedly experienced outdoors some of the highest musical low-bass sound pressure levels yet created. For myself and many others, the experience is extremely physical, pleasurable and ultimately cathartic.

Bass – the acoustic background

Rudimentary audio knowledge talks of music requiring a –3dB bandwidth that begins at 20Hz or even higher. It is true that few traditional instruments generate fundamentals below 40Hz, but this does not mean that they produce no output. Moreover, pipe organs, gongs and synthesizers, as well as diverse sounds that have been sampled and pitch shifted, as used in today's House and Ambient music, reach down to at least 20Hz, if not below 10Hz.

** I believe the misunderstanding arises from Lyall Watson's first book[17] in which he reported how experiments with continuous subsonic waveforms – not musical 'sub-bass' – demonstrated destructive and fatal qualities.*

In domestic reproduction, the relevance of frequencies below 100Hz and the ability to recreate them in any ordinary living room is debatable. In many cases, they are not realisable on at least three counts. First, the LF response of two-way domestic speaker systems is stymied by acceptable size. Second, the maximum sound pressure level that can be developed in the bottom decade of domestic speakers is usually severely limited by driver excursion limits. Particularly in sealed box enclosure, the cone moves proportionately further with decreasing frequency for a given sound pressure level.

A continuum of trade-offs exists between piston area, excursion and allowable harmonic, **Fig.1**, and Doppler distortion. So the average domestic speaker capable of 105dB at 1kHz is likely to be limited to 70dB or less below 50Hz. A speaker that has been tuned to go lower will have less maximum output since extension is traded against excursion.

Third, and often overlooked, Robinson and Dadson curves, **Fig. 2**, show that the threshold of human perception at 20Hz – the minimum audible frequency – is around 75dB pressure level. This suggests that a speaker has to produce more than 70dB at 20Hz for programme at this frequency to even begin to be audible.

On the other hand, a number of specialist bass enclosures and sub-woofers are capable of producing useable bass to well below 40Hz with dimensions suitable a domestic setting[1] and smaller than elementary theory suggests is possible.

It is sometimes thought that our perception of bass qualities in familiar music is greatly influenced by the ear's ability to synthesise missing or attenuated frequencies using the mid frequency harmonics as cues. It follows that accurate mid-frequency reproduction has as much influence on bass sonics as the bass *per se*.

Synthesis of the fundamental explains how mini and micro-monitors and the majority of small domestic speakers can satisfactorily portray bass lines while being incapable of reproducing the fundamental audibly. Still, the mind has to work to achieve this, and some of music's higher qualities are lost.

Physical presence of the lower octaves, say –3dB at or below 50Hz, is found more relaxing and preferable to most listeners. Equally, deficiencies, excesses or imbalances in the HF are inversely reciprocated by our senses: Listeners may hear an excess of low bass as a lack of treble, and *vice-versa*[2].

It is also not widely appreciated that the rule of thumb that 1dB is the smallest perceptible change to the average listener is true only at conversational pressure levels, at mid-frequencies. Smaller changes are exaggerated at higher and particulary lower frequencies. At mid-frequencies, a 10dB increase from 65dB to 75dB is perceived as a doubling of loudness, but the same change at 20Hz would be perceived as more than a loudness quadrupling.

This is evident from Fig. 2, by following the distance between the curves. It follows that whereas electronics and audio both use log 20 decibel scales for discussions and comparisons of attenuation, gain and amplitude, the *meaning* of audio equipment figures in decibels must be considered.

Ambient sounds are important cues, and while not musical *per se*, they are part of the hologram of up-market hi-fi reproduction. They include the 'feel' – almost a throb – of the wooden stage or floor that the musicians are on. These are difficult to describe, but obvious when present.

System roll-offs, capacitors and delays

Most audio equipment contains too many dc blocking capacitors. The only positively mandatory series capacitors in audio are one for analogue disc, usually RIAA/IEC, and one for each standard of tape equalisation. At most, up to three series capacitors are essential. These are for up to –18dB/octave high-pass filtration to roll-off unwanted bass and even these should be preferably bypassable.

Every series capacitor in the chain creates a high-pass pole. As a result, the response falls off below xHz at -6dB/octave times the number of series capacitors – assuming all roll-offs are the same within an octave or two. Since filtering caused by dc blocking is wholly passive, it is at least highly damped (sub-Bessel).

With active high-pass filters, non linearities in degrees phase per hertz (with frequency plotted linearly) are compounded by Q and their gross effect reaches higher into the audio spectrum with every pole.

Setting very low -3dB roll-off points, well below 20Hz, and minimising the number of roll-off as is important on several counts. One is group delay or phase dispersion. In 1990, I published possibly the first analysis of the effects of audio signal chains in the time and frequency domains[3].

Figure 3 shows a portion of lf music program variably delayed by 0 to 8ms. The delay depends on the dominant component after the signal has passed through each of 24 stages having a modest 3dB low-pass roll-off, or f_{3L}, of 2.1Hz. Such response is typical of a conventional, complex recording studio signal path, so it is a feature of most recordings. It was also a feature of earlier hi-fi systems. Fortunately not everyone is sensitive to it.

A well designed modern domestic replay system can have as little as one series capacitor, giving an ultimate roll-off in the electron-

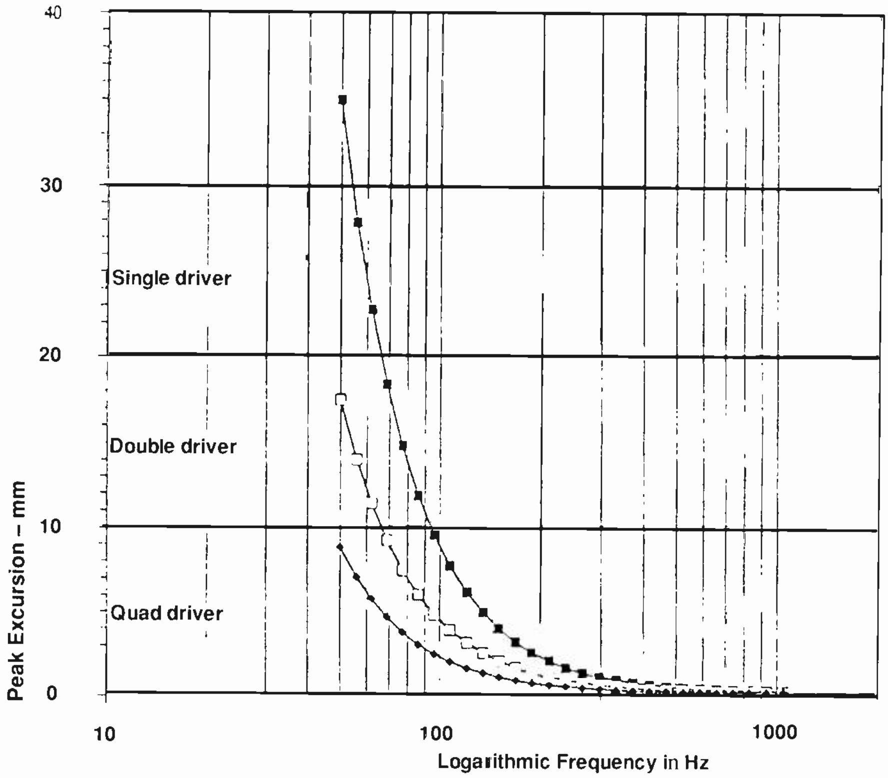


Fig. 1. Cone excursion required to produce 135dB sound-pressure level. Even using four 15in drivers the excursion needed at 40Hz is considerable.

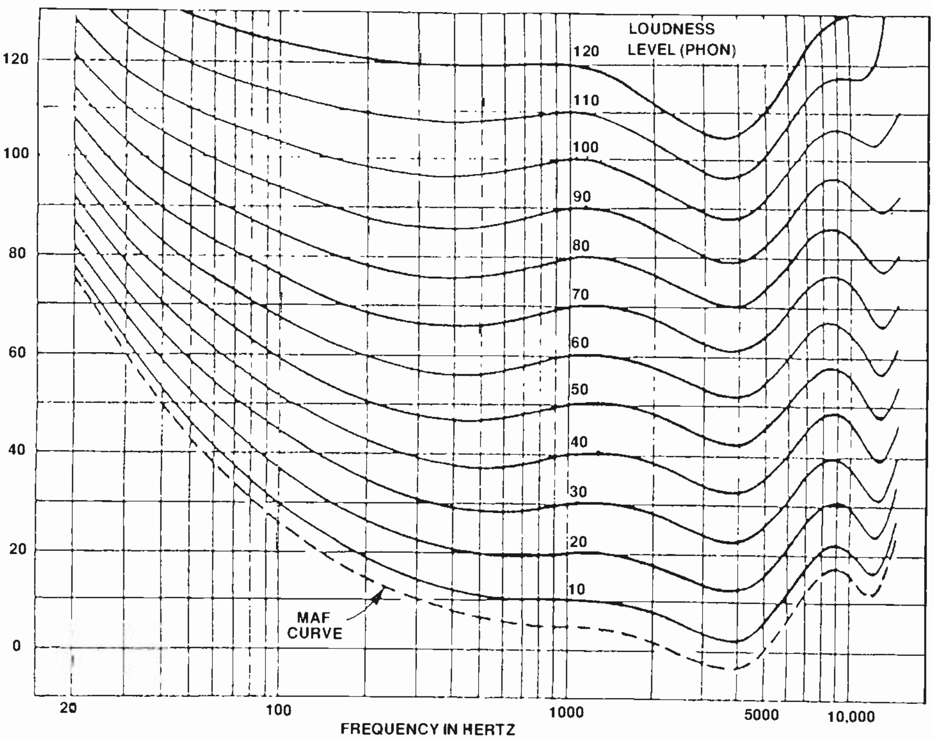


Fig. 2. Often overlooked equal loudness contours from Robinson and Dadson show that a speaker has to produce over 70dB to be detectable at 20Hz.

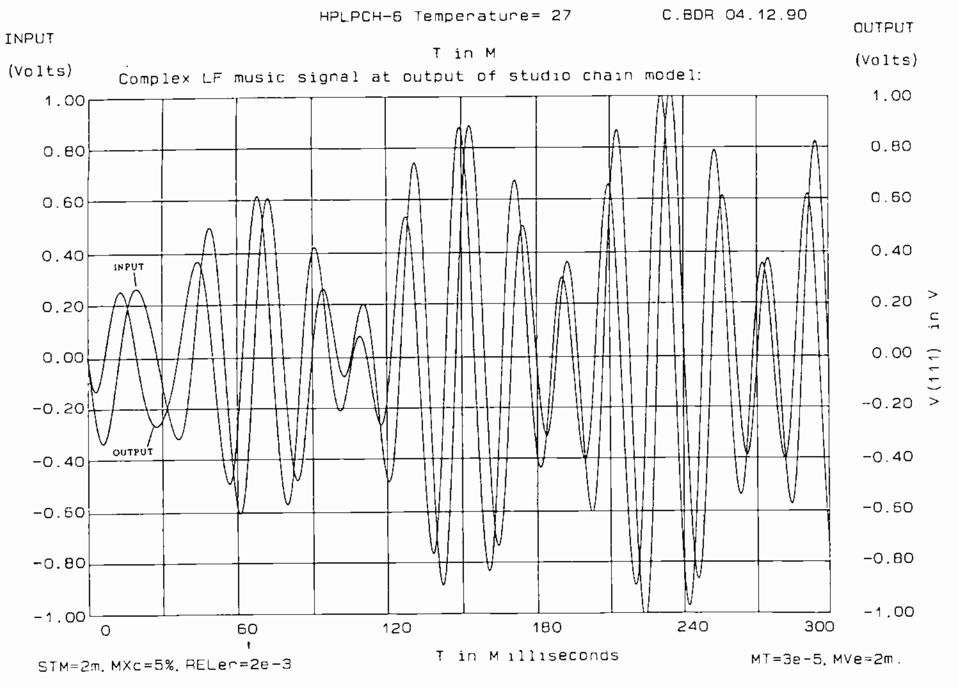


Fig. 3. Phase dispersion illustrated as varying delay in the output of a studio signal chain, compared to the input. Delay varies up to 10ms with dominant instantaneous frequency. In a fully direct coupled system, the delay would be absent.

ics limited to –6dB/octave. Phase dispersion embedded in the recording cannot be fixed this way, but further damage is avoided. In recording studio equipment chains, and wherever more than one series capacitor is essential, phase dispersion is kept at bay by setting F_{3L}<<1Hz.

Properties of real capacitors lead to the same conclusion. An electrolytic capacitor can act as a filter with a huge temperature coefficient, frequency-dependent capacitance and complex, non-linear parasitics[4,5]. The threshold where they cease to act this way is properly set by considering psychoaccoustics, taking the outer limits, and adding a margin of 300% (just 10dB, remember).

For dc blocking in purist audio equipment, electrolytic capacitor values are practically as large as can be borne by the circuit-board area available, while maintaining a resonant frequency that is comfortably above 20kHz. In the few places I use DC blocking, an audio grade, low inductance 1000μF electrolytic capacitor is combined with 10kΩ, giving an F_{3L} of 0.016Hz.

Why not use film capacitors, whose characteristics are nearly perfect ? Well, to attain even a modest F_{3L} of say 1Hz using moderate resistor values, the values needed are unacceptably physically large and expensive – 16μF, for 10kΩ. A smaller and more acceptably priced 150nF would require partnering with 1MΩ for an F_{3L} of 1Hz. But a resistance this high will cause problems with microphony, electrostatic pickup, dc offset and possibly thermal noise.

Microphony is by far the most serious problem. The capacitor allows a distorted acoustic signal to egress at a random point in the chain, causing tonal defects and smearing. This is a common defect in vacuum tube amplifiers, with their high impedance networks. In discrete transistor topologies, some capacitor locations such as bootstraps can cause bias shifts and 'tails' when driven or momentarily overdriven. Asymmetrical music signals can cause this even though the system may behave perfectly with a sinewave (see panel).

It is rarely acknowledged that the temperature coefficient of resistors and particulary the capacitors that set F_{3L}s will cause the system F_{3L}, as well as the phase dispersion, to change as a large audio system warms up. Assume that the net system F_{3L} is set too high at 30Hz – a common feature of mass-market equipment. To astute listeners equipped with a suitable system, a plausible 1dB change in bass response as the system warms up will be perceivable as around a doubling in loudness. But if net F_{3L} is 16mHz, then the same shift will lie below audibility.

Music – the ultimate test signal

Musical information spans at least ten octaves, while mankind's entire visual world is drawn from just the one octave of visible light.

Music signals are the antithesis of the pure sine wave routinely used to test audio equipment. The sine wave has a crest factor of 3dB. The most adventurous test signal used for the majority of 'objective' audio testing is filtered and slightly clipped pink noise. This can crudely imitate music's variable crest factor or 'peak-to-mean ratio', which averages 15dB but varies from around 8dB up to 25dB.

Music is made up of 'shifty' sine waves, many of them harmonically related. Their periodicity, amplitude, frequency and rate of change of these parameters, hence envelope, is some combination of the regular, the rhythmic, the self-similar, the nested, the random and the chaotic.

Direct coupling

In the past decade, the construction of directly coupled audio circuits has been made possible by the arrival of op-amps that combine good ac performance with high DC precision. PMI's *OP37* was the first, followed by Linear Technology's *LT1037* and *LT1028* in 1983.

ll this was made possible by George Erdi's work in combining bias-current cancellation with low-noise input pairs[6,7]. Since then, Analog Devices/PMI, Harris, and Texas have all produced even better parts. But what about drift ? My *AMP-01* preamplifier design of 1983[8] contained up to five direct-coupled opamps in a line. These were individually nulled. After ten years, I can report that drift causing dc levels to build up, to make pots 'scrapey' or cause significant clicks when switching, has not been an issue.

Since then, I have designed a crossover for use in some of the world's foremost recording studios. It has twelve direct coupled stages and requires absolute reliability over thousands of hours operation. Using Harris *HA3-5221-5s* to replace the *NE5534s*, no nulling was required. In addition, bass sonics are vastly improved over the MkI design, which was mostly ac coupled. Servos may be used to force offsets to zero. But this technique just moves the capacitor, adds another, and adds an op-amp with its attendant cost, supply consumption and noise injection.

Power Supply optimisation

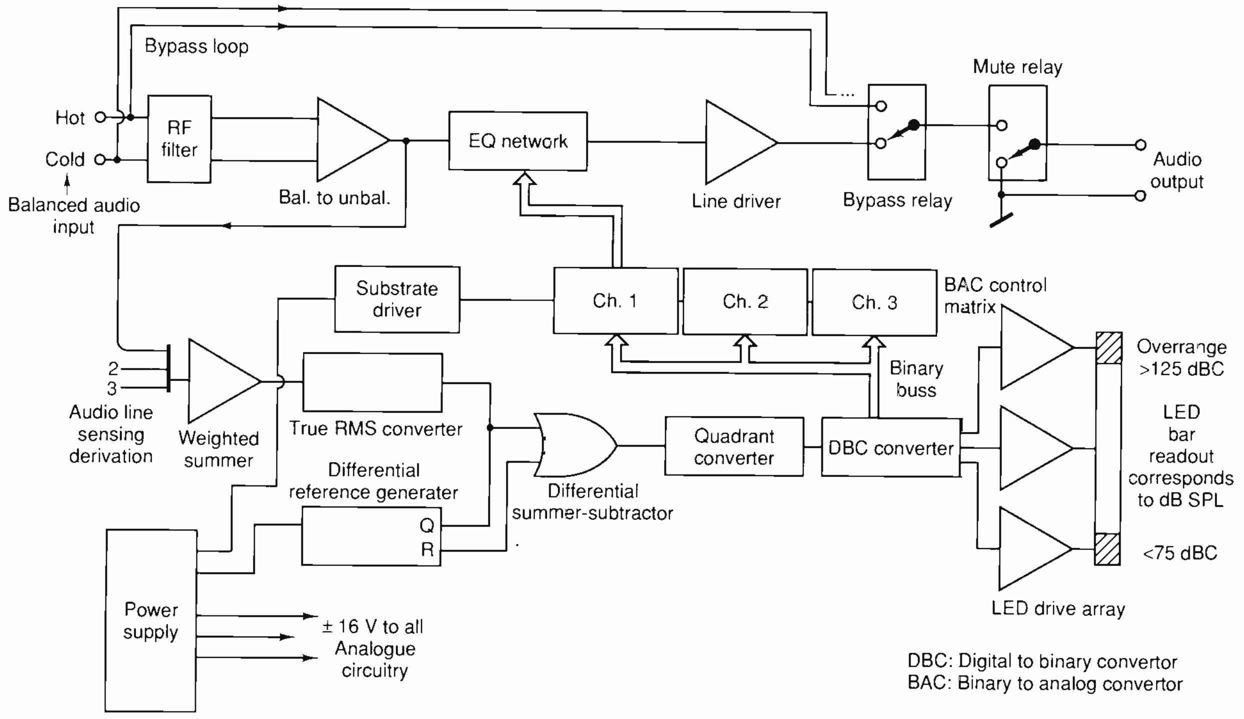
Benefits of regulated supplies with regard to low frequency capability and purity have been covered in some depth[9,10]. Considering envelope modulation, the 'sample rate' of a conventional 50/60Hz power supply interacts with the bass frequencies dominating the music envelope.

In 1986, I helped create the *DVT-300*. This was the first high power – 600W per channel – audio amplifier to use a quasi-resonant switching supply. Switching was at 80kHz, which placed the power supply recharge rate well above the audio band.

Today, several UK amplifier makers use this and other power conversion techniques above 20kHz. Bass clarity has been shown to be improved by exchanging a given output stage circuit between linear and 80kHz switching power supplies.

Dynamic tonal correction

Knowing about the ear's non-linear perception of the 3D change-of-SPL versus SPL versus frequency, instead of frequency leads to the conclusion that sonic accuracy in audio systems hinges on reproducing music at precisely the original sound pressure level (this same effect causes musicians think in terms of pitch). At any other level, the music will be



tonally and dynamically different, and the low bass content will be most affected by the most common practice of listening to reproduced music at 10 or 20dB below the original.

On this basis, turned-down volume controls act to make the music recede into the distance, rather than simply making the musicians shrink. Small tonal differences can masquerade as dynamic differences. The 'loudness' switch on down-market audio equipment attempts to compensate for this. It increases LF (and HF) at low levels, but its operation is based for the most part on false premises[11].

At the very least, any tonal compensation for the purposes of replay at lower than real levels requires some reference to the original sound level. The correct equalisation is certainly not the Fletcher and Munson curve. In 1986, I designed the *Inflexor* – an automatic tonal compensator, **Fig. 4**, which was more correctly referenced psychoaccoustically[12].

While the curves accepted professionally[13] most serious domestic listeners, preferring the minimum signal path, either compensate for the tonal errors in their heads, or listen at what seems to be the original level.

Dynamics

With the exception of COG ceramic capacitors, all components used in analogue electronics vary in characteristics due to the temperature changes that occur in real equipment.

None is more acutely sensitive and fast responding than the semiconductor, particulary one with small junctions. Conventionally, thermal distortion is considered to be an affliction restricted to power amplifiers at bass frequencies. By bass I mean signals that can have periods long enough, above say 10ms, to cause considerable cycle-to-cycle and even sub-cycle temperature changes.

Higher level errors generated by inter-component and inter-stage thermal feedback are mostly limited to monolithic ICs. But even small signal transistors in long tailed pairs reflect their thermal environment in the sonic end-result[14].

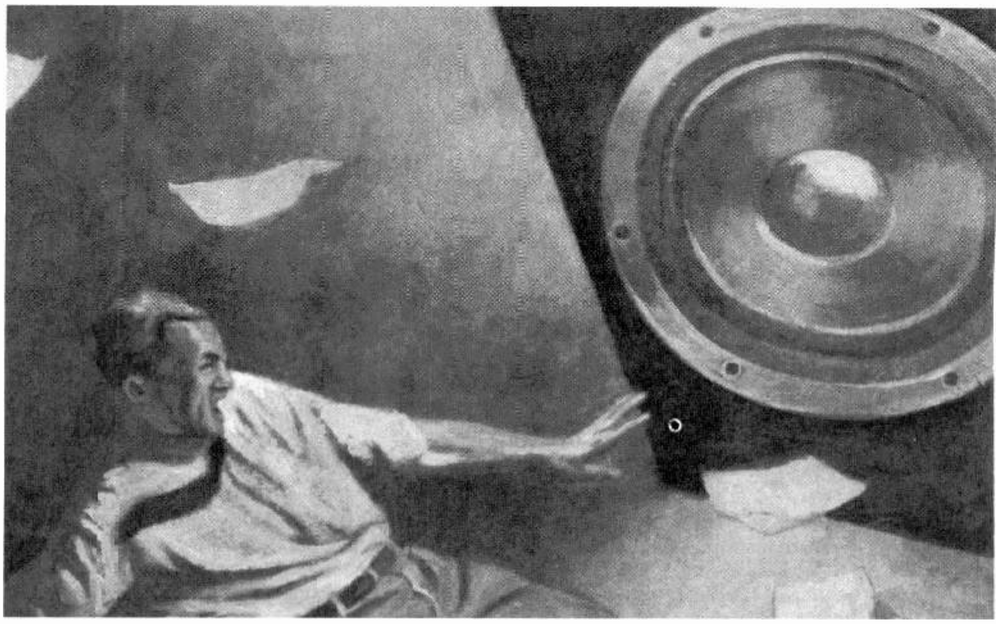
Fig. 4. The Inflexor, a dynamic loudness compensator, allows music to retain its bass vitality when reproduced at up to 40dB below the original sound-pressure level. The short audio signal path can be seen at the top, and below, the analogue computing elements used to track compensatory equalisation. This overcomes the ear's non-linear perception of change-of-SPL versus SPL versus frequency.

Thermal distortion can be measured with sine waves on an open-loop fixture[15], but its real effect on music is more complex, as it can be highly asymmetric (see panel). So for example, one half of an output stage will experience greater cycle-to-cycle changes, creating an error that manifests as even–harmonic residue. Large geometry transistors with thermally conductive epoxy casing have been used to reduce the audibility of thermal tails.

Both analogue and digital ICs can be as sensitive to vibration, by bass in particular, as capacitors. Spectral analysis has demonstrated the value of shock mounting a CD player's digital electronics. In a UK university, laser interferometry has showed that the legs of semiconductor devices dance when passing music, but whether this is caused by acoustic excitation, or by magneto-restriction effects, is as yet unclear. Bob Pease's anecdote[16] about integrated circuits being transparent to infrared radiation and hence modulation by 50Hz ambient lighting flicker or noisy leds, is an example of the potential for left-of-field bass contamination. ■

References

1. Butler, T., *In search of 3D sound, Hi-Fi News*, Nov '93.
2. Philip Newell, *Hi-fi & monitoring: hearing is believing, Studio Sound*, Oct 1993.
3. Duncan, B., *The signal chain, Studio Sound*, June 1991†.
4. Duncan, B., *With a Strange Device*, Hi-Fi News, Parts 2-5, Jun, July, Sept, Nov 1986. (Available in one volume, as *Audio Quality Investigations*, Vol.1 from Audio Synthesis, 99 Lapwing Lane, Manchester M20 0UT. Tel. 061 434 0126.)
5. Hawksford, M., *The Essex Echo, tracks 3 & 4, Hi-Fi News*, Oct 1986 & Feb 1987.
6. Erdi, G. & Cakhnokhi, Y., *A bipolar op-amp with a noise resistance of less than 50 ohms*, IEEE solid-state circuits conference, Feb 1986.
7. Erdi, G., Jung, W., Bernardi, S., Schwartz, T., *Op-amps tackles noise, Electronics Design International*, Dec 1980.
8. Duncan, B., *AMP-01 modular preamplifier, Hi-Fi News*, May-Nov 1984.
9. Duncan, B., *PSU regulation boosts audio performance, EW+WW*, Oct 1992.
10. Hawksford, M., *The Plot Thickens, EW+WW*, Letters, Mar 1991.
11. Holman, T., & Kampmann, F., *Loudness Compensation: Use & abuse, Journal of the AES* July/Aug 1978.
12. Duncan, B., *Dynamic Loudness Compensation*, Reproduced Sound 3, Nov 1987, Institute of Acoustics.
13. Dibble, K., *Greystone Inflexor - psychology of volume restriction, Disco & Club Trade International*, Feb '88.
15. Graeme, J. G., *Designing with op-amps, application alternatives*, pp. 229-231, McGraw Hill, 1977.
16: Pease, R., *Troubleshooting analogue circuits*, Butterworth, 1991.
17. Watson, L., *Supernature – a natural history of the Supernatural*, Hodder & Stoughton, 1973



THE BASS RACE

Over the years, designers have battled with the laws of physics to eke more bass from ever more compact enclosures. Acoustic design consultant David Berriman examines the compromises involved in trying to squeeze out that last decibel.

Ever since the earliest loudspeakers – literally loud-speaking telephones – engineers have tried to extend the bandwidth of the audible spectrum at both ends. While high frequencies have been well-served by small and light diaphragms driven by various means, low frequencies have invariably involved bulky cabinets and/or low conversion efficiencies.

At lower frequencies, sound pressure level is directly related to the amount of air the loudspeaker diaphragm can move. Volume velocity is the product of the radiating element's velocity and area: the higher the figure, the greater the sound pressure. A large cone is therefore desirable but most people want more bass from a smaller box

The obvious alternative is to make the smaller diaphragm move further. Extended travel needs either a very long voice coil in a short gap or a short coil in a long gap. The former wastes amplifier power while the latter requires a much larger, more costly magnet.

Increasing cone area improves coupling between the cone and the air load, increasing efficiency. However the resulting cone is also usually heavier, offsetting the sensitivity gain.

So large area diaphragms are usually required to produce deep, powerful bass. There is no easy way of overcoming this general rule, though many schemes have been devised to achieve the best possible bass for a given size of driver and/or box.

Woofers can be designed to suit specific loudspeaker systems, (or cabinets optimised for woofers) by using a calculator, Thiel Small parameters and a few basic formulae. Far better are modern purpose-designed cad packages such as *Leap*, which achieve the same ends much faster and more accurately.

One thing is certain. Woofers need some sort of enclosure to work properly – and there are plenty to choose from.

Enclosures

Open baffle. At its most basic, this is a large board with the woofer bolted in a hole at the centre. Unfortunately, the pressure wave from the rear of the woofer passes around the baffle and because it is 180° out of phase with the frontal radiation it causes cancellation at lower frequencies. At high frequencies the wavelength is shorter and cancellation does not occur.

Between the two frequency extremes there is a rather uneven low-frequency roll off. Though the woofer on its own is basically a second-order system (and thus rolls off asymptotically to 12dB per octave below resonance) the roll-off for most real open baffles starts well above this frequency and reduces the output by a half for every halving in frequency, which is equal to 6dB per octave. For a circular baffle of radius r, the rear wave must travel $2r$ to reach the front. So, there is a roll-off in output at frequencies where the wavelength is longer than the baffle diameter ($f=c/baffle\ diameter$ where c is the speed of sound in air).

Increasing baffle size lowers the turn-over frequency, but giant baffles are domestically unacceptable. One solution is boosting the output at 6dB per octave below the 'knee', but this dramatically increases the cone excursion. Boosting can be achieved by adding a large inductor or by active compensation.

Bass from such a system can sound very natural. Even so, open-baffle bass loudspeakers are limited to the esoteric end of the hi-fi market and will only ever have a limited following. For popular consumption, most systems come in more compact packages.

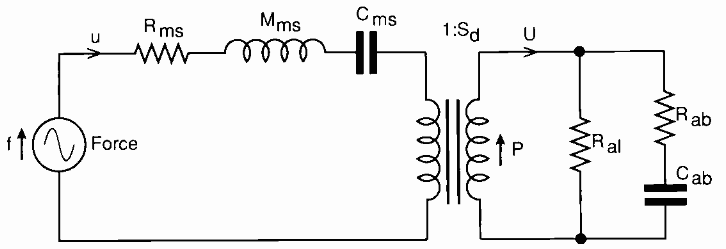


Fig. 1. Mechanical/acoustical circuit for a woofer in a sealed box. R_{ab} *and* R_{al} *represent box and leakage losses respectively. The cone is represented by a transformer of ratio 1:*S_d*, where* S_d *is cone area, thus converting mechanical force to pressure and velocity u to volume velocity U. Input is force from the voice coil.*

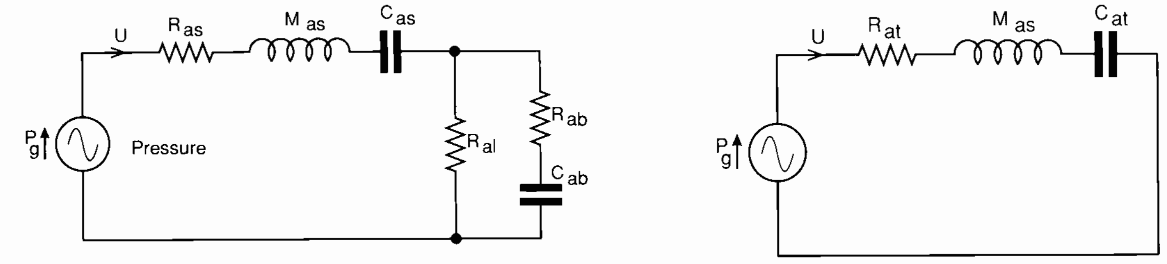


Fig. 2. Acoustical circuit for a woofer in a sealed box (a). Mechanical impedances in Fig. 5 have been converted to acoustic impedances. Input is now a pressure source. In (b) is a simplified acoustic circuit for a woofer in a sealed box, combining woofer, box and leakage losses to form R_a*. These total capacitances representing driver* C_{as} *and box compliances* C_{ab} *to form* C_{at}*. This is the minimum form. Sound pressure is proportional to the product of U and frequency. Provided Q is low enough in this circuit, U will rise at 6dB per octave below resonance and fall at 6dB per octave above. Sound pressure is proportional to the product of U and frequency, giving a flat output above resonance and a high pass 12dB per octave filtering action below.*

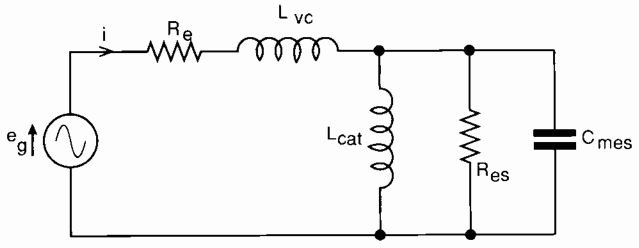


Fig. 3. Electrical circuit for sealed box woofer system with L_{cat} *representing woofer compliance and* C_{mes} *the woofer diaphragm moving mass. Impedance peaks at resonance with a sharpness dependent on losses.*

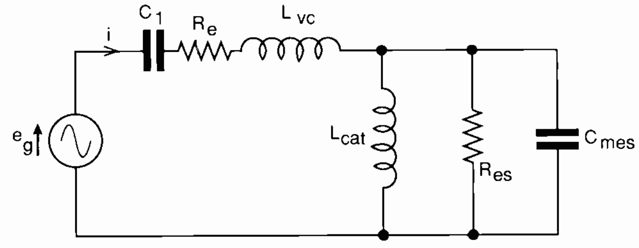


Fig. 4. Capacitive tuning of a sealed box extends bass by interacting with C_{at} *and* C_{mes}*, while also forming a third order high-pass electro/acoustic filter. Only the electrical equivalents are shown here.*

Sealed enclosure. For the simple sealed-box loudspeaker , the acoustical compliance of air in the cabinet, C_{ab} and the mechanical compliance of the woofer suspension C_{ms}, **Fig. 1**, behave acoustically like series capacitors. These are C_{as} for the speaker and C_{ab} for the box, as shown in the acoustical circuit, **Fig. 2**. When transferred to the electrical 'side' of the model, **Fig. 3**, they become parallel inductors of total value L_{cat}.

Conversely, the woofer's moving mass M_{ms} behaves like an inductor on the acoustical side, but is transformed via the voice coil and cone to a capacitor C_{mes} on the electrical side as in Fig. 3. The resulting inductance and capacitance form a damped resonant circuit. Clearly, the presence of the box increases the stiffness on which the moving mass resonates, thus raising the resonance frequency.

From the acoustical viewpoint of the equivalent circuit, placing one capacitor, represented by the enclosure, in series with another, the suspension, reduces the capacitance and raises the resonance frequency. From the electrical equivalent viewpoint this is like placing two inductors in parallel. Their total value is reduced while their resonance frequency is increased. Whichever way it is viewed, the end result is the same.

With a small box, a very compliant suspension is needed to keep the resonance frequency low and assure a decent low-frequency output. If the *BL* product – representing the woofer's electrical/magnetic system – is too high, the total electrical Q of the speaker (Q_{ts}) is too low and electrical damping is too great. Efficiency is gained, but at the expense of bass loss. A *BL* product which is too small (higher Q_{ts}) looses efficiency and can cause a rise in bass output near and above resonance. In addition, transient response can become poor and bass may tend to boom.

Final Q of the speaker in a closed cabinet is known as Q_{tc}. Generally in a sealed box, Q_{tc} values above unity can tend to boom, while Q_{tc} values below 0.5 can sound very dry. Critical damping occurs at a Q_{tc} of 0.5, (–6dB sound pressure level at resonance). For the so-called Butterworth alignment (–3dB sound pressure at resonance) occurs when Q_{tc} is 0.7.

Capacitance loading. A development of the acoustic suspension, sealed, or totally enclosed box, is the use of a series capacitor to modify the frequency response, **Fig. 4**. Series electrical capacitor C_1 interacts with the capacitive electrical equivalent of the woofer mass C_{mes} and inductive electrical equivalent of the driver/air compliance L_{cat} to form an electrical/acoustical filter. This modifies the input current, and hence volume velocity, at low frequencies.

The net result of a correctly aligned filter is that the low-frequency roll-off changes from second order to third order and the –3dB frequency is reduced. This makes capacitance tuning a handy way of extending bass without incurring the poor sub-bass woofer loading which a reflex port causes. Expressed another way, the loudspeaker can be more compact and protected from being over-driven.

Input to the woofer is reduced at very low frequencies, thus curtailing excursion and improving power handling below resonance. However, a major disadvantage is the deterioration in sound introduced by the large-value capacitor of a few hundred microfarads. Typically, for economic reasons this is an electrolytic type, which is hardly ideal in terms of linearity.

Reflex. In reflex loading, a hole, or port,

Driver characteristics

Assuming no influence from output from the rear of the speaker, sound output above resonance is flat. Below it falls off by 12dB/octave. Strength of the motor is determined by flux density and wire length in the gap. This is expressed as $B \times L$ in newtons or tesla metres.

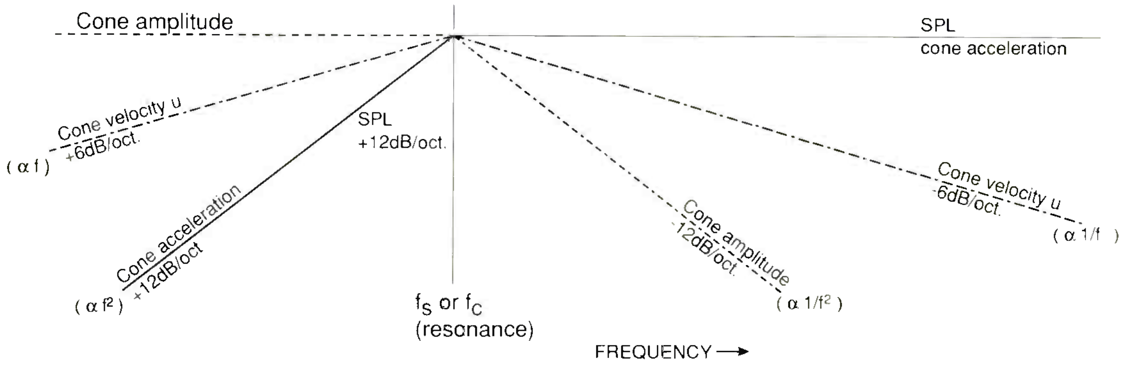
Voice coil and cone are interfaces between the acoustical, mechanical and electrical parts of the woofer. To analyse electro/acoustic systems, acoustics engineers have turned to electrical engineering and borrowed ac circuit analysis models which have proved ideal

Acoustical parameters must first be converted into electrical analogues. The results of these transformations are circuits are quite different from electrical circuits. They are not directly comparable to ac filters, but should be thought of as electro/acoustic filters.

Due to interaction between motor/generator, mechanical and acoustical parts, the loudspeaker's modulus of impedance curve shows a peak at low frequencies where resonance occurs. The shape and centre frequency of this curve is determined not only by electrical parameters, but also by the acoustical and electrical ones.

For more involved systems, such as reflex loaded woofers, the impedance curve is more complex. Normally it shows two humps representing the higher and lower frequency resonances either side of a dip at port resonance.

Until Thiel and Small's work in the early 1970s analysing drive units was difficult. They derived Thiel/Small parameters which are now used by virtually all loudspeaker designers to help them align woofer systems.

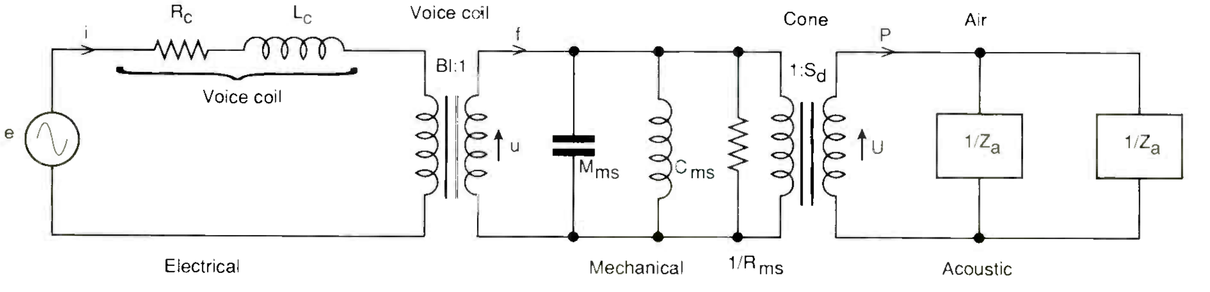


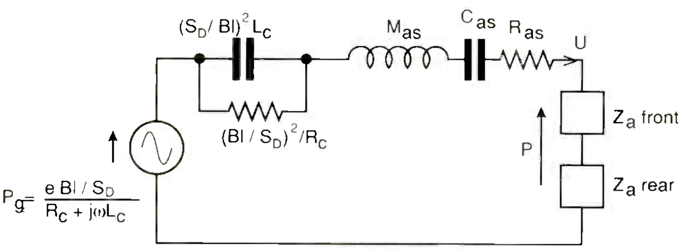
Cone amplitude, cone velocity and acceleration versus frequency around a speaker's resonance point. Above resonance the woofer diaphragm is controlled by its own mass and the much smaller mass reactance of the air load. With constant force from the voice coil, acceleration is constant and amplitude drops at 12dB per octave with increase in frequency. Cone velocity drops at 6dB per octave as frequency rises. As frequency is reduced below resonance, the cone becomes under control of the suspension stiffness and the stiffness of air trapped in an enclosed box. Cone amplitude levels off, velocity drops at 6dB per octave and acceleration at 12dB per octave.

Sound pressure p at distance r is proportional to frequency and is

$$P_r = \frac{f\rho_0}{2r}|U|$$

Clearly, P_r follows the acceleration curve, giving rise to the classic 12dB/octave high-pass filter curve for a woofer in a sealed box (or very large baffle). This holds good provided the wavelength of sound is greater than π_r (or frequency is less than c/π_r). At and above this transition, radiation impedance on the diaphragm changes from predominantly reactive to resistive. Sound dispersion narrows and cones normally cease to behave as pure pistons, so the neat theory starts to fail. For bass, the theory works well.





Interfacing between an audio signal and air via a loudspeaker voice coil and cone, above. Models such as this are excellent loudspeaker design tools. Note that this is a mobility circuit, in which the parameters are the inverse of their impedances. Acoustic load on the diaphragm affects its ability to move under the influence of the mechanical force of the 'motor' created by the input current to the voice coil and magnet system. Working as a generator, acoustical qualities are transformed to mechanical via diaphragm and then to electrical ones via the voice coil. In impedance circuit, left, the various parameters have been converted into their acoustic equivalents. Here, the output we are concerned with is the volume velocity U.

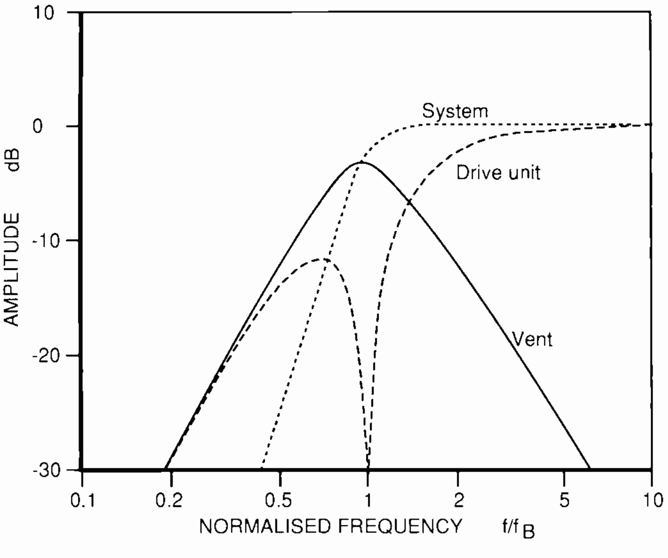


Fig. 5. Outputs of woofer and vent in a reflex loudspeaker. The vent peaks as woofer output diminishes. In a correctly-designed loudspeaker these should combine to form a smooth curve with a forth-order high-pass filter shape.

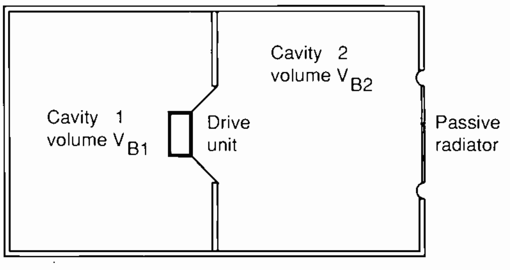


Fig. 6. In a double-cavity, or coupled-cavity loudspeaker, either a port or passive radiator couples the woofer to the air. Volume velocity from the port/passive radiator can be higher than that from the woofer alone, boosting sound output over a band of frequencies. Unlike reflex speakers, the coupled cavity has a bandpass characteristic.

located in one of the cabinet's panels, introduces another second-order resonant system to the simple second-order closed box. Air mass in the port behaves in the acoustical circuit like another inductor, which resonates with the enclosed air's acoustical capacitance.

Inductance in the acoustical circuit representing the driver's moving mass, M_{as}, also resonates with this capacitance. In the electrical equivalent, the roles of capacitance and inductance are reversed. The upshot is that the extra tuned circuit due to the port, coupled to that of the woofer and internal air volume, further complicates the electrical equivalent circuit filter. The total sound pressure radiated is the vector sum of that from the cone and the port. In other words, this is derived from the volume velocity of the port, added to the woofer volume velocity.

The two equivalent second-order tuned circuits interact electro/mechanically and combine acoustically to create fourth-order high-pass filter, **Fig. 5**. In the reflex system, the single peak in the impedance curve presented by the open backed and sealed box loudspeakers is replaced by two peaks, representing upper and lower resonances. These are either side of a dip at the port resonance, where restricted cone motion generates minimal back EMF.

Acoustically, at the port resonance, cone velocity and excursion are at a minimum due to extra loading by the high pressure of air in the cabinet. This is produced by the air in the port, which rushes into the cabinet simultaneously with the cone and at greater velocity – an excellent example of resonance amplitude exceeding the excitation. The increase in port output here fills the 'hole' created by the lack of cone output at resonance, thus transferring the main radiating zone from diaphragm to port, Fig. 5.

In a well-designed reflex, the combined response can be smooth down to the low-frequency roll-off. With a poor one, there are unwanted peaks and troughs and an inferior transient response. In a well-designed reflex loudspeaker, power handling is improved and cone excursion reduced, at the same time as moving more air at low frequencies.

For a given sensitivity, the reflex loudspeaker can be given a wider bandwidth than a comparable sealed box. Conversely, it can have a greater sensitivity than a sealed box having a comparable bandwidth. The main drawback is that below the port resonance, the cone excursion rises rapidly. Because it is unrestrained by the box's air spring, the cone moves further than that of a comparable acoustic suspension design. This can be overcome by capacitive loading, as with the capacitively-coupled sealed box, which reduces the electrical input at low frequencies, while extending bass and creating a fifth-order high-pass filter characteristic.

Passive Radiator Passive radiator loudspeakers are a sub-division of the reflex category. The difference is that the port's air mass is replaced by a subsidiary drone cone with its own suspension. Mass of this drone resonates with the air spring in the cabinet, and the compliance of its own suspension. The main advantage of this type of bass loading is that it avoids the sound of air chuffing through the reflex port.

Unfortunately, passive radiator loudspeakers have, in the past, been even more difficult to design than good reflex loudspeakers, so the breed has gained a rather poor reputation, perhaps unfairly.

Coupled cavity. This type of enclosure is a cross between a sealed enclosure and a reflex. Output from a conventional reflex loudspeaker comprises that of the port, driven by the rear of the cone, plus that from the front of the cone. In contrast, output from a coupled cavity can be considered to be solely that from the port driven by one side of the cone. The other side of the cone couples to a sealed box, which disposes of the sound from this side and acts merely like an acoustic spring, **Fig. 6.**

Figure 7 shows the acoustic equivalent circuit for a coupled cavity using a passive radiator. The coupled cavity is not a high-pass filtering reflex. It acts as a band-pass filter with lower and upper slopes of 12dB per octave. Though reflex in concept, it is no sharper acting than a sealed box and , in theory at least, achieves automatic low-pass crossover filtering.

In theory, no crossover filter is required. However, unlike the interior of sealed box loudspeakers, the front cavity cannot be completely filled with absorbent wadding. Consequently, air resonances at higher frequencies, even when damped as far as is practicable, can rather spoil the sound.

Advantages of coupled-cavity loading are many. The main gains are a great improvement in the amount of air that can be shifted by the woofer at low frequencies and the bandpass nature of the design. Because the coupled cavity is a tuned system, the port output can exceed that of the cone alone, while cone movement is restricted. This reduces distortion and enables more power to be accepted before the maximum linear excursion is reached.

For sealed rear cavities, cone excursion is limited below resonance, which is good for power handling. By altering the port and cavity dimensions, sensitivity can be traded for bandwidth, or vice verse. This makes it a very flexible system but one which really needs cad.

Transmission line. This alternative is named after the transmission line of classical electrical theory with lumped absorption, mass and compliance (lumped resistance inductance and capacitance). In theory, it attenuates the wave from the cone rear, preventing it from being reflected at the open end to influence the woofer, or radiating from the open end.

In reality, the transmission line could not be more different. Often, the line is lightly damped and tuned, rather like an rf co-axial 'stub'. Sound takes a finite time to travel down the line which is chosen to be a quarter, or sometimes an eighth of a wavelength of sound in the line at frequencies where bass reinforcement is required.

The delay brings the output from the end of the line roughly opposite in phase to that from the rear of the cone. Thus it is more or less in-phase with the front of the cone over quite a wide range of frequencies and reinforces the low-frequency output of the loudspeaker at around the woofer resonance and below. In addition, the reflected wave from the mis-terminated end travels back through the line to increase the air load behind the cone.

Similarly to reflex loudspeakers, this reduces the excursion and simultaneously reinforces bass output, but over a wide range, not a sharply tuned frequency band. Commercial transmission lines thus behave a little like a reflex loudspeakers, but not quite. They also add their own pipe-like resonances.

Horn loudspeakers. While the previous examples behave like relatively simple high-pass filters, the horn loudspeaker is rather more complex, behaving also like an acoustic transformer. This is because the flare of the horn couples the large air load at the mouth on to the smaller diameter of the drive unit end. The driver diaphragm is very much better coupled to the air than without the horn. Thus a relatively small diaphragm is almost magically provided with a very large radiating area without the penalty of a large and heavy diaphragm.

The horn's acoustic impedance is largely resistive, **Fig. 8**, and very high in value. As a result the driver becomes resistance controlled over a wide bandwidth, instead of mass controlled. The resistive load adds considerable damping to the diaphragm, which also reduces resonances within it.

As always there is a trade off – in this case size. Below the horn's cut-off frequency, its resistance falls away sharply, with a corresponding rapid drop in output. At near to cut-off, with a finite-length horn, impedance can fluctuate by quite a wide margin, thus giving rise to large unwanted variations in output and corresponding audible colorations (honking). Careful design can help minimise these variations, but they can never be eliminated.

Ideally, to accommodate the required flare profile and work properly down to 30Hz, a horn would need to be over 9m long. In practice, designers have brought cabinet sizes down considerably by folding the horn (which adds colouration) and using the room corner as an extension of the horn.

Quart from a pint pot?

By now it should be apparent that deep bass means large woofers and cabinets. So, it is perhaps not surprising that designers have tried various ideas to extract deeper bass from cabinets than would normally be possible. In particular, the 'Holy Grail' of deep bass from very small boxes has been with us for many years. Once the established techniques of using a small long-throw, low sensitivity woofer in a small cabinet and reflex or capacitive loading have been investigated, other means must be sought to extend bass or shrink the box.

Equalisation

This involves boosting bass output at frequencies where it would otherwise roll off. For instance, with a sealed box loudspeaker, having a Q_{ts} (system Q at resonance) of 0.7, and –3dB at system resonance (the so-called Butterworth alignment), electrical boost of 3dB could be applied at resonance, rising asymptotically to 12dB per octave as frequency is reduced.

At some low bass frequency the electronic boost is arrested and changed to a low-pass characteristic. Thus the frequency response is levelled down to a cut-off frequency lower than the system resonance, below which the bass rolls off at greater than 12dB per octave, due to the electronic filter.

Putting voice-coil thermal considerations aside, the main limitation to power handling at low frequencies is cone excursion. With a normal sealed box, or open baffle loudspeaker, cone excursion increases at 12dB per octave as frequency is reduced towards resonance. At resonance there is a transition: below this frequency the woofer diaphragm is under stiffness control: and the increase in the excursion levels off

However, while electronic equalisation boosts bass, it also increases cone excursion below resonance, placing much greater demands on the woofer and limiting low frequency power output. Even within the power limit, distortion is bound to rise as frequency drops.

With some manufacturers, the sub-resonance boost is made programme dependent in an attempt to minimise the excursion problem and the distortion or damage which over-driving can cause. In other words, bass boost is made greater for quiet bass sounds and smaller for loud bass sounds. However, sudden deep bass notes can

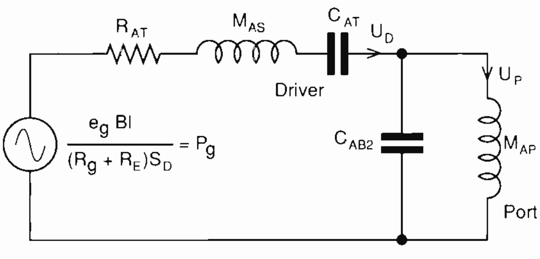


Fig. 7. Simplified acoustic analogue circuit for the bandpass, or double (coupled) cavity system. Input is pressure from the cone, output is volume velocity from the port.

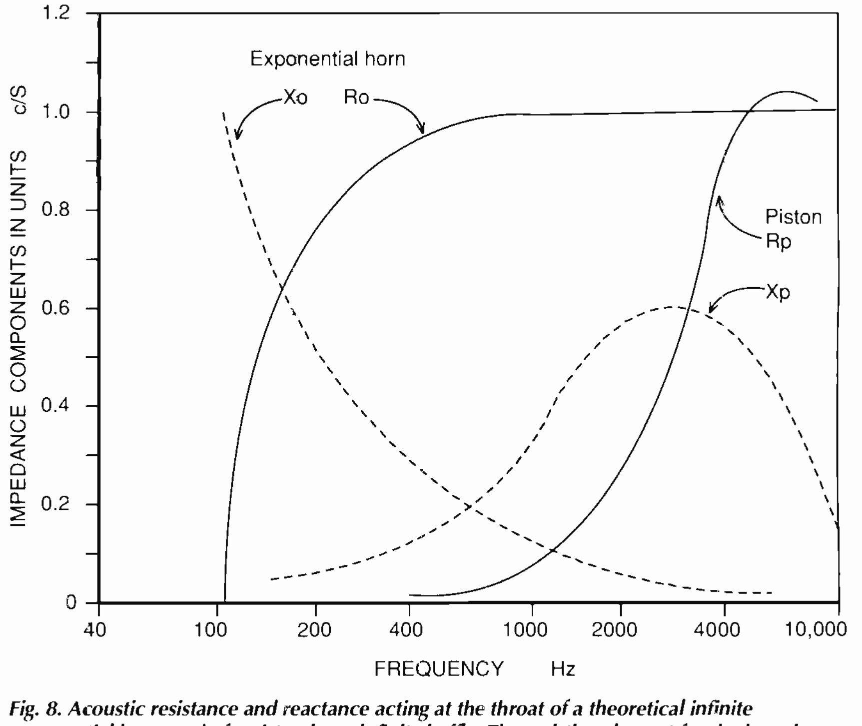


Fig. 8. Acoustic resistance and reactance acting at the throat of a theoretical infinite exponential horn, and of a piston in an infinite baffle. The resistive element for the horn is higher and very a wide band, while reactance drops very low at high frequencies. A horn can load a diaphragm with a high resistive air load, reducing amplitude of movement while coupling the diaphragm more effectively to the air. This gives greater efficiency.

catch the system out and as a result they can often be heard working.

Bass equalisation can just as easily be applied to reflex loudspeakers. There is a whole family of bass alignment curves which are well-suited for this application. Of course, this results in an even sharper high-pass filter cut-off than an equalised sealed box, with even greater phase shifts. Such systems must be designed very carefully if bass quality is not to suffer.

Negative Output Impedance. Another approach is to make the power amplifier behave as part of the loudspeaker. One way of doing this is to use negative feedback to create a negative output impedance, cancelling out the voice coil dc resistance. This enables the loudspeaker driver and cabinet to be designed as if it had no dc resistance losses at all. With a reflex loudspeaker, this allows the Q of the port resonance to be made higher, because there is less waste of energy through damping.

Though the use of negative output impedance was used to improve woofer control as long ago as 1950s, this idea was resurrected a few year's ago by Yamaha, who launched Active Servo Technology, which is still used in some of their equipment . The main disadvantage of this technique is that the power amplifier and loud speaker must be designed for each other, so it is hardly a universal solution. However, it does have the advantage that standard bass units can be used, which minimises the cost increase.

Motional feedback

None of the above bass systems requires anything other than standard drive units for their operation. However, the only way to control the woofer absolutely is to include it in the feedback loop in a totally active system. To achieve this, a transducer must be used to detect cone motion – either position, acceleration or velocity. Output from the transducer is modified and used, in classic control-system fashion, to derive an error signal for a feedback amplifier.

The extra complexity and cost of such systems has restricted motional-feedback woofer loudspeakers in the past to mainly studio applications. A variety of sensing systems may be used, from capacitive displacement sensors, to accelerometers. Alternatively the signal from an extra winding on the voice coil can be electronically converted to displacement or acceleration.

Not only can motional feedback extend bass response below the normal cut-off, it can also greatly reduce non-linear distortions – important if the voice coil and cone are moving a long distance. Woofer distortion always rises at low frequencies, so motional feedback can help solve one of the main problems in bass reproduction.

Despite numerous 'come backs', motional feedback woofer systems seem to have receded in recent years, though Tannoy has recently announced a new system. Perhaps with today's digital technology it could be due for a revival. Certainly the extra signal processing required could be included on a chip with relative ease for use within a loudspeaker providing on-board dacs, digital crossover filters and internal power amplifiers. Similar so-called digital loudspeakers have already been introduced by Philips and Meridian.

Over the years we may see new attempts to get deeper bass from smaller boxes. Most of these 'new' ideas are likely to have already been tried in one form of another, though the technology used will undoubtedly be advanced. Ultimately designers are up against the same laws of physics as their predecessors, who so many times used those same laws to their advantage. ■

Sloping vees for VHF and UHF

Armed with performance figures, Richard Formato contends that the sloping vee antenna is unjustly ignored for high-frequency use.

When antennas for VHF and lower UHF band are mentioned, most engineers think of whips, Yagis, log-periodics. The sloping vee is conspicuously absent from this list. But the vee happens to be a superb antenna – especially at shorter wavelengths. Yet it isn't well known or widely used by amateur operators on the higher frequencies.

The sloping vee is inexpensive, mechanically and electrically simple and easily transported and installed. Most importantly, it provides excellent gain-bandwidth performance – particularly for single-band operation. Because the radiating elements are inclined wires, vees also provide the added bonus of inherent polarization diversity.

A sloping vee comprises two radiating wires diverging from the antenna feed point, **Fig. 1**. A typical installation is shown in **Fig. 2.** Note that Fig. 1 is a perspective view in which both resistors R are at the same height H_t. These non-inductive resistors terminate the radiating elements. The resistance value is half the antenna input resistance, and the power rating is typically 10 to 20% of the maximum antenna input power.

Incident energy that has not been radiated into space is absorbed by the resistors. This suppresses reflections which would otherwise generate standing waves and create strong resonances. The resistors are connected by a shorting wire to complete the current path.

RF design software

If you want to experiment with VHF/UHF vees, a sloping vee modelling program is essential. It is the only way to investigate trade-offs as various antenna or ground parameters are changed. Radiation patterns in this article were computed using inexpensive PC compatible antenna design software from Phadean Engineering Co Inc. The company is at PO Box 611, Shrewsbury MA 01545-8611 (SAE for prices).

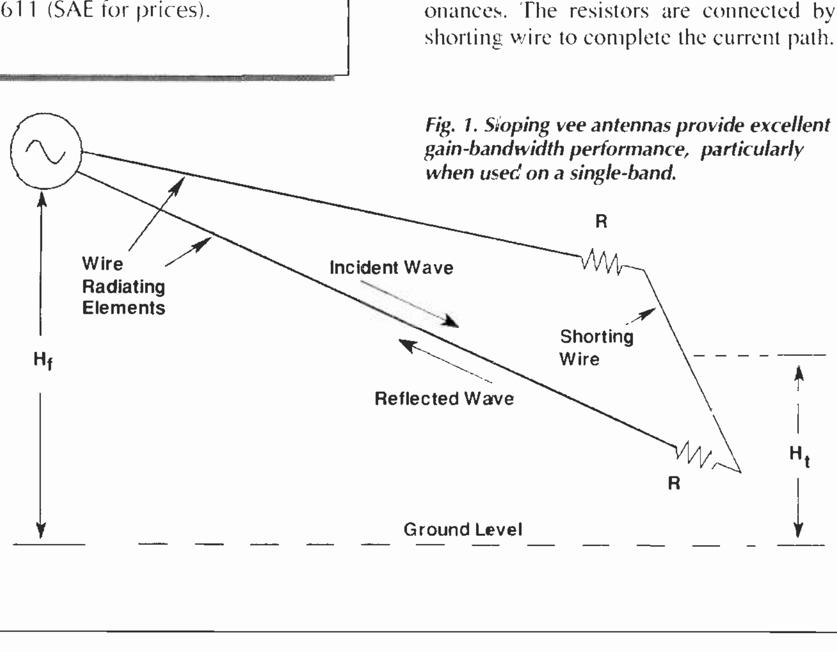


Fig. 1. Sloping vee antennas provide excellent gain-bandwidth performance, particularly when used on a single-band.

Since the sloping vee is a balanced radiating system any unbalanced feed line, such as coaxial cable, requires a balun. The balun should have the lowest possible insertion loss and flattest possible response over the vee's operating frequency range. As an engineering rule-of-thumb, the vee is considered a 600Ω antenna. To match to a 50Ω feed system, a balun with a 12:1 impedance ratio is required (square of the turns ratio). In practice, the "600Ω" antenna may actually turn out to be a 400Ω or even a 900Ω system, which, of course, changes the balun requirements.

To give you an idea of how good a vee can be, the three plots show computed vertical radiation patterns for a 6m sloping vee. Although not optimised, the antenna performs very well. The design frequency range is the 6m amateur band (50 to 54MHz). It is assumed that the antenna will be deployed over average ground with an electrical conductivity of 0.002S/m and a relative permittivity (dielectric constant) of 8.

Antenna parameters are as follows. Diameter of the radiating element is 3.2mm while the apex angle – the angle between wires at the feed point – is 15°. Feed point height above ground is 6m and above ground terminating resistor height is 8m.

Input resistances of the vee are 455, 446 and 437Ω at frequencies of 48, 52 and 56MHz respectively. Taking the average value of 446Ω as representative, each terminating resistor should have a value of 223Ω. In practice, 200 or 250Ω is close enough. Since computed input resistance varies only 4% between 48 and 56MHz, this design should provide essentially flat VSWR from 50 to 54MHz.

The plots show total power gain in dBi (decibels relative to an isotropic radiator). To convert to dBd (decibels relative to a dipole), subtract 2.15 since gain of a half-wave dipole in free space is 2.15dBi. Note that the total power gain includes both horizontal and vertical radiated fields, as well as antenna radiation efficiency. Patterns were computed at 48, 52 and 56MHz for three radiating element lengths of 20, 40 and 60m as annotated on the curves. These radiation patterns are in a vertical plane bisecting the elements (zero azimuth angle). They are plotted on linear scales which provide a more detailed view than polar plots.

Key computed performance parameters are

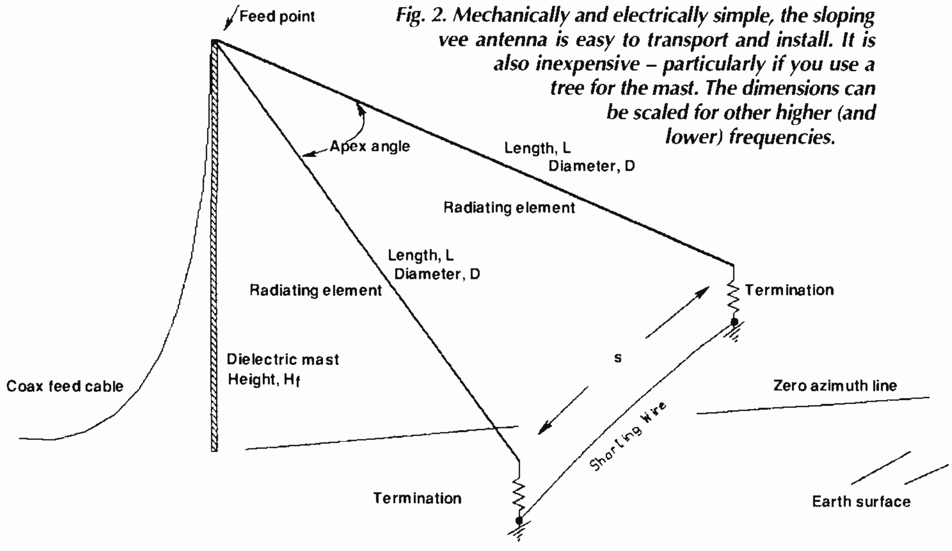


Fig. 2. Mechanically and electrically simple, the sloping vee antenna is easy to transport and install. It is also inexpensive – particularly if you use a tree for the mast. The dimensions can be scaled for other higher (and lower) frequencies.

Specialist antenna components

Non-inductive film power resistors for termination: Power Film Systems Inc, Yellville, AR 72687.

Stranded 7x9 phosphor-bronze cable (avoids kinks and tangles): Astro Industries Inc, Dayton OH 43432.

Strong non-metallic masting: J. T. Ryerson Co., PO Box 1111, Boston MA 02103.

Toroidal ferrite balun cores: Radio Kit Inc, PO Box 97, Pelham NH 0076.

The following table summarizes key computed performance parameters:

L (m)	Gmax (dBi)	Angle (deg)	–3dB BW (deg)	1st SL (dBi)	1st SL (dB//Gmax)
Frequency = 48MHz					
20	7.7	12	12.6	0.5	7.2
40	13.3	11	12.0	-2.3	15.6
60	16.3	11	10.6	-1.1	17.4
Frequency = 52MHz					
20	8.5	11	11.9	1.9	6.6
40	14.2	10	11.0	-1.5	15.7
60	17.2	10	9.9	0.3	16.9
Frequency = 56MHz					
20	9.3	10	11.0	3.3	6.0
40	15.0	10	10.4	-0.9	15.9
60	18.0	9	9.5	1.8	16.2

summarised in the table. It is evident that this simple antenna exhibits exceptionally good performance. With the largest element 160m long, main lobe gain varies from 16.3 to 18dBi between 48 and 56MHz. Maximum gain for all lengths occurs at take-off angles between 9 and 12°, which is a suitable range for long-distance links. Take-off angle can be controlled by adjusting the radiating element lengths and feed point and termination heights. As expected, the shortest element of 20m provides the lowest gain, but even its performance is very respectable (7.7 to 9.3dBi).

This design example shows how well the sloping vee performs at VHF and UHF. As the example illustrates, the physical size of a high gain vee can be large. But its dimensions are not so imposing, compared to the size of a Yagi providing the same gain. Of course at higher frequencies, the shorter wavelengths result in much smaller designs.

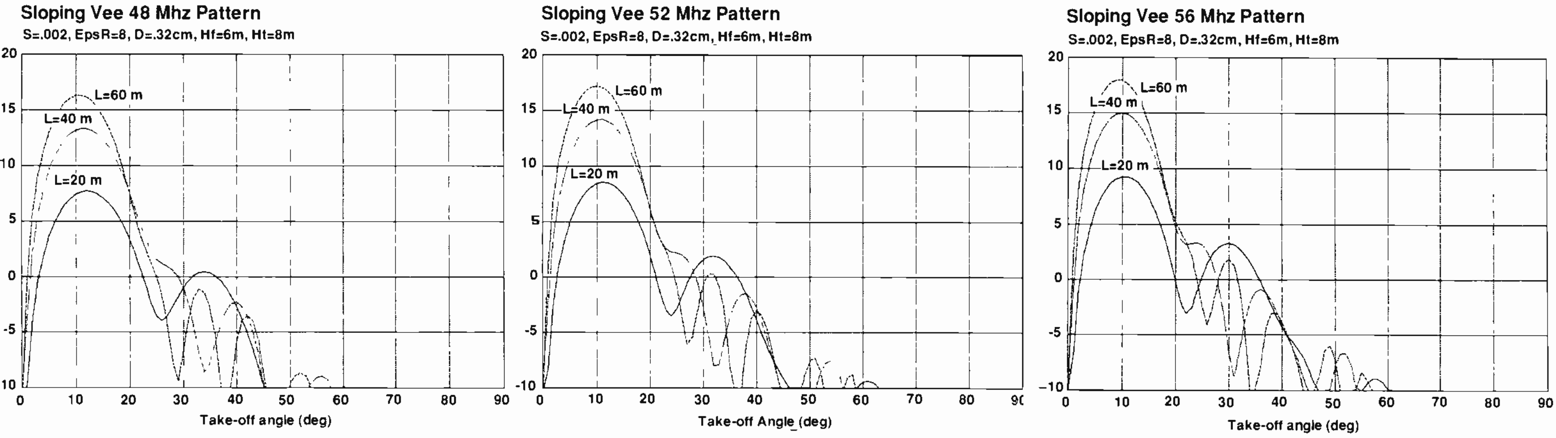
Another advantage provided by the vee is simple installation. The three different size antennas in the design example could be deployed in a variety of places, for example, between trees, or hung from a building or other structure. The range of possibilities is limited only by your imagination. Most antennas do not provide the installation flexibility that the vee does. About the only *caveat* to bear in mind is that, like any antenna, the vee's performance is influenced by nearby metallic structures. If they are too close to the radiating elements, parasitic effects may become a problem.

The vee's electrical performance is the same whether an exotic stranded cable or a plain single-conductor wire is used. The main difference is convenience. As far as masts go, trees provide the same results as fancy dielectric ones, with somewhat less convenience perhaps, but probably more fun.

It is evident that the sloping vee antenna exhibits exceptionally good performance, despite its simplicity. Maximum gain for all lengths occurs at take-off angles between 9 and 12° which is a suitable range for long-distance links

Radiating element length is L and main lobe maximum gain is G_{max}. Take-off angle for maximum gain is given in degrees above the horizon while approximate main-lobe beam width is in degrees between points 3dB down from maximum gain. First side-lobe level is given in dBi and relative to the maximum gain (decibels down from the main lobe). ■

This curve family shows antenna gain (dBi) versus take-off angle for three lengths over three frequencies. It assumes the optimum case where the feed point at the apex is lower than the termination side. The more convenient arrangement shown in the drawings may incur a gain penalty of up to 3dB.



Think of students

I was very interested to read the letter about your review of Electronic Workbench (*EW+WW*, December 1993), and found myself firmly in agreement with virtually all of it. But I feel Reg Williamson's favourable bias toward the program should perhaps be balanced by a student's view.

A student rarely has months to become familiar with a software package, a program like Workbench being used for maybe one or two units of a course, and here lies the main problem.

Workbench requires the dexterity of a watchmaker to place connections, and there seem to be a few inconsistencies in the way the circuits operate in the package. For example, in the digital version one of two identical circuits failed to function when macroed. Couple this with the fact that the help and print facility do not always produce the desired results, and perhaps it is possible to appreciate that this package takes a considerable amount of time to master.

I have spent around three hours trying to find a problem in a Workbench circuit that I could have built trouble free in one hour. How many students can afford this time?

At previous educational establishments I have attended, there was a very restricted access to anything other than word processing software, and even now I cannot guarantee to get any extra time with a technical package outside normal classes.

Should I wish to investigate a typical one-to-twenty active device educational circuit, one could be fabricated from a collection of TTL chips or a few small signal semiconductors fairly easily and quite cheaply. This can even be done and tested at home if the interest, inclination, and a few pounds worth of basic test equipment are there.

To prototype the same circuit using *Workbench*, I have to either ensure that the circuit is completed, tested and all results obtained within 1 to 2 hours, or try to arrange extra laboratory sessions that will not clash with the timetables.

I have been on the receiving end of PC oriented courses for a few years and, in general, after getting acquainted with syntax, file handling, and other little quirks, have found the PC quite an aid to design and development. Throw in the odd bug (nearly every package has them), add a few badly written batch files, courtesy of lecturers' and technicians' menu writing skills, restrict the time available with the technology, and the student ends up frustrated, angry and, towards the term end when the obligatory kilogram of printouts are due in, panic stricken.

Perhaps colleges and universities might like to bear in mind that the average student cannot afford the cost of a package such as *Workbench*. Unless they contemplate computer access over and above normal course requirements, a more practical approach to digital design would perhaps be more advantageous.

For analogue work I see no reason to consider *Workbench* at all, as there is a limited node version of *Pspice* and *Probe* freely available for students' own use.

I have spent quite some time with *Workbench* and, like Mr Williamson, I consider the package has some very good points. The trouble is it takes a fair few mouse miles to appreciate them.

H. Neary
Stoke on Trent, Staffs

Solid air

Dick Manton was quite right to point out (*EW + WW*, November 1993) for the benefit of those who might have been confused that my comments on coaxial cable referred to solid dielectric types. Air spaced types offer higher velocity ratios, approaching unity. For example, Andrew Heliax air dielectric coaxial cable type *HJ4-50* (0.5in corrugated solid copper outer, air-spaced inner supported on a helical spacer cord) has a 50Ω impedance and a velocity of 91.4% of c.

Feeding these into the formula in my article, L (per metre)=$Z_0 \times b$ gives 152nH/m for this cable, well below the free space value of 1256nH/m.

Likewise, I would imagine that C/m for an air spaced 1200Ω balanced line is well below the free space value. As for velocity of propagation in a line, coaxial or otherwise, this can vary widely.

In a coaxial delay cable as used for delaying an oscilloscope's Y amplifier signal, a spiral wound inner is used. This reduces the velocity of propagation way below the $0.65c$ typical of a solid dielectric coaxial cable. In a loaded telephone line the velocity of propagation may be as little as a 20th of c.

However, when I said the values of L and C per metre in free space are the lowest that can ever be achieved I meant just that – in free space. If a space wave is propagating in a medium where the value of C is greater than 88.5–12/m, that is in a dielectric medium, it will be slowed down. This effect is used in certain types of radar lenses.

Similarly, a wave propagating in a medium where L is greater than $4 \times p \times 10^{-7}$/m will likewise be slowed, an effect used in certain waveguide structures.

Further to my comments in the

Silent problems

Five recent papers on distortion in power amplifiers by Douglas Self make very interesting reading and it is good to have the various aspects of the subject considered as an entity. They will be recommended reading for my final year undergraduate audio systems engineering class.

He stresses, quite correctly, the need to avoid the generation of high order harmonic and intermodulation products, but appears to accept that for stability of the feedback loop, the high-frequency open-loop gain of an amplifier should fall at –6dB/oct, starting from a frequency well within the audio range.

Do not overlook that audible intermodulation products can arise from spurious signals with frequencies above human hearing. The linearising benefits of negative feedback are therefore reduced just where they are really wanted.

He also sees little advantage in using fets as output devices as opposed to bipolar transistors. I agree they are not perfect devices and are in some respects inferior to bipolars, but when used in suitable circuits they offer the possibility of considerably wider bandwidth output stages. This is their most significant advantage.

In this case it is possible to employ a considerable amount of feedback and have an open-loop bandwidth covering the whole audio range. The high frequency roll-off needs careful design to have an average rate of about –9dB/oct. This can be achieved by a succession of poles and zeros in the response, rather than one dominant time constant.

The immediate reaction of listeners to my amplifiers designed this way is that they have an enhanced treble response. Further listening, and comparison with other types, alters this assessment to one of greater clarity and realism, creating the illusion of more treble. A stereo pair with suitable speakers are capable of excellent imaging.

Ivor Brown
University of West London
Uxbridge

Douglas Self replies:

"I thank Mr Brown for his comments. It is quite true that in general I expect the open loop gain of an amplifier ultimately decrease at 6dB/octave, though I do not assume that this rolloff begins "well inside the audio range". The actual frequency at which rolloff starts, as far as I can see, of very little importance in itself; what matters is the amount of feedback available at the upper end of the audio band, because only here will the distortion of a Blameless amplifier be measurable, and also the amount and slope of rolloff at the unity loop gain frequency, because this sets HF stability.

As I showed in the article on the voltage amplifier stage, with the aid of a cunningly placed resistor, you can make the start of the rolloff occur at almost any frequency you like (20Hz if you wish) without affecting the most critical part of the open loop gain characteristic. This is why I prefer to quote NFB factors at 20kHz.

I do indeed take a rather pessimistic view of fets in output stages, and have always found the greater bandwidth to be more of a hindrance than a help. They may promise a higher bandwidth output stage, but do they deliver, especially when capacitively loaded? If Mr Brown has some data on this, I hope he will share it with us. The worst drawback of fets is that they are so depressingly nonlinear, despite what you sometimes read in the hi-fi press.

I am well aware that there are other kinds of compensation, as future parts of the series will show. However, I don't much care for the 9dB/octave approach, because if this slope is maintained up to the unity-gain frequency, it directly reduces the stability margins. It also requires a series of alternate poles and zeros that are not easy to fit into the conventional amplifier topology I have been discussing. I think a better method uses two pole compensation because: it is cheap and easy to implement if you know and avoid the snags; gives a stunning reduction in HF distortion; allows the gain slope to be returned to 6dB/octave before the unity gain frequency is reached giving HF stability no worse than standard dominant pole compensation."

article concerning a per-unit generator that always supplies a total of 2W, wherever it is dissipated and whatever the load, no one has so far written in to quote an earlier reference. While they are admittedly academic constructs rather than useful circuit arrangements, perhaps the Hickman generators types I and II really are novel after all.

Ian Hickman
Waterlooville, Hants

Theories of science

Judging from the sort of arguments presented in these columns over the past year or so, the jolt A. M. New (*EW+WW*, October 1993) gave us was much needed. In the simplest of language he reminded us of a truth that most of us seem to want to forget – all scientific theories are fiction.

This puts into perspective such disingenuous questions as how we can accept theory based on the constancy of the speed of light in view of the experimental evidence that it is bent by gravity.

Those who propose such questions seem entirely blind to the fact that they may be applied to other theories. We might equally ask how, in view of the experimental evidence that neither mass nor time are invariants, we can possibly accept a theory based on the assumption that they are. And yet we accept and strongly defend such a theory on a daily basis.

As far as the question about special relativity is concerned, Einstein dealt with the matter in 1916, long before any experimental evidence existed. In his book Relativity, the special and the general theory he points out that general theory predicts the bending of light, which must call the assumption of the constancy of its velocity into question, and examines what this means for his earlier theory. He says, "We can only conclude that the special theory of relativity cannot claim an unlimited domain of validity; its results hold good only so long as we are able to disregard the influences of gravitational fields on the phenomena, for example of light."

If necessary, we might well paraphrase this and say, "We can only conclude that Newtonian theory cannot claim an unlimited domain of validity. Its results hold good only so long as we are able to disregard the influences of relative motion on the phenomena, for example mass and time."

To fail to do this is to hold double standards, a failing that seems not uncommon even among physicists. The harsh fact is that all theories have shortcomings by their very nature. Some may be shorter than others.We ought to be spending our energy on determining what those are and how they influence the domain of validity of any particular theory, not on arguing whether any one theory is the truth. Truth in this case is – dare I say it – entirely relative?

Alan Watson
Mallorca, Spain

Radio ham baloney

While I endorse most of the remarks contained in your editorial comment (*EW+WW*, December 1993), I must take issue on several of your opening points.

First, you quite rightly state that Tony Hancock's radio amateur was "pompous, petty and technically incompetent", but then you go on to say that this interpretation "encapsulates the truth about the hobby". I strongly object to being described by any of these adjectives and would think the majority of other radio amateurs agree with me, particularly when the remarks are made by a fellow amateur who edits a respected technical periodical.

Secondly, it is open to debate whether "everyone... enjoys the Hancock sketch." The fact that many people found the sketch offensive and insulting was evidenced by letters to editors of various magazines at the time, which I suggest was long before you obtained your licence.

Hancock was, in fact, demonstrating the "self conscious and inane chatter" you attribute to the new influx of radio amateurs, but unlike them was broadcasting to a far wider audience. Mr and Mrs Average are unfamiliar with amateur radio and therefore assume Hancock's parody mirrors the truth. On the other hand, the trivial prattle heard on many amateur bands is heard only by other radio amateurs, most of whom are capable of treating it with the contempt it deserves.

While I do not condone 'cheque-book engineering' and am unhappy with many of the evolutionary changes that have affected amateur radio over the years, I am a realist and accept that times change. However much we might like to believe it, the likelihood of a radio amateur working in his garden shed coming up with an idea that will revolutionise radio communication is fatuous nonsense.

We, as radio amateurs, must not fall into the trap of taking ourselves too seriously and thus forget that amateur radio is a hobby to be enjoyed and not some vast army of latterday Maxwells or Heavisides.

To think otherwise is as pompous as Hancock's portrayal of a typical radio amateur, but then you think this encapsulates the truth anyway. Perhaps I have been wasting my leisure time over the last 34 years on the air enjoying my hobby when I should have been engaged in more productive pursuits whereby the frontiers of knowledge could have been pushed further forward.

As a chartered engineer, I am content in the knowledge that I have contributed to the advance of knowledge during the working day and see no necessity to carry on in the same vein while pursuing my hobby. We should concentrate our efforts into improving the image of the hobby rather than hankering after passed glories that are unlikely to be repeated in the late 20th century.

Far more could be gained by cleaning up the language and improving operating procedures on our hard won frequencies than will ever be achieved by bemoaning the fact that revolutionary advances in spread-spectrum techniques are not being developed by the amateur fraternity.

A. C. Wadsworth G3NPF
Horsham, West Sussex

You make some excellent points in you editorial on amateur radio (December 1993). However you fail to mention the fact that amateur radio is a hobby and that in itself justifies its existence. Just as people still climb Everest, we can all experience the thrill of personal firsts long after the achievement has become commonplace.

G. P. Stanley G3MCK
Staines Middlesex

Not going round in circles

It has been repeatedly demonstrated by men such as Poor, Ives, Dingle, Marinov, Beckmann, Sachs and Hayden that Einstein's relativity involves circular logic, and is quite impossible. The idea of the constancy of the speed of light in all directions, independent of source and observer, can be shown to violate the second law of thermodynamics.

Many alleged conclusive proofs of relativity have been shown to be based on fudged data, such as Eddington's on the gravitational bending of light, the perihelion precession of Mercury, the Navstar project, and Hafele and Keating's on time dilation. Relativity is riddled with anomalies and paradoxes: the Ehrenfest paradox, the paradox of self reference, the curvature of space paradox, the ruler paradox, the clock paradox, the simultaneity paradox etc.To this day, all Michelson-Morley type experiments are unable to detect even a smidgen of the earth's purported 67,000mile/h translational velocity around the sun. As Professor Lincoln Barnett said, "We cannot feel our motion through space; indeed no experiment has ever shown that the earth actually is in motion".

Relativity is refuted by the aberration of starlight, the everyday operation of numerous electro-optical engineering devices, the Sagnac effect, and the results of the Michelson-Gale experiment, which are all exactly what one would expect in a geocentric universe.

There is much more recent evidence from astronomy and quantum physics to show that the earth is indeed the preferred frame of reference in the universe, located at or near its centre.

Bertrand Russel said, "Whether the earth rotates once a day from west to east as Copernicus taught, or the heavens revolve once a day from east to west as his predecessors held, the observable phenomena will be exactly the same: a metaphysical assumption has to be made".

See Geocentricity, Gerardus Bouw 1993, "The earth is not moving – 400 years of deception exposed", Marshall Hall 1992, and The cosmos, Einstein and the truth, Walter van der Ramp, 1993.

Amnon Goldberg
London

Black hole error

In my article on gravity and electric force in a black hole (*EW+WW*, February 1993) a small printing error managed to creep through. The symbol h of the last equation was misprinted. The correct equation is:

$$G = c^5\alpha^2(2-\alpha)^2(e/4\pi^2)^4/\pi h$$

D. Di Mario
Milan, Italy

MATHCAD 4

16 bits of difference?

Over the five years since its launch, Mathcad has become faster and evolved from DOS to Windows. But according to Allen Brown it remains quirky and has progressed little in real problem solving power.

One of the foremost equation processing packages – *Mathcad* from Mathsoft – has been upgraded to version four. The principal difference between this version and its predecessor version *3.0* is its ability to take full advantage of the 32bit architecture of the *386/486* PC. Surprisingly, 32bit *386* PCs have been available for at least five years, yet very little software is actually coded in 32 bits. Most *PC* software is still coded in 16bit format which is a waste of the extended data bus width.

Installation of *Mathcad 4.0* is straightforward. The only notable point is the creation of a sub-directory of WINDOWS/SYSTEM called WIN32S. It contains the dynamic link libraries (DLLs) used in the 32bit processing.

Mathcad is a mathematical scratch pad allowing you to express equations directly on screen. Solutions can be displayed in tabular form or graphically in either two or three-dimensions. Additionally, equations can be solved to give either numerical or analytical solutions using the *Maple* equation processor. In fact *Mathcad* can be envisaged as a sort of super calculator.

Version 4 for Windows is installed like any other Windows software and possesses the general familiar features, including buttons and dialogue boxes. However it does require the virtual memory facility for swapping between hard disk and system ram. As the illustrations show, the screen display is rich in icons, permitting easy access of *Mathcad*'s many features. The Windows version also allows you to use the range of Windows fonts for creating text alongside equations, tables, graphs or diagrams.

For modelling linear systems, *Mathcad* is a very attractive tool. It can be effective for designing infinite-impulse response (IIR) digital filters, **Fig. 1**, since they are derived from a linear system model. The same applies to finite impulse response (FIR) filters based on the window design - again a linear system model. Things get a little more awkward when dealing with coupled linear systems since one has resort to using *Mathcad*'s matrix facilities.

Within *Mathcad* is a large array of easily accessible built in functions. For example if you want to determine the condition for an optical fibre to support single mode then you need the first root of the equation.

$$J_0(x) = 0$$

where J_0 is a Bessel function. **Figure 2** shows a plot of this equation and from the plot the function is zero when x is around 2. By using the ROOT function with your guess of 2, *Mathcad* will give the correct value of x=2.404.

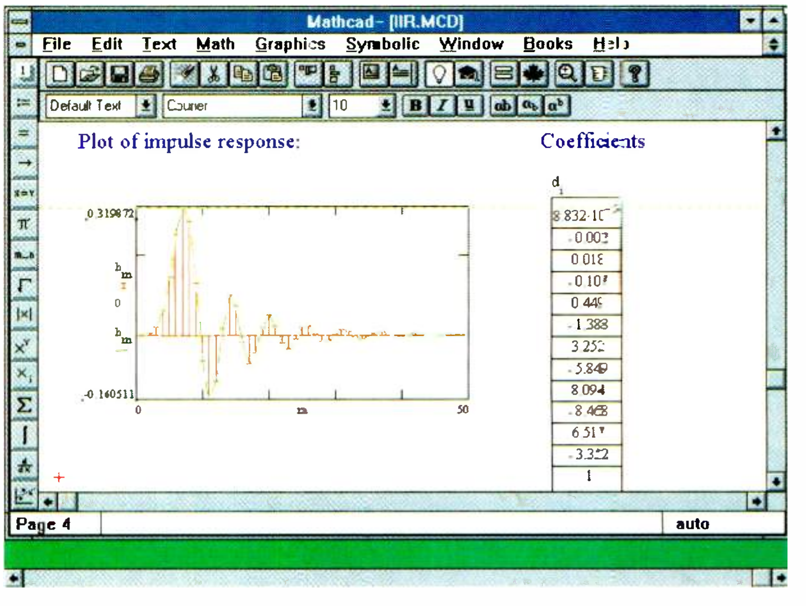


Fig. 1. Mathcad can be quite effective at modelling infinite impulse response filters.

Symbolic Calculator

One of the attractive features of *Mathcad* is *Symbolic Calculator*. Developed by Waterloo Maple Software of Ontario, this is a derivative of *Maple* which performs algebra and calculus analytically or symbolically.

Given an integral, you no longer have to work it out. **Figure 3** shows an integration and an expansion of a series. Even when manipulating matrices, variables can be

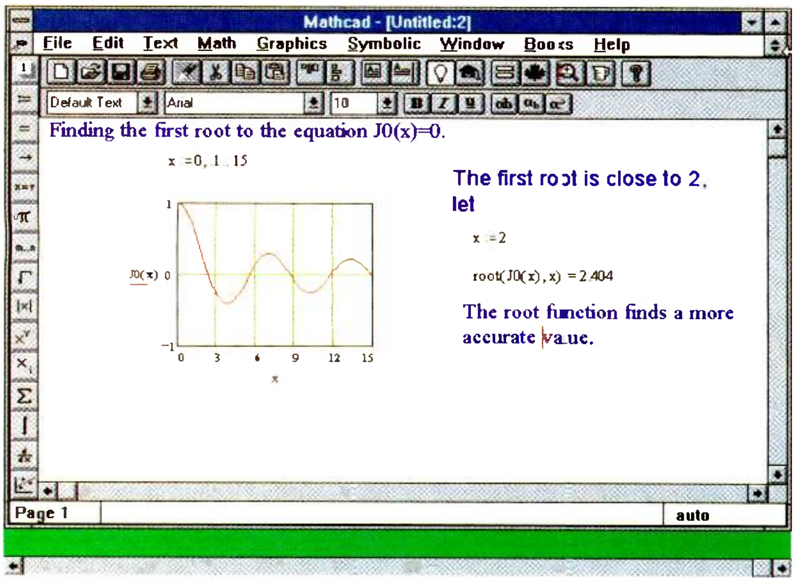


Fig. 2. There is a wide choice of built-in functions. This is an example of finding roots to aid determining the condition for an optical fibre to support single mode.

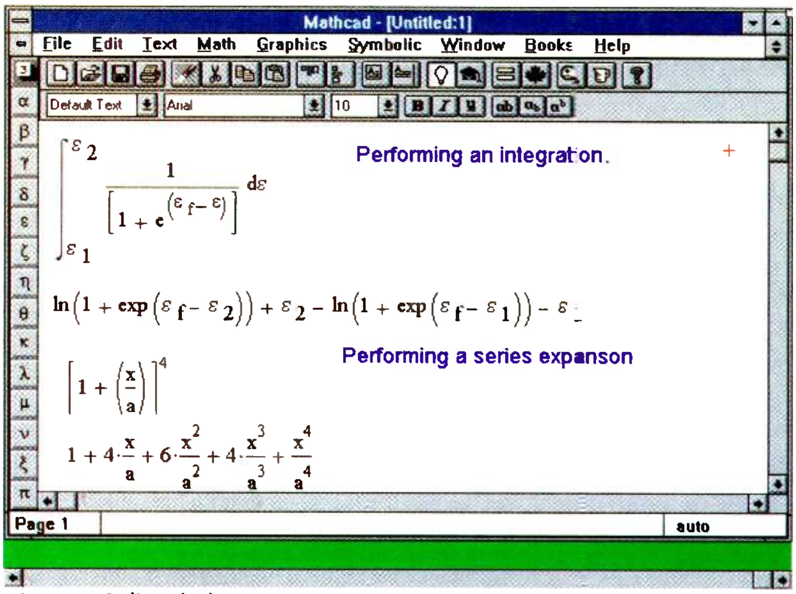


Fig. 3. Symbolic Calculator in action – integration and expansion of a series.

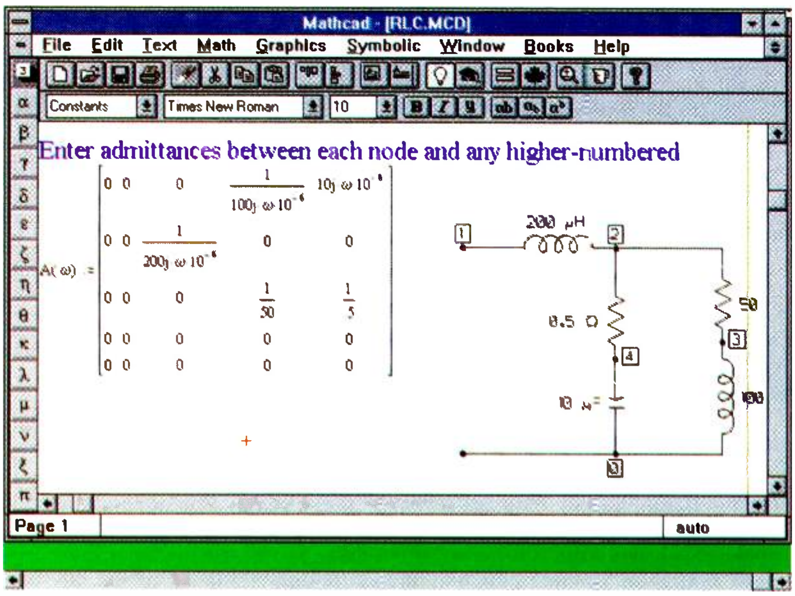


Fig. 4. An example of network analysis with Mathcad. When manipulating matrices, variables can be used as elements in the matrix.

used as elements in the matrix – handy when you are performing small scale network analysis, **Fig. 4.**

Using *Symbolic Calculator* for evaluating analytical expressions you also have the option of mixing constants with numbers. For example if you put an actual value on the upper limit of an integral sign and a variable on the lower limit, *Mathcad* will perform the integration, carrying the variable through. With the previous version of *Mathcad*, *Symbolic Calculator* was rather slow but version 4 makes it noticeably faster.

SmartMath

An interesting variation on the *Symbolic Calculator* is *SmartMath* which uses an expert system to best combine symbolic and numerical evaluations. The ordinary *Symbolic Calculator* dutifully takes every expression literally and works out the answer. *SmartMath* on the other hand examines what you have entered and works out the optimum method of solution according to a set of expert system rules. It is difficult to divine why *Mathcad 4* should carry the *Symbolic Calculator* at all since *SmartMath* appears to do the same job in a fraction of the time.

Graph Plotting

Even with the first version of *Mathcad*, graph plotting was particularly easy to perform. This was especially true for two dimensional graphs. Later versions of the product had the provision for drawing 3D surface plots but it was not particularly easy to generate these since the user was expected to create a matrix of the surface first.

The same awkward and unappealing technique is used in version 4. However the new version can produce polar plots with the same ease as two dimensional plots and is useful for estimating field patterns from radiators, Fig. 5. In addition there is a new facility for contour plots which has contour labelling. This is common feature with commercial graphics packages and it is reassuring to see parts of *Mathcad* keeping up with its competition.

Electronic books

When *Mathcad* is installed two electronic books are loaded onto the hard disk, the tutorial and the standard handbook. The handbook contains information drawn from the *Rubber Handbook*[1]. Although useful, it contains only a relatively small snippet from it. Maybe a future CD-rom version of *Mathcad* will contain all the non-chemical information from the *Rubber Handbook*. The tutorial book on the other hand should prove very useful, especially for newcomers to Mathcad and those upgrading from the dos version.

Deficiencies

Initially I was very keen about *Mathcad* but with subsequent versions my enthusiasm has waned due to its lack of progress. True, the screen presentation is better under Windows, and it now runs faster. Computationally however, it cannot do a great deal more than the dos version 2 release.

The *Symbolic Calculator* is certainly novel and could serve as a powerful teaching aid. However it is the absent features that worry me. For instance you cannot directly solve differential equations – even linear ones. In fact they are not even mentioned in the user guide. As any practising engineer knows virtually all modelling is performed with differential equations.

With *Mathcad*, if you want to model a simple *LCR* circuit for instance then you would be struggling. One way is to generate a pair of difference equations from your *LCR* equation and solve them using matrix notation, **Fig. 6**, which is very messy.

My reservations regarding 3D plotting have already been mentioned. If you have a function $f(x,y)$ you should be able to plot it directly without first having to go through the tiresome process of converting the function into a matrix. I cannot help thinking that in order to get *Mathcad* to solve a difficult problem you have to spend so much time trying to evade the limitations imposed by the software.

Operation of the package for complex tasks is not intuitive and becomes rather quirky. For example if you want to perform an autocorrelation function and you enter the equation,

$$r_n = \frac{1}{N}\sum_k x_k \cdot x_{n+k}$$

Mathcad will refuse to perform the operation. Everyone knows that if you have N samples then you are going to run into problems when $n+k$ exceeds N. So why cannot the software issue a warning and perform the calculation up to $n+k=N$ automatically? Instead it stops dead and does nothing. In order to carry out the task, you have to enter conditionals.

This is typical of problems with the first release five years ago. One does not expect to have the same deficiency five years on. There is also a concern regarding memory consumption. It is questionable whether Mathcad makes as efficient use of memory as it should. For example when generating data for a graph, should it use high numerical precision for something which is purely visual?

User guide

The user guide is a single volume manual which is indispensable – even for the veteran user of the product. After several iterations it has become a well crafted document whose layout is appealing and opulently illustrated with valuable screen dumps.

As stated, much of *Mathcad*'s operation is rather quirky and the guide is invaluable for learning the product's peculiarities. Maybe Mathsoft should issue the user guide as a text book as I believe there is a ready market for it.

Conclusion

When *Mathcad* was first introduced about five years ago it was novel, and in its day powerful. It was very popular because of its ease of use and its graphic front end.

Several versions later its presentation has improved and its compatibility with Windows makes it an attractive package. But it is still fundamentally a maths tool for solving small scale linear problems only. It has not grown with expectation.

The main redeeming features are the additions of *SmartMath* and the electronic books. What *Mathcad* does it does very well, but what worries me is what it does not do. However for small scale linear modelling there is no better product on the market and this new version will surely find many enthusiasts – especially among the student population. ■

Reference

CRC handbook of chemistry and physics, CRC Press, (regularly updated).

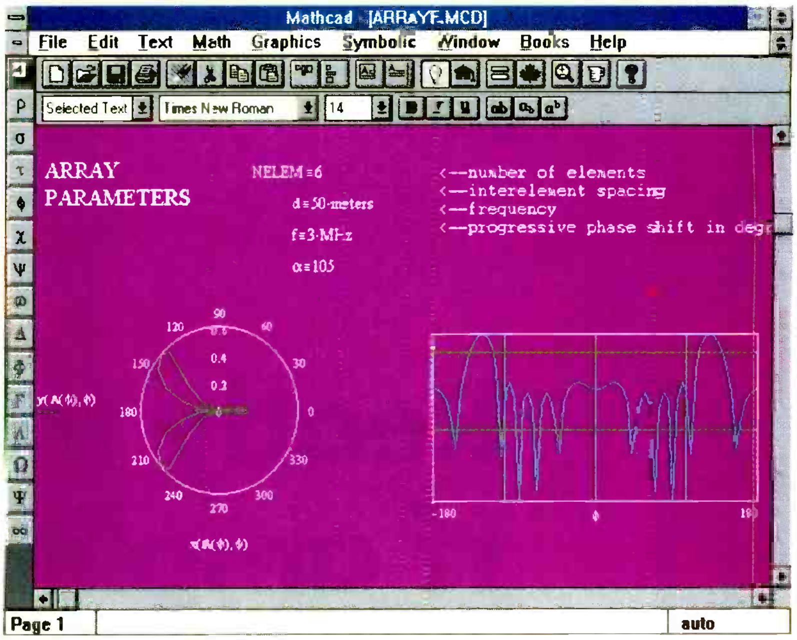


Fig. 5. Polar plots are useful for displaying field patterns from radiators.

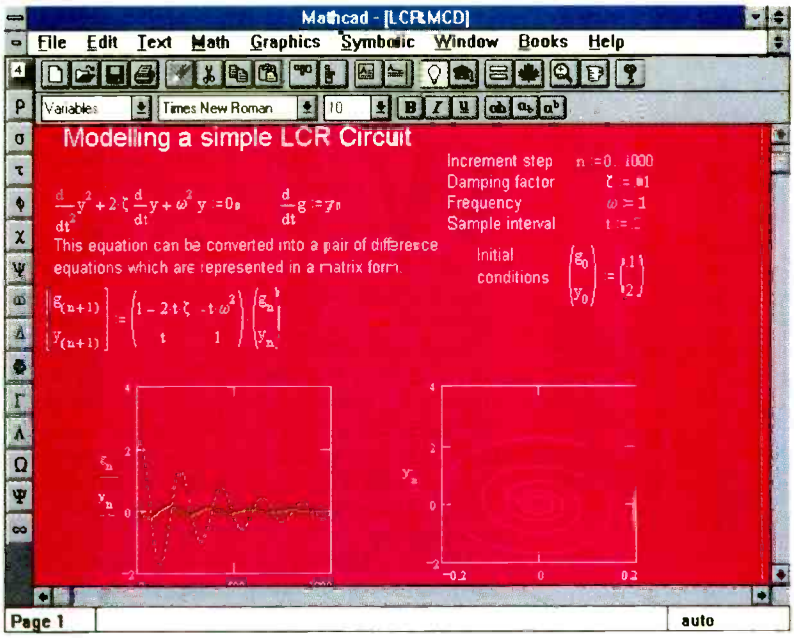


Fig. 6. In Mathcad, modelling an LCR circuit is difficult. One method is to generate a pair of difference equations and solve them using matrix notation.

SYSTEM REQUIREMENTS

PC compatible (Apple version also available), *386* or *486* processor, maths coprocessor, Microsoft mouse, Windows 3.1, 4Mbyte ram, 10Mbyte hard disk, 8Mbyte virtual hard disk space.

SUPPLIER DETAILS

At £375 plus £10 delivery, Mathcad is available from Adept Scientific Micro Systems, 6 Business Centre, West Avenue One, Letchworth Hertfordshire SG6 2HB. Tel. 0462-480055, fax. 0462-480213. Apple version is same price.

DESIGN BRIEF

Ready to use rf amplifiers

Electronic component design has yet to reach the point where rf building blocks can be applied as easily as their logic counterparts. As Ian Hickman shows, the MAR series of rf amplifiers require the minimum of skill to apply.

While it may pay to design a clever rf amplifier stage, especially if constraints such as a current consumption apply, an off-the-peg solution can be attractive. In these circumstances, the ready-to-use rf amplifiers described here can fill the bill perfectly.

The devices referred to are the *MAR* series from Mini Circuits (a Division of Scientific Components Corporation) and the performance offered by the various members of the family is summarised in **Table 1**. It is an open secret that these are basically an Avantec range of components, but sold by Mini Circuits at supermarket prices, making them an attractive buy.

One way to take a quick look-see what an amplifier can do is to connect its output back to its input, to implement an oscillator. As **Fig. 1** shows, the integrated two-stage amplifier is inverting, the component values having been carefully designed to give a nominal match to 50Ω at both input and output (type *MAR8* excepted). Thus it will oscillate at a frequency F_0 when its output is connected back to its input via a length of 50Ω coax whose electrical length is $\lambda/2$ at F_0, as shown in **Fig. 2a**. With the length of coax shown and assuming it has a wave velocity of 65% that of free space, the expected frequency is 0.65 x 300/(2 x 0.965) = 101MHz. As the loss in the feedback network is negligible, the excess loop gain is virtually equal to to stage's forward gain, so the waveform would not be expected to be very good, as Fig. **2b** confirms. Given that the 1ns wide spikes on the edges of the waveform are way beyond the 250MHz bandwidth of the oscilloscope, their true amplitude must be even more horrendous than it appears.

The hard limiting in the amplifier is also responsible for excess phase delay (at this frequency the device exhibits 9° of excess phase anyway, even under small signal conditions) or, put another way, the circuit is almost a relaxation oscillator, which always results in a lower frequency of oscillation than if the loop gain barely exceeds unity. In consequence, the actual frequency of oscillation is less than 100MHz, Fig. **2c**, which shows high amplitudes of harmonics: the tenth harmonic is as large as the second, both barely more than 20dB down on the fundamental, while the third harmonic is only 8dB down – definitely not a clean oscillator.

Fig. 3a shows the interesting effect of reducing the supply voltage to the circuit of Fig. 2a from +12 to +8V. The narrow spikes are no longer so evident, but

Table 1. Performance summary of MAR series amplifiers. (The colour dot referred to, in addition to denoting the type number, also indicates the input lead).

Model No.	Color Dot	FREQ. MHz	GAIN, dB Typical (at MHz) 100	500	1000	2000	(Note 4) MIN	MAXIMUM POWER, dBm Output (1dB) Compression	Input (no damage)	DYNAMIC RANGE Intercept pt. dBm NF dB Typ.	3rd Order Typ.	VSWR In	Out	MAXIMUM RATING (25°C) I(mA)	P(mW)	DC POWER at Pin 3 Current (mA)	Volt. Typ.
MAR-1	Brown	DC-1000	18.5	17.5	15.5	—	13.0	0	+10	5.0	15	1.5:1	1.5:1	40	100	17	5
MAR-2	Red	DC-2000	13	12 8	12.5	11	8 5	+ 3	+15	6.5	18	1.3:1	1.6:1	60	325	25	5
MAR-3	Orange	DC-2000	13	12.8	12.5	10.5	8.0	▲+ 8	+15	6.0	23	1.6:1	1.6:1	70	400	35	5
MAR-4	Yellow	DC-1000	8.2	8 2	8.0	—	7 0	+11	+15	7.0	27	1.9:1	2.1	85	500	50	5
MAR-6	White	DC-2000	20	19	16	11	9	0	+15	2.8	15	2:1	1.8:1	50	200	16	3.5
MAR-7	Violet	DC-2000	13.5	13.1	12.5	10.5	8.5	+ 4	+15	5 0	20	2:1	1.5:1	60	275	22	4
MAR-8	Blue	DC-1000	33	28	23	—	19	+10	+15	3.5	27	□	□	65	500	36	7.5

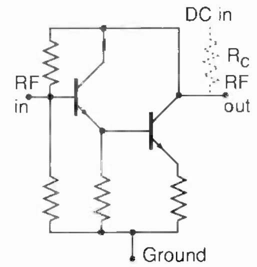
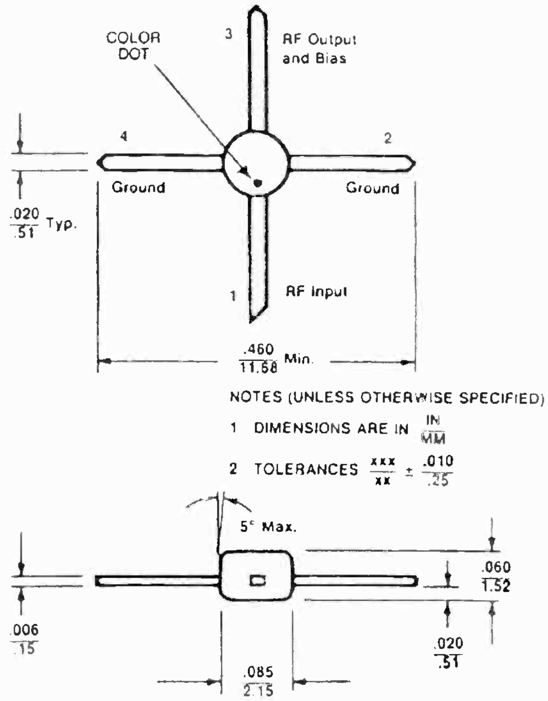


Fig. 1 a) The internal circuit of an MAR series amplifier. The resistor Rc is not part of the device, but provides an external DC feed path, while doubling as an rf choke.



b) Biar resistor values for MAR amplifiers. An rf choke may be advisable in addition, in those cases where the value of R_c is fairly low (i.e. with the lower supply voltages).

Amplifier	Bias Current	Bias Voltage	Approximate Bias Resistor (Ohms)				Resistor Dissipation (Watts)
	I_B (mA)	$+V_O$	+5V	+9V	+12V	+15V	$+V_{CC}$=12V
MAR-1	17	~5	—	235	412	588	.12
MAR-2	25	~5	—	160	280	400	.18
MAR-3	35	~5	—	114	200	286	.25
MAR-4	50	~6	—	60	120	180	.30
MAR-6	16	~3.5	98	344	531	719	.14
MAR-7	22	~4	45	227	364	500	.18
MAR-8	36	~8	—	—	111	194	.14

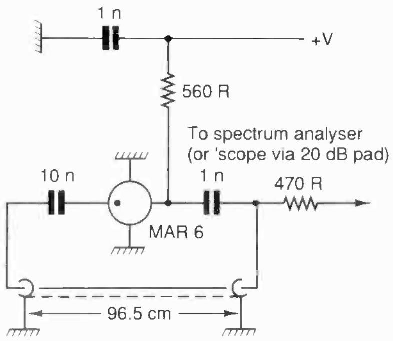
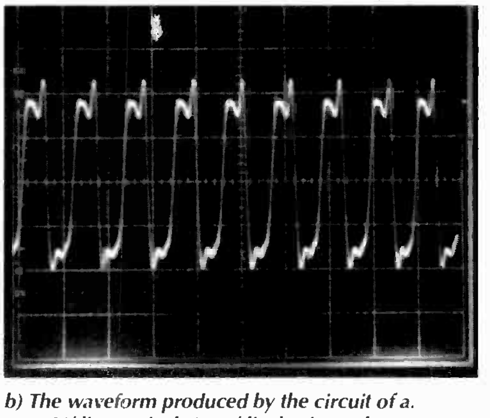
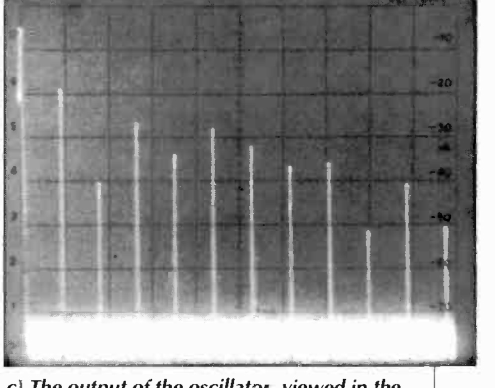


Fig. 2 a) Connecting the amplifier's output back to its input via a halfwavelength of coax causes it to oscillate. In view of the wildly excessive loop gain, this circuit is sheer cruelty to helpless ICs.



b) The waveform produced by the circuit of a. (10mV/div. vertical, 10ns/div. horizontal



c) The output of the oscillator, viewed in the frequency domain. (Ref. level +10dBm but effectively higher due to the 470Ω resistor) vertical 10dB/div, horizontal 100MHz/div, 1MHz IF bandwidth, video filter off.

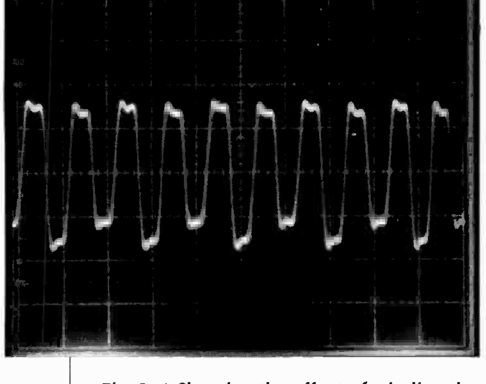
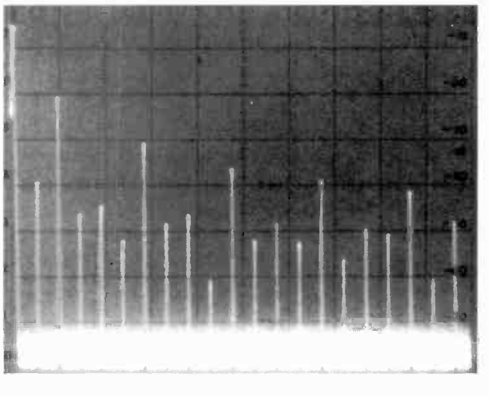
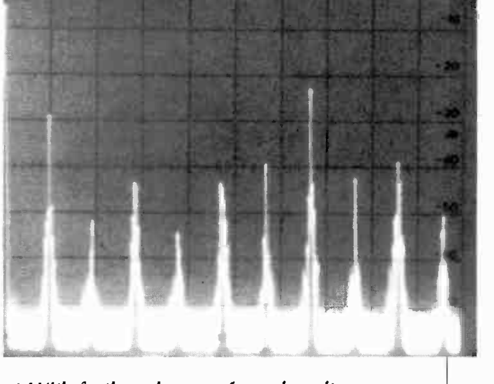


Fig. 3 a) Showing the effect of winding the supply voltage to the circuit of Fig. 2 a down from +12 to +8V, frequency halving is evident.



b) As a), in the frequency domain.



c) With further change of supply volts, behavious is even more chaotic.

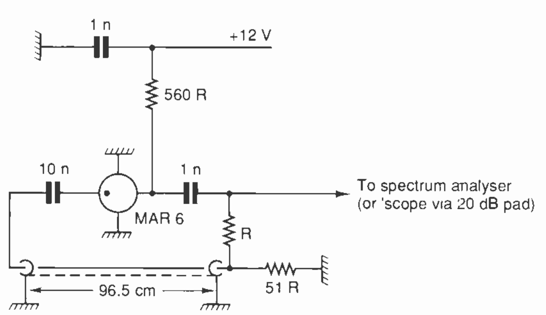
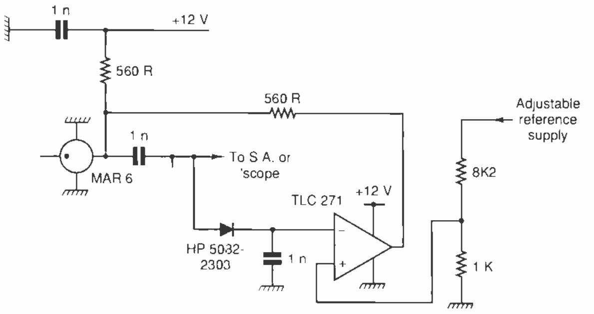
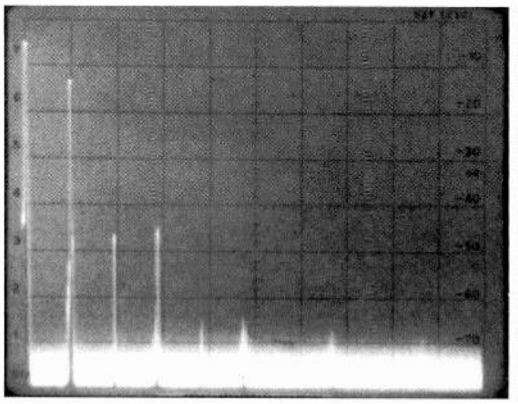


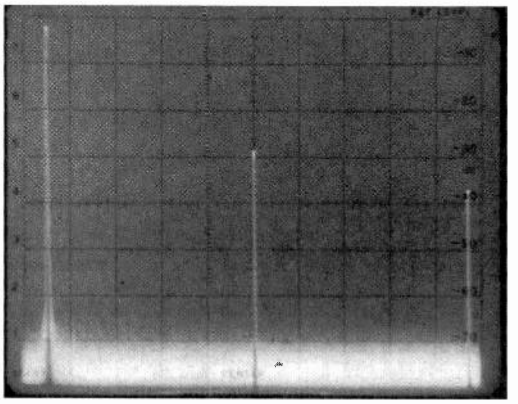
Fig. 4 a) Circuit of Fig. 2 modified to reduce the excessive loop gain. With R = 150Ω the circuit oscillates, but not with R = 220Ω. Second harmonic 30dB below fundamental, third harmonic 20dB down. The reduced output loading due to R enables the amplifier to supply almost its rated output power to an external 50Ω load circuit.



b) An add-on to a, to define the amplitude of oscillation.

c) Performance of a plus b, showing the clean output obtained. Second and third are the only significant harmonics, both well over 30dB down. (Ref. level 0dBm, other settings unchanged.





d) Performance of a alone with reduced line length, giving an operating frequency of 930MHz. (Ref level 0dBm, centre frequency 930MHz, 100MHz/div, horizontal, other settings unchanged.

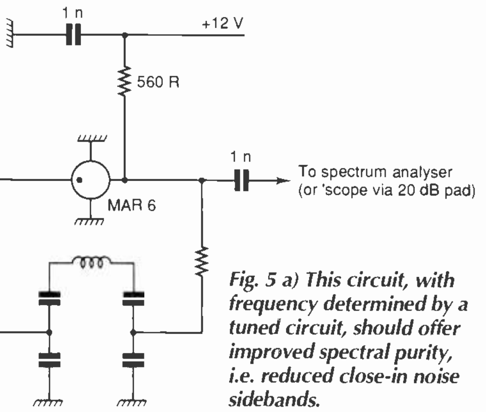
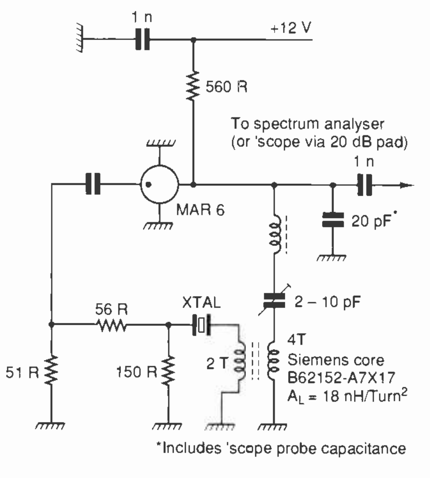
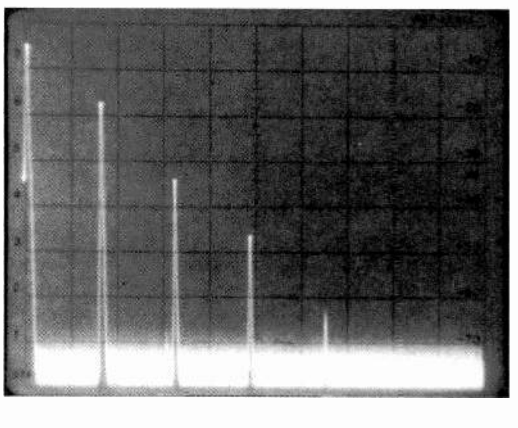


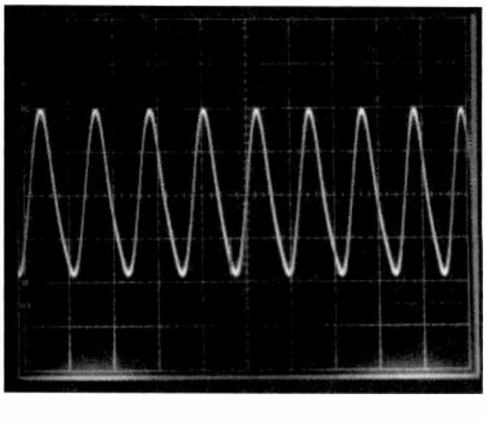
Fig. 5 a) This circuit, with frequency determined by a tuned circuit, should offer improved spectral purity, i.e. reduced close-in noise sidebands.



b) Circuit modified for control by a series resonant overtone crystal.

c) Output of 85MHz crystal controlled oscillator. (Ref level +10dB, 50MHz/div horizontal, 0Hz at LHS, other settings unchanged.





d) Output waveform of the 85MHz crystal oscillator.

circuit behaviour is beginning to be chaotic. It exhibits a tendency to frequency halving (clearly shown in the frequency domain in Fig. **3b** – the circuit has reached the "first bifurcation point", described in Ref.1. A further slight change in supply volts prompts even more chaotic behaviour, with wide noise-like sidebands appearing around the fundamental and harmonics and the seventh harmonic actually greater than the fundamental, Fig. **3c**.

To produce a more sanitary oscillator, attenuation was added in the feedback loop, to reduce the excess loop gain, **Fig. 4a**. With $R = 220\Omega$, the circuit did not oscillate, but did so with $R = 150\Omega$. The amplitude control loop of Fig. **4b** was grafted on and the performance was then as illustrated in Fig. **4c**. The output is very clean; second and third are the only significant harmonics, both well over 30dB down. The circuit of Fig. 4a was then run without the amplitude control loop and with a 2 - 10pF trimmer in parallel with the 51Ω resistor in the feedback network. This provided a 7MHz tuning range, and it was noticeable that at the 10pF setting, the harmonics were substantially lower than at 2pF, due to filtering action on the feedback signal; the levels of 2nd - 4th harmonics seen were respectively –22dB, –28dB, –48dB, higher harmonics being negligible.

To check the performance obtainable at much higher frequencies with such a crude and simple oscillator, the circuit of Fig. 4a was used, with the trimmer removed and the length of line drastically reduced. The circuit oscillated at 930MHz (Fig. **4d**) which shows how well the gain of the *MAR6* is maintained with frequency, since R was still set at 150Ω. Note, however, that the amplitude was substantially reduced. The frequency of oscillation was well below that predicted by the line length, due to the excess phase shift through the amplifier at this frequency, amounting to some 70° according to the data sheet.

Crystal control

The spectral purity of the simple oscillators described above, with their frequency controlled by a length of transmission line, will not of course compare with that obtainable with an oscillator controlled by a high Q tuned circuit. This is because in the latter case, the change of phase shift around the loop with change of frequency is much more rapid than with a $\lambda/2$ transmission line. **Fig. 5a** shows a possible configuration with the necessary 180° phase reversal provided by a tuned circuit, provided with matching for both ports of the amplifier.

Even greater stability and spectral purity will result from crystal control and an 85MHz crystal was connected into the circuit of Fig. **5b**. Being an overtone crystal operating at series resonance, it cannot conveniently be arranged to provide a phase reversal in the same way as a parallel resonant crystal, so a small two hole balun core was used to provide the phase reversal. It was also arranged to step up the impedance presented by the crystal circuit to the amplifier's output, while a series tuned circuit set to resonate at the desired frequency was interposed between the amplifier output and the reversing transformer, to suppress oscillations at any but the intended overtone. Excess loop gain was avoided by fitting a pad between the other side of the crystal and the amplifier's input. As Fig. **5c** shows, the circuit produced an output of –7dBm with low harmonic content, the waveform being shown in Fig. **5d**. This is

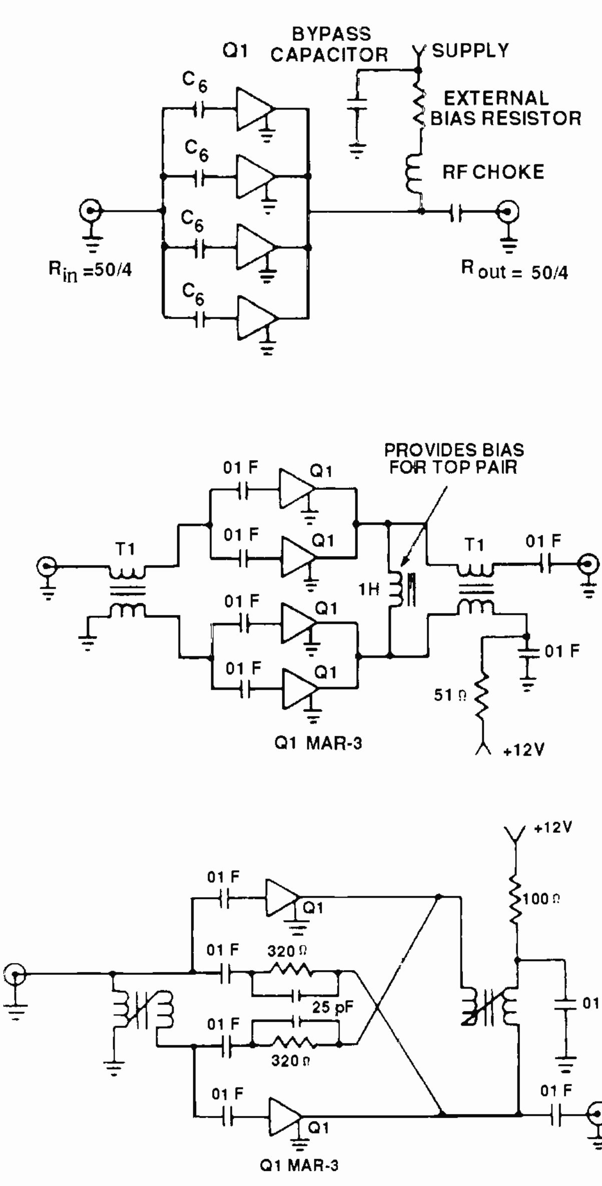


Fig. 6 a) Being unconditionally stable, MAR series amplifiers may be simply paralled for higher output power.

b) A parallel push-pull arrangement offers lower even order harmonic output.

c) A push-pull stage may be unilateralised for greatly increased reverse isolation.

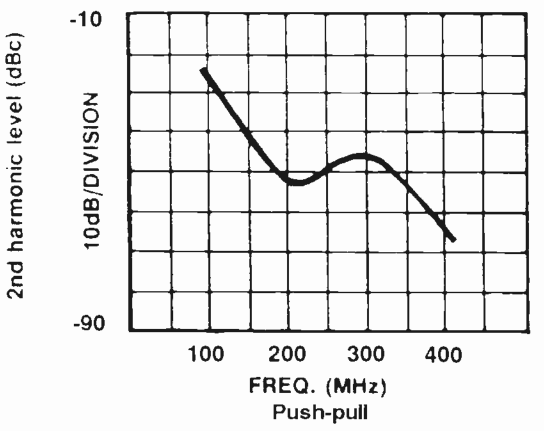
visibly a bit "secondish", not so very different from an asymmetrical triangular wave. By taking the output from the point shown, it has not had the benefit of the filtering action of the frequency selective components. An output with lower harmonic content could be obtained from an additional buffer connected downstream of the series tuned LC circuit, or even from the pad downstream of the crystal.

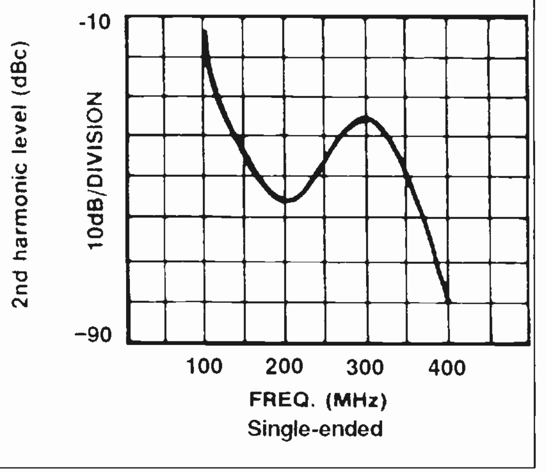
Other considerations

The experiments suggest that these amplifiers are delightfully tame and easy to apply, provided the two earth leads are connected directly to a ground plane. Microstrip construction is recommended with all transmission lines and ground plane running flush to the package, which means mounting it in a hole in the board. But for the experiments reported here, a fairly cavalier approach was adopted. The device was sat on top of the ground plane and the two ground leads were cranked down to connect to connect to it, resulting in lead lengths of a millimeter or more, while other components were mounted using fresh air. Even so,

Fig. 7 a) Comparing the performance of a single ended stage with that of push-pull stages with and without unilateralisation.
b) Level of second harmonic output, dBc, for the push-pull stage.
c) Level of second harmonic output, dBc, for the single ended stage.

Configuration	Freq. (MHz)	Gain (dB)	P_{-1dB} (dBm)	2nd Harmonic @ P_{-1dB} (dB below carrier)
Single-ended	100	12	+10	−15
Push-pull (Unilateralized)	100	15	+13.5	−26
Push-pull	100	12	+17	−34





no problems of instability were encountered.

The results indicate that when the amplifiers are run at well below their output compression point, second harmonic distortion predominates (e.g. Fig. 5c). With overdrive resulting in heavy compression, third harmonic is the largest, as the waveform approaches a squarewave, Fig. 2b and c for instance. Where it is desired to obtain more output than available from a single device while retaining low harmonic levels, *MAR* series amplifiers may be paralleled as in **Fig. 6a** – this is possible since they are (*MAR8* excepted) unconditionally stable. The input and output impedances of the paralleled amplifiers fall within the range that is conveniently accommodated by standard 4:1, 9:1 and 16:1 broadband line transformer configurations. The bandwidth of such a paralleled stage will be limited by the bandwidth of the necessary matching transformers. In narrower bandwidth applications, other matching and combining techniques, such as quarterwave transformers and n-way Wilkinson splitters/combiners can be considered.

Note that the gain of such a compound amplifier is only the same as that of the component individual amplifiers, so to get the desired increased output, additional drive power must be applied. Paralleled amplifiers offer only increased output power, not lower levels of harmonics (unless derated). The four amplifiers in **Fig. 6a** could advantageously be redeployed into the slightly more complex circuit of Fig. **6b**. In this circuit, owing to the push-pull arrangement, even order harmonics will tend to cancel out.

A push-pull pair of *MAR* series amplifiers also lends itself to unilateralisation, Fig. **6c**. (Unlike neutralisation, in which only the reactive components of the devices' internal feedback are cancelled, unilateralisation is a technique in which both the real and the imaginary terms of the feedback elements are cancelled. Consequently, unilateralisation tends to be effective over a wider frequency range than neutralisation.

The reason that these amplifiers are so easily and effectively unilateralised is that the Q of their internal feedback paths is low compared to conventional amplifiers. In an amplifier that has been unilateralised, the reverse isolation is greatly increased, so that variations in the load impedance will no longer affect the input impedance nor variations in the source impedance affect the output impedance. But unilateralisation tends to increase both the amplifier's input and output impedances, so careful attention must be paid to the effects of unilateralisation on input and output match.

Fig. 7a compares the performance of a single ended *MAR* series amplifier with that of a push-pull pair with and without unilateralisaton. It is clear that the only major advantage of unilateralisation is the increased reverse isolation, the straighforward push-pull pair being better on other counts. In particular, unilateralisation has lowered the 1dB compression point by 3.5dB. This is partly due to the power lost in the resistive components of the cross-coupled feedback networks and partly to the effect on input and output impedances. Fig. **7b** and **c** compare the level of second harmonic in dBc for the single ended and push-pull amplifiers respectively. ■

References.

1. Chaos and engineering R. Dettmer IEE Review Sept. 1993 pp 199 - 203
Mini Circuits. For further information contact Tel. 0252 835094 .

OSCILLOSCOPES

HEWLETT PACKARD 54600A – 100 MHz DIGITISING **£1400**
HEWLETT PACKARD 1740A, 1741A, 1744A – 100MHz DUAL CHANNEL **£350**
HEWLETT PACKARD 1707A, 1707B – 75MHz DUAL CHANNEL **£275**
HEWLETT PACKARD 182c – 100MHz FOUR CHANNEL **£350**
TEKTRONIX 453, 454, 465, 465B, 475, 475A, 466, 468 – 100 to 500MHz DUAL CHANNEL **from £250**
TEKTRONIX 7603, 7613, 7623, 7633, 7313, 7704A, 7844, 7904 – 100 to 500 MHz TWO or FOUR CHANNEL **from £350**
(Various plug in options available)
TELEQUIPMENT DM63, D75, D83-50MHz DUAL CHANNEL **from £175**
GOULD MODELS OS 245, OS 4000, OS 4020, OS 4200 **from £125**
GOULD 400-20MHz BAND WIDTH – 100Ms/S SAMPLING RATE – AS NEW **£1350**
PHILIPS 3211 3212, 3217, 3226, 3240, 3243, 3244, 3261, 3262 – 15 to 120MHz TWO AND FOUR CHANNEL **from £125 to £350**
SOLARTRON SCHLUMBERGER CD 1740 – 20MHz – FOUR CHANNEL **from £250**
DUMART 1062 – 50MHz – DUAL CHANNEL **£250**

SPECTRUM ANALYSERS

HEWLETT PACKARD 3580A – 5Hz to 50KHz **£1250**
HEWLETT PACKARD 3582A – 25KHz DUAL CHANNEL **£3000**
HEWLETT PACKARD 3585A – 20Hz to 40KHz (GPIB) **£4250**
HEWLETT PACKARD 182T – MAINFRAME WITH 8559A (10MKHz to 21GHz) **£3750**
HEWLETT PACKARD 8590A – 10MKHz to 1 50 GHz PORTABLE **£4250**
AILTECH 727 – 20GHz **£2200**
MARCONI 2370 – 110MHz **£1250**
ADVANTEST TR4131 – 10KHz – 3 5 GHz **£4500**
HEWLETT PACKARD 141T WITH 8552B + 8554B (500KHz – 1250MHz) **£1500**
HEWLETT PACKARD 8757C – SCALAR NETWORK (20GHz) ANALYSER **£4500**
TEKTRONIX 7603 WITH 7L14 (10KHz – 1 8GHz) **£2000**

MISCELLANEOUS

HEWLETT PACKARD 3701A/3702A/3703A – MICROWAVE LINK ANALYSER **£1000**
HEWLETT PACKARD 3702B/3705A/3710A/3716A – MICROWAVE LINK ANALYSER **£1500**
HEWLETT PACKARD 3779A PRIMARY MULTIPLEX ANALYSER **£1000**
HEWLETT PACKARD 415E – S.W.R. METER **£275**
HEWLETT PACKARD 8750A – STORAGE NORMALISER **£400**
HEWLETT PACKARD 5342A FREQUENCY COUNTER (18GHz) **£1400**
HEWLETT PACKARD 8443A TRACKING GENERATOR COUNTER **£450**
HEWLETT PACKARD 8445B AUTOMATIC PRE-SELECTOR **£700**
HEWLETT PACKARD 4953A – PROTOCOL ANALYSER **£2500**
HEWLETT PACKARD 3456A DIGITAL VOLTMETER **£675**
HEWLETT PACKARD 8011A PULSE GENERATOR **£500**
HEWLETT PACKARD 8013B PULSE GENERATOR **£750**
KEITHLY 197AUTO D M M WITH IEEE **£400**
MARCONI 2830 PCM MULTIPLEX TESTER **£1250**
MARCONI 2831 CHANNEL ACCESS SWITCH **£500**
RACAL-DANA 9081 SYNTHESISED SIGNAL GENERATOR (520 MHz) **£550**
RACAL-DANA 9084 SYNTHESISED SIGNAL GENERATOR **£500**
RACAL-DANA 9303 TRUE RMS – R/F LEVEL METER **£650**
ANRITSU – ME588C – MICROWAVE SYSTEM ANALYSER (AS NEW) **£3500**
AVO – RM215 L/2 – BREAKDOWN/LEAKAGE + IONISATION TESTER **£500**
DATRON 1061A – AUTOCAL MULTI-FUNCTIONAL VOLTMETER (6 5 DIGITS) **£950**
DATRON 1071A – AUTOCAL MULTI-FUNCTIONAL VOLTMETER (7 5 DIGITS) **£1150**
FARNELL RB1030-35 ELECTRONIC LOAD **£450**
GALLENKEMP 300 + SERIES THERMOSTAT/CYCLE TIMER/TEMPERATURE CYCLING OVEN **£750**
ROHDE + SCHWARTZ QDM BN 36711 DIGITAL 'Q' METER **£400**
PHILIPS 5167 1-10 MHz FUNCTION GENERATOR **£400**
PHILIPS PM6672 TIMER COUNTER 1GHz WITH IEEE **£650**
SOLARTRON/SCHLUMBERGER 1170 – FREQUENCY RESPONSE ANALYSER **£500**
SOLARTRON/SCHLUMBERGER 3530 – ORION DATA LOGGING SYSTEM **£750**
WAYNE KERR B905 – AUTOMATIC PRECISION BRIDGE **£800**
WAYNE KERR 3245 – PRECISION INDUCTANCE ANALYSER **£2500**
WILTRON 560 – SCALAR NETWORK ANALYSER **£900**
YOKOGAWA YEW 3655 ANALYSING RECORDER **£650**
B & K – 2511 VIBRATION METER **£1250**
B & K – 2512 VIBRATION METER (2511 + 1621) **£2000**
B & K – 2515 VIBRATION METER (FFT) **£4500**
HEWLETT PACKARD 5335A – UNIVERSAL COUNTER WITH IEEE **£450**
PHILIPS PM2525 – MULTIMETER WITH IEEE **£850**
PHILIPS PM8272 XYT CHART RECORDER **£600**
TEK TM503/SG503/PB506/TG50 SCOPE CALIBRATOR **£2200**
TEK 834 DATA COMMS ANALYSER **£500**
ANRITSU ML93B OPTICAL POWER METER WITH SENSOR **£2000**

MANY MORE ITEMS AVAILABLE – SEND LARGE S.A.E. FOR LIST OF EQUIPMENT
ALL EQUIPMENT LISTED HERE IS USED, WITH 30 DAYS GUARANTEE
Please check availability before ordering. Carriage + VAT to be added to all goods

TELNET

8 Cavans Way, Binley Ind. Estate, Coventry CV3 2SF
(Premises situated close to Eastern Bypass in Coventry with easy access to M1, M6, M40, M42, M69, M45)
TELEPHONE 0203 650702 FAX 0203 650773

CIRCLE NO. 115 ON REPLY CARD

Z8 Z80 Super8 6502 65816 68HC11 68000... 8051...

from only **£69**

and many more

COMPILERS CROSS-ASSEMBLERS SIMULATORS

Call for data sheets

Tel: 081-441 3890
Fax: 081-441 1843

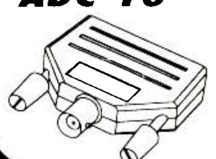
SMART COMMUNICATIONS

CIRCLE NO. 116 ON REPLY CARD

Low cost data acquisition for IBM PCs & compatibles

All our products are easy to install - they connect directly to either the printer or serial port and require no power supply. They are supplied with easy to use software which collects data for either display or print-out.

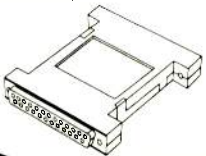
ADC-10



- *8 - bit resolution*
- *one channel*
- *10-25K samples per second*
- *Oscilloscope/Voltmeter software*
- *0-5V input range*
- *Connects to printer port*

£49

ADC-11



- *10 - bit resolution*
- *11 channel*
- *5-10K samples per second*
- *Data logger software*
- *0-2.5V input range*
- *Connects to printer port*

£75

ADC-16

- *8, 12, 16 - bit resolution + sign*
- *8 s/e or 4 differential inputs*
- *2 16 or 300 8-bit samples per second*
- *± 2·5V input range*
- *Data logger software*
- *Connects to serial port*

£99

All prices exclusive of V.A.T.

PICO TECHNOLOGY LTD
Broadway House, 149 - 151 St Neots Road, Hardwick, Cambridge CB3 7QJ
Access VISA **TEL: 0954 - 211716 FAX: 0954 - 211880**

CIRCLE NO. 117 ON REPLY CARD

KESTREL ELECTRONIC COMPONENTS LTD

☆ All items guaranteed to manufacturers' spec.
☆ Many other items available.
'Exclusive of V.A.T. and post and package'

	1+	100+		1+	100+
EPROMS			**STATIC RAMS**		
2764A-250	2.50	2.20	62256ALP-10	3.30	2.60
27C64-150	2.50	2.20	6264ALP-10	1.95	1.48
27128A-200	2.40	2.20	6116ALP-10	1.10	0.80
27256-250	2.30	2.10	628128LP-10	12.00	10.00
27C256-150	2.90	2.30	HD63B21P	1.95	1.50
27C512-150	3.30	2.55	6522P	2.40	1.80
27C010-150	5.80	4.20	65C02P2	2.90	2.50
27C020-150	9.00	7.60	65C21P2	2.90	2.50
68B50P	1.40	0.85	65C22P2	2.75	2.40
D8749H	4.40	3.75	Z80A CPU	1.30	0.99
MM58274CN	7 00	3.95	Z80A P10	0.95	0.75
80C31-12meg	2 60	2.10	Z80A CTC	0.90	0.70
75176BP	1 90	0.95	Z80A DART	2.20	1.38

74LS, 74HC, 74HCT Series available
Phone for full price list
All memory prices are fluctuating daily, please phone to confirm prices

178 Brighton Road,
Purley, Surrey CR8 4HA
Tel: 081-668 7522. Fax: 081-668 4190.

CIRCLE NO. 118 ON REPLY CARD

In the penultimate part of his series on power amplifier design, Douglas Self presents a worked example delivering 0.0015% total harmonic distortion. But this performance is only attainable with correctly chosen compensation...

Distortion in power amplifiers

7: Frequency compensation and real designs

The distortion performance of an amplifier is determined not only by open loop linearity, but also the negative feedback factor applied when the loop is closed. In most practical circumstances doubling the NFB factor halves the distortion. To date, this series has focused on basic circuit linearity. I have assumed that open loop gain falls at 6dB/octave due to a single dominant pole, with the amount of NFB permissible at hf being set by the demands of hf stability. Because of this, the distortion residuals from a 'blameless' amplifier are comprised almost entirely of crossover artifacts due to their high frequency content. Audio amplifiers using more advanced compensation are rather rare. However, certain techniques do exist...

This series has stuck close to conventional topologies, because even commonplace circuitry has been shown to have little known aspects and interesting possibilities. This implies a two-gain-stage circuit (unity gain output stages not being counted) with most of the feedback applied globally, but smoothly transferred to the voltage amplifier stage alone as frequency increases. Other configurations are possible; a one stage amplifier is an intriguing possibility – they are common in cmos op-amps – but is probably ill-suited to power amp impedances. See reference 1 for an eccentric three-stage amplifier with an open loop gain of just 52dB (due to the dogged use of local feedback) and only 20dB of global feedback. Most of the section below refers only to the conventional two-stage structure.

Making a pole dominant

Dominant pole compensation is the simplest kind, though its implementation involves subtlety. Simply take the lowest pole to hand (P1), and make it dominant, ie so much lower in frequency than the next pole P2 that the total loop gain (the open loop gain as reduced by the attenuation in the feedback network) falls below unity before enough phase shift accumulates to cause hf oscillation. With a single pole, the gain must fall at 6dB/octave, corresponding to a constant 90° phase shift. Thus the phase margin will be 90° giving good stability. **Figure 1a** shows the traditional Miller method of making a dominant pole. The collector pole of Tr_4 is lowered by adding the Miller capacitance C_{dom} to that which

Fig. 1. Implementing dominant-pole compensation. (a) Miller capacitor, (b) Shunt-lag circuit (c) Output-stage Inclusive Miller compensation. (d) How to implement 2-pole compensation. *See p 140.*

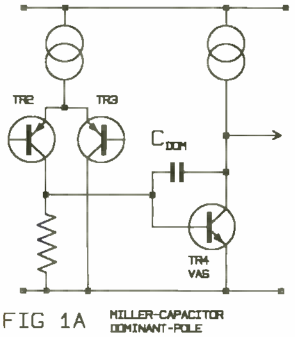


FIG 1A MILLER-CAPACITOR DOMINANT-POLE COMPENSATION.

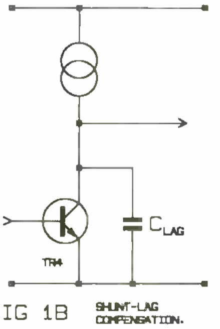


FIG 1B SHUNT-LAG COMPENSATION.

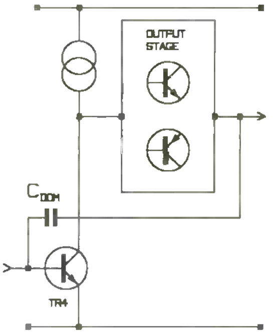


FIG 1C OUTPUT-STAGE-INCLUSIVE MILLER COMPENSATION.

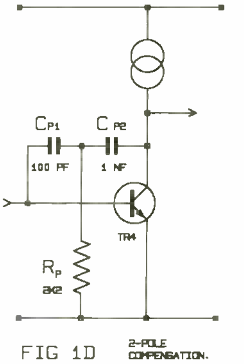


FIG 1D 2-POLE COMPENSATION.

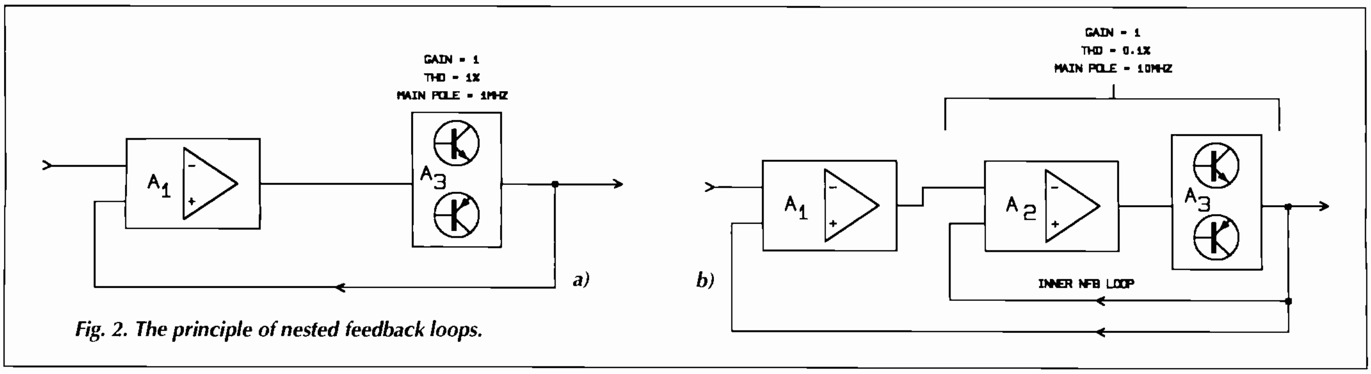


Fig. 2. The principle of nested feedback loops.

unavoidably exists as the C_{bc} of the VAS transistor. However there are other beneficial effects; C_{dom} causes "pole splitting", in which the pole at Tr_2 collector is pushed up in frequency as P1 moves down – most desirable for stability. Simultaneously the local NFB through C_{dom} linearises the vas.

Assuming that input stage transconductance is set to a plausible 5mA/V, and stability considerations set the maximal 20kHz open loop gain to 50dB, then from the equations in Part 1, C_{dom} must be 125pF. This is more than enough to swamp the internal capacitances of the vas transistor, and is a realistic value.

The peak current that flows in and out of this capacitor for an output of 20V rms, 20kHz, is 447μA. Recalling that the input stage must sink C_{dom} current while the vas collector load sources it, and likewise the input stage must source it while the vas sinks it, there are four possible places in which slew rate might be limited by inadequate current capacity. If the input stage is properly designed then the usual limiting factor is vas current sourcing. In this example a peak current of less than 0.5mA should be easy to deal with, and the maximum frequency for unslewed output will be comfortably above 20kHz.

Figure **1b** shows a much less satisfactory method – the addition of capacitance to ground from the vas collector. This is usually called shunt lag compensation, and as Peter Baxandall aptly put it, "The technique is in all respects sub-optimal[2]."

We have already seen in Part 3 that loading the vas collector resistively to ground is a very poor option for reducing LF open loop gain, and a similar argument shows that capacitive loading to ground for compensation purposes is an even worse idea. To reduce open loop gain at 20kHz to 50dB as before, the shunt capacitor C_{lag} must be 43.6nF, which is a whole different order of things from 125pF. The current flowing in C_{lag} at 20V rms, 20kHz, is 155mA peak, which is going to require some serious electronics to provide it. This important result can be derived by simple calculation, and I have confirmed it with Spice simulation. The input stage no longer constrains the slew rate limits, which now depend entirely on the vas.

A vas working under these conditions is almost certain to have poor linearity. The current variations in the stage, caused by the extra loading, produces more distortion and there is now no local NFB through a Miller capacitor to correct it. To make matters worse, the dominant pole P1 will probably need to be set to a lower frequency than for the Miller case, to maintain the same stability margins, as there is now no pole splitting to raise the pole at the input stage collector. Hence C_{lag} may have to be even larger, and require even higher peak currents. Takahashi has produced a fascinating paper on this approach[3], showing one way of heaving about the enormous compensation currents required for good slew rates. The only thing missing is an explanation of why shunt compensation was chosen in the first place.

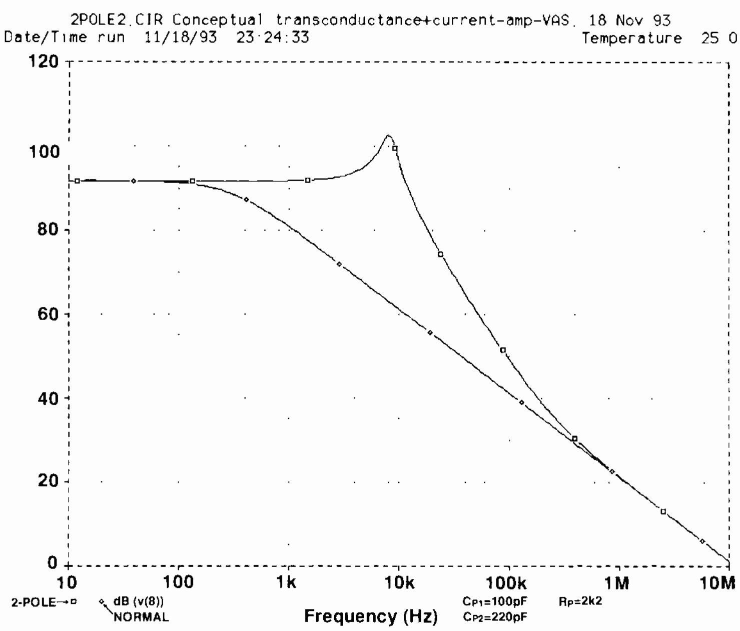


Fig. 3. J3 Spice plot of the open-loop gain of a 2-pole compensated amplifier. The difference between the two plots shows the amount of extra NFB possible.

Including the output stage

Miller capacitor compensation elegantly solves several problems at once, and the decision to use it is not hard. However the question of whether to include the output stage in the Miller feedback loop is less easy. Such inclusion (see Fig. **1c**) presents the desirable possibility that local feedback could linearise both the vas and the output stage, with just the input stage left out in the cold as frequency rises and global NFB falls. This idea is most attractive as it would greatly increase the feedback available to linearise a Class B output stage.

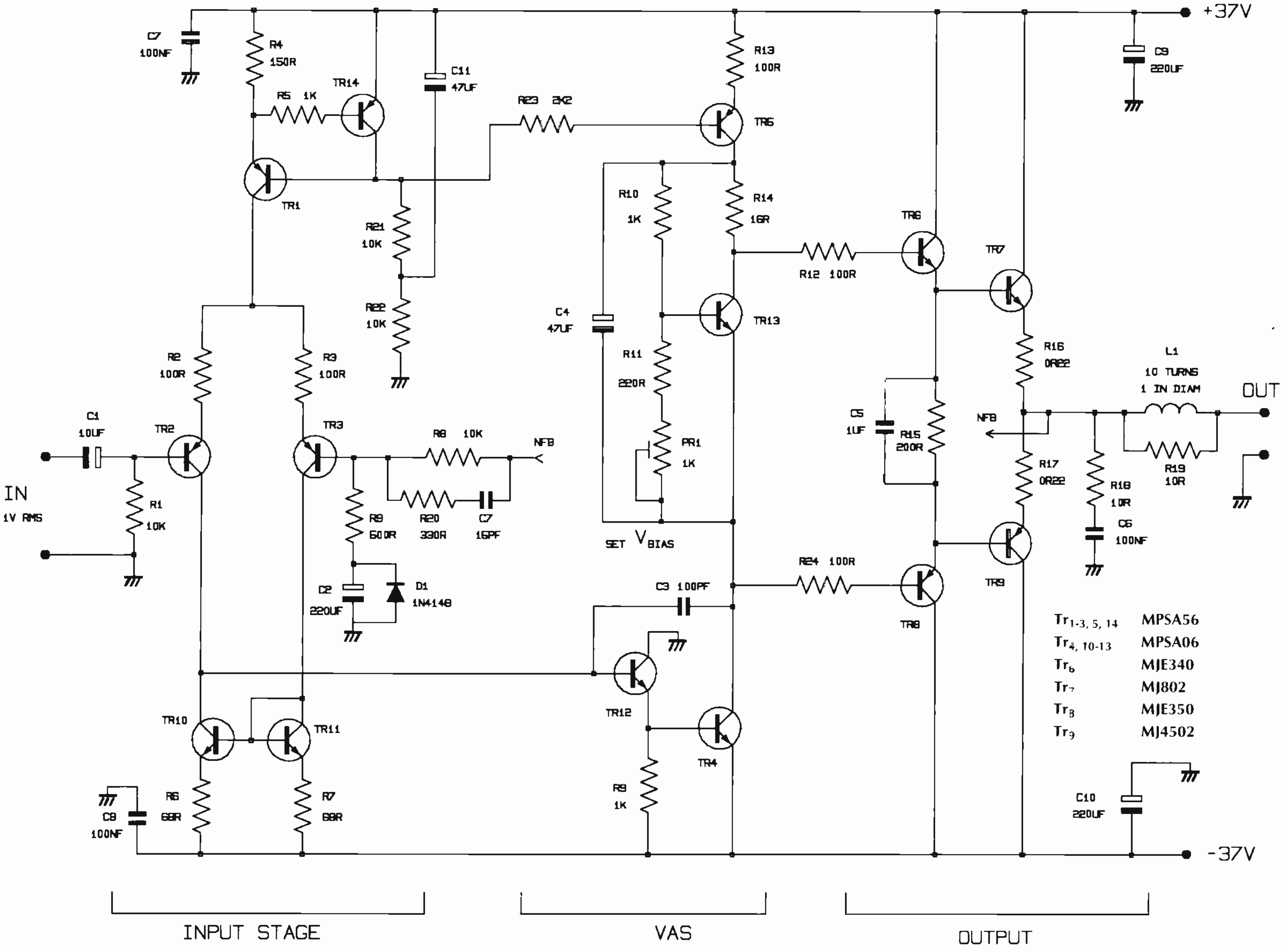
There is certainly *some* truth in this where applying C_{dom} around the output as well as the V_{as} reduced the peak 1kHz THD from 0.05% to 0.02%[4]. However it should be pointed out that the output stage was deliberately under biased to induce crossover spikes because, with optimal bias, the improvement was too small to be either convincing or worthwhile. Also, this demonstration used a model amplifier with TO-92 "output" transistors. In my experience this technique just does not work with real power bipolars because it induces intractable HF oscillation.

The use of local NFB to linearise the vas demands a tight loop with minimal extra phase shift beyond that inherent in the C_{dom} dominant pole. It is permissible to insert a cascode or a small signal emitter follower into this local loop, but a sluggish output stage seems to be pushing the phase margin too far; the output stage poles are now included in the loop, which loses its dependable HF stability. Bob Widlar has stated that output stage behaviour must be well controlled up to 100MHz for the technique to be reliable[5]. This would appear to be virtually impossible for discrete power stages with varying loads.

While I have so far not found "Inclusive Miller compensation" to be workable myself, others may know different. If anyone can shed further light I would be most interested.

Nested feedback loops

Nested feedback is a way to apply more NFB around the output stage without increasing the



global feedback factor. The output has an extra voltage gain stage bolted on, and a local feedback loop is closed around these two stages. This NFB around the composite bloc reduces output stage distortion and increases frequency response, to make it safe to include in the global NFB loop.

Suppose that bloc A_1 (**Fig. 2a**) is a distortionless small signal amplifier providing all the open loop gain and so including the dominant pole. A_3 is a unity gain output stage with its own main pole at 1MHz and distortion of 1% under given conditions: this 1MHz pole puts a firm limit on the amount of global NFB that can be safely applied.

Fig **2b** shows a nested feedback version; an extra gain bloc A_2 has been added, with local feedback around the output stage. A_3 has the modest gain of 20dB so there is a good chance of stability when this loop is closed to bring the gain of A_3+A_2 back to unity. A_2 now experiences 20dB of NFB, bringing the distortion down to 0.1%, and raising the main pole to

Fig. 4. 50W Class B amplifier circuit diagram. Transistor numbers correspond with the generic amplifier in Part 1.

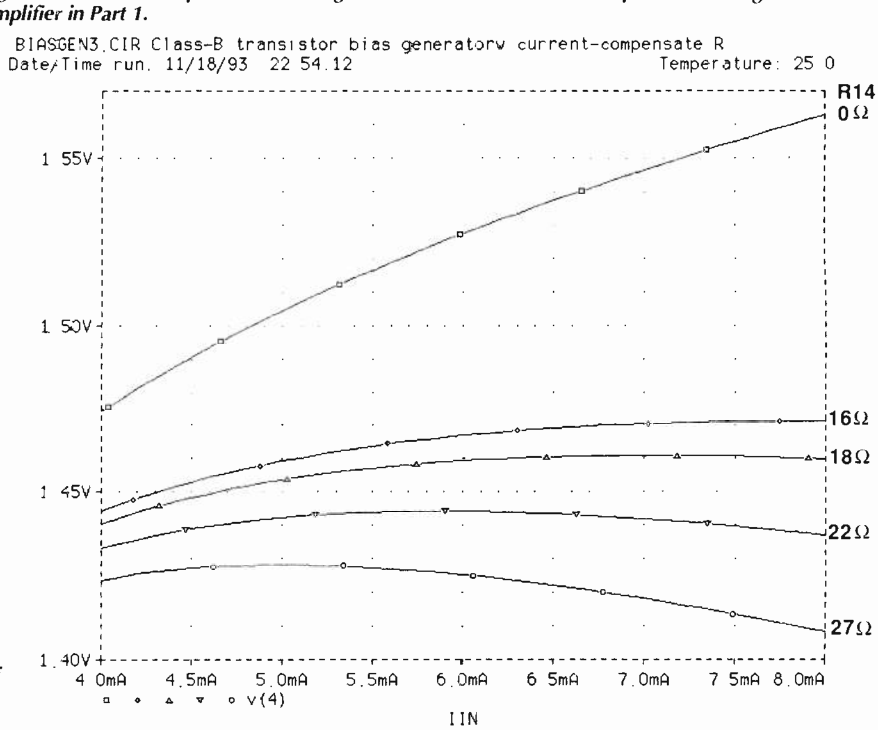


Fig. 5. Spice plot of the voltage-peaking behaviour of a current-compensated bias generator.

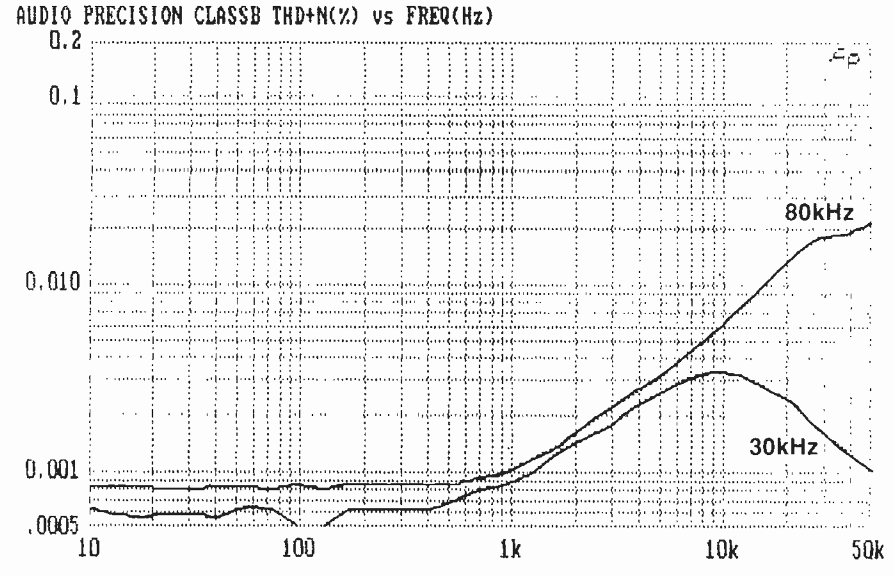


Fig. 6. Class B amplifier: THD performance at 50W/8-ohm; measurement bandwidths 30kHz and 80kHz.

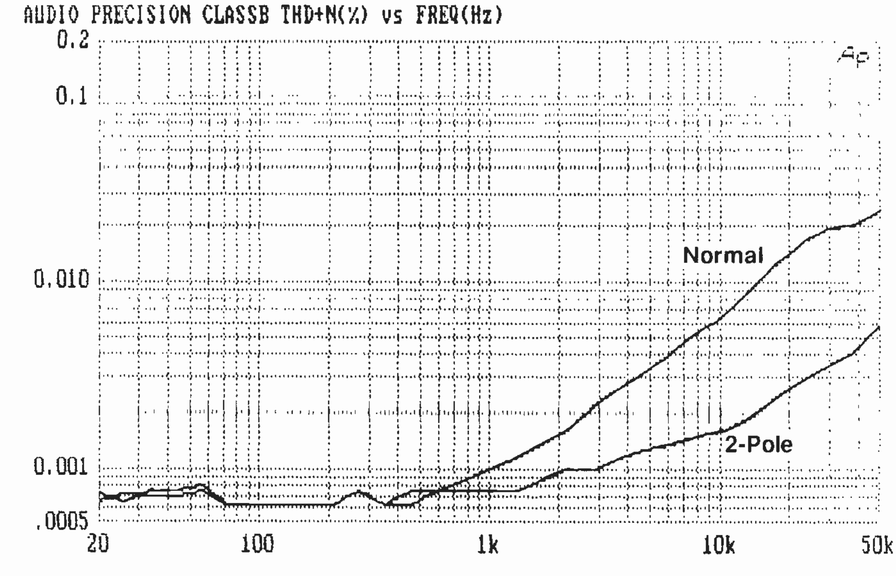


Fig.7. The dramatic THD improvement obtained by converting the Class B amplifier to 2-pole

10MHz, which should allow the application of 20dB more global NFB around the overall loop that includes A_1. We have thus decreased the distortion that exists before global NFB is applied, and simultaneously increased the amount of NFB that can be safely used, promising that the final linearity could be very good indeed. For another theoretical example see reference 6.

Real life examples of this technique in power amps are not easy to find, but a variation is widely used in op-amps. Many of us were long puzzled by the way that the much loved *5534* maintained such low THD up to high frequencies. Contemplation of its entrails appears to reveal a three-gain stage design with an inner Miller loop around the third stage, and an outer Miller loop around the second and third stages; global NFB is then applied externally around the whole lot. Nested Miller compensation has reached its apotheosis in cmos op-amps – the present record appears to be three nested Miller loops plus the global NFB[7]. Don't try this one at home.

Two pole compensation

Two pole compensation is a mildly obscure technique for squeezing the best performance from an op-amp[8,9], but it has rarely been applied to power amplifiers. I know of only one example[5]. An extra HF time constant is inserted in the C_{dom} path, giving an open loop gain curve that initially falls at almost 12dB/octave, but which gradually reverts to 6dB/octave as frequency continues to increase. This reversion is arranged to happen well before the unity loop gain line is reached, and so stability should be the same as for the conventional dominant pole scheme, but with increased negative feedback over part of the operational frequency range. The faster gain roll off means that the maximum amount of feedback can be maintained up to a higher frequency. There is no measurable mid band peak in the closed loop response.

One should be cautious about any circuit arrangement which increases the NFB factor. Power amplifiers face loads that vary widely: it is difficult to be sure that a design will always be stable under all circumstances. This makes designers rather conservative about compensation, and I approached this technique with some trepidation. However, results were excellent with no obvious reduction in stability. **Figure 7** shows the result of applying this technique to the Class B amplifier described below.

The simplest way to implement two pole compensation is shown in Fig **1d**, with typical values. C_{P1} should have the same value as it would for stable single pole compensation, and C_{P2} should be at least twice as big; R_p is usually in the region 1k-10k. At intermediate frequencies C_{P2} has an impedance comparable with R_p, and the resulting extra time constant causes the local feedback around the vas to increase more rapidly with frequency, reducing the open loop gain at almost 12dB/octave.

At HF the impedance of R_p is high compared with C_{P2}, the gain slope asymptotes back to 6dB/octave, and then operation is the same as conventional dominant pole, with C_{dom} equal to the series capacitance combination. So long as the slope returns to 6dB/octave before the unity loop gain crossing occurs, there seems no obvious reason why the Nyquist stability should be impaired.

Fig. 3 shows a simulated open loop gain plot for realistic component values; C_{P2} should be at least twice C_{P1} so the gain falls back to the 6dB/octave line before the unity loop gain line is crossed. The potential feedback factor has been increased by more than 20dB from 3kHz to 30kHz, a region where THD tends to increase due to falling NFB. The open loop gain peak at 8kHz looks extremely dubious, but I have so far failed to detect any resulting ill effects in the closed loop behaviour.

There is however a snag to the simple approach shown here, which reduces the linearity improvement. Two-pole compensation may decrease open loop linearity at the same time as it raises the feedback factor that strives to correct it. At HF, C_{P2} has low impedance and allows R_p to directly load the vas collector to ground. This worsens vas linearity as we have seen. However, if C_{P2} and R_p are correctly proportioned the overall reduction in distortion is dramatic and extremely valuable. When two pole compensation was added to **Fig. 4**, the crossover glitches on the THD residual almost disappeared, being partially replaced by low level 2nd harmonic which almost certainly results from vas loading. The positive slew rate will also be slightly reduced.

This looks like an attractive technique, as it can be simply applied to an existing design by adding two inexpensive components. If C_{P2} is much larger than C_{P1}, then adding/removing R_p allows instant comparison between the two kinds of compensation. Be warned that if an amplifier is prone to HF parasitics then this kind of compensation may exacerbate them.

Design example: a 50W class B amplifier
Figure 4 shows a design example of a Class B amplifier intended for domestic audio. Despite its conventional appearance, the circuit delivers a far better distortion performance than that normally associated with the arrangement.

With the supply voltages and values shown it delivers 50W/8Ω from 1V rms input. In previous articles I have used the word *blameless* to describe amplifiers in which all distortion mechanisms, except the apparently unavoidable ones due to Class B, have been rendered negligible. This circuit has the potential to be *blameless*, but achieving this depends on care in cabling and layout. It does not aim to be a cookbook project; for example, overcurrent and DC offset protection are omitted.

The investigation presented in parts 4 and 5 concluded that power fets were expensive, inefficient and non linear. Bipolars make good output devices. The best BJT configurations were the emitter follower type II, with least output switch-off distortion, and the complementary feedback pair (CFP) giving best basic linearity and quiescent stability.

I assume that domestic ambient temperatures will be benign, and the duty moderate, so that adequate quiescent stability can be attained by suitable heatsinking and thermal compensation. The configuration chosen is therefore emitter follower type II, which has the advantage of reducing switch-off nonlinearities (Distortion 3c) due to the action of R_{15} in reverse biasing the output base emitter junctions as they turn off. The disadvantage is that quiescent stability is worse than for the CFP output topology, as there is no local feedback loop to servo out V_{be} variations in the hot output devices.

The NFB factor was chosen as 30dB at 20kHz, which should give generous HF stability margins. The input stage (current source Tr_1,Tr_{14} and differential pair $Tr_{2,3}$) is heavily degenerated by R_2,R_3 to delay the onset of third harmonic Distortion 1. To assist this the contribution of transistor internal r_e variation is minimised by using the unusually high tail current of 4mA. $Tr_{10,11}$ form a degenerated current mirror that enforces accurate balance of the $Tr_{2,3}$ collector currents, preventing second harmonic distortion. Tail source $Tr_{1,14}$ has a basic PSRR 10dB better than the usual two diode version, though this is academic when C_{11} is fitted.

Input resistor R_1 and feedback arm R_8 are made equal and kept as low as possible consistent with a reasonably high input impedance, so that base current mismatch caused by beta variations will give a minimal DC offset. This does not affect Tr_2-Tr_3 V_{be} mismatches, which appear directly at the output, but these are much smaller than the effects of I_b. Even if $Tr_{2,3}$ are high voltage types with low beta, the output offset should be within ±50mV, which should be quite adequate, and eliminates balance presets and DC servos. A low value for R_8 also gives a low value for R_9, which improves the noise performance.

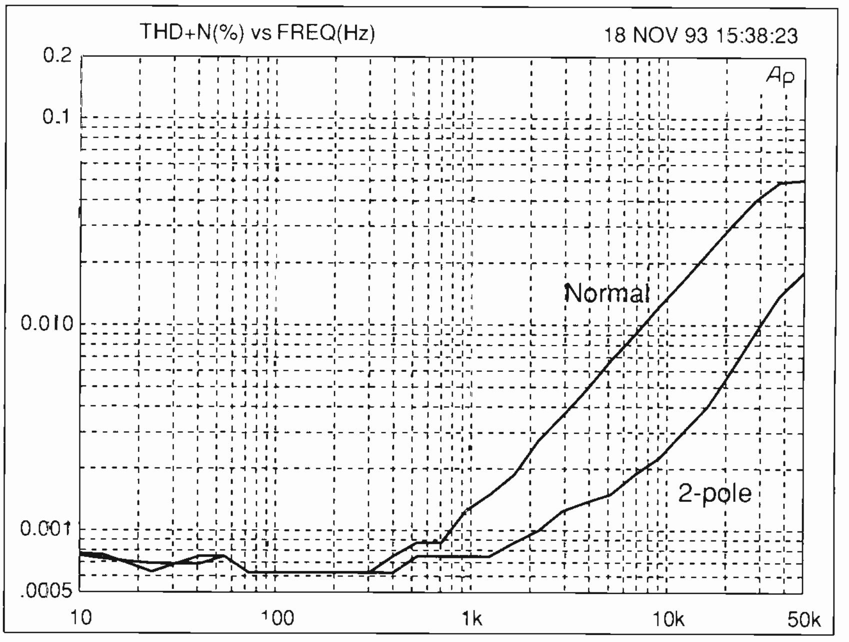


Fig. 8. Class B amplifier with simple quasi-complementary output. Lower trace is for two-pole compensation (80kHz bandwidth).

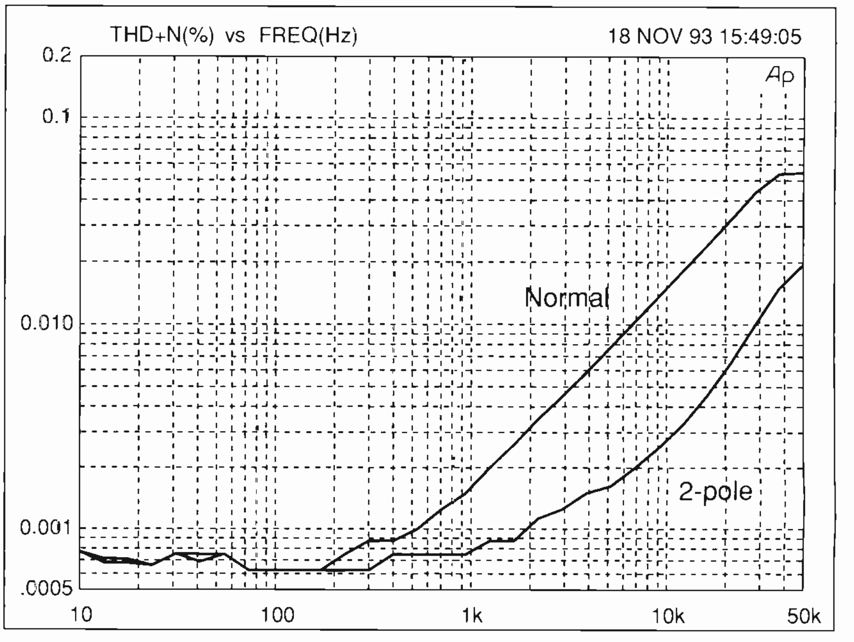


Fig. 9. Class B amplifier with quasi-comp plus Baxandall diode output. Lower trace is the 2-pole case (80kHz bandwidth).

The value of C_2 shown (220μF) gives an LF roll off with R_9 that is –3dB at 1.4Hz. The aim is not an unreasonably extended sub-bass response, but to prevent an LF rise in distortion due to capacitor non linearity.

For example, 100μF degraded the THD at 10Hz from less than 0.0006% to 0.0011%. Band limiting should be done earlier, with non electrolytic capacitors. Protection diode D_1 prevents damage to C_2 if the amplifier suffers a fault that makes it saturate negatively; it looks unlikely but causes no measurable distortion[10]. C_7 provides some stabilising phase advance and limits the closed loop bandwidth; R_{20} prevents it upsetting Tr_3.

The vas stage is enhanced by an emitter follower inside the Miller compensation loop, so that the local NFB which linearises the vas is increased by augmenting total vas beta, rather than by increasing the collector impedance by cascoding. The extra local NFB effectively eliminates vas nonlinearity (Distortion 2).

Increasing vas beta like this presents a much lower collector impedance than a cascode stage due to the greater local feedback. The improvement appears to make a vas buffer to

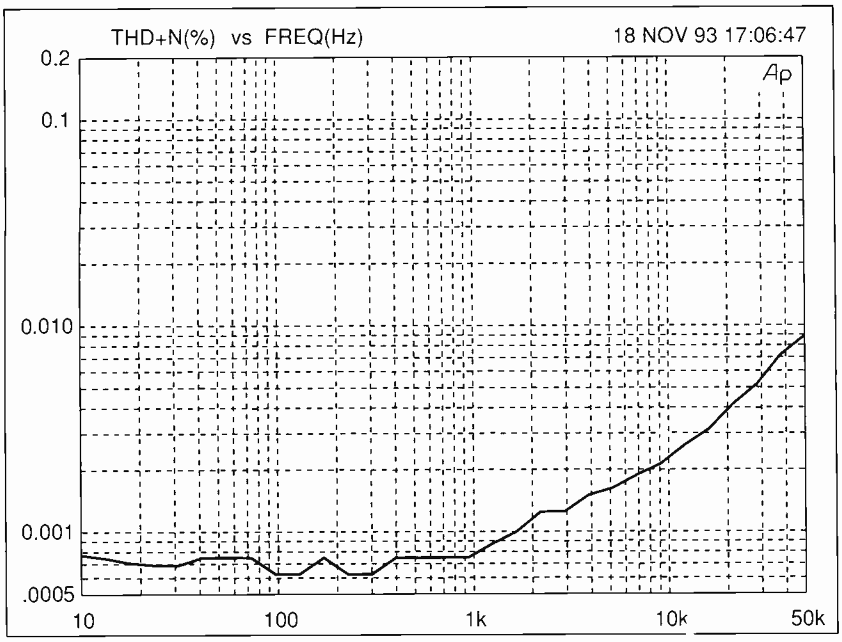


Fig. 10. Class B amplifier with complementary feedback pair (CFP) output stage.

eliminate Distortion 4 (loading of vas collector by the nonlinear input impedance of the output stage) unnecessary. C_{dom} is relatively high at 100pF, to swamp transistor internal capacitances and circuit strays, and make the design predictable. The slew rate calculates as 40V/μsec. The vas collector load is a standard current source, to avoid the uncertainties of bootstrapping.

Quiescent current stability

Since almost all the THD from a *blameless* amplifier is crossover, keeping the quiescent optimal is essential. Quiescent stability requires the bias generator to cancel out the V_{be} variations of four junctions in series; those of two drivers and two output devices. Bias generator Tr_{13} is the standard V_{be} multiplier, modified to make its voltage more stable against variations in the current through it. These occur because the biasing of Tr_5 does not completely reject rail variations: its output current drifts initially due to heating thus changing its V_{be}. Keeping a Class B quiescent stable is hard enough at the best of times, and so it makes sense to keep these extra factors out of the equation.

The basic V_{be} multiplier has an incremental resistance of about 20Ω; in other words its voltage changes by 1mV for a 50μA drift in standing current. Adding R_{14} converts this to a gently peaking characteristic that can be made perfectly flat at one chosen current; see **Fig. 5**. Setting R_{14} to 22Ω makes the voltage peak at 6mA, and standing current now must deviate from this value by more than 500μA for a 1mV bias change. The R_{14} value needs to be altered if Tr_5 is run at a different current. For example, 16Ω makes the voltage peak at 8mA instead. If TO3 outputs are used, the bias generator should be in contact with the top or can of one of the output devices, rather than the heatsink, as this is the fastest and least attenuated source for thermal feedback.

Output stage

The output stage is a standard double emitter follower apart from the connection of R_{15} between the driver emitters without connection to the output rail. This gives quicker and cleaner switch-off of the outputs at high frequencies; this may be significant from 10kHz upwards dependent on transistor type. Speed up capacitor C_5 improves the switch-off action. C_6, R_{18} form the Zobel network while L_1, damped by R_{19}, isolates the amplifier from load capacitance.

Figure 6 shows the 50W/8Ω distortion performance, about 0.001% at 1kHz, and 0.006% at 10kHz. The measurement bandwidth makes a big difference to the appearance, because what little distortion is present is crossover derived, and so high order. It rises at 6dB/octave, the rate at which feedback factor falls. The crossover glitches emerge from the noise, like Grendel from the marsh, as the test frequency increases above 1kHz. There is no precipitous THD rise in the ultrasonic region, and so I suppose this might be called a high speed amplifier.

Note that the zigzags on the LF end of the plot are measurement artifacts, apparently caused by the Audio Precision system trying to winkle distortion from visually pure white noise. Below 700Hz the residual was pure noise with a level equivalent to approx 0.0006% at 30kHz bandwidth. The actual THD here must be microscopic.

This performance can only be obtained if all seven of the distortion mechanisms are properly addressed; Distortions 1-4 are determined by the circuit design, but the remaining three depend critically on physical layout and grounding topology.

Figure 7 shows the startling results of applying 2-pole compensation to the amplifier. C_3 remains 100pF, while C_{P2} was 220pF and R_p 1kΩ. The extra NFB does its work extremely well, the 10kHz THD dropping to 0.0015%, while the 1kHz figure can only be guessed at. There were no unusual signs of instability on the bench, but I have not tried a wide range of loads.

This experimental amplifier was rebuilt with three alternative output stages: the simple quasi-complementary, the quasi-Baxandall and the CFP. The results for both single and two pole compensation are shown in **Figs 8, 9,** and **10**. The simple quasi comp generates more crossover distortion, as expected, and the quasi Baxandall version is not a lot better, due to remaining asymmetry around the crossover region. The CFP gives even lower distortion than the original EF-II output. **Figure 10** shows only the result for single pole compensation; in this case the improvement with two pole was marginal and the trace is omitted for clarity. ■

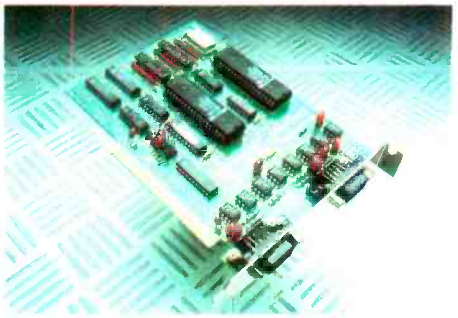
The AP plots in earlier parts of this series were mostly done with an amplifier similar to Fig. 6, though of higher power. Main differences were the use of a cascode-vas with a buffer, and a CFP output to minimise distracting quiescent variations. Measurements at powers above 100W/8Ω used a version with two paralleled output devices.

Next month: In the final part of this series, Douglas Self presents a full Class A amplifier design.

References

1. Otala, *An Audio Power Amplifier for Ultimate Quality Requirements.* IEEE Trans on Audio & Electroacoustics, Vol AU-21, No.6, Dec 1973.
2. Baxandall, *Audio Power Amplifier Design.* Part 4. *Wireless World,* July 1978, p76.
3. Takahashi, *et al, Design & Construction of High Slew-Rate Amplifiers.* AES 60th Convention, Preprint No. 1348 (A-4) 1978.
4. Self, *Crossover Distortion & Compensation.* Letters, *Electronics & Wireless World,* Aug 1992, p. 657.
5. Widlar, *A Monolithic Power Op-Amp. IEEE Journal of Solid-State Circuits,* Vol 23, No 2, April 1988
6. Bonello, *Advanced Negative Feedback Design for High Performance Amplifiers.* AES 67th Convention, Preprint No. 1706 (D-5) 1980.
7. Pernici, *A CMOS Low-Distortion Amplifier with Double-Nested et al Miller Compensation. IEEE J. Solid-State Circuits,* July 1993, p. 758.
8. *Fast Compensation Extends Power Bandwidth* Linear Brief 4, National Semiconductor Linear Applications Handbook 1991.
9. Feucht, *Handbook of Analog Circuit Design.* Academic Press 1990, p. 264.
10. Self, *An Advanced Preamplifier. Wireless World,* Nov 76, p. 43.

Dual port serial communications for your PC



The PC 47,48 and 49AT boards each provide two independent serial ports for any PC/XT/AT(ISA) computer. RS232, RS422 and RS485 standards are supported and all combinations are possible with this range of low power boards.

9 pin D connectors are provided for the RS422 and 485 ports. RS232 can be connected via 9 or 25 way D connectors and charge pump circuitry ensures signal levels in excess of 7V are transmitted.

Interrupt and base addresses are independently selectable for each port, full, half duplex and multidrop communications are fully supported and all boards use the industry standard 82C450, UART.

Designed and manufactured by Amplicon Liveline each board is supplied with a comprehensive technical manual and interrupt driven device driver software is available.

Write in number 1

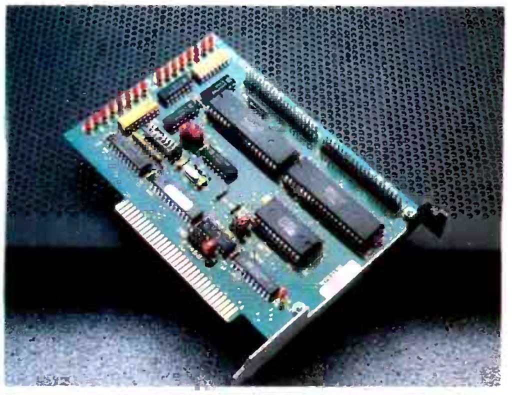
48 lines TTL I/O with 16bit counter timers

PC 14AT available from Amplicon Liveline is a high quality general purpose interface board for PC/XT/AT(ISA) computers.

The board has 48 lines of TTL compatible digital I/O provided as six 8 bit ports four of which can be programmed to be all inputs or all outputs and two which can be split to be 4 inputs and 4 outputs if required. There are also three 16bit counter timers which can be used to generate and measure pulses at up to 5MHz.

An on board 4MHz crystal oscillator is also included on PC 14AT along with LEDs showing the status of a selection of the I/O ports, these can be used as programming and debugging aids.

PC 14AT is suitable for a wide variety of interface, monitoring and control applications



and an LP (low power) version PC 14LP is available for installation in laptop computers, both boards have flexible base address and interrupt support. **Write in number 3**

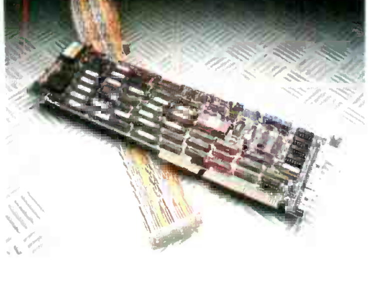
750kHz professional data acquisition with Windows support

PC226 from Amplicon Liveline provides 16 true differential 12 bit analog inputs each with dynamically programmable gain. PC226 has sample rates up to 400kHz multichannel and 750kHz single channel, programmable scanning hardware, flexible triggers and a 2048 sample FIFO to ease programming in high speed applications.

PC 226 is supplied with menu driven software for DOS and Windows, LabTech Notebook drivers and comprehensive well documented libraries for most popular high level languages.

Optional software support for Microsoft Windows includes a complete Dynamic Link Library (DLL) and two icon driven packages, Signal Centre for signal capture and analysis and TRACS for process monitoring, visualisation and control.

Write in number 2



CIRCUIT IDEAS

SEND YOUR CIRCUIT IDEAS TO THE EDITOR, ELECTRONICS WORLD, QUADRANT HOUSE, THE QUADRANT, SUTTON, SURREY SM2 5AS

DO YOU HAVE A £100 CIRCUIT? EACH MONTH'S TOP CIRCUIT IDEA AUTHOR WILL RECEIVE £100. ALL OTHER PUBLISHED IDEAS WILL BE WORTH £25. WE ARE LOOKING FOR INGENUITY AND ORIGINALITY IN THE USE OF MODERN COMPONENTS

Smart fet battery charger

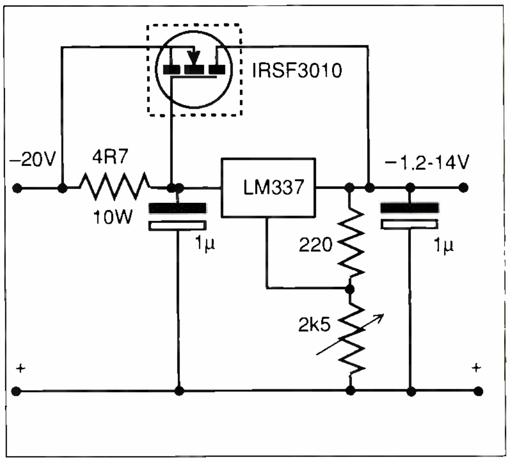
£100 WINNER

This circuit uses an external transistor wrap-around to boost the current capability of a voltage regulator for a constant voltage lead-acid battery charging application. Using the International Rectifier *IRFS3010* smart mosfet (*EW+WW* Dec 93 p990) in this position confers current, voltage and thermal protection to the circuit. Standard devices will also work but will self-destruct under short circuit and other unfavourable conditions.

The normal configuration would use a positive variable regulator of the *317* type together with a pnp bypass. Since the IR fet is an n-channel device, it requires the use of a negative rail regulator producing a negative rail but this shouldn't be problem if the raw DC supply is made floating.

Operation is self explanatory. The cut-in point for the external fet will be determined by the value of the input resistor to the variable regulator. It should be chosen so that the voltage across it fully enhances the channel before the *337* reaches its own 1.9A current limit. The DC supply should have enough overvoltage to allow for this. Although the devices have inherent thermal protection, they require appropriate heatsinking for continuous operation.

Nick Wheeler
Sutton
Surrey



10MHz/1MHz marker generator

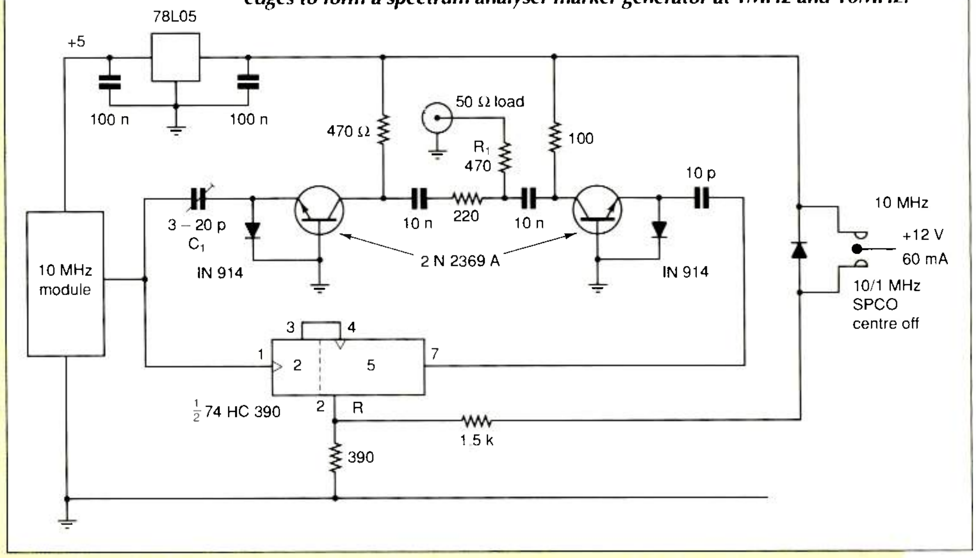
A common-base amplifier with a diode connected inversely across base and emitter and fed with TTL input produces narrow negative-going pulses at the collector. Two such devices fed with 10MHz and 1MHz input generate marker pulses – in my case for a 300MHz spectrum analyser.

Signal from a 10MHz oscillator goes to the 10MHz pulse generator directly and, by way of a decade divider, to the 1MHz generator. Depending on the position of the power switch, the voltage supply goes either to the 10MHz circuit alone or to both circuits to produce a 10MHz/1MHz "comb".

Resistor R_1 isolates the load, causing a 20dB loss. The 10MHz output is –50dBm, that at 1MHz being –70dBm.

D Hutchinson
Bromsgrove
Worcestershire

Common-base amplifiers produce fast, narrow pulse when fed with TTL edges to form a spectrum analyser marker generator at 1MHz and 10MHz.



A-to-D card for PCs

GEC Plessey's *ZN448* has a built-in clock running at up to 1MHz, has its own reference and is an 8-bit device, resolving to 39mV. This card converts a ±5V input to 256 bits, 128 bits corresponding to 0V.

Since the *ZN448* is a single-channel design, a *74HC4351* 8-channel analogue multiplexer precedes it, the output on Z being selected by $Y_{0\text{-}7}$ when LE latches the input-select data on $S_{0\text{-}2}$. The analogue voltage at Z goes to the analogue input of the *ZN448* via the potentiometer chain.

Conversion starts when WR\ goes low, RD\ enabling the data latch, valid data appearing 8-9 clock cycles after the conversion has started.

When the address data on the P inputs of the *74LS688* comparator corresponds to that on the 8-way dip-switch, the P+Q\ output enables the *74LS245* bidirectional bus buffer and gates read and write signals to the *ZN448*. IOR\ also controls the direction of the buffer for read and write.

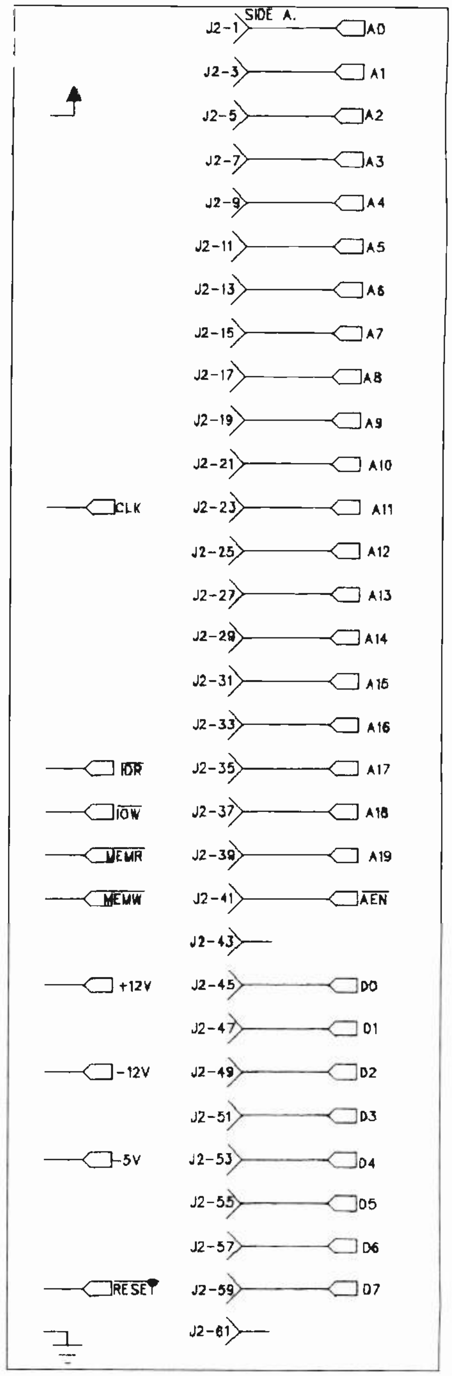
Addresses must be in the 300-31F range. For 300, switch 1 is off ($A_{8\text{-}9}$ high), 2 and 4 on ($A_{5\text{-}7}$ low) and setting the rest on alters the address upwards. Base address plus 1 enables the multiplexer latch.

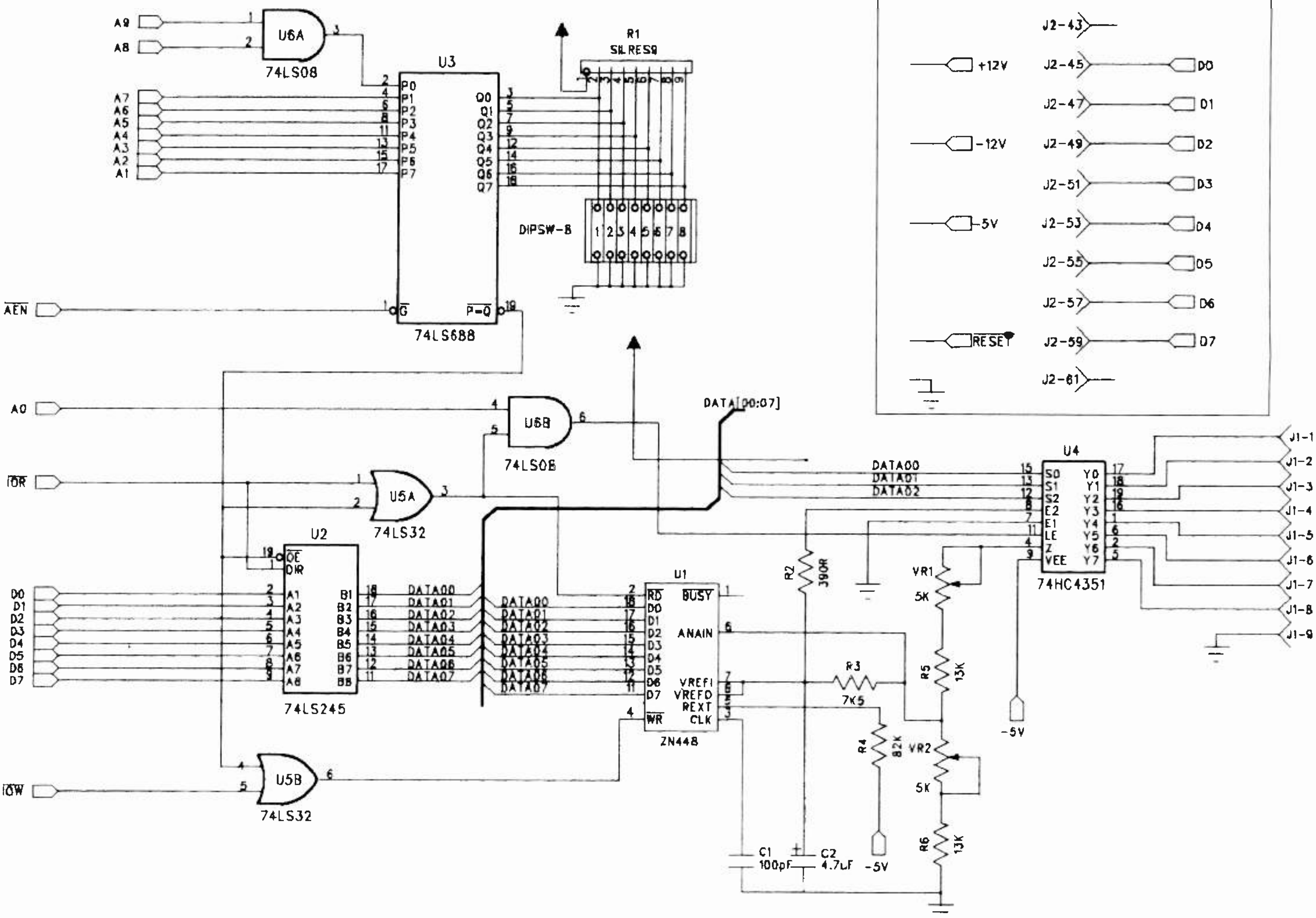
To calibrate the card, apply +5V and adjust VR_2 until D_7 flicks between 0 and 1 to give a reading on the screen of 254-255. Now apply -5V and adjust VR_1 for a flicker between 0 and 1 on D_0, the screen showing 0-1. The adjustments are interactive.

This program continuously reads the *ZN448* and displays the result on screen.

```
10  base=#300:base1=#301 rem;
    for example!
20  out base1,X rem; select any
    input channel
30  out base,Y rem; write to
    ZN448 to start
40  for t=1 to 50:next T rem; if
    necessary
50  A=inp(base) rem; collect
    result
60  print A rem; display result
70  goto 20 rem; start again
80  rem; X=0-7: Y=any value
```

Mel Saunders
Leicester





A-to-D converter on a PC card resolves to 39mV. Configuration shown measures ±5V inputs

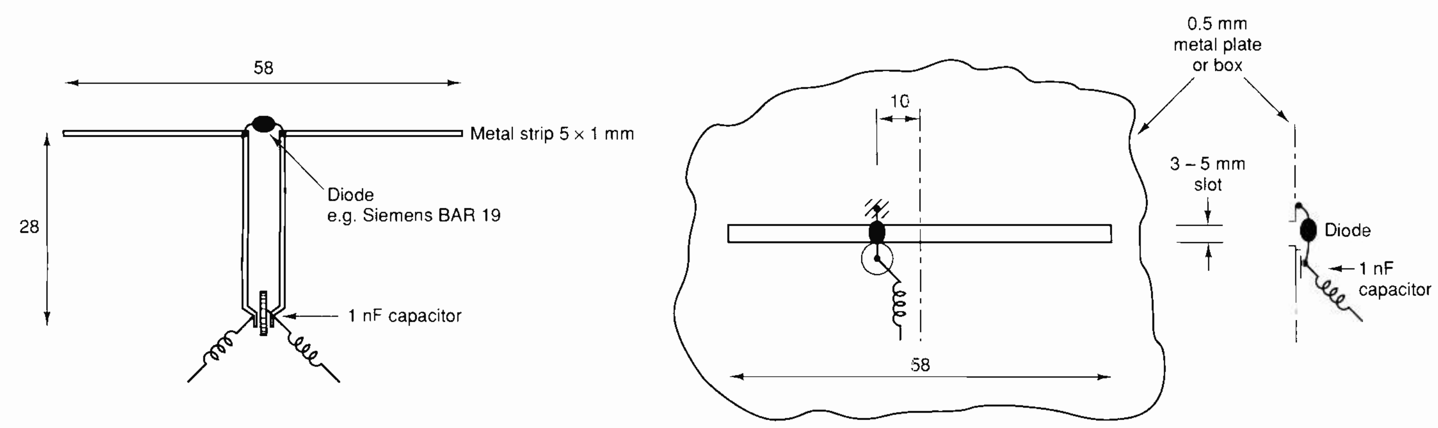


Fig.1. Dipole dimensioned for the microwave heating frequency, 2.45GHz; the diode is a Schottky type. A wire dipole is somewhat subject to static charges, which are relatively unimportant in the slot antenna.

Radiation detector

Microwave heating in domestic ovens and in industrial processes must comply with radiation standards. New equipment does so, at least when properly loaded, but older ovens can leak through deteriorating door and service openings, particularly when improper loading generates harmonics of the magnetron's 2.45GHz. When calibrated, this detector measures 0.01-10mW/cm^2 power density.

Figure 1 shows the simplest type of detector – a simple, half-wave dipole with a Schottky diode in the gap, of correct dimensions for 2.45GHz, although these are not critical. Depending on the diode and meter used, sensitivity is 1 to 10mW/cm^2. The slot antenna is an improvement, being as sensitive to RF power, but much less vulnerable to static charges. Adjust the distance from the oven and the angle so that a maximum can be seen. If the meter shows full-scale at 1m, it is showing up to 1W/cm^2 and that is dangerous.

Adding an amplifier, as in **Fig. 2**, improves sensitivity to around 0.01mW/cm^2. Ceramic "radar" diodes work as well as Schottkys, whereas glass seal diodes have their problems.

A more comprehensive circuit, shown as **Fig. 3**, is provided with two detectors, one horizontal and the other vertical, each with its own amplifier. Comparators, set to produce an output when preset levels are exceeded, activate the audio oscillator alarm and trip the relay supplying power to the magnetron.

Ji í Polívka
Mexican National Autonomous University
Mexico

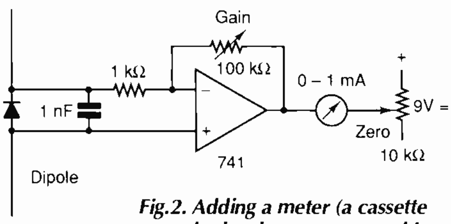


Fig.2. Adding a meter (a cassette recorder level meter was used in the prototype) increases sensitivity to 0.01mW/cm2.

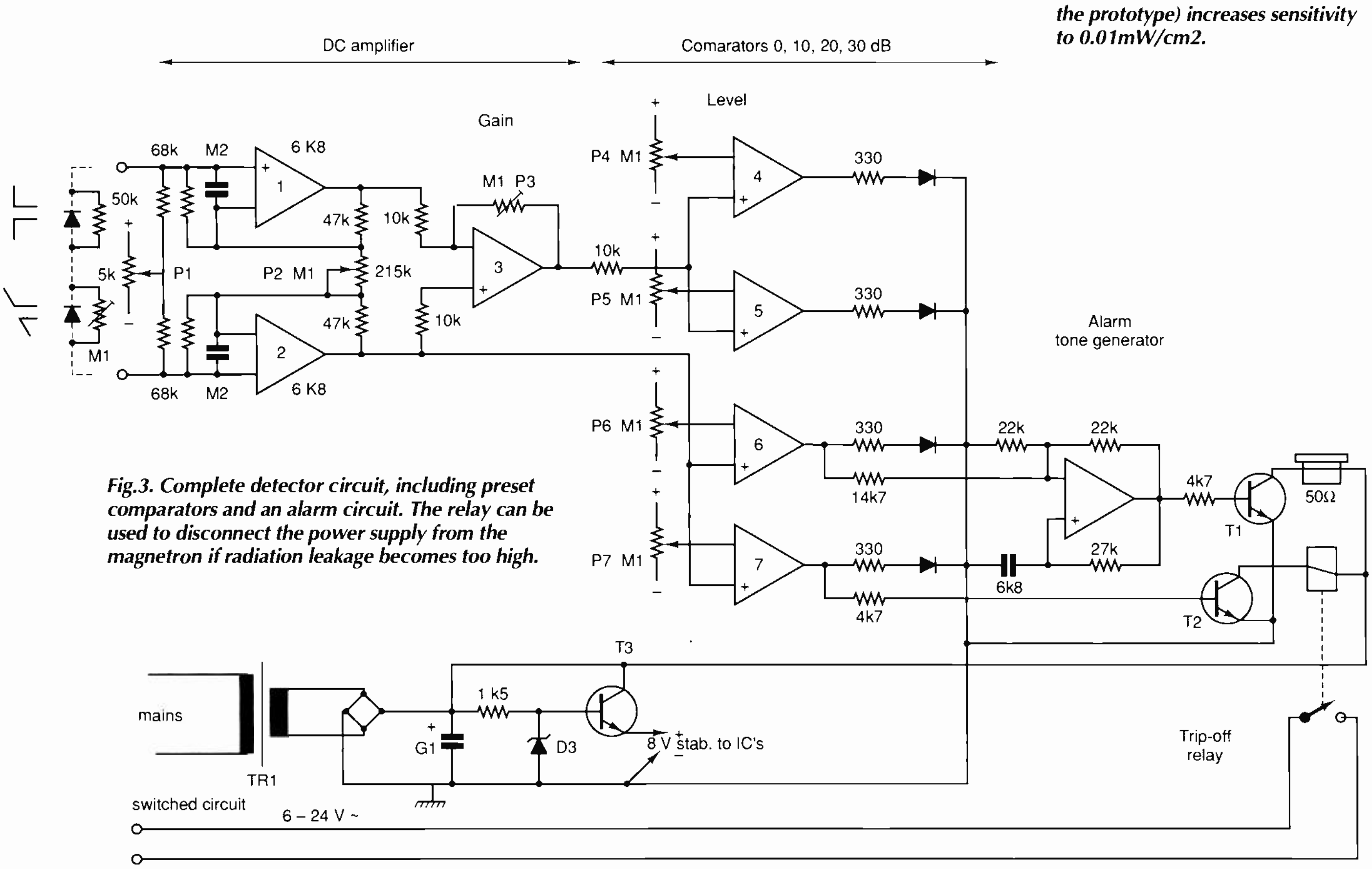


Fig.3. Complete detector circuit, including preset comparators and an alarm circuit. The relay can be used to disconnect the power supply from the magnetron if radiation leakage becomes too high.

Single pot tunes Wien oscillator

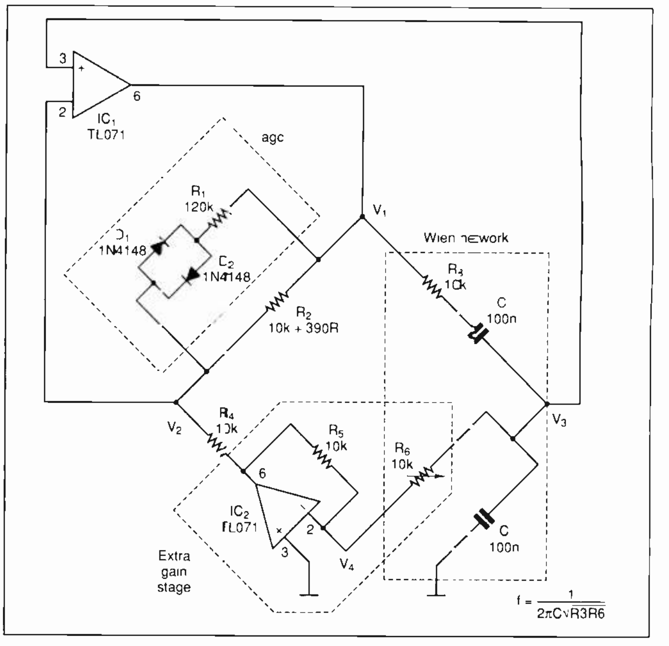
Varying one of the resistors in a Wien bridge alters the frequency and also the attenuation. In the circuit shown here, the tuning resistor also varies the gain of a compensating amplifier to compensate exactly for the varying attenuation.

Since the inverting input of IC_2 is a virtual earth, the attenuation of the bridge is determined by the setting of R_6, which is also the input arm of the compensating amplifier feedback network, the gain of which is now R_5/R_6. Since the frequency varies as $1/(R_3R_6)$, R_6 must have a resistance range of 100:1, increasing the amplifier gain in the same proportion.

Diodes $D_{1,2}$ and R_1 form the AGC circuit. As the amplitude of output increases towards the distortion region, The diodes begin to conduct on peaks, bringing R_1 in to circuit in parallel with R_2 which, with R_4, sets the gain of the maintaining amplifier IC_1. Gain thereby reduces and amplitude stabilises. This function is usually performed by a thermistor or small light bulb with, perhaps, a little less distortion but with a certain amount of "bounce".

An upper limit to the frequency is fixed by the gain of IC_2 falling at higher frequencies – when more gain is needed for compensation.

W A Cambridge
Richmond
Surrey



Wien-bridge oscillator with one variable resistor, which copes with varying bridge attenuation by adjusting compensating amplifier gain.

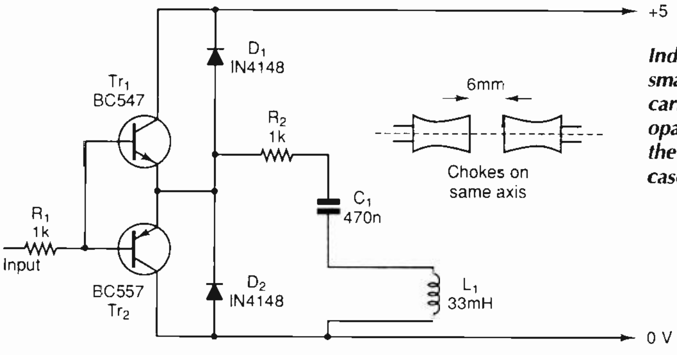
Inductively isolated data link

Inductive coupling between two small chokes up to 6mm apart has the advantage over optical coupling that the link can be made across an opaque barrier, such as through the wall of a sealed plastic case. Inductance values shown here work for 1200baud transmission, but 9600baud should be possible with smaller chokes.

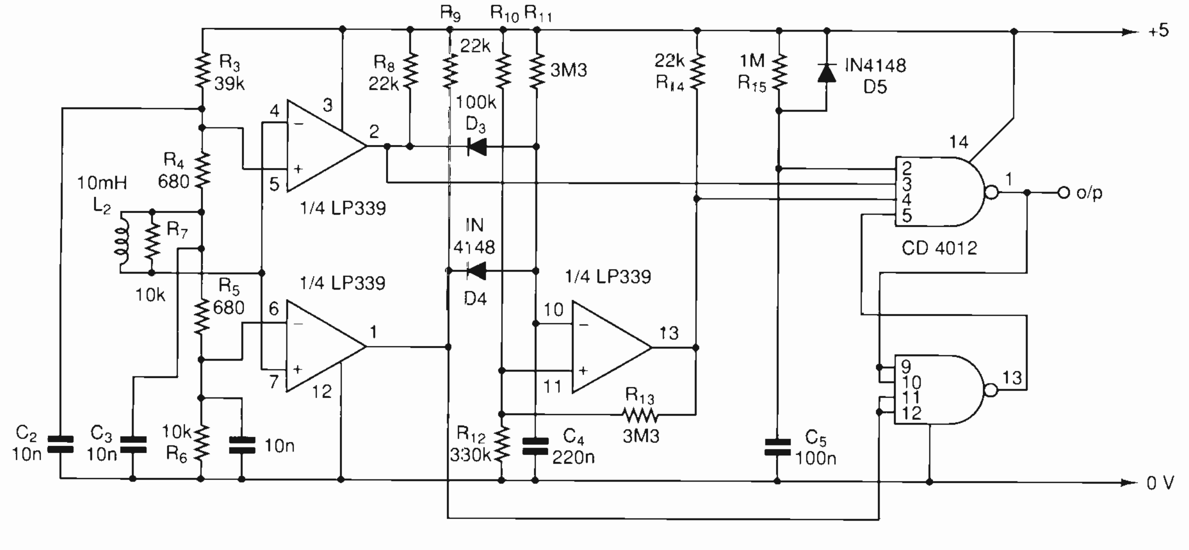
Complementary emitter-followers buffer the input and drive the overdamped *LCR* circuit $R_2C_1L_1$, in which short current pulses flow at transitions without causing any baseline shift in non-return-to-zero data. Inductor L_1, placed in line across the barrier, must be sensed not to invert the data.

Two comparators see the induced voltage across L_2 and produce low pulses for positive and negative data transitions, R_7 eliminating ringing. These pulses trigger and retrigger the flip-flop, made from *4012* gates, to reconstitute the data. During breaks in data, C_4 charges and forces the third comparator's output low, resetting the flip-flop to a known state, which is Mark for RS232. Resistor R_{16} and C_5 do the same at switch-on.

S J Kearley
(Would Mr Kearley please contact our editorial office? – ed)



Inductive link using two small chokes serves to carry data through an opaque barrier, such as the wall of an instrument case.



NEW PRODUCTS CLASSIFIED

ACTIVE

A-to-D & D-to-A converters

Sample-rate converters. SamplePort stereo, asynchronous sample-rate converters from Analog, namely *AD1890/91*, solve sample rate and digital data interconnection problems in computer connections and audio applications. They are used to convert a digital input sample stream at an arbitrarily clocked or changing sample rate to an output sample rate set by the user, input clock frequency being sensed automatically. Incompatible equipment such as MiniDisc, CD and DCC players, HDTV and digital speakers will interface simply using these devices. *AD1890* is the professional device, taking sample widths up to 20 bits, while the *1891* for consumer use takes up to 16 bits at a slightly reduced functionality. Analog Devices Ltd, 0932 253320.

Discrete active devices

Dual zeners. ITT announces a series of two-zener packages, each having a common cathode. They are silicon planar diodes to the E24 standard and come in surface-mounting form. *DZ23* devices dissipate 300mW at 25°C and cover the 2.7V-51V range of voltages, while the *DZ89* series has a range of 3.9V to 200V, dissipates 600mW and takes maximum test currents of 100mA down to 5mA. Packages are SOT-23 and SOT-89A. ITT Semiconductors, 0932 336116.

SOT23 n-p-n. Of surface-mounted n-p-n transistors in Zetex's *FZT* range, the *FZT853* handles a collector/emitter voltage of 100V and collector current of 6A while the *FZT857* copes with 300V/3.5A, with a minimum gain of 100. Saturation voltage of the *853* is 340mV at 5A. All devices in the *FZT* range will dissipate 3W at 25°C and all take a 10A peak current. Zetex plc, 061-627 5105.

Linear integrated circuits

Dual audio op-amp. Exhibiting a voltage noise density of 5.2nV/√Hz and total harmonic distortion of 0.004% at a gain of 1 at 1kHz with 1Vpk-pk output, Analog's *SSM2135* dual op-amp is meant for +5V single-supply operation in audio applications. It will drive 24Ω headphones directly. Its applications include balanced line driving and receiving and sigma-delta A-to-D buffering and it can be used as a low-pass filter and current-to-voltage converter at the output of an 18-bit D-to-A converter, for example. Analog suggests it is most suited for use with stereo codecs in computer audio systems. Analog Devices Ltd, 0932 253320.

Temperature controller. Analog Devices's *TMP01* is a temperature sensor that generates a voltage output proportional to absolute temperature and a control signal from one of two outputs to indicate when the device is above or below a set temperature range. An on-board reference gives a stable 2.5V output and a sensor output with a temperature coefficient of 5mV/K at an accuracy of ±2%. Window comparators provide an open-collector output to signal when high or low thresholds are exceeded, trip points being resistor-programmable. *TMP01* is in an 8-pin plastic mini-dip, an 8-pin SOIC or an 8-lead TO-99 can. Analog Devices Ltd, 0932 253320.

Buffers. A range of wide-band, low-power buffers by Calogic includes the *CLM4122/4222/4322* ultra-low power types which exhibit a 2000V/µs slew rate, 180MHz bandwidth and need only 4mW from the supply, while delivering 60mA peak drive current. These devices are meant to drive coaxial and twisted-pair cables in open-loop application, being specified to drive into 50Ω loads. *CLM4102/4202/4302* devices slew at 2500V/µs at 250MHz and take only 5mA from the ±3V to ±18V supply. The *4102* is said to be an improved version of the Elantec *EL2002* buffer. Calogic Corporation, 0256 51569.

Micropower op-amps. *MAX 417* and *MAX418/9* dual and quad op-amps from Maxim use a novel output stage to enable them to operate at a supply current of 1.2µA maximum per amplifier. With rail-to-rail output swings and single or dual rail working, they are well suited to battery-powered equipment. Each output sources up to 2mA and drives a 1000pF load with no external components; input bias is less than 0.1pA input voltage range extending from the negative rail to within 1V of the positive rail. Unity-gain-stable, quad *MAX418* has an 8kHz GB product and 5V/ms slew rate, while the dual and quad *417* and *418* are stable at gains over 10, have a 150kHz GB product and 80V/ms slewing. Maxim Integrated Products Ltd, 0734 845255.

Voltage dropper. Semtech's electronic zener (EZ) dropper, formerly only in TO-220, is now available in SOT-23 and TO-92 packages. The devices convert 5V to 3.3V or 3V for mixed circuitry, passing currents from 0.1A to 1A. No filtering is needed and, since dissipation is 2W, neither is a heat sink, although if one is used, the device handles more than 8A. Semtech Ltd, 0592 773520.

Bifet op-amps. Up-graded families of bifet op-amps from TI provide 10MHz bandwidth and 40V/µs slew rates. TH distortion is reduced to 0.008% and wide-band noise voltage to 11.6nV/√Hz. Both the *TLE2070* and *TLE2080* are available as single, dual and quad devices. Texas Instruments, 0234 223252.

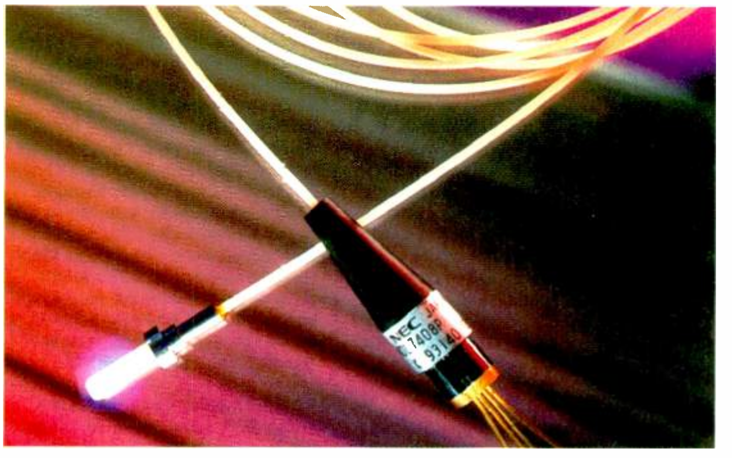
Logic building blocks

Window comparator. Hysteresis programmable from 3mV to 20mV and 150mW drive capability enables Harris's *CA3098* low-cost comparator to function as a programmable Schmitt trigger, as a window comparator in signal processing and in automotive sensing applications, where it will switch semiconductor or inductive devices without interfacing. Hysteresis control is by means of a bias variation, other characteristics such as input and quiescent power also being controllable; the same input is used for strobing or squelching. Power needed is +5V to +16V or ±6V at a maximum of 800µA. Harris Semiconductor (UK), 0276 686886.

Fast SCSI controller. *M53CF94* fast SCSI controller design by *MJE* is compatible with *53CF94* devices and supports both SCSI-1 and SCSI-2 protocols. It reduces demands on the CPU by implementing common SCSI sequences from a single command and is microcode free. The design is supplied on tape to enable it to be incorporated into a asic, being part of the MEJ MacroWare range of designs. MEJ Electronics Ltd, 0483 505895.

Mixed-signal ICs

Modem chipsets. High-speed modem chipsets from AT&T offer



Narrow-spectrum lasers. NEC claims its *NDL7408Px* multiple-quantum well lasers to have the narrowest spectral width of any available, at 1.3nm. In addition, the lasers work at temperatures up to 85°C with no need for semiconductor cooling. Since these devices exhibit low relative noise intensity, low intermodulation distortion and linear transfer characteristics, they are well suited to analogue systems such as cable television. Conforming to CCITT requirements, the lasers are available in 0.2mW and 1mW versions. The absence of cooling reduces current requirement from 1A to around 35mA. NEC Electronics (UK) Ltd, 0908 691133.

Please quote "Electronics World + Wireless World" when seeking further information

cellular, voice and V.32terbo facility and include an integrated microcontroller and datapump with voice compression. The HSM devices cover 9600bit/s and 14400bit/s data rates in form factors for desktop, laptop and PCMCIA use. MNP4/5/10 error-correction is included. *HSM* chipsets now offer three choices: data/fax, data/fax/voice or data/fax/cellular options. AT&T Microelectronics, 0732 742999.

Serial-data interface. Maxim's *MAX562* is a three-driver, five-receiver serial-data interface that copes with data rates up to 250kbit/s, guaranteed slew rate being 4V/µs. It is designed for use in notebook and palmtop computers and meets EIA/TIA-562; it is therefore compatible with RS-232 interfaces. Acceptable power supply is 40mW from 2.7V-5.2V in normal use, all five receivers remaining active in low-power shut-down mode in which only 60µA is needed (10µA in complete shut-down). Packaging is 28-pin SO and SSOP, with driver outputs and receiver inputs on the same side of the device. Maxim Integrated Products Ltd, 0734 845255.

Optical devices

Optical-fibre modules. FORCE model *267* optical-fibre datacomm modules transfer NRZ digital data on single-mode fibre at speeds up to 2.5Gb/s, supporting an optical-loss budget of 10dB. Transmitter uses a 1300nm laser diode and the receiver a pin-diode detector, each unit being in an RFI/EMI-shielded enclosure. Minimum full-specification distance is 10km, but up to 25km is obtainable with care in connections and splicing. Both units are intended for direct PCB soldering, interfacing with PicoLogic, 10k or 100k ECL levels. No setting up is needed. Aerotech World Trade, 0628 34555.

Power semiconductors

Microwave power. A microwave power transistor from Motorola, the *MRF2000-5L* is meant mainly for use in up to 2GHz, in large-signal output and driver linear amplifier stages. It delivers 5W output for 1W input, operating from 20V supply, as a Class-A common-emitter amplifier. Motorola Inc., (USA) 602 994 6561.

HV power mosfets. A number of new power mosfets by Motorola use a patented high-voltage power technique – a multiple-ring layout that provides field shaping towards the edge of the die, giving enhanced voltage blocking to protect against surface charges that commonly occur on the junction termination at high voltages. Motorola's 400-600V set of mosfets have been upgraded to use this technique and new 800V and 1000V devices are now included. Three 800V devices have drain currents of 4A and 7A, depending on the package, two of them having on resistances of 3Ω and the third 1.4Ω. Six 1000V devices have drain currents of 1-10A with on resistances of 1.3Ω-10Ω. Motorola Ltd, 0296 395252.

Transient suppressors. A range of surface-mounted transient-voltage suppressors handling peak powers of 300W, 600W and 1500W from Semtech includes an array, the SMDA in an SO-8 package and two singles, *SMBJ* and *SMCJ*, in DO-214AA and DO-214 AB packages. They are suitable for data, signal and supply bus protection and come in unidirectional and bidirectional forms. Reverse stand-off is 5-24V for the array and 5-170V for the singles; leakage current 1-100µA, 5-800µA or 5-1000µA, depending on version. Semtech Ltd, 0592 773520.

PCMCIA power switch. First in a series of PCMCIA power switching ICs by Siliconix, the *Si9710CY* is a PCMCIA power interface switch offering an on resistance down to 150mΩ. In response to voltages at the cmos-compatible inputs, it switches either 3.3V or 5V to the supply pin in the computer's PCMCIA slot and 3V, 5V or 12V to the flash memory program voltage pin, thereby providing all the voltage switching needed for a single PCMCIA slot. The device is packaged in SO-16 form. Siliconix/TEMIC Marketing, 0344 485757.

PASSIVE

Connectors and cabling

Memory card connectors. Interconnection and packaging for memory cards made by Elco meet PCMCIA and JEIDA standards, being fitted with dual contact beams, first-mate/last-break contacts on the headers and high-temperature insulators to cope with reflow soldering. Elco says it can design and make these connectors to individual specification. Elco Europe Ltd, 0638 664514.

Filters

Ceramic chip resonator. Operating at 2-20MHz and incorporating built-in load capacitors, AVX's *PBRC-B* series of ceramic chip capacitors measures 7.4 by 2.6 by 2mm, ceramic casings allowing reflow soldering and washing. Resonant resistance to 8MHz is 30Ω and 150Ω up to 20MHz; stability with temperature ±0.5% from 2 to 8MHz and ±0.2% from 8 to 20MHz for –40°C to 85°C. A new catalogue of timing devices is available on request. AVX Ltd, 0252 336868.

Ceramic resonators. Tougher and smaller than crystals, Murata's new range of ceramic resonators includes two and three-terminal types, equivalent to series crystals and three-terminal crystal filters, the latter having built-in load capacitors. The filters cover the frequency range 3-33MHz in standard frequencies, with custom designs available. Initial frequency tolerance is ±0.5% with a stability of ±0.3%-±0.4% with temperature. Two-terminal units operate between 1.8MHz and 33.86MHz, again with custom designs offered. Initial centre-frequency tolerance is ±5% with stability of ±0.3%-±0.4% Surtech Interconnection Ltd, 0256 51221.

Hardware

RFI/EMI shielding. ElectroCoat coatings come in chrome, satin chrome, brass, nickel, copper and gold finishes on suitable plastics.

IVC's *ElectroCoat 280* is a multilayer coating of nickel and copper deposited by electroless plating for shielding to over 80dB with a 1.5µm layer of copper and for abrasion-resistance properties. The 380 version has a top layer of chrome to provide corrosion resistance and is meant for use on sub-miniature connectors. Jigs exist for many connectors and customers may not need to invest in tooling. Inco Vacuum Coatings, 021-511 1115.

Literature

Amplicon Liveline. This has been redesigned completely, now containing sections on test and measurement, GPIB and industrial communications, data acquisition and control, power conversion and panel instruments. Newest products described include a low-cost neural networking board for PCs, an opto-isolated digital i/o boar, a 488.2-compatible GPIB controller for PCs, and two software packages: TRACS process monitoring and control and Signal Centre data acquisition and analyse, both these being for Windows. Amplicon Liveline Ltd, (Free)0800 525335.

Farnell catalogue. 2500 new items are contained in the Farnell Electronic Components catalogue, which includes new sections on networking and data communications and safety, security and warning devices, including CCTV systems. ICs from nearly 30 manufacturers are listed and the new Psion series 3a pocket computer is on offer. Editions are published in European languages and currencies and one in Australasian currency. Farnell Electronics plc, 0532 636311.

Resonators. IQD's Crystal Products Data Book for 1984 is increased to 272 pages, now being split into leaded and surface-mounted devices. New products this year include very low profile SM crystals for PCMCIA cards, a new range of plastic-packaged, SM clock oscillators and some ceramic-packaged SM oscillators for high-end and PCMCIA card use, measuring 2.3mm high by 5mm by 7.5mm. Ceramic resonators are available in the frequency range 190kHz-20MHz. IQD Ltd, 0460 77155.

Instrument rental. Livingston has a new mini-catalogue of instrumentation and data recording. Instruments from Racal, Yokogawa, Sony, Fluke, Siemens and Graphtec are described, but additionally the catalogue provides an assessment of the economics of renting or otherwise acquiring data recorders, bearing in mind utilisation, obsolescence and depreciation. Livingston Hire Ltd, 081-943 5151.

Power supplies

DC-DC converters. Three-watt DC-to-DC converters from Gresham accept 2:1 inputs from 4.5V to 72V, which makes them suitable for use from 5V bus supplies, 12V and 24V batteries and in 48V communications systems. They are plastic encapsulated, are the same size as standard 24-pin dil packages, nut need only convection cooling and suffer no derating at temperatures from –25°C to 71°C. There are six single and dual output voltages of 5V, 12V and 15V and four input-voltage ratings. Gresham Power Electronics Ltd, 0722 413060.

3.3V/5V converters. Linear Technology's *LT100* step-up DC-DC converter is programmable for either 3.3V or 5V, provides 250mA at 5V from a 2V supply and takes a quiescent current of 120μA, or 10μA during shut-down. Its power switch exhibits a saturation voltage at 1A of 170mV. Since operating frequency is a minimum of 150kHz, inductors and capacitors around the circuit may be small, surface-mounted types. Efficiency is up to 88% at 1.8V input. The device comes in 8-pin dip or 8-lead SOIC packaging. Micro Call Ltd, 0844 261939.

Radio communications products

8GHz IC attenuator. Contained in an SOIC plastic package, Samsung's *HMP-100008-2* is a voltage-variable attenuator in a GaAs monolithic microwave integrated circuit (MMIC). Using two analogue voltages of -3V to 0V at less than 30μA to control attenuation, the device provides up to 60dB over the 0-8GHz range of frequencies, without external circuitry. Units are cascadable for greater attenuation. Operating temperature is –55°C to 85°C. Anglia Microwaves Ltd, 0277 630000.

RF power amplifier. *RF2013*, a GaAs heterojunction bipolar transistor by RF Micro Devices, provides up to 135mW output power from a 3V supply and up to 800mW from 6.3V; average power output for a two-tone input signal is 400mW from a 6.3V supply. Total gain, depending on the output matching, is 25-30dB, flatness being ±3dB from 800MHz to 1000MHz (±0.75dB from 800MHz to 950MHz). Input voltage standing-wave ratio of less than 2:1 and better than –125dBm/Hz output noise power are combined with an efficiency of 47% from 6.3V and 40% from a 3V supply. The device is intended for use in mobile radio and wide-area networks. Anglia Microwaves Ltd, 0277 630000.

Mini mixer. A 200MHz-3000MHz mixer by Mini-Circuits is packaged in a ceramic surface-mount measuring 0.25in by 0.31in by 0.275in. *RMS-30* has an IF response down to zero frequency and can be used for up and down conversion of RF as well as bi-phase, QPSK and I&Q modulators and phase detectors. The devices conform to MIL-M-28837 standards for resistance to shock and vibration and the solder pads have solder over nickel barrier for leach resistance and solderability. Mini-Circuits Europe, 0252 835094.

Transducers and sensors

Hall-effect switches. Hall-effect ICs capable of operating at temperatures up to 150°C in automotive or industrial environments are introduced by Allegro. Types *3121/2/3* interface with bipolar or cmos logic circuitry and incorporate a voltage regulator to handle supplies of 4.5-24V, a reverse battery protection diode, a quadratic Hall-voltage generator, amplifier, Schmitt trigger and open-collector output sinking up to 25mA. They are unipolar-switching devices for operation with bar magnets and all three are identical except for differing magnetic switch points. Allegro Microsystems, 0932 253355.

COMPUTER

Computer board level products

PC modules. *Apex* embedded PC modules by Blue Chip are PC/AT building blocks which enable a manufacturer quickly to develop a target system while allowing him to concentrate on the application's requirements. Functional support modules may be added to a processor to form a sub-assembly or a complete system. Modules contain a fully featured 80386SX or CX486slc PC/AT system with AMI BIOS, watchdog, DMA and programmable counter/timers. There are also two asynchronous serial ports, a mouse, keyboard and utility connectors. 512K of flash memory is available and VGA 640 by 480 resolution through a support module. Two hard-disk and two floppy drives are provided. Blue Chip Technology, 0244 520222.

Data acquisition board. A 68-pin version of National's *AT-MIO-16X* high-performance, high-resolution, multi-function data-acquisition board is announced. It is an analogue, digital and timing i/o board for PCs and compatible computers, having a 16-bit sampling A-to-D converter with 16 analogue inputs configurable as single-ended, pseudo-differential or fully differential inputs. National's *NI-PGA* instrumentation amplifier is used in the new version to enable the board to settle to high accuracy at 100ksample/s at all gains when scanning multiple channels. Cable assemblies are available in lengths up to 10m to connect the board to an SCSI chassis and to a wide range of

Instrumentation

Programmable PSUs. *PL-P* series power supplies from Thandar are controllable by RS-232 or GPIB, which also provide readback to the serial port of a PC without an interface card. The series includes single, dual and triple models, all being made in bench-top or rack-mounted form. Main output is variable from 0 to 32V, with a variable current limit to 3.1A. A 4V-6V logic output is included. Providing power levels to 360W, the other new series, the *TSX-P*, is available in 35V/10A and 18V/20A versions. The hardware and software of the RS-232 interface also supports an extended multi-instrument mode ARC, in which up to 32 instruments may be linked and computer-controlled. Thandar Instruments, 0480 412451.

Standard signal generator. Programmable AM/FM standard signal generator *SG-5260* by Trio Kenwood uses PLL synthesis to produce a resolution of 100Hz over the 10kHz-260MHz frequency range, its output being from –20dBμ to 132dBμ in 0.1dB steps. A digital display provides a 7-digit frequency readout, a 4-digit output EMF figure and amplitude or frequency modulation setting. Modulation is internal or external and a non-volatile memory contains up to 100 conditions to be quickly set up. Trio Kenwood UK Ltd, 0923 816444.

Digital potentiometer. The *Digipot* series of digital potentiometers by Control Transducers now includes model *MD*, a miniature unit that detects direction, position and speed. Output is a square wave at a resolution of between 100 and 1024 pulses per revolution, dual channel, with an optional marker/index pulse. Shaft sizes from 3mm to 8mm are acceptable and there are many mounting choices. The unit handles shaft play of ±0.01in and has a screwed housing. Power requirement is +5V at 40mA. Control Transducers, 0234 217704.

Please quote "Electronics World + Wireless World" when seeking further information

50-pin signal-conditioning and termination accessories. A full range of software is offered. National Instruments UK, 0635 523545.

Development and evaluation

80386EX simulator. A range of 32-bit development tools by Systems and Software now includes the *386EX/SIM*, a software simulation of the complete chip. It allows engineers to develop software in assembler, C or PL/M, including routines that use the on-chip device with source-level debug facilities while hardware is still being developed, before chips and boards become available. Code-debugging can go ahead, in many cases showing faults not detected by ICE. The simulator replicates all devices, including master and slave 8529 interrupt controllers, 8524 interval timers, 8250 uarts, synchronous serial units, etc, and up to 4Gbyte of memory. Computer Solutions Ltd, 0932 829460.

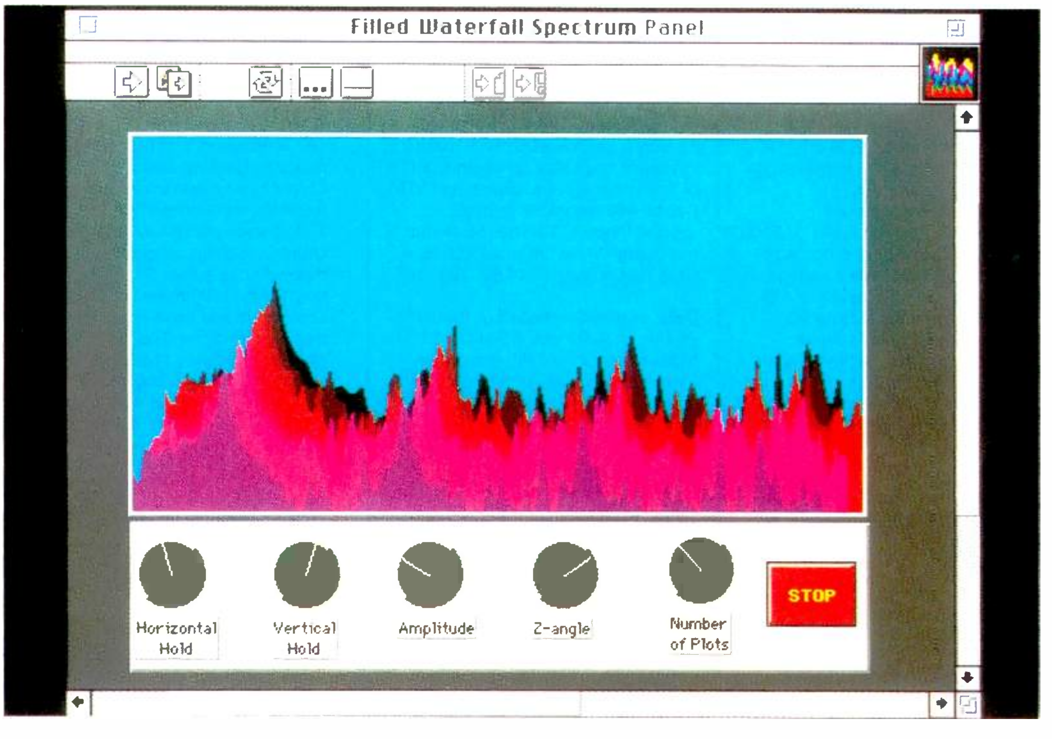
87C750 development. Micro Computer Control offers the *87C750-SDK*, which is a software development kit for Philips's 87C750 microprocessor, containing all tools needed to create and execute programs. There is a multi-window text editor and an assembler that finds syntax errors, converts source code to machine code and generates program listings. The kit also provides a software simulator and debugger to run programs on a PC without extra hardware. An optional Micro-C compiler is available. Logicom Communications Ltd, 081 756 1284.

8051 emulator. For those working with the 8031/51 microcontroller, the Micro AMPS in-circuit emulator operates up to 16MHz, has 64Mbyte of battery-backed memory partitionable in 4K blocks between program and on-board or off-board data, 64K hardware breakpoints in program and extended data memory, single-step and software trace. Micro AMPS Ltd, 0483 268999.

In-circuit emulator. *Biceps51* is an in-circuit emulator that supports almost all 8051 derivatives, combining the features of an eprom emulator with those of a full-function ICE. It replaces the eprom in a test circuit with 64Kbyte of emulation ram that may be partitioned in 4K blocks. The package includes high-level debugging, hardware breakpoint capability, cross assembler and real-time trace buffer that can be interrogated on the fly. A single adaptor allows the unit to emulate almost all 8051 derivatives. Micro AMPS Ltd, 0483 268999.

Eprom emulator. *MicroRom* from Squarewave is a conventional eprom emulator, but is contained in an 11mm high module that plugs into a PC's eprom socket, eliminating the usual ribbon cable and often obviating the need to remove the target board. After programming, the device may remain in circuit, being non-volatile. It is usable with any computer fitted with a Centronics printer port and down-loads a 512Kb file in two seconds. Squarewave suggests it be regarded as an eprom with a built-in programmer and an unlimited number of write cycles. Squarewave Electronics Ltd, 081-880 9889.

LabVIEW graphics. Using National's new *LabVIEW Picture Control Toolkit*, users are able to generate their own front-panel displays for the *LabVIEW* computer-instrumentation software. Possible displays include bar graphs, polar plots and Smith charts, and objects such as robot arms can be animated. Images are described by graphics instructions stored as a series of drawing commands. There is also a library that implements a set of functions that take an existing picture and add new instructions to create a more elaborate picture. National Instruments UK, 0635 523545.



Software

Windows data acquisition. Visual Designer is a software package for data acquisition designers that allows users to develop a PC-based system by generating custom applications without any programming. It runs under Windows 3.1, icons representing function blocks such as i/o, graphic displays, waveform generators, measurement and comparison functions, maths and logic operations. The package is termed an application generator, holding the middle ground between fixed-function application software and complex programming languages. Custom software can be developed simply by drawing a block diagram on the screen and specifying parameters; each block then represents perhaps hundreds of lines of C code. Intelligent Instrumentation, 0923 896989.

Spice for RF. A new model library for RF devices that is three times the size of the earlier version is announced by Intusoft of California. Device models provided include pin diodes, monolithic microwave ICs, H-P GaAs mesfets, Philips's BJTs and an ideal coupler. These models are usable with any Berkeley Spice-compatible simulator on any computer. This approach to modelling RF devices is claimed to be superior to that usually adopted, in that no unrealistic parameters are needed to "force-fit" the device behaviour; custom sub-circuits account for all parasitics and match published s-parameters in magnitude and phase. Any Spice program is thereby enabled to simulate linear and non-linear RF circuits in frequency, time and DC domains. Intusoft, 0101 (310)833-9658.

Loudspeaker design. Audiosoft of Melbourne, Australia, has the *Calsod 3.00*, the latest version of a software package for loudspeaker design and system optimisation. It simulates sound pressure and impedance response of individual drivers, multiple systems and crossovers, effects of geometric layout of various types of enclosure being included in the analysis. Models of the systems include loss parameters for leakage, absorption and port or passive radiator losses. Multiple off-axis observation points for response calculation can be specified, overlaid colour screen plots being produced for each location. A circuit optimiser is included, both active and passive crossovers being optimised, and Thiele-Small parameters of drivers are determined from two impedance measurements under mass or compliance perturbation conditions. *Calsod* imports data files and supports SYSid, System One, MLSSA, IMP and LMS systems. Munro Associates, 071-379 7600.

Autorouter. *ULTIroute GXR* is a ripup-and-retry autorouter by ULTImate Technology which runs under Windows and allows the user to influence the way the router performs, so customising it to his own type of design. "Keep-out" areas can be specified, both for vias and the general trace. New versions of *ULTIboard*, a PC board designer with real-time design-rule checks, and *ULTIcap* are 32-bit packages using a Windows-compatible dos extender, so that they will run in Windows or dos with no performance loss. ULTimate Technology Ltd, 0734 812030. ■

Global positioning core in one IC

All the active circuitry needed to convert global positioning information in rf spread-spectrum form to 4.309MHz final IF is contained in a single IC called the *GP1010*. Its manufacturer, GEC Plessey, has published application note *AN139* containing details on how to simplify the device's evaluation.

Input to the evaluation circuit described is C/A code signals on the GPS L1 carrier at 1575.42MHz. Peripheral functions needed are a low-noise preamplifier, a 10MHz reference and a sample clock for the output digitiser.

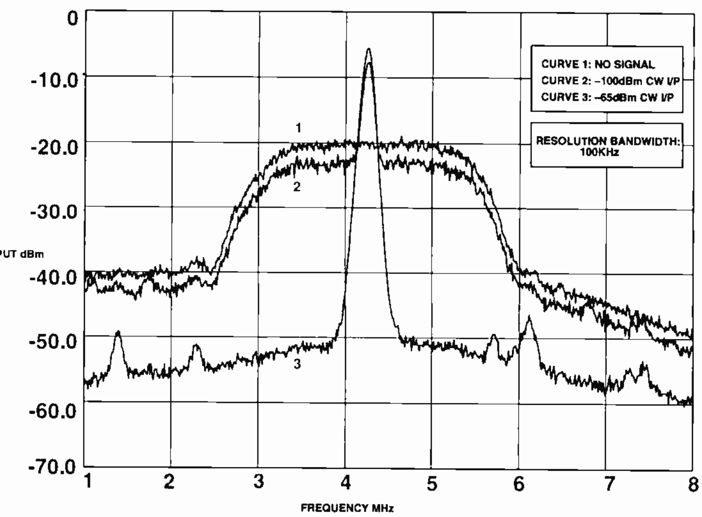
Local oscillators of 1.4GHz, 140MHz and 31.1MHz are provided by the *GP1010* on-chip synthesizer while clocking at 40MHz is available for an external processor. There are rf and IF amplifiers with external first and second IF filters. The third IF stage needs no external filtering.

Output magnitude data controls the on-chip AGC loop whose time constant is set by an external capacitor. For testing, the AGC can be forced to maximum gain.

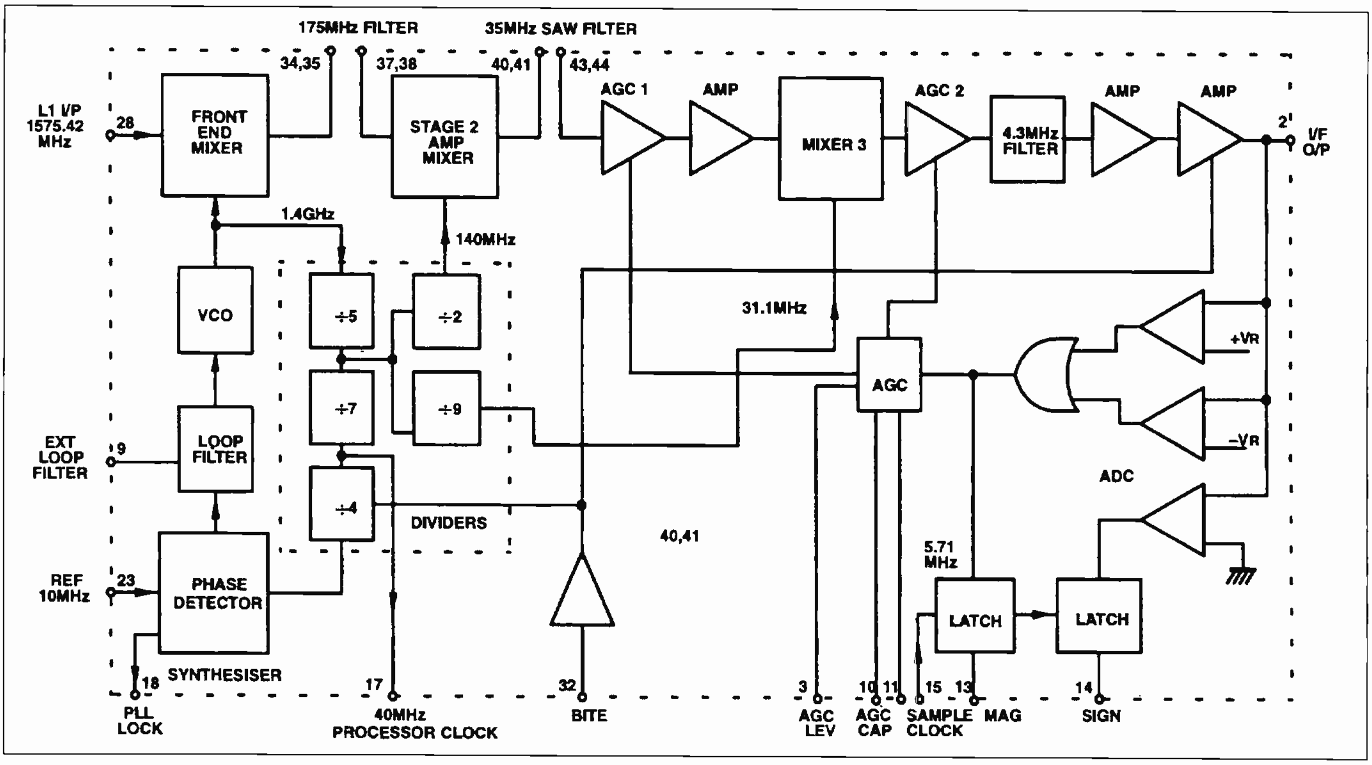
In commercial GPS applications, the first IF filter, at 175MHz, can be implemented using *LC* networking. This provides a 3dB bandwidth of approximately 20MHz. If severe out-of-band interference is expected, a SAW filter may be substituted for the *LC* version. The device is designed to cope with large interfering signals without compression. This may be important where the GPS receiver is co-located with interfering signal sources such as Inmarsat transmitters.

Second 35.42MHz IF filtering is via a custom SAW device that provides a 1.8MHz

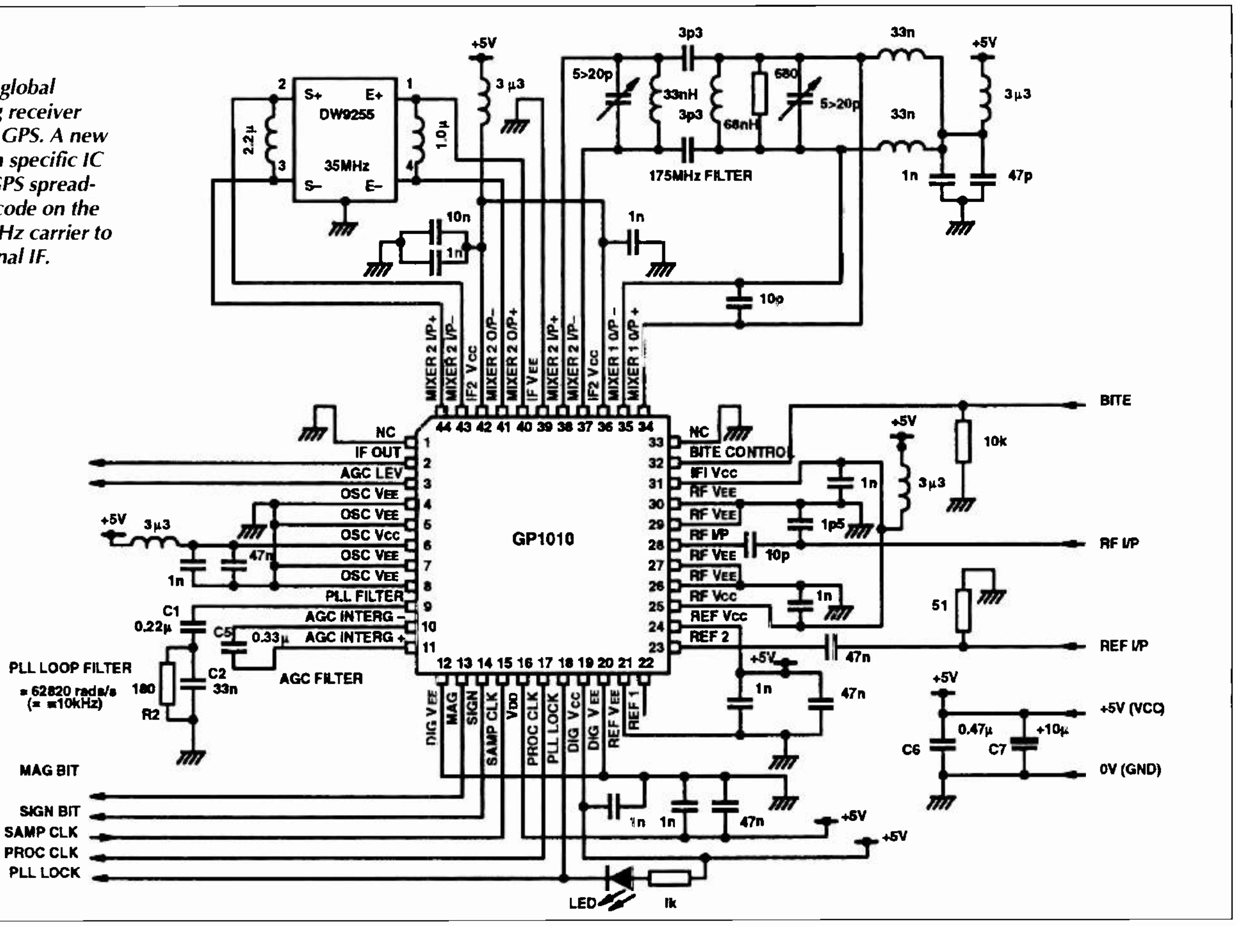
Intermediate frequency output spectra for the GPS receiver. Digital data is output in the final IF of 4.309MHz.



GPS core on a chip. This device provides rf and IF amplifiers as well as the three local oscillators needed to convert convert the carrier down to 4.309MHz final IF.



Heart of a global positioning receiver system for GPS. A new application specific IC converts GPS spread-spectrum code on the 1575.42MHz carrier to digitised final IF.



bandwidth with 1dB ripple and 40dB out-of-band rejection. Tuning indicators are needed at the SAW filter output *and* input.

A reference source for the oscillator for the local oscillator synthesizer can be configured on-chip by simply adding an external crystal. However the frequency accuracy and stability of global positioning system receivers demands better characteristics than can be provided by such a simple source. For this reason the evaluation board is designed to accept an external source. This source can be derived from a high-performance signal generator or fixed frequency temperature controlled crystal oscillator.

To minimise signal breakthrough the 5V supply is separately decoupled to the rf, IF and VCO stages via 3.3μH inductors.

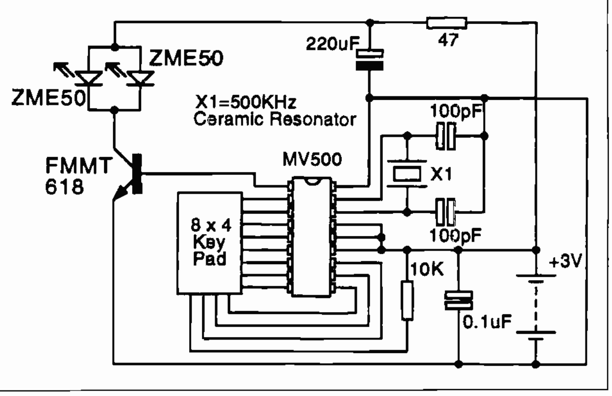
GEC Plessey Semiconductors, *Cheney Manor, Swindon, Wiltshire SN2 2QW. Tel. 0793 518000, fax 0793 518411.*

Uses for a 2.5A transistor in SOT23

By reducing saturation voltage and redesigning the lead frame, Zetex has produced a pair of SOT23 transistors that outperform much larger SOT223 devices. One is a 20V device with a 2.5A continuous current rating, the other 50V at 2A. Gain of the former is at least 200 at 2A while that of the latter is 200 at 1A. Both have a power rating of 625mW which is over 200mW higher than the industry norm.

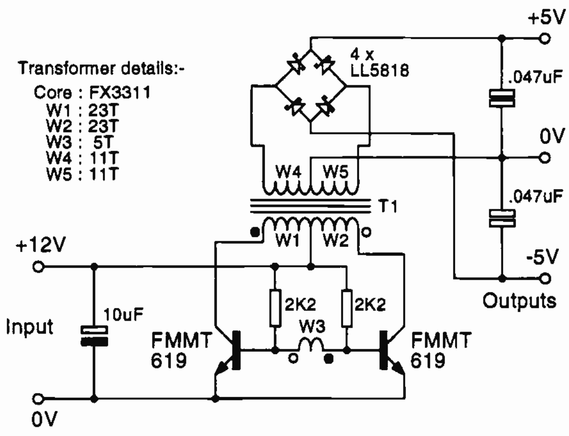
At 2.5A, the *FMMT618* has a saturation voltage of typically 130mV so it is a good choice for switching in battery applications since losses will be minimal. These circuits are from a document entitled *Features and applications of the FMMT618 and FMMT619.*

To produce useful output over extended distances, infra-red leds in remote control units need to be pulsed at up to 2A each. High gain of the FMMT618 means that no additional interfacing is needed and low saturation minimises losses.

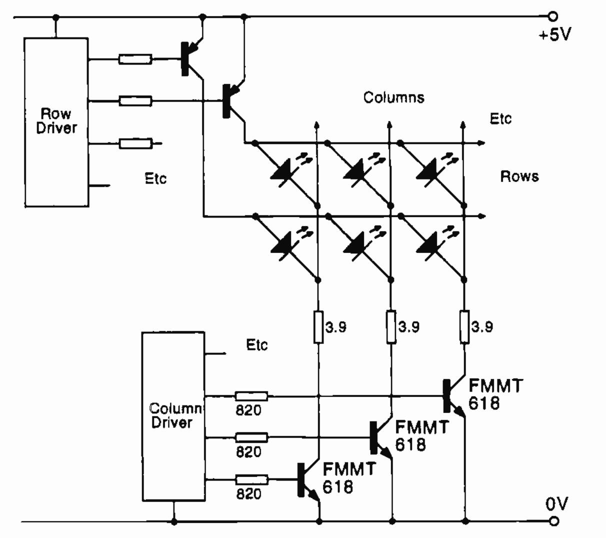


The first circuit is a remote-control transmitter. To maximise the range of the transmitter, each photodiode needs to be pulsed at between 1 and 2A so the combined load is 2 to 4A. Transmitter pulses are 16ms wide. Although 4A exceeds the *FMMT618*'s continuous current rating, it is within its pulsed limit of 6A.

Low saturation is also a useful parameter in power converter applications such as the one in the second diagram. This is an unregulated isolating power supply switching at about 90kHz. Although tiny, the circuit has a theoretical output of 10W



DC-to-DC converter with isolation. Given a smaller transformer core, this circuit could be built into a 0.3in wide DIL package. Note that even though the diodes are Schottky types, they are the main contributors to losses.



Part of a 32 by 32 matrix display. Column drive current is up to half an amp so transistors with good gain at high current are needed to minimise component count.

but this is limited by rectifier losses and power dissipation capability of the substrate. Efficiency with values shown will be around 85%, with most of the loss due to the rectifier diodes.

In moving message displays such as the one in the third diagram, interface circuitry needs to be as small and simple as possible due to the large number of row and column drivers. The circuit shown is a small portion of a 32 by 32 matrix. Each LED needs about 15mA so total column current is just under half an amp. Because of its high gain at high current, the *FMMT618* needs only 5mA of base current to saturate to below 50mV. As a result, it can be used for direct interfacing to low power logic registers.

Other applications outlined are printer stepper motor driving and 1.5 to 5V power conversion. It is also suggested that the devices would be useful as pin drivers and FPLA programmers due to their low saturation voltage.

__Zetex__, Fields New Road, Chadderton, Oldham, Lancashire OL9 8NP. Tel. 061 627 5105, fax 061 627 5467.

Inductorless voltage booster provides intermediate voltages

Charge-pump circuits are convenient for doubling or inverting a voltage. They are also cheap and easy to design since they need no inductors. On the other hand, they do not regulate or make it easy to provide intermediate voltages.

Adding a comparator and reference can provide a degree of regulation and offer the ability to produce intermediate voltages without significantly increasing complexity. As this configuration from the latest Maxim *Engineering Journal* shows, it is possible to produce a 3V to 5V converter that varies by only 0.1V for loads down to 50Ω.

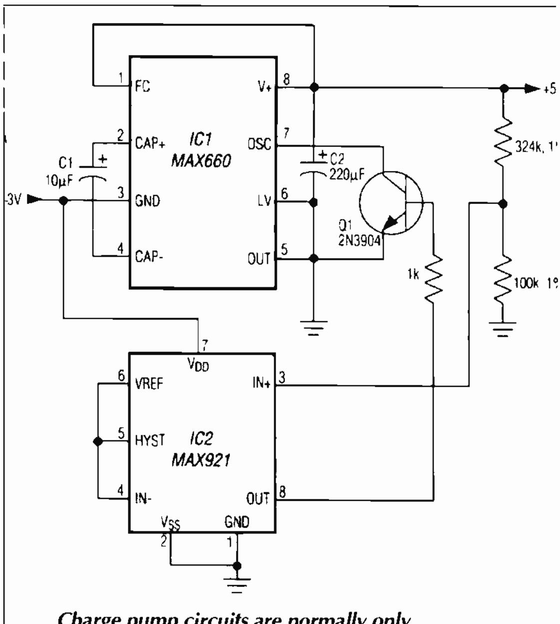
Both reference and comparator functions are provided within IC_2. Charge pump IC_1 has an internal oscillator whose 45kHz switching transfers charge from C_1 to C_2, causing the regulated output to rise. When feedback voltage at pin 3 of IC_2 exceeds 1.18V, the comparator turns off the oscillator via the transistor.

Because the control loop needs no hysteresis, it is set to zero but hysteresis can easily be added via IC_2. At turn on, the oscillator generates two cycles, which is always enough to drive output voltage slightly above the desired level before feed back turns the oscillator off again. Resulting output ripple depends mainly on input voltage and output current.

Output ripple can be reduced at the expense of efficiency by adding a small resistor of about 1Ω in series with C_1. Ripple also depends on the value and ESR of C_1. Smaller values of C_1 transfer less charge to C_2, producing smaller jumps in output voltage.

Output parameters versus load for the regulating, variable voltage charge-pump circuit. These figures assume a 3V supply.

Load (Ω)	Output (V)	Ripple (mV p-p)
∞	5.00	30
10k	5.00	35
1k	5.00	100
100	4.96	100
50	4.59	150



Charge pump circuits are normally only good for doubling or inverting a supply voltage. Adding a comparator and reference provides both regulation and intermediate output voltages.

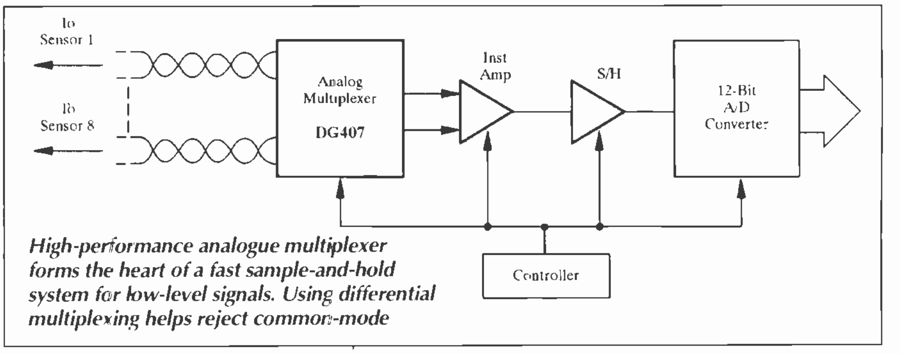
__Maxim__, 21C Horseshoe Park, Pangbourne, Reading RG8 7JW, Tel 0734 845255, fax 0734 .

Fast, high-performance sampling

In a sample-and-hold system, sampling time is limited by two consecutive events - the transition time of the multiplexer and the settling time of the sampled signal at the output.

Application hints in the *DG406/7* data sheet from Siliconix describe how to increase the accuracy of low-level signal measurements by using differential multiplexing.

The *DG406* is a 16-channel high-performance analogue multiplexer while the *407* is identical except for being configured as a dual eight-channel device.



High-performance analogue multiplexer forms the heart of a fast sample-and-hold system for low-level signals. Using differential multiplexing helps reject common-mode

In a sample and hold system such as the one shown, transition time is that of the multiplexer, in this case 300ns maximum. Settling time at the load depends on several parameters, including $r_{DS(on)}$, of the multiplexer, source impedance and multiplexer and load capacitances. Charge injection of the multiplexer and required accuracy also play a role.

Settling time for the multiplexer alone can be derived from the model shown. Assuming a low impedance source, such as presented by an operational or buffer amplifier, settling time of the *RC* network for a given accuracy is $n\tau$.

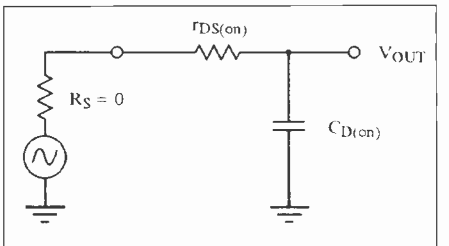
% accuracy	# of bits	n
0.25	8	6
0.012	12	9
0.0017	15	11

Maximum sampling frequency of the multiplexer is,

$$f_s = \frac{1}{x(t_{Settling} + t_{Trans})}$$

where x is the number of channels to scan and,

$$t_{Settling} = n\tau = n \times r_{DS(on)} \times C_{D(on)}$$



Simplified model of a single-channel multiplexer. With this model, settling time for the input multiplexer of the sample-and-hold circuit can be derived.

Given a DG406 at room temperature, for 12 bit accuracy using maximum limits,

$$f_s = \frac{1}{16(9 \times 100\Omega \times 100 \times 10^{-12} F) + 300 \times 10^{-12} s}$$

or f_s=694kHz.

From the sampling theorem, to properly recover the original signal, the sampling frequency should be more than twice the maximum frequency of the original signal. This assumes perfect band limiting. In a real application, sampling at three or four times the frequency of the filter cut-off frequency is advisable so,

$$f_c = \frac{1}{4} \times f_s = 173kHz$$

This shows that the *DG406* can sample sixteen different signals with a maximum component to 173kHz. Using two channels to double sample the incoming signal also doubles cut-off frequency.

In the block diagram is a typical data-acquisition front end suitable for low-level analogue signals. Differential multiplexing of small signals is preferred since it helps to reject common-mode noise. This is especially important when sensors are located at a distance.

A low $r_{DS(on)}$, low leakage multiplexer helps to reduce measurement errors. The *DG407* has lower than usual power dissipation so on-chip thermal gradients are reduced. These gradients can cause errors due to temperature mismatch along parasitic thermocouple paths.

***Siliconix**, East Hampstead Road, Bracknell, Berkshire RG12 1LX. Tel. 0344 485757, fax 0344 427371.*

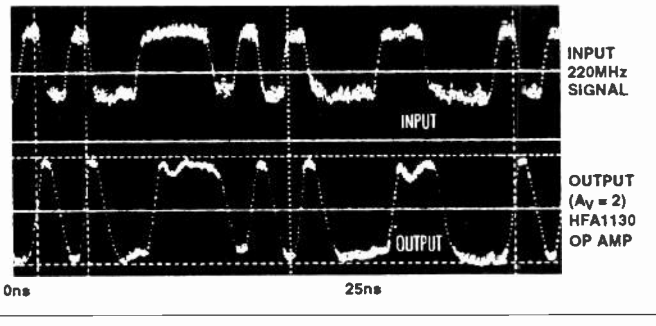
Ultra-fast op-amp with clamp

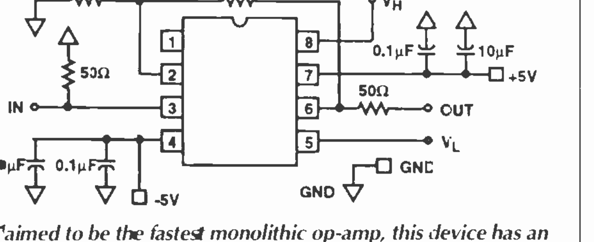
Claimed to be the fastest monolithic amplifier available, the *HFA1130* from Harris has a –3dB bandwidth of 850MHz and a slew rate of 2300V/µs. Distortion is low, at –56dBc, and the device has user programmable output clamping to protect later stages from damage or input saturation.

According to the device data sheet, there is an evaluation board for the device following this circuit. Discussed within the sheet are optium feedback resistance, clamp operation, PCB design and recovery from overdrive. Comprehensive performance figures are also presented.

Overdrive recovery of the device is typically less than a nanosecond while settling time is 11ns. Gain flatness at 100MHz is 0.14dB.

***Harris Semiconductor,** Riverside Way, Camberley, Surrey GU15 3YQ, Tel. 0276 686886.*





Claimed to be the fastest monolithic op-amp, this device has an 850MHz bandwidth and 2300V/µs slew rate.

DAB – delivery, delay or debacle?

Digital Audio Broadcasting is seen by many as the logical progression in sound transmission for the next century. It undoubtedly represents magnificent technical progress. However, the packing of channels which is part of the system reduces the independence of individual station operators. It delivers the quality but will it deliver the audience? Norman McLeod reports.

The politics of DAB

I overheard a delegate at the Radio Academy's technical conference last November mutter "Will DAB be another RDS...?" RDS – the Radio Data System – was launched in a blaze of glory six years ago, as nothing less than "the greatest improvement in radio sets since the invention of the transistor" and "the key to the future of radio listening[1]". Few would make such hyperbolic claims for this decidedly minor miracle today.

DAB – Digital Audio Broadcasting – first came to public attention amid similarly glowing praise for its properties of high sound quality and immunity to interference. It has been popularly reported as a spectacular technology under test and demonstration conditions, but without much anticipation of its wider implications and real-life practicality as a consumer product.

Eureka DAB – the European version of digital audio broadcasting – is an impressive attempt at tackling the technical shortcomings of present-day FM radio broadcasts.

The fall of FM

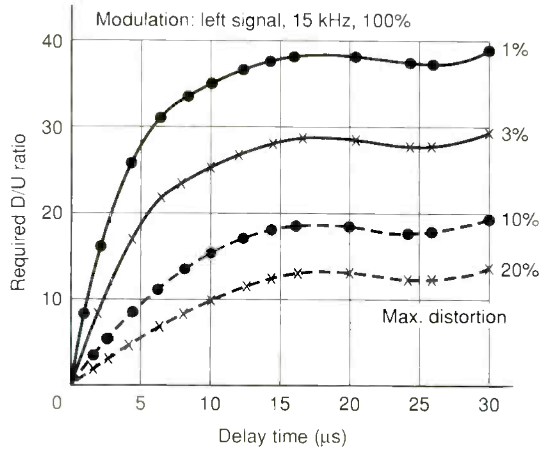
FM has not lived up to the ambitions expressed by early pioneers for an interference-free high quality service intended eventually to replace AM broadcasting on long and medium wave. In the fifties, when VHF broadcasting first started, excellent mono results could be obtained from fixed VHF/FM receivers connected to an outdoor rooftop aerial, as was the accepted norm for television reception.

What was not foreseen at the time was the advent of portable and mobile VHF receivers with modest, street-level aerials. And stereo was in its infancy – the Zenith-GE multiplex system unheard of. Both these factors heavily compromise FM reception quality today.

Any prospective broadcasting service must work well in portable and car radios. The signals picked up by these receivers, already much lower than the rooftop signals for which VHF broadcasting was originally planned, fluctuate randomly by up to 20dB as the receiver is moved. This causes swishes and pops of noise whenever the signal falls too low for reasonable quieting.

FM broadcasts are also affected by multipath distortion caused by time-delayed components, due to signal reflections from buildings and other obstacles, which arrive at the antenna varying in phase and amplitude with respect to the direct signal[2]. Multipath distortion of the FM signal at the antenna translates

Multipath consequences for FM: This graph shows how susceptible FM stereo reception is to delayed reflections. The required direct to unwanted (D/U) signal ratio is shown for various time delays and resultant distortion. Serious problems start with delays of less than 5μs (1.5km extra path length). Eureka DAB, on the other hand, is completely resistant to multipath content – indeed benefits from it – until the signal delay reaches 300μs. Source: "Analysis of Multipath Distortion in FM Sound Broadcasting", NHK Technical Research Laboratories, June 1979, quoted by Hawker[2].



into unpleasant nonlinear distortion of the received audio. When the Zenith-GE pilot tone stereo system is in operation, matters get considerably worse:

Stereo operation worsens the signal-to-noise ratio of the transmission by 20dB;

Multiplexing makes co- and adjacent channel interference more troublesome – some 15 to 20 dB more protection is required when the stereo decoder is operational;

Separation between the channels is incomplete and highly vulnerable to small errors on the multiplex anywhere in the transmission chain from coder to decoder.

Multipath reflections can distort the supersonic 'S' signal to the extent that reception which is tolerable in mono can be unlistenable in stereo, even when the signal level is such that noise is not a problem.

The Eureka DAB alternative positively thrives on multipath effects, has no penalty for stereo working, and offers the flexibility to trade off quantity against quality when it comes to allocating bits to services. It performs magnificently.

But it is being called into question, not by engineers, but by the people who will have to work with it and foot the bill for the new technology.

Standard struggle

While DAB shines technically, it is clear that there is a political imperative to beat US technical proposals: IBOC (in-band on-channel) is currently being researched by USA Digital, AT&T Bell Communications and Amati Communications Corp. IBOC systems piggyback a digital carrier on top of existing analogue FM (or even AM) transmissions. They do not require a new frequency band, and place firm control of the new medium in the hands of existing operators with their own transmitters, sites and frequencies.

Although US work is at a much earlier stage of development than Eureka, IBOC has such strong attractions for established broadcasters that if they do prove viable, Eureka will almost certainly be eclipsed. So there is a rush bordering on desperation to have Eureka accepted and up and running before IBOC has the chance to prove itself one way or the other. Adoption of Eureka inevitably means centralised control of a broadcasting channel for its interleaved services. The technical aspects of this system make it mandatory to supply a multiplex of services from centrally-assembled data streams. The 'frequency diversity' properties which allow the multiplex to resist drop-outs in the incoming signal, which are typically between 30 and 300kHz wide, make it necessary to occupy a wide bandwidth with an interleaved signal.

Frequencies must be planned on the basis of 1.75MHz wide chunks, as opposed to segments an eighth of that size on FM. This makes Eureka planning a much less flexible

COFDM: transmitting the digits

COFDM (coded orthogonal frequency division multiplex) describes the process used by Eureka DAB to send data over the radio transmission path[14]. The essential feature of this system is that a large number of closely-spaced carriers are used, each of which carries a relatively slow data rate. In Mode 1 DAB, as demonstrated by the BBC on 226MHz, a total of 1536 carriers are used 1kHz apart .

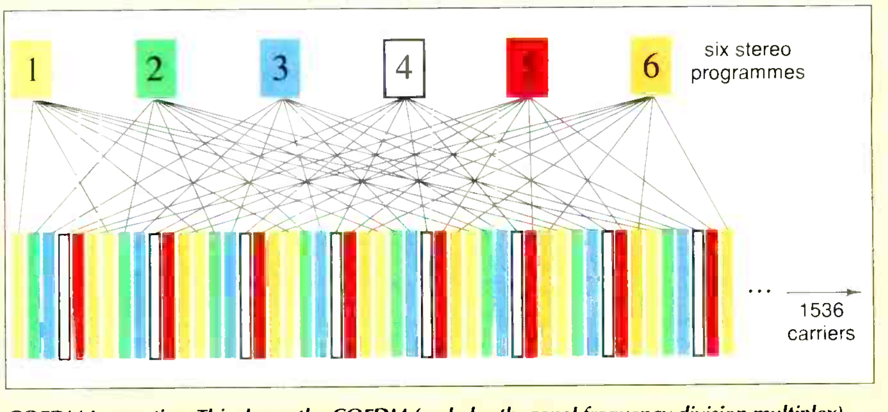
The total RF bandwidth occupied is therefore 1.537 MHz, but a further 250 kHz guard band is needed between adjacent blocks, as the DAB signal does not cut-off very cleanly at the edges of the spectrum, and is prone to spread due to intermodulation effects. It has been noted that TV transmitters carrying DAB may be backed off some 9dB on pep output for good IM performance.

Two bit QPSK symbols are transmitted on each carrier at intervals of 1.246ms, made up of 1ms active symbol duration, and a 246μs guard interval period. The bit rate per carrier is therefore just over 1600 bits per second, making the total available data rate some 2.46Mbits/s. Of this, 1.09Mbits/s was used in the BBC demonstration to transmit audio from the five national radio networks at three different quality grades – the rest is accounted for by error correction and identification data.

Each slow COFDM carrier is unruffled by delayed signals arriving up to 1.2 times the guard interval later than the direct signal – in this case 300μs, corresponding to an extra path length for the delayed signal of 90km. Indeed delayed signals caused by multipath reflections add to, rather than detract from, the received signal quality. The 300μs permissible delay is well beyond the delay encountered in multipath conditions from a single transmitter, and opens up the possibility of Single Frequency Networks (SFNs). Transmitters carrying the same programme selection can all operate on the same frequency, and provided the difference in path length between two transmitters does not exceed 90km, and the transmitted signals are synchronised with each other, the effect on the receiver of multiple signals is wholly beneficial.

COFDM in Mode 1 is limited to applications below about 350MHz because it is sensitive to frequency shifts caused by the Doppler effect. A shift of $^{1}/_{20}$ of the carrier spacing (50Hz) increases the required carrier-to-noise ratio by 4dB. It is convenient to remember that at 300MHz the Doppler shift in hertz has the same numeric value as the speed of a moving receiver in m/s. So signals picked up in a speeding car or a fast train moving at 50m/s (about 112mile/h) directly away from or toward the transmitter will suffer a 50Hz shift at 300MHz, and *pro rata* at higher or lower frequencies. At the 226MHz frequency proposed as a 'parking band' for DAB, the 'velocity loss' will not exceed 4dB until the vehicle is travelling at nearly 150mile/h.

Modes 2 and 3 define carriers 4 and 8kHz apart, and can be used without unacceptable Doppler limitations up to about 1.4 and 2.8GHz respectively. Mode 2 may be applicable to local broadcasting and limited-area single-frequency networks, and allows a guard interval of 62.5μs. Mode 3 suits satellite or hybrid systems in the 'L' and 'S' bands – the guard interval here is 31.25μs.



COFDM in practice: This shows the COFDM (coded orthogonal frequency division multiplex) system working in Mark 1 (FDMA) mode. Each programme permanently occupies one carrier in every six. FDMA was used in the early Rennes and Geneva DAB demonstrations.
The Mark 2 TDMA mode – used in the BBC and NAB DAB demonstrations – uses all the carriers interleaved in the time domain for all the programmes, and seems to be more resistant to impulsive interference. But it would preclude radiating a set of skeleton carriers in an attempt to introduce more flexibility for individual channel operators.

process when it comes to matching local radio services to particular communities. It means for instance that a 'one-for-one' guarantee – that every current FM service can have a comparable DAB outlet – would be impossible to achieve economically. The shape and structure of the radio market will change fundamentally with Eureka DAB.

Private fears

Eureka means bundled broadcast channels. Independent operators running just one or two services do not want another four or five starting up in parallel with them, while stand-alone services in rural areas or small markets do not want to have to pay for a six-service installation when all they require is one, always assuming that a frequency band would be available for them. Eureka DAB can run single-frequency networks with great ease over wide areas, but the benefits of this sort of working turn to dust when a different programme is required in a different area, and the protection ratios required end up much the same as FM[3].

A recent report from Germany, the birthplace of DAB, began with Ursula Adelt of the Private Broadcasters' Association sending shock waves through the recent IFA show in Berlin by declaring that "for the private radio operators, Eureka 147 is dead[4]." Already ARD, the German state broadcaster, has caused gloom in the Eureka lobby by announcing that it could not afford to implement DAB until 1997.

Now private broadcasters are beginning to fret about who is going to pay for it, whether listeners really want it, or whether it might share the fate of Digital Satellite Radio, with just 100,000 listeners after five years. Reports from Washington also put the very existence of a US digital radio business in doubt given the current business environment[5].

The development of DAB

The Eureka Project EU147 on digital broadcasting was launched at the European Conference of Ministers in Stockholm in 1986, initially for a four-year plan of research and development between 1987 and 1991. Late in 1991, the second phase began. It has been looking into system specifications, application-specific integrated circuits and details of mass communication services additional to broadcasting which could be provided. These might include public or subscription services carrying data, for which there is some spare

Musicam: cutting the bit budget

A decade of work on bit rate reduction has made the compact disc, shifting 1.4Mbit/s for two channels of audio, look highly extravagant. Television's nicam system manages with 728 kbits/s using a simple scaling technique, but Musicam – in the form known by the snappy title "ISO-MPEG Layer 2: IEC CD 11172-3" offers between 192-256kbits/s per stereo channel for quality closely approaching that of the original source.

The block diagram of the audio encoder is shown below. The heart of the process is the 'psychoacoustic model' – a device which exploits the fact that the threshold of hearing is programme-dependent. In the presence of a 500Hz tone at 70dBA, for example, the threshold of hearing is raised 10dB between 300Hz and 1kHz, while between 400 and 600Hz energy needs to be above 40dBA to be detected. Ultimately, 'perceptual coders' such as this one aim to produce a noise floor which pumps up and down in 32 sub-bands in such a way that it is always masked by the programme material above it[12,13].

The bit allocation for each sub-band is determined by a calculation involving input from a 1024-point FFT scan capable of detecting the difference between sinusoidal and noise-like energy and adjusting masking thresholds to suit. The data from the sub-band filter is re-quantised to maintain just enough resolution (maybe only two or three bits) for the quantisation noise to be inaudible below the masking threshold for each band.

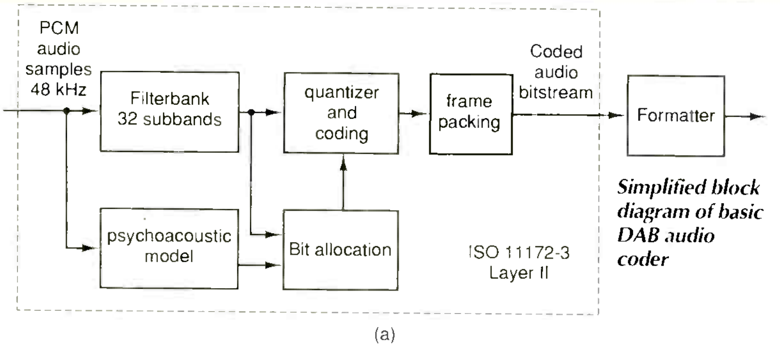
The perceived sound quality of this very complex process can be traded against bit rate to a fine degree. The BBC demonstrations offered the following options:

256kbit/s independent stereo for Radio 3;
224kbit/s joint stereo for Radios 1 & 2;
192kbit/s joint stereo for Radios 4 & 5.

The 'joint stereo' process exploits the redundancy between left and right signals of a stereo pair to provide better quality for a given bit budget than could be obtained with two independent mono channels each running at half the rate. The overall bit budgets that can be assigned vary from 32kbit/s to 192kbit/s for a single channel, the latter allowing headroom for further processing after reception, or from 64kbit/s to 320kbit/s for a stereo channel or pair of services.

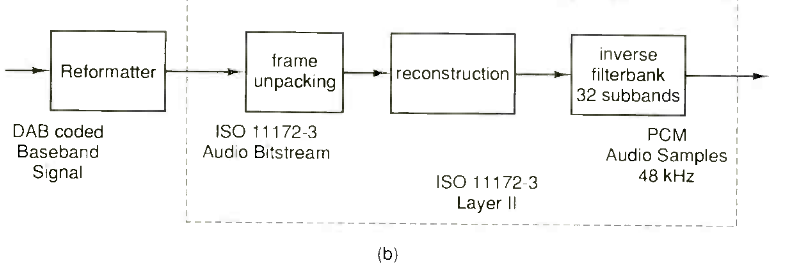
It is very difficult to assess the performance of Musicam at the bit-rates offered when the original, uncompressed signal is not available for comparison, and there is currently such a shortage of DAB receivers that it was not possible to do an A/B listening test inside Broadcasting House.

Any lack of transparency has a fundamentally different quality from familiar analogue shortcomings, although listeners who have taken part in demonstrations say that once the ear has learnt what to listen for, deficiencies can be more readily detected. In the BBC coach, any sonic difference between DAB and undegraded FM was very subtle at the bit rates chosen for demonstration.



Simplified block diagram of basic DAB audio coder

(a)



(b)

Simplified block diagram of basic DAB decoder
Source: see Ref. 13.

Demonstrating DAB

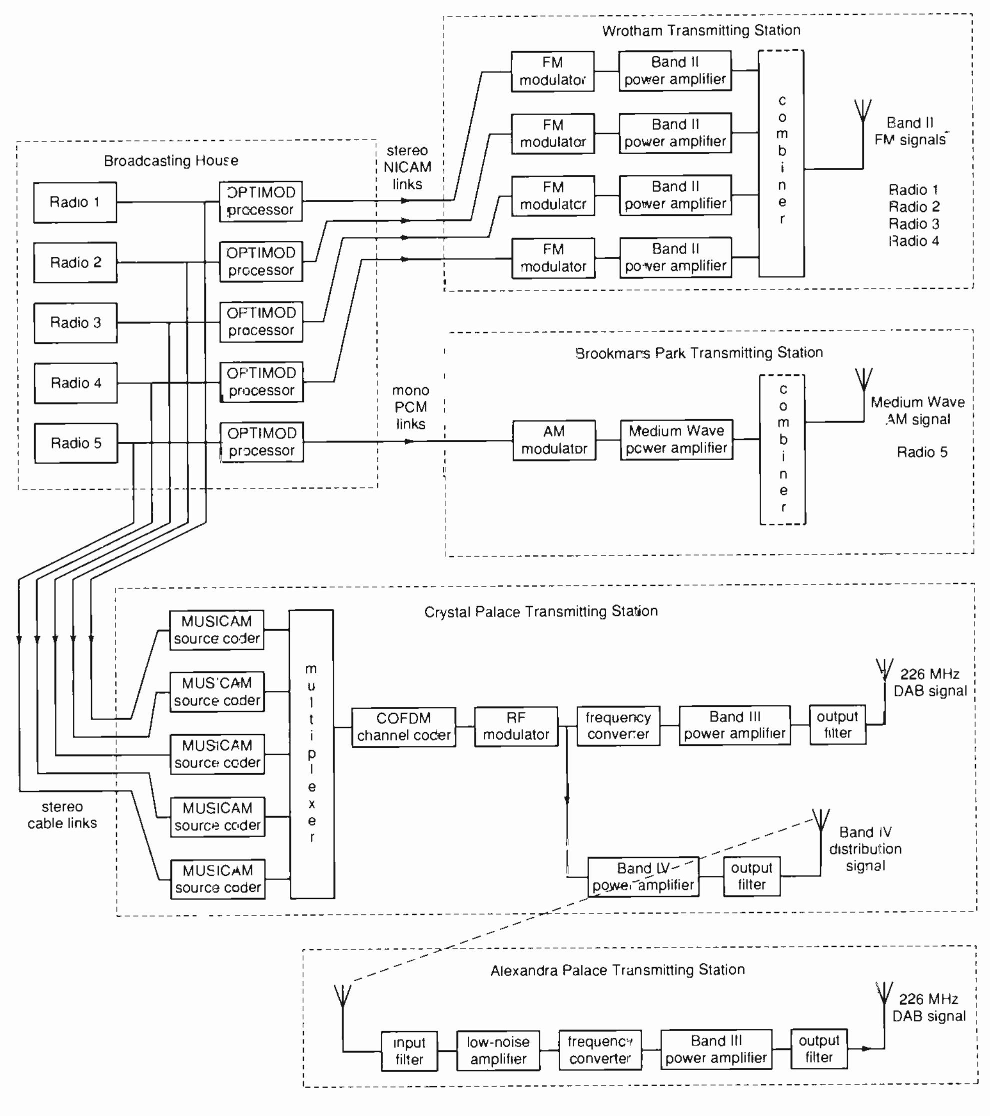
"The digits deliver the goods – they solve everything," said one engineer at the BBC demonstration of Digital Audio Broadcasting in central London in early December. From a technical point of view, the coach ride – with DAB or FM selectable on headphones – was impressive, but with reservations.

We were invited to compare FM reception from Wrotham, 20 miles to the East in Kent, with DAB reception from much closer transmitters at Crystal Palace and Alexandra Palace running simultaneously. DAB reception was solid all the way, while FM was subject to the familiar noise and fading which mars much city reception. But we were not, it has to be said, comparing like with like.

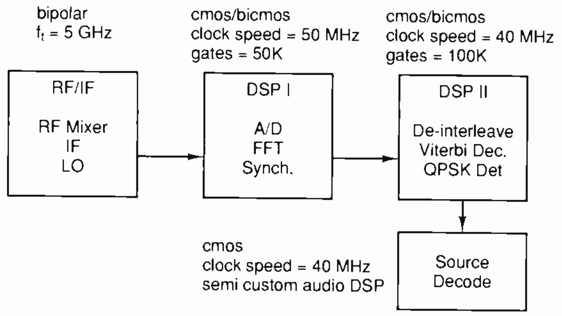
I put this to the BBC, suggesting that a fairer demonstration would have been to compare DAB with FM reception from Crystal Palace (the BBC's GLR transmitter on 94.9 MHz), or alternatively to radiate DAB only from Wrotham so that it suffered the same degradation as the FM network signal. I was told that the BBC wanted to show DAB on the network services with their various programme content, for which there is no central London transmitter, and that to bring DAB into central London from Wrotham would require a transmitter power of at least 10kW, which was not at the BBC's disposal.

Originally, the tests were wired to provide unprocessed, uncompressed feeds to the DAB transmitter while leaving the 'Optimod' multi-band signal processors in circuit on the FM system. This proved to be an unwise move: listeners commented on differences in the sound quality of the two services, attributing them (wrongly) to inherent characteristics of FM versus DAB, whereas in fact they were observing the effects of the Optimod units, which make programme material seem denser, louder and superficially more impressive. So DAB got the same processed feeds as FM after the first day.

This meant that, interference apart, DAB audio quality sounded practically identical to FM. It may indeed have been sold short by having been fed first from a Nicam link sampling at 32kHz and therefore limiting the audio bandwidth on both systems to 15kHz. To get digital quality above, and not just on a par with, FM may well need some re-engineering of distribution networks in the future. And we have been reminded that most people prefer limited dynamic range on their radio material, with the exception of a few readers of *Hi-Fi News*[11], so that CD levels of dynamic range are positively unwelcome to most consumers.



Arrangements for the London DAB demonstration. This shows the arrangements made to distribute and demonstrate DAB by the BBC in central London, with two important omissions. The input to the Musicam coders was transferred to the output of the 'Optimod' processors to avoid misleading listeners hearing the effects of the processor. And the process of distributing audio to the DAB network was handled by a Nicam system operating at 32kHz sampling rate, which is not shown on the diagram.



Proposed four-chip DAB receiver. This shows a possible architecture for a practical DAB receiver. RF functions are handled by high speed bipolar technology using an oxide or trench isolated process. The first DSP incorporates the A/D converter, FFT algorithm and synchronisation circuitry. The next two ICs take in the Viterbi decoder, de-interleaver, QPSK detector and source decoder. The hope is that source decoders will become off-the-shelf devices if the DAB Musicam audio compression standard is widely implemented elsewhere. Source: Ref. 6

capacity. The project is a consortium of partners including broadcasters, PTT administrations and consumer electronics interests.

In the UK, Michael Heseltine announced the setting up of a DAB Forum in the Commons on 16 February 1983, declaring that he was anxious to ensure the benefits and opportunities of DAB were made available in the UK. Broadcasters, equipment manufacturers and other interested parties were invited to join. At a meeting with opposite numbers representing France, Germany and the Netherlands, they considered factors necessary for the successful early introduction of DAB. They concluded that success depended on the following:

- an early move to a single frequency band with sufficient spectrum for all public and private broadcasting systems;
- new programmes on DAB, not just simulcasts of current channels;
additional services, such as data broadcasting;
- new ways of financing digital broadcasting, including encryption and pay-per-listen;
the support of car manufacturers to create a pool of DAB car radios;
- a legal framework which provides incentives and allows for innovation; conditions which will provide sufficient rewards for DAB pioneers.

Alongside these committee recommendations remains the fundamental question: how much will DAB cost to transmit?

At the transmission end of the chain, Mike Thorne, of UK transmission providers NTL, the privatised wing of the old IBA, has been bold enough to put a figure on transmitting DAB to a large urban area, although he would be the first to admit that these are projected costs at this stage, and not a bid for a contract.

NTL's costs assume that an existing transmitter site can be used, with space available on the mast for the 230 MHz antenna system, and room below in the building for the apparatus. Given this, construction costs for a single 1kW station are put at £450,000, and the running costs – use of mast, rent of building, electricity, BT lines and maintenance etc. – would amount to £70,000 pa. To provide a 'total broadcast contract', including paying back the capital costs over some 8 - 10 years, and a sixth of the running costs above, would cost each of six operators £25,000 pa.

To put together five 250W transmitters in a single frequency network (SFN) would cost each operator about double the amount for a single transmitter (£40 – 50,000 pa), but would provide much more solid coverage and better frequency use. This is not dissimilar to the present-day costs of a comparable FM service from the same company.

Receiver costs

As for the receiver, the current state of development is known as the third generation, and consists of a substantial rack of equipment drawing 2.5A from a 12V supply, and costing over £3000. A four-chip future receiver selling for the price of a good CD player (£150 – £200) has been sketched out but is still a long way off[6].

More immediately, an assessment of likely parameters for a start-up phase in 1995[7] anticipates that first generation consumer DAB products will contain around ten specific integrated circuits together with their peripheral components.

The first sets will be expensive, on a par, perhaps, with the early fax machines or mobile phones. It is highly unlikely that a £50 DAB 'Walkman' radio will be a practical proposition for a few years. DAB and FM will not compete initially for mass-appeal receivers.

Estimates of demand and penetration levels have been produced[9]. The most hopeful projection does not foresee DAB reaching 50% of the marketplace before 2010. This is three years later than the date when some are recommending that FM stations are closed down to make way for it[8].

DAB and the radio business

In the long term, Eureka DAB, with its centralised transmission system, will profoundly affect the nature of the marketplace in radio for both transmission and programme services, favouring national networks and large-area broadcasters over smaller companies. That DAB is seen not as an addition to FM but in the long term as a replacement for it implies a possible threat to FM stations.

If an existing operator merely re-broadcasts current programmes on DAB, he possibly faces the additional costs of transmitting the new service, but without the audience and revenue to fund it. There may however be some profit in subsidiary data services. Previous attempts to market Radiotext data services on top of FM broadcasts have been unsuccessful.

On the other hand, if an established or new operator decides to offer a separate audio service on DAB in the hope of attracting new listeners and sources of revenue, he will have to bear the costs of producing this alternative programme. This is in addition to transmitting it while the service builds up to profitable proportions.

Future developments

These possible business shortcomings in the DAB system have focussed thoughts on modifications. Dr. Brian Evans[10] has suggested that were the Mark 1 FDMA interleaving process to be adopted, it ought to be possible to erase five of the six sets of carriers from the mix in order to permit individual transmitters to provide individual services without the need for inter-broadcaster synchronisation of sources.

There are penalties with this system in that it is no longer possible to reach the theoretical minimum spacing between carriers, and the FDMA system is generally less efficient than a full TDMA multiplex at resisting interference. But it does address the handicap of centralised transmission without losing the key quality features of DAB.

Improvements to DAB's non-graceful degradation are under examination. To have a system fail completely, even if only 1% of locations are affected, is still a serious handicap compared to the more graceful degradation of analogue systems. AM carriers or some analogue 'helper' signals which could be added to the DAB multiplex to save the day without themselves taking up too much bandwidth or power might smooth out this technical rough edge to everyone's satisfaction.

DAB spectrum use and frequency planning

Plans for the UK interim DAB service on Band III propose two 1.75MHz-wide blocks for national services, one set for the BBC and one for the commercial sector, and five further blocks for local and regional coverage. A maximum of 42 services could be potentially available if all blocks could be received in a given location.

In terms of programme packing density per megahertz, DAB is very broadly equivalent to FM when set against the local radio sections of the FM band (94.6 – 97.6 and 102 – 105MHz). Typically six DAB services are provided in 1.75MHz of spectrum. Where Band II is well-used, it is also possible to pick out six or so FM services across 1.75MHz on a reasonably good tuner and aerial. The noise bandwidth per service used for DAB s/n calculations is typically 300kHz, slightly greater than the bandwidth of an FM transmission (240kHz).

National networks

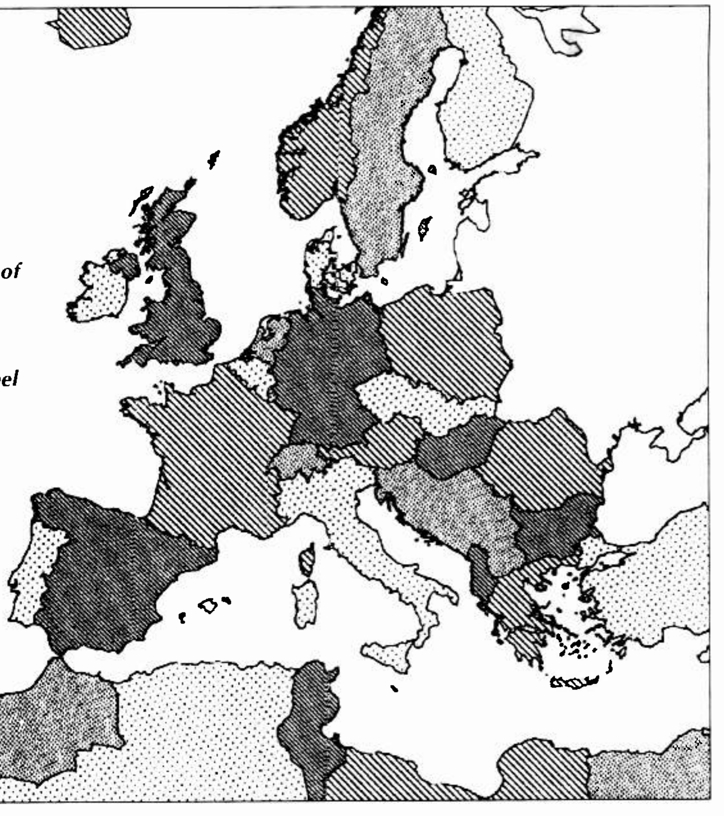
For national networks, DAB looks splendidly efficient at frequency use, since the same 1.75MHz slab of spectrum can be re-used *ad infinitum* across the UK to provide five or six national services. The current five national FM services consume 11MHz between them – more than half of the Band II allocation.

The ability to extend national networks over an SFN stops, of course, at national boundaries. Maps like the one shown have been produced showing Europe painted according to the four-colour map theory. This would enable four blocks to be used for national SFNs with only minor problems in places like Luxembourg or Liechtenstein. National blocks assigned to other countries could, of course, be re-used by smaller regional or local services away from the frontiers.

DAB fails – abruptly and completely – when the incoming carrier-to-noise ratio falls below 10 - 14dB. Above that, there is solid reception independent of signal strength. FM systems degrade gracefully as the signal fails, so that for planning purposes a median curve (signal exceeded in 50% of locations, 50% of the time) can give a good rule of thumb for setting 'coverage'. Anyone receiving less than a fair share of signal may get degraded reception, but will still be able to follow the service.

Because DAB does not behave in this manner, calculations for DAB planning have to be based on generous margins – up to 99% probability – on top of median values which look impressive in isolation. Calculations have shown that assuming a receiver noise figure of 8dB operating at 200MHz from a practical car antenna, the required DAB field strength must be at least 22dBμV/m in the vicinity of the aerial, which may be barely 2m off the ground[15]. The familiar planning curves for FM broadcasting assume the use of an antenna 10m high[16]. A correction for this discrepancy is open to debate but we will assume 12dB, bringing the signal requirement at 10m to a modest 34dBμV/m (50μV/m). On a fixed FM tuner with an outdoor aerial this level of strength will produce tolerably quiet mono reception, but mobile FM reception would be choppy even with a good receiver.

Onto this figure must be added a factor to increase the probability of the signal reaching this minimum value from 50% to, say, 99%. This would amount to 19dB, bringing the required median level to 53dBμV/m, very similar to the field strength required for stereo reception of the FM service. However, where an SFN is in operation (and this is the most likely situation) there is a factor working to improve matters because the receiver aerial is being illuminated by signals from more than one direction, greatly increasing the chances of successful reception. An advantage of 6 – 10dB might be expected from this source. But against this, propagation curves for Band II (CCIR Rec 370) need to be scaled down by 6dB when applied to 230MHz predictions; this is a function of the frequency, of course, and not the system, but it works against DAB in Band III (though would work in its favour on Band I).



European frequency reuse: just four DAB frequency blocks could provide a national network of programmes – up to six per block – in each country without co-channel interference.

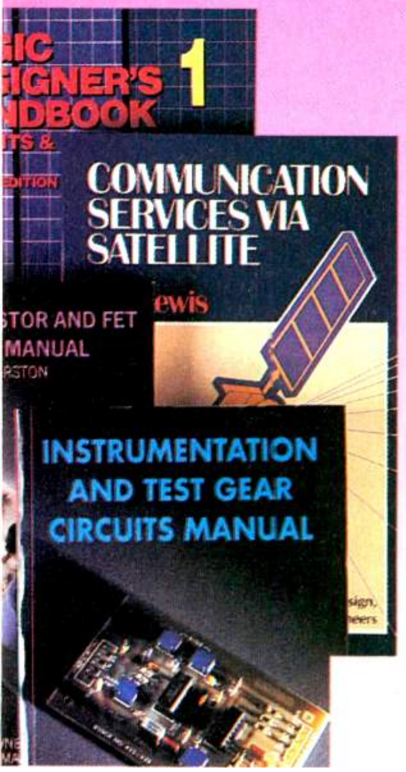
Man-made interference in populated areas is becoming significant. This can increase the noise floor of the received signal by 7dB or more. The effect is worst in dense urban areas, but generally signals are planned to be stronger in these locations as a matter of course. Nevertheless, a value of 4dB has been suggested as an overall margin to allow for current levels of pollution from computer and digital electronic equipment.

In general, it seems that a DAB SFN of perhaps five 250W DAB transmitters could provide substantially better coverage over a large urban area than a single FM transmitter running 2-5kW. Were the DAB transmitters to be co-sited with existing UHF television transmitters where aerial space above and accommodation below are already available, initial projections show that DAB may offer slight savings over a comparable number of FM services. But there are still trade-offs and margins in DAB planning. ■

References

1. *RDS - an ear to the future*, BBC Engineering Training Department, 1988
2. Pat Hawker, *How serious is multipath distortion?, Wireless World*, April 1980, p.45
3. Gerd Petke, Institut fur Rundfunktechnik GmbH, *Planning aspects and plans for terrestrial DAB implementation*, Proc. of Montreux DAB Symposium, 1992, p.167 (EBU)
4. *Fundamental Questions Hamper German DAB, Radio World*, 23 Nov 1993, p.24
5. USA *DAB Confronts New Obstacles, Radio World*, 23 Nov 1993, p.20
6. McComas, Kady & Warren, Delco Electronics Corp., *Manufacturing DAB automobile receivers*, Proc. of Montreux DAB Symposium, 1992, p.235 (EBU)
7. Manfred Halbe, Philips Consumer Electronics International bv, *DAB receiver technology and implementation considerations*, Proc. of Montreux DAB symposium, 1992, p.121 (EBU)
8. *FM Stations may close in spectrum shake-up, Electronics World and Wireless World*, September 1993, p.708
9. M. Carrasco & A. Lange, Idate consortium, *What markets for DAB in Western Europe*, Proc. of Montreux DAB Symposium, 1992, p.245 (EBU)
10. Dr. Brian Evans, Tantara Tek Ltd., *Ensuring a bright future for independent DAB*, September 1993
11. Reg Williamson, *The Limiting Factor, Hi-Fi News & Record Review*, October 1993, p.73
12. Fred Wylie, APT, *Digital Audio Data Compression, Audio Media*, September 1993, p.84
13. Gerhard Stoll, Institut for Rundfunktechnik, Munich, *Source Coding for DAB and the evaluation of its performance: a major application of the new ISO Audio Coding Standard*, Proc. of Montreux DAB Symposium, 1992, p.83 (EBU)
14. B. Le Floch, CCETT, *Channel coding and modulation for DAB*, proc. of Montreux DAB Symposium, 1992, p.99 (EBU)
15. C.P.Bell & W.F.Williams, BBC Research Dept., *DAB: Digital Audio Broadcasting coverage aspects of a single-frequency network IBC*, July 1992
16. *Recommendations and reports of the CCIR*, Volume V, Rec. 370-5, ITU, 1990

BOOKS TO BUY



Programmable Logic Handbook
Geoff Bostock
Logic circuit designers are increasingly turning to programmable logic devices as a means of solving problems. This book, for the established electronics engineer, student and technician, is a thorough introduction to programmable logic. Geoff Bostock will take you to a level where you, as a designer, can take full advantage of the growing product range of ASICs and other self-programmable arrays used in computer and control systems. Paperback 256 pages.
Price £19.95 0 7506 0808 0

Understand Electrical and Electronic Maths
Owen Bishop
People who find maths difficult often have, as a result, difficulty in grasping electrical and electronics theory. This book has been written to help such students to understand the mathematical principles underlying their subject so that they can go on with confidence to tackle problems in practical circuits. Paperback 256 pages.
Price £14.95 0 7506 0924 9

CIRCUIT MANUALS
Ray Marston
A series of books dealing with their subjects in an easy-to-read and non-mathematical manner, presenting the reader with many practical applications and circuits. They are specifically written, for the design engineer, technician and the experimenter, as well as the electronics student and amateur. All the titles are written by Ray Marston, a freelance electronics design engineer and international writer.

Op-amp Circuits Manual
Paperback 224 pages
Price £13.95 0 434 912077

Audio IC Circuits Manual
Paperback 168 pages
Price £13.95 0 434 912107

CMOS Circuits Manual
Paperback 192 pages
Price £13.95 0 434 912123

Electronic Alarm Circuits Manual
Paperback 144 pages
Price £13.95 0 7506 00640

Timer/Generator Circuits Manual
Paperback 224 pages
Price £13.95 0 434 912913

Diode, Transistor and FET Circuits Manual
Paperback 240 pages
Price £13.95 0 7506 0228 7

Instrumentation and Test Gear Circuits Manual
Ray Marston
Modern instrumentation and test gear circuits of value to the industrial, commercial, or amateur electronic engineer or designer make up this book. Almost 500 outstandingly useful and carefully selected practical circuits are in here. This is one book you must have if you need access to practical working circuits ranging from simple attenuators and bridges to complex digital panel meters, waveform generators, and scope trace doublers. Paperback 400 pages.
Price £16.95 0 7506 0758 0

Logic Designers Handbook
Andrew Parr
Easy to read, but none the less thorough, this book on digital circuits is for use by students and engineers and provides an accessible source of data on devices in the TTL and CMOS families. It's a 'Designers Handbook' that will live on the designer's bench rather than on the bookshelf. The basic theory is explained and then supported with specific practical examples. Paperback 488 pages.
Price £25.00 0 7506 0535 9

Digital Audio and Compact Disc Technology
Luc Baert, Luc Theunissen & Guido Vergult
Essential reading for audio engineers, students and hi-fi enthusiasts. A clear and easy-to-follow introduction and includes a technical description of DAT (digital audio tape). Contents includes principles of digital signal processing, sampling, quantization, A/D conversion systems, codes for digital magnetic recording, principles of error correction, the compact disc, CD encoding, opto-electronics and the optical block, servo circuits in CD players, signal processing, digital audio recording systems, PCM, Video 8, R-DAT and S-DAT. Paperback 240 pages.
Price £16.95 0 7506 0614 2

NEWNES POCKET BOOKS
A series of handy, inexpensive, pocket sized books to be kept by your side and used every day. Their size makes them an ideal 'travelling' companion as well.

Newnes Electronics Engineer's Pocket Book
Keith Brindley
Hardback 319 pages
Price £12.95 0 7506 0937 0

Newnes Electronics Assembly Pocket Book
Keith Brindley
Hardback 304 pages
Price £10.95 0 7506 0222 8

Newnes Television and Video Engineer's Pocket Book
Eugene Trundle
Hardback 384 pages
Price £12.95 0 7506 0677 0

Newnes Circuit Calculations Pocket Book
T Davies
Hardback 300 pages
Price £10.95 0 7506 0195 7

Newnes Data Communications Pocket Book
Michael Tooley
Hardback 192 pages
Price £12.95 0 7506 0427 1

Newnes Telecommunications Pocket Book
JE Varrall & EA Edis
Hardback 400 pages
Price £12.95 0 7506 0307 0

Newnes Z80 Pocket Book
Chris Roberts
Hardback 185 pages
Price £12.95 0 7506 0308 9

Newnes 68000 Pocket Book
Mike Tooley
Hardback 257 pages
Price £12.95 0 7506 0309 7

Newnes Electrical Pocket Book
21st edition
E A Parr
Paperback 526 pages
£12.95 0 7506 05138

Newnes Electric Circuits Pocket Book Linear IC
Ray Marston
Hardback 336 pages
Price £12.95 0 7506 0132 9

Newnes Guide to Satellite TV
D J Stephenson
A practical guide, without excessive theory of mathematics, to the installation and servicing of satellite TV receiving equipment for those professionally employed in the aerial rigging/TV trades. Hardback 256 pages.
Price £17.95 0 7506 0215 5

Newnes Practical RF Handbook
Ian Hickman
Pressure on the RF spectrum has never been greater and it's people with knowledge and skills of RF design who are now in demand in the electronics industry to design, produce, maintain and use equipment capable of working in this crowded environment. This practical introduction to modern RF circuit design will equip you with the necessary RF knowledge and skills to enable you to compete effectively in the industry. Paperback 320 pages.
Price £16.95 0 7506 0871 4

Troubleshooting Analog Circuits
R A Pease
Bob Pease is one of the legends of analog design. Over the years, he's developed techniques and methods to expedite the often-difficult tasks of debugging and

troubleshooting analog circuits. Now, Bob has compiled his 'battle-tested' methods in the pages of this book. Based on his immensely popular series in EDN Magazine, the book contains a wealth of new material and advice for Digital/Analog electronics engineers on using simple equipment to troubleshoot. Paperback 217 pages.
Price £14.95   0 7506 16326

PC-Based Instrumentation and Control
M Tooley
Do you need information to enable you to select the necessary hardware and software to implement a wide range of practical PC-based instrumentation and control systems? Then this book is for you. Paperback 320 pages.
Price £14.95   0 7506 1631 8

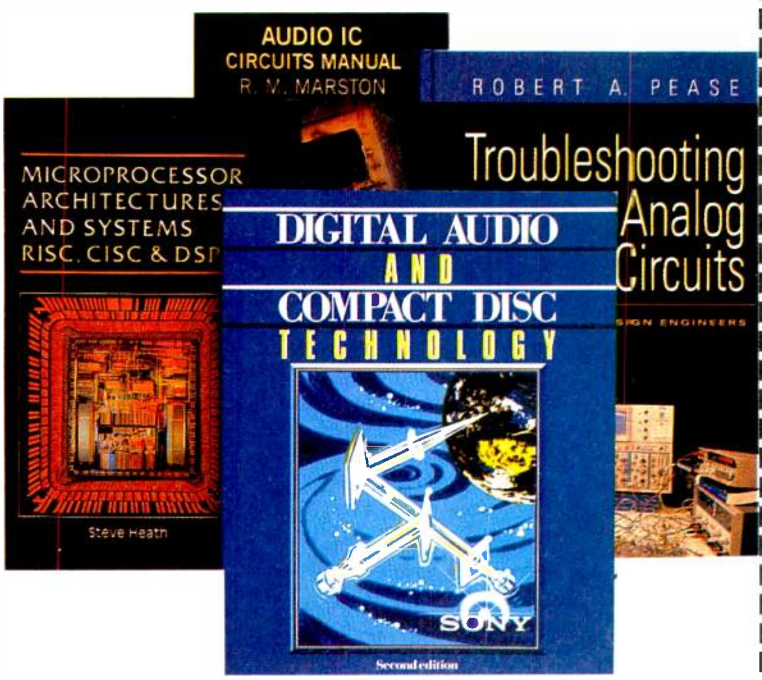
Electronic Circuits Handbook
M Tooley
Provides you with a unique collection of practical working circuits together with supporting information so that circuits can be produced in the shortest possible time and without recourse to theoretical texts. Paperback 345 pages.
Price £24.95   0 7506 0750 5

Communication Services via Satellite
G E Lewis
DBS is already with us, and will create a series of new technical problems for engineers/technicians in television and communication services. This book gives you the solutions to these problems by: explaining how the system functions; describing several actual systems and giving several analyses and design rules. You can't afford to be without this invaluable technology update if you're a systems design engineer, service engineer or technician. Paperback 400 pages.
Price £25.00   0 7506 0437 9

Digital Logic Design
Brian Holdsworth
As one of the most successful and well established electronics textbooks on digital logic design, this book reflects recent developments in the digital fields. The book also covers new functional logic symbols and logic design using MSI and programmable logic arrays. Paperback 448 pages.
Price £19. 50   0 7506 0501 4

The Circuit Designers Companion
T Wiliams
This compendium of practical wisdom concerning the real-world aspects of electronic circuit design is invaluable for linear and digital designers alike. Hardback 320 pages.
Price £25 00   0 7506 1142 1

Credit card orders accepted by phone
081 652 3614



Return to: Lorraine Spindler, Room L333, Quadrant House, The Quadrant, Sutton, Surrey SM2 5AS

Please supply the following titles:

Qty	Title	ISBN	Price
	Programmable Logic Handbook	07506 0808 0	19.95
	Understanding Electrical & Elec Maths	07506 0924 9	14.95
	Op-amp Circuits Manual	0434 912077	13.95
	Audio IC Circuits Manual	0434 912107	13.95
	CMOS Circuit Manual	0434 912123	13.95
	Electronic Alarm Circuits Manual	07506 0064 0	13.95
	Power Control Circuits Manual	07506 06908	13.95
	Timer/Generator Circuits Manual	0434 91291 3	13.95
	Diode, Transistor & FET Circuits Man	07506 0228 7	13.95
	Instrumentation & Test Gear Circuits Man	07506 0758 0	16.95
	Logic Designers Handbook	07506 0535 9	25.00
	Digital Audio and Compact Disc	07506 0614 2	16.95
	Newnes Elec Engineers Pkt Bk	0 7506 0937 0	12.95
	Newnes Elec Assemby Pk Bk	07506 0222 8	10.95
	Newnes TV and Video Eng Pkt Bk	07506 0677 0	12.95
	Newnes Circuit Calculations Pkt Bk	07506 0427 1	10.95
	Newnes Data Communications Pkt Bk	07506 0308 9	12.95
	Newnes Telecommunications Pkt Bk	07506 0307 0	12.95
	Newnes Z80 Pkt Bk	07506 0308 9	12.95
	Newnes 68000 Pkt Bk	07506 0309 7	12.95
	Newnes Electrical Pk Bk	07506 05138	12.95
	Newnes Electric Circuits Pocket Bk	07506 0132 9	12.95
	Newnes Guide to Satelite TV	07506 0215 5	17.95
	Newnes Practical RF Handbook	07506 0871 4	16.95
	Troubleshooting Analog Circuits	07506 16326	14.95
	PC-Based Instrumentation and Control	07506 1631 8	14.95
	Electronic Circuits Handbook	07506 0750 5	24.95
	Communication Services via Satellite	07506 0437 9	25.00
	Digital Logic Design	07506 05014	19.50

PLEASE ADD £2.50 FOR POSTAGE ______
Add VAT at local rate ______
NB ZERO RATE FOR UK & EIRE ....TOTAL ______

Business purchase: Please send me the books listed with an invoice. I will arrange for my company to pay the accompanying invoice within 30 days. I will attach my business card/letterhead and have signed the form below.
Guarantee: If you are not completely satisfied, books may be returned within 30 days in a resaleable condition for a full refund.

Remittance enclosed £ ______
Cheques should be made payable to **Reed Book Services Ltd.**
Please debit my credit card as follows:

Access/Master   Barclay/Visa   Amex   Diners

Credit Card No. ______ Exp date ______
NAME (Please print) ______
ORGANISATION ______
STREET ______
TOWN ______
COUNTY ______ POST CODE ______ COUNTRY ______
DATE ______ TELEPHONE NUMBER ______
SIGNATURE ______

T3000

VAT RATES
6% Belgium, 25% Denmark, 5.5% France, 7% Gemany, 4% Greece, 4% Italy, 3% Luxembourg, 6% Netherlands, 5% Portugal, 3% Spain. FOR COMPANIES REGISTERED FOR VAT, PLEASE SUPPLY YOUR REGISTRATION NUMBER BELOW (customers outside the EEC should leave this part blank)
VAT NO. ______
If in the UK please allow 28 days for delivery. All prices are correct at time of going to press but may be subject to change.
Please delete as appropriate. I do/do not wish to recieve further details about books, journals and information services.
Reed Business Publishing - Registered Office - Quadrant Hse The Quadrant Sutton Surrey SM2 5AS   Registered in England 151537

USING RF TRANSISTORS

Biased view

***Norm Dye and Helge Granberg explain the ins and outs of bias circuits, and show how to handle temperature compensation. The authors also look at using devices in pulse mode. From the book* RF Transistors: principles and practical applications.**

In bipolar transistors, base current is equal to $I_{c(peak)}/h_{FE}$. So the base bias supply must be able to supply this current without too much effect on the base-emitter voltage between the no-signal and the maximum signal conditions. The supply should also be a constant voltage source, as variations of a few millivolts represent a large portion of the nominal 0.63-0.67V typical value.

But specific applications demand many other requirements of the base bias voltage source.

In some instances a large value capacitor can be connected across the voltage supply, further reducing its ac impedance. But this makes impedance dependent on frequency of modulation, and is a practical solution only where the modulating frequency is in the medium to high audio frequency range.

One of the simplest biasing circuits for bipolar transistors (**Fig. 1**) uses a clamping diode to provide a low impedance voltage source.

Forward current of the diode must be greater than the peak base current of the transistor. In the circuit, current is adjusted with R_2 and the resistance of RFC_1 and R_1 is used to reduce the actual base voltage to a slightly lower value than the forward voltage of D_1. Mechanical connection to the heat sink or the transistor housing performs a temperature compensating function for Tr_1 — an adequate solution although for perfect temperature tracking, Tr_1 and D_1 should have similar dc parameters.

A disadvantage of the circuit is its inefficiency, especially in biasing high power devices: $(V_{cc}-V_b)\times I_{b(max)}$ will always be dissipated in the dropping resistors.

But the loss can be overcome by amplifying the clamping diode current with an emitter follower (**Fig. 2**). Two series diodes (D_1 and D_2) are used so that one can compensate for the $V_{BE(f)}$ drop in Tr_1. In this case low current signal diodes can be used and their forward current is equal to $I_{(bias)}/h_{FE(Tr1)}$.

For best result, Tr_1 should have a linear h_{FE} up to the peak bias current required. In higher power systems Tr_1 must be cooled, and ideally, it and one of the series diodes should remain at ambient temperature. The other diode (D_1 or D_2) can be used for temperature compensation of the rf device.

When the diode (having long leads) is located near the rf transistor the result is an effective fast-responding system. The leads can be formed to allow the body of the diode to be pressed against the ceramic lid of the rf transistor and fastened in place with thermally conductive epoxy.

Resistor R_1 sets the bias idle current and R_2 limits its range of adjustment – the value of R_2 depends on the supply voltage employed. Capacitor C_1 and rf choke *rfc* are there simply to prevent the rf signal from getting into Tr_1.

Another fairly simple bipolar bias source (**Fig. 3**) has its output voltage equal to the base emitter junction drop of Tr_1 plus the drop across R_3. R_1 must be selected to provide sufficient base drive current for Tr_2, set by its h_{FE}. Normally this current is in the range of a few milliamperes, and Tr_1 can be any small signal transistor in a package that can be easily attached to the heat sink or rf transistor housing for temperature compensation. The only requirement is that its $V_{BE(f)}$ at that current must be lower than that of the rf transistor at its bias current level. Maximum current capability depends on Tr_2 and R_2.

Power dissipation of Tr_2 can be up to a few watts. In most cases it should be heat sunk.

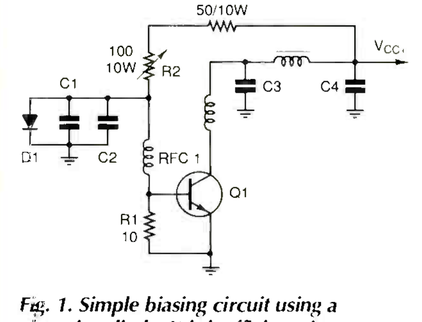


Fig. 1. Simple biasing circuit using a clamping diode. It is inefficient since a minimum of $I_{B(peak)}$ must go through D_1. Tr_1 is the transistor to be biased.

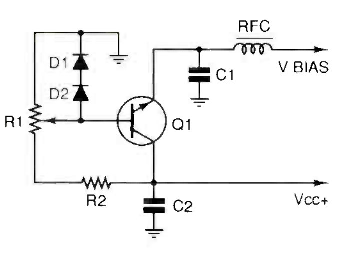


Fig. 2. A clamping diode scheme is used as in Fig. 1, but a low current flows through the diodes and is then amplified by an emitter follower.

But it must be electrically isolated from ground. The value of R_2 can be calculated as:

$(V_{CE}-V_{CE(sat)})/I_b$.

Capacitors C_{1-3} are a precaution to suppress high frequency oscillations, but may not be necessary depending on the transistors used and the physical circuit lay-out.

Output source impedances for the circuit, when used in conjunction with a 300W amplifier, have been calculated as low as 200-300mΩ.

Biasing mosfets

Gate threshold voltages of mosfets are 5-6V, so some gate bias voltage is needed in most applications.

Class C operation is possible (zero gate bias), though there is a cost in low power gain: the input voltage swing must be big enough to overcome the gate voltage, from zero to over the threshold level. But drain efficiency is usually higher than in other classes. Operation can also approach class D – especially if over-driven.

Zero bias is often used in amplifiers intended for signals not needing linear amplification — such as fm signals and some forms of cw signals. Efficiencies in excess of 80% are not uncommon.

In class B, the gate bias voltage is set just below the threshold, resulting in zero drain idle-current flow. Power gain is higher than in class C, but drain efficiency is 10-15% lower. Class B is also suitable only for non-linear amplification.

Between classes, the decisions to be made are whether the system has power gain to spare and the importance of efficiency.

At higher frequencies, such as uhf, a good compromise may be class B or even class AB. In class AB the gate bias voltage is somewhat higher than the device threshold, with drain idle-current flow resulting.

The idle current required to place the device in the linear mode is usually given in a data sheet. In this respect, mosfets are much more sensitive to idle current than bipolar transistors, and also require somewhat higher current levels compared to bipolars of similar electrical size.

Temperature compensation

Temperature compensation of mosfets can most readily be accomplished with networks of thermistors and resistors – the ratio of the two must be adjusted for thermistor characteristics and the g_{fs} of the fet. Changes in the gate threshold voltage are inversely proportional to temperature and amount to approximately 1mV/°C. They have a larger effect on the I_{DQ} of a fet with high gfs than one with low gfs. Unfortunately the situation is complicated by the fact that g_{fs} is also reduced at elevated temperatures, making the drain idle-current dependent on two variables.

In spite of this dependence, this compensation method can operate satisfactorily and is repeatable for production. The thermistor is thermally connected into a convenient location in the heat source in a similar way to the compensating diodes with bipolar units. **Figure 4** shows a simple mosfet biasing circuit.

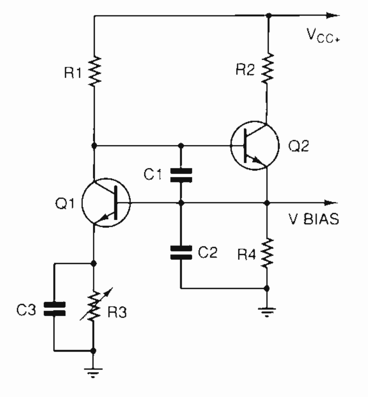


Fig. 3. Bias circuit featuring the lowest source impedance of the less complex bias circuits. It is recommended for high power device biasing and for other demanding applications.

Most mosfet device data sheets give $V_{gs(th)}$ versus I_D data. But the values are only typical. In some cases gfs can vary as much as 100% from unit to unit. In production, the devices should have gfs values within 20% of each other, or every amplifier will have to be indi-

Operating transistors in a pulse mode

RF energy, in the form of pulses, is exploited in many applications including medical electronics, laser excitation and various types of radar. In each, the specifications of carrier frequency, pulse repetition rate and duty cycle vary.

The carrier frequency is usually much higher than the pulse repetition rate, resulting in the generation of bursts of rf at the carrier frequency whose lengths depend on the pulse width. Pulse repetition rates are typically in the audio range and duty cycles range between 0.05 and 10%.

For low duty-cycle applications – such as radar – special devices have been developed to operate at higher peak powers, but with relatively low average powers, reducing dissipation. These transistors (uhf to microwave) are almost exclusively bipolar.

In general, bjts have higher peak power capabilities than mosfets. Peak power performance can be further improved by reducing the emitter ballast resistor values to lower than normally required for cw.

The epitaxial layer that controls the transistor's saturated power, is also made thinner than normal since the problem of ruggedness is partly eliminated by the low average power.

Increasing the pulse width increases the dissipation, and at pulse widths of 1ms and wider the device can be considered to operate like a cw signal: the temperature time constant of a medium size rf power die is around 1ms, beyond which more heat will be transferred into the bulk silicon and through it to the transistor housing.

If the pulses are short but the repetition rate approaches 1kHz (1ms period), the effect is the same. Transistors made exclusively for pulse operation can produce peak power levels of five to six times the cw rating for a die of a similar size.

Standard transistors designed for cw have a multiplying factor more of the order of three to four.

Mosfets for pulsed power

Mosfets can be used for pulsed power operation, but they have disadvantages as well as advantages compared to bjts. Disadvantages include pulse drooping, where the trailing end of the pulse has a lower amplitude than the leading end. It is caused by the decreasing g_{fs} of a mosfet with temperature. Corrective circuitry can compensate, but adds to circuit complexity.

Advantages include smaller phase delays and faster rise and fall times. Choice depends on the application and on what the designer decides which is the most suitable.

Additional considerations in design of a pulsed amplifier are energy storage near the device, and minimising inductance in the emitter leads. Both affect rise time of the pulse and prevent droop resulting from voltage decay during its duration.

Some trade-off will be required because as the emitter inductance to ground is reduced, wideband matching is made more difficult. Also a minimum amount of inductance is needed in the collector circuit to achieve adequate decoupling. But pulses with rise times of the order of tens of nanoseconds can be obtained with devices that deliver up to several hundred watts of power over bandwidths of at least 20 to 30%.

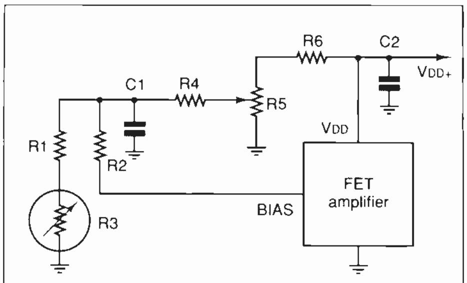


Fig. 4. Mosfet bias circuit using a thermistor-resistor network for temperature compensation.

vidually checked for temperature tracking. Some manufacturers such as Motorola supply rf power fets with specified ranges of g_{fs} matching.

A closed loop system for mosfet biasing (**Fig. 5**) could provide an automatic and precise temperature compensation for any mosfet regardless of electrical size and g_{fs}.

No temperature sensing elements need be connected to the heat sink or to the device housing. In fact, fets with different gate threshold voltages can be changed in the amplifier without affecting the idle current, so the gate threshold voltage can vary within wide limits over a short or long time for a variety of reasons.

Other factors affecting $V_{gs(th)}$, in addition to temperature, might be moisture levels and atmospheric pressure.

Principle of automatic bias

In the automatic bias circuit, the idle current of the mosfet amplifier is initially set to class A, AB, or anywhere in between these bias limits by R_7 – which also provides a stable voltage reference to the negative input of the operational amplifier U_1.

Current flows through R_1 with a consequent voltage generated across it. The voltage is fed to the positive input of U_1, resulting in the output of U_1 following it in polarity but not in amplitude. Due to the voltage gain in U_1, which operates in a dc open loop mode, its output voltage excursions are much higher than those generated across R_1. So if the current through R_1 tends to increase for any reason, part of the output voltage of U_1 fed to the amplifier gate bias input will adjust to a lower level, holding the current through R_1 at its original value. A similar self adjustment will also take place in the opposite direction.

Values for the resistive voltage divider $R_{4\text{-}5}$ have been selected for a suitable range, sufficient to control the amplifier fet gate with the full voltage swing at the output of U_1.

When the amplifier is rf driven, the current through R_1 increases and the bias voltage to the amplifier tends to decrease along with the voltage to the positive input of U_1.

But at the same time, Tr_1 will start conducting, lowering the effective value of R_1 since Tr_1 is in parallel with it. The turn-on gate voltage for Tr_1 is obtained from the voltage drop across R_2.

Typical values for R_1 are 5-10Ω and for R_2, 0.1-0.2Ω. The values must be selected on the characteristics of Tr_1, the exact application and the currents in question.

The higher the current drawn by the amplifier, the harder will Tr_1 be turned on. For example if R_1 is 5Ω and Tr_1 is fully turned on with its $r_{DS(on)}$ of 0.2Ω, the effective value of R_1 will vary between 5Ω and less than 0.2Ω, depending on the current drawn. So the current variable resistor (Tr_1-R_1) makes it possible to keep the output of U_1 and the resulting amplifier bias voltage relatively stable under varying current conditions.

The circuit is ideal for class A amplifiers, where the drain current remains constant regardless of the rf drive. Tr_1, R_2 and R_3 can be omitted for class A amplifiers and the value of R_1 can be made as low as 0.05-0.1Ω. ■

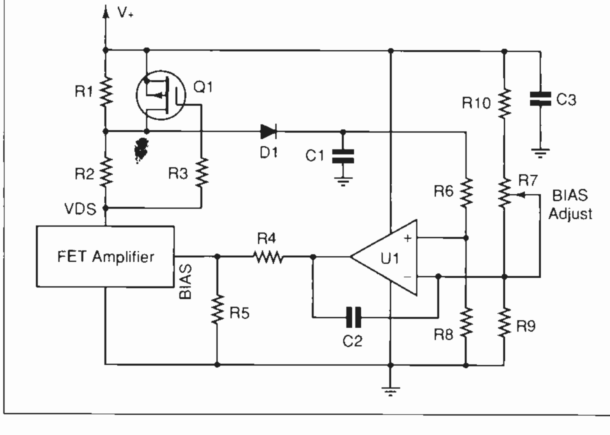


Fig. 5. Automatic bias tracking system for mosfet power amplifiers. It provides automatic temperature compensation without sensors as well as versatility for substituting a variety of electrical sizes of fets operating at any supply voltage.

CLASSIFIED

TEL 081 652 8339 **FAX 081 652 8931**

ARTICLES FOR SALE

Cooke International

SUPPLIER OF QUALITY USED TEST INSTRUMENTS

ANALYSERS, BRIDGES, CALIBRATORS, VOLTMETERS, GENERATORS, OSCILLOSCOPES, POWER METERS, ETC. ***ALWAYS AVAILABLE***

SPECIALIST REPAIR WORK & CALIBRATION UNDERTAKEN

ORIGINAL SERVICE MANUALS FOR SALE
COPY SERVICE ALSO AVAILABLE

EXPORT, TRADE AND U.K. ENQUIRIES WELCOME, SEND LARGE "A3" S.A.E. + 50P POSTAGE FOR LISTS OF EQUIPMENT AND MANUALS.

ALL PRICES EXCLUDE VAT AND CARRIAGE
DISCOUNT FOR BULK ORDERS *SHIPPING ARRANGED*

OPEN MONDAY-FRIDAY 9AM-5PM

Cooke International

ELECTRONIC TEST & MEASURING INSTRUMENTS
Unit Four, Fordingbridge Site, Main Road, Barnham, Bognor Regis, West Sussex, PO22 0EB
Tel: (+44) 0243 545111/2 Fax: (+44) 0243 542457

HIGH END TEST & COMMUNICATIONS EQUIPMENT PURCHASED

CIRCLE NO. 133 ON REPLY CARD

TURN YOUR SURPLUS TRANSISTORS, ICS ETC, INTO CASH
Immediate settlement.
We also welcome the opportunity to quote for complete factory clearance.
Contact:
COLES-HARDING & CO. 103 South Brink Wisbech, Cambs PE14 0RJ.
ESTABLISHED OVER 15 YEARS
Buyers of Surplus Inventory
Tel: 0945 584188 Fax: 0945 475216

CIRCUIT BOARDS

TELEKINETIKS
RS232 CONTROLLED I/O FOR IBM COMPATIBLES
INEXPENSIVE DATA ACQUISITION AND CONTROL
Single board kit comprises:
Populated PCB with prototyping area applications with source code in Turbo Pascal include
A/D Conversion, D/A Conversion Stepper Motor Control, VFET Applications Waveform Synthesis
Price of kit including documentation £100 (all inclusive) Allow 21 days for delivery
Sole Proprietor: B. Teleki
(make cheques payable to B. Teleki)
8 Victoria Court, Brampton Road, Maybank, Newcastle-Under-Lyme, Staffs ST5 0SL.
Telephone (0782) 662099

FREE CLASSIFIED

SELLING NEW PORTABLE CB trasceiver eurosonic FM, P.L.L. 40 channels 4/1 watts, UK-model, ultra-slim (7.24H x 2.76W x 1.73D inch) squelch/tone control, jack for external micro/earphone/ power supply. BNC antenna connector, rubber antenna: Tel: 01043/512/260371 (after 8pm, please ask for Christoph).

FOR SALE HP 8556A HP8553B plug-in units for 141T O.I.R.O. £550. Tel: 0442 842457 wanted parabolic antenna and feed > 1.2m good for 12G.

WHY NOT FAX YOUR ADVERT ON 081 652 8931

COMPELEC
1994 BARGAIN LIST
THE ELECTRONIC ESSENTIAL, MILLIONS OF COMPONENTS AT UNBEATABLE PRICES.
FOR A FREE COPY PHONE, FAX, OR WRITE (TRADE ONLY)
COMPELEC
14 CONSTABLE ROAD, ST. IVES, HUNTINGDON, CAMBS. PR17 6EQ
0480 300819
WE ALSO PURCHASE ALL TYPES OF ELECTRONIC COMPONENTS. LISTS TO ABOVE ADDRESS.

ARTICLES WANTED

WANTED
Test equipment, receivers, valves, transmitters, components, cable and electronic scrap and quantity. Prompt service and cash.
M & B RADIO
86 Bishopgate Street, Leeds LS1 4BB
Tel: 0532 435649
Fax: 0532 426881 3956

STEWART OF READING
110 WYKEHAM ROAD, READING, RG6 1PL.
TEL. 0734 268041
FAX: 0734 351696
TOP PRICES PAID FOR ALL TYPES OF SURPLUS TEST EQUIPMENT, COMPUTER EQUIPMENT, COMPONENTS, etc. ANY QUANITYT 103

WANTED
Receivers, Transmitters, Test Equipment, Components, Cable and Electronic, Scrap. Boxes, PCB's, Plugs and Sockets, Computers, Edge Connectors.
TOP PRICES PAID FOR ALL TYPES OF ELECTRONICS EQUIPMENT
A.R. Sinclair, Electronics, Stockholders, 2 Normans Lane, Rabley Heath, Welwyn, Herts AL6 9TQ. Telephone: 0438 812 193. Mobile. 0860 214302. Fax: 0438 812 387 Telephone: 0763 246939 780

WANTED
High-end Test Equipment, only brand names as Hewlett-Packard, Tektronix, Rhode & Schwarz, Marconi etc. Top prices paid.
Please send or fax your offer to:
HTB ELEKTRONIK
Alter Apeler Weg 5, 2858 Schiffdorf, West Germany
TEL: 01049 4706 7044
FAX: 01049 4706 7049

ARTICLES FOR SALE

NEW VMEDSP MPV901 Boxed. Offers: 0703 733312.

WANTED VALVES especially KT66, KY88, PX4, PX25 (also transistors IC's capacitors, valve radios/hi-fi). If possible send written list for offer by return to Billington Export, 1E Gilmans Ind Estate, Billinghurst, Sussex, RH14 9EZ. Tel 0403 784961. Fax 0403 783519.

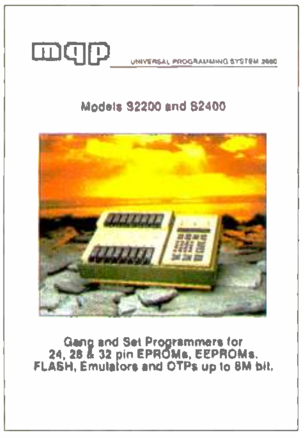
WE WANT TO BUY !!

IN VIEW OF THE EXREMELY RAPID CHANGE TAKING PLACE IN THE ELECTRONICS INDUSTRY, LARGE QUANTITIES OF COMPONENTS BECOME REDUNDANT. WE ARE CASH PURCHASERS OF SUCH MATERIALS AND WOULD APPRECIATE A TELEPHONE CALL OR A LIST IF AVAILABLE. WE PAY TOP PRICES AND COLLECT.
R.HENSON LTD.
21 Lodge Lane, N.Finchley, London N12 8JG.
5 Mins, from Tally Ho Corner.
TELEPHONE 081-445-2713/0749
FAX 081-445-5702.

ELECTRONIC UPDATE

Contact Pat Bunce on
081-652 8339

A regular advertising feature enabling readers to obtain more information on companies' products or services.



The system 2000 is an ideal programmer for the production environment. Fast programming results in high throughput and rigorous verification leads to improved quality control. Single key functions and checks against misoperation facilitates its use by unskilled staff

MQP ELECTRONICS LTD.
Tel: 0666 825146
Fax: 0666 825141

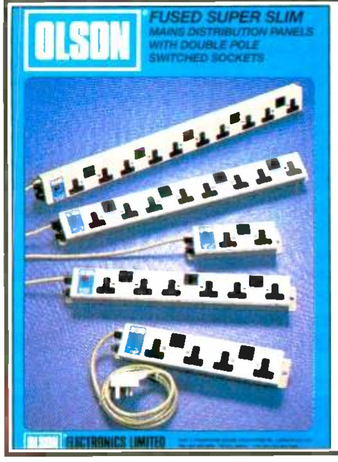
CIRCLE NO. 142 ON REPLY CARD

FREE VXI BROCHURE

The National Instruments VXI brochure describes the company's embedded PC and GPIB controllers, MXIbus interface kits for multiple platforms, and NI-VXI, LabWindows, and LabVIEW software for developing and controlling VXI instrumentation systems.

NATIONAL INSTRUMENTS
Tel: 0800 289877

CIRCLE NO. 146 ON REPLY CARD



OLSON ELECTRONICS LIMITED is a leading manufacturer in the field of mains distribution panels of every shape and size to suit a variety of needs. For use in Broadcasting, Computing, Data Communications, Defence, Education, Finance, Health etc. All panels are manufactured to BS5733. BRITISH AMERICAN, FRENCH, GERMAN CEE22/IEC and many other sockets. Most countries catered for.

All panels are available ex-stock and can be bought direct from OLSON.

Olson Electronics Limited
Tel: 081 885 2884
Fax: 081 885 2496

CIRCLE NO. 143 ON REPLY CARD

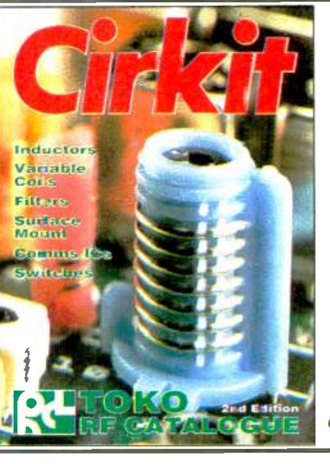
DATAUPDATE is Electronics Weekly's section for advertisers to market their product information. From catalogues to newsletters Data Update is designed to present your product information in a clear and attractive manner, whilst our colour coded enquiry numbers help readers to obtain the information they need fast.

CIRCLE NO. 147 ON REPLY CARD

IR Group, Europe's leading supplier of used instrumentation, has published the latest update of models available. With a range from power supplies to network analysers, most items are available on short delivery and come with a 12 month parts and labour warranty. For a detailed quotation call

0753 670000.

CIRCLE NO. 144 ON REPLY CARD



2nd EDITION TOKO RF CATALOGUE

Cirkit have just published the 2nd Edition of the Toko RF Catalogue, featuring details of Tokos' extensive range of RF coils, inductors, filters and comms ICs.

The 128 page catalogue includes many new products such as; Surface mount high current inductors, surface mount multilayer inductors, helical filters at 2.5GHz and a new section of push button and tact switches.

Cirkit Distribution Ltd, Park Lane, Broxbourne, Herts, EN10 7NQ
Tel: (0992) 441306
Fax: (0992) 441306

CIRCLE NO. 148 ON REPLY CARD

ELECTRONIC UPDATE is Electronic World and Wireless World's section for advertisers to market their product information. From catalogues to newsletters, Data Update is designed to present your product information in a clear and attractive manner while our "CIRCLE NUMBERS" help readers to obtain the information they need fast.

CIRCLE NO. 145 ON REPLY CARD

High Speed EPROM & FLASH Programming from your PC

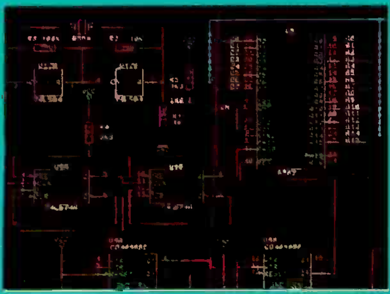
- ☐ Programs EPROMs to 4 Mbits/ 32-pins
- ☐ Superfast 8, 16 & 32-bit programming
- ☐ Approved algorithms
- ☐ Menu driven software included
- ☐ Sophisticated editor functions
- ☐ Easy file management
- ☐ FREE demo disk available

Stag Programmers Limited
Martinfield Welwyn Garden City, Hertfordshire, AL7 1JT UK
Tel: (0707) 332148
Fax: (0707) 371503

CIRCLE NO. 149 ON REPLY CARD

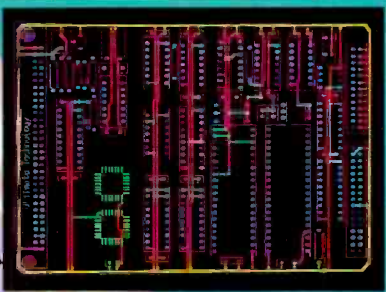
INDEX TO ADVERTISERS

FROM CONCEPT TO ARTWORK IN 1 DAY

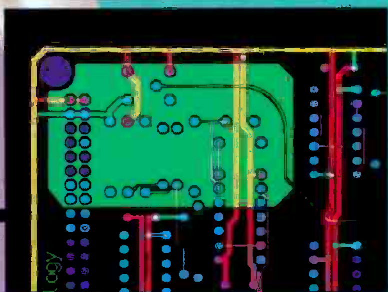


Your design ideas are quickly captured using the ULTIcap schematic design Tool. ULTIcap uses REAL-TIME checks to prevent logic errors. Schematic editing is painless; simply click your start and end points and ULTIcap automatically wires them for you. ULTIcap's auto snap to pin and auto junction features ensure your netlist is complete, thereby relieving you of tedious netlist checking.

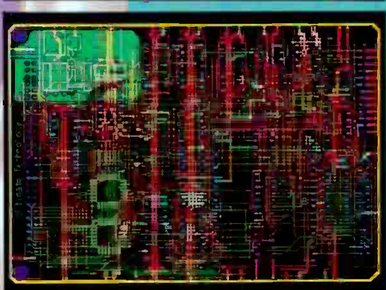
ULTIshell, the integrated user interface, makes sure all your design information is transferred correctly from ULTIcap to ULTIboard. Good manual placement tools are vital to the progress of your design, therefore ULTIboard gives you a powerful suite of REAL-TIME functions such as, FORCE VECTORS, RATS NEST RECONNECT and DENSITY HISTOGRAMS. Pin and gate swapping allows you to further optimise your layout.



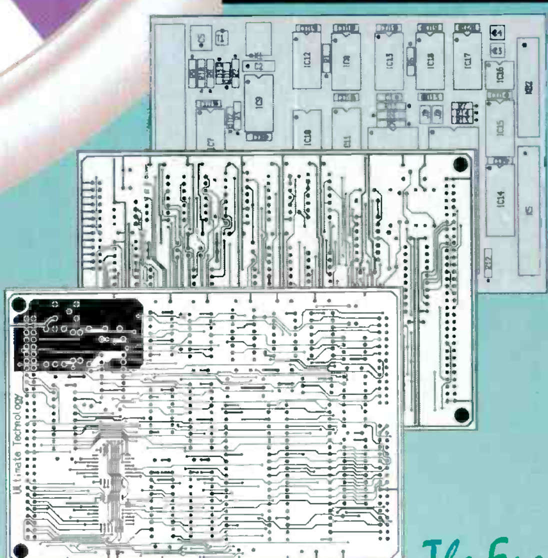
Now you can quickly route your critical tracks. ULTIboard's REAL-TIME DESIGN RULE CHECK will not allow you to make illegal connections or violate your design rules. ULTIboard's powerful TRACE SHOVE, and REROUTE-WHILE-MOVE algorithms guarantee that any manual track editing is flawless. Blind and buried vias and surface mount designs are fully supported.



If you need partial ground planes, then with the Dos extended board systems you can automatically create copper polygons simply by drawing the outline. The polygon is then filled with copper of the desired net, all correct pins are connected to the polygon with thermal relief connections and user defined gaps are respected around all other pads and tracks.



ULTIboard's autorouter allows you to control which parts of your board are autorouted, either selected nets, or a component, or a window of the board, or the whole board. ULTIboard's intelligent router uses copper sharing techniques to minimise route lengths. Automatic via minimisation reduces the number of vias to decrease production costs. The autorouter will handle up to 32 layers, as well as single sided routing.



ULTIboard's backannotation automatically updates your ULTIcap schematic with any pin and gate swaps or component renumbering. Finally, your design is post processed to generate pen / photo plots, dot matrix/laser or postscript prints and custom drill files.

CIRCLE NO. 100 ON REPLY CARD

NEW

ULTIboard/ULTIcap evaluation system:
- all features of the bigger versions
- full set of manuals
- design capacity 350 pins

Price incl. S & H, excl. VAT: **£ 75**

Purchase price is 100% credited when upgrading to a bigger version. • *Also suitable for study & hobby*

ULTIboard PCB Design/ULTIcap Schematic Design Systems are available in low-cost DOS versions, fully compatible with and upgradable to the 16 and 32 bit DOS-extended and UNIX versions, featuring unlimited design capacity.

The European quality alternative

ULTICAP + ULTIBOARD = MAXIMUM PRODUCTIVITY

ULTImate Technology UK Ltd. • 2 Bacchus House, Calleva Park, Aldermaston Berkshire RG7 4QW • Fax: 0734 - 815323 • Phone: 0734 - 812030

4 Mbit upgrade only £95! (plus VAT)

The legendary S4 - the smallest, most powerful personal programmer you can buy - and only £495!

Plus V.A.T.

From engine management to Antarctic survey teams, you can find S4s the world over, up and running where the competition is left far behind. S4 gets the job done in every situation you might expect - and quite a few you wouldn't!

Size:
186 x 111 x 46mm
Weight: 515g

S4 CAPABILITIES

A 32 pin ZIF socket programs a huge library of EPROMs, EEPROMs and FLASH devices. Dataman S4 programs devices up to 8Mbits and the unique, loadable Library means that new parts can be added quickly *without extra cost!* Serial EPROMs, 40 pin EPROMs and micro-controllers are all supported with optional modules.

S4 EMULATION

With Dataman S4's powerful emulation system you see your code running *before* committing yourself to an EPROM. Simply download your code to S4, press 'EMULATE', and your target system runs, *in real time*, as if an EPROM was plugged in.

S4 REMOTE CONTROL

Dataman S4 has its own internal processor and memory, but can also be operated remotely from your PC at speeds up to 115,200 Baud. S4 is supplied with a free disk containing custom terminal software and a pop-up TSR communications utility.

S4 - THE PACKAGE

Dataman S4 is shipped *ready to use*, complete with a mains charger, emulation lead, write lead, personal organiser/instruction manual, MS-DOS communications software, spare Library ROM - and a 3 year guarantee.

DATAMAN

Dataman Programmers Ltd

Credit Card Hotline:

(0300) 320719

for same-day dispatch

Access · MasterCard · VISA

Station Road, Maiden Newton, Dorset, DT2 0AE, UK. Telephone: 0300 320719 · Fax: 0300 321012 · Telex: 418442 · BBS: 0300 321095 24hr · Modem: V32bis/16.8K HST.
22 Lake Beauty Drive, Suite 101, Orlando, FL 32806, USA. Telephone: (407) 649-3335 · Fax: (407) 649-3310 · BBS: (407) 649-3159 24hr · Modem: V32bis/16.8K HST.

CIRCLE NO. 131 ON REPLY CARD