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May 1994 £2.25

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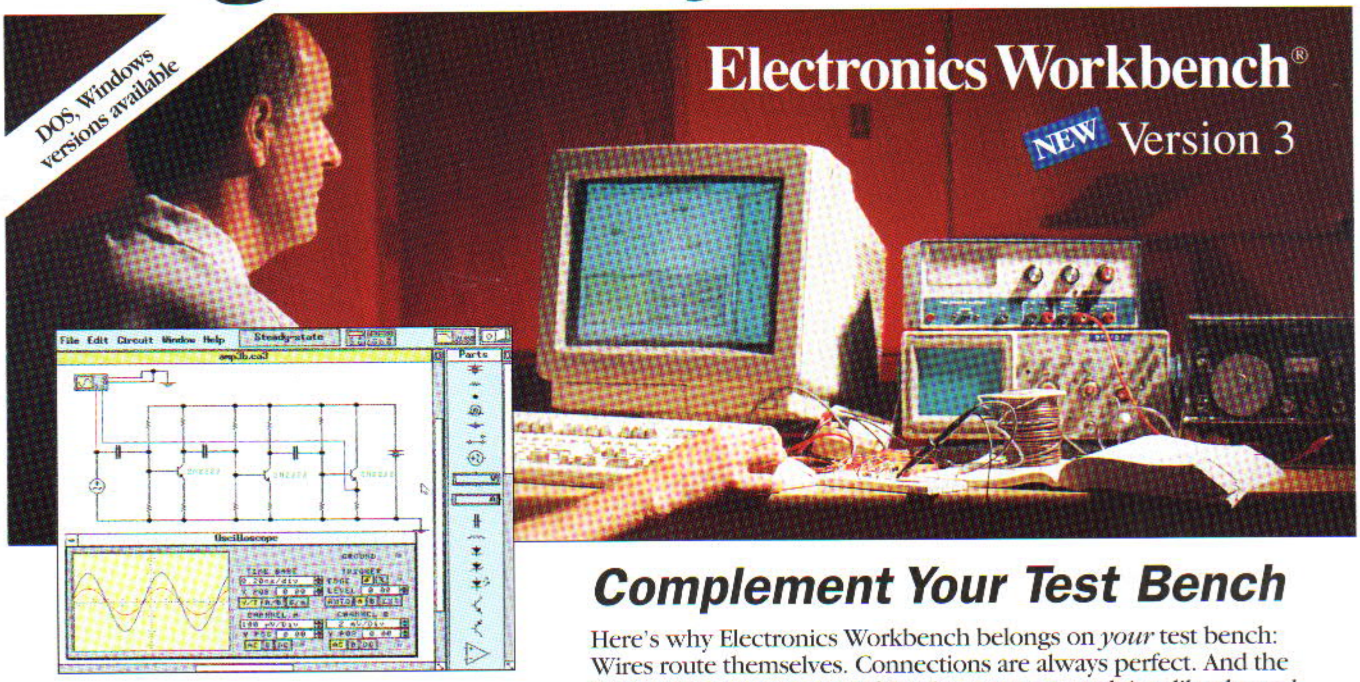

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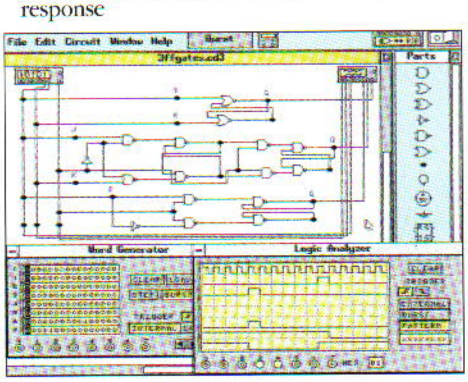

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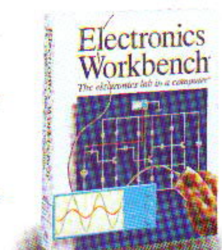

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INTERACTIVE

ELECTRONICS in ACTION

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The musical sound of technology

Thank you for your comments regarding the magazine. I know you have been telling friends about us and what a refreshing change our mag is. Do let us know if there any newsagents where we cannot be found.

There continues to be a growing interest in Electronic Music Technology as those colleges and universities already running courses will verify. You will see amongst these pages another college offering such a course. It is good to see that this branch of electronics education is quickly expanding. The means to explore musical creativity using electronic instruments and computers has never been greater and is within easy reach financially to many people. It is right therefore that the best way to get maximum benefit from such highly complex technology is to use the facilities and guidance of a college course. Keep an eye out in future editions of this mag for the college that might be able to help you.

Finally we hope this special audio edition will help those at home, at college or university to enjoy the fruits of audio electronics and music.

Paul Freeman-Sear

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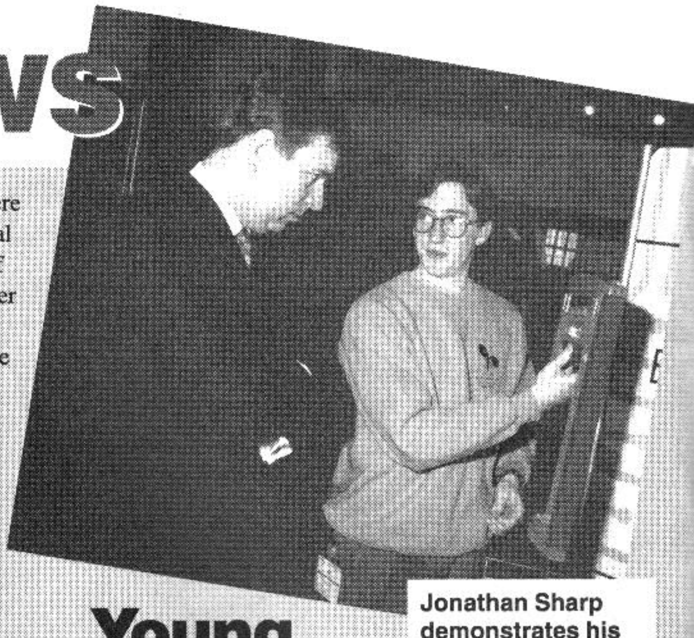
NEWS



Jonathan Sharp demonstrates his award winning project to the Duke of York

Young Electronic Designer Awards

This years Awards were presented by His Royal Highness The Duke of York at a special dinner for finalists at the Science Museum at the end of March.

This is the ninth annual occasion that the Awards have been made for young minds with a talent for electronic design. The scheme was set up to help bridge the gap between the two different worlds of science and commerce where very often few clever inventions ever leave the laboratory bench owing to poor marketing and packaging.

The Young Electronic Designer Awards scheme is open to students in secondary schools, colleges and universities between the ages of 12 and 25, and challenges young designers to invent and produce an electronic device that meets an everyday need.

The event is organised by the YEDA Trust, a registered charity, and sponsored by Texas Instruments Ltd. and Mercury Communications Ltd.

For further information on how to enter for next year contact:

The YEDA Trust, 24 London Road, Horsham, West Sussex, RH12 1AY Tel: (0403) 211048 Fax: (0403) 210770

The following is a list of this years finalists and the winners together with their ideas.

SENIOR CATEGORY

Jonathan Sharp (23) **1st Prize Winner**
Brunel University, Egham, Surrey
A rechargeable electronic foot measuring device for use in shoe shops.

Samantha Haines (19)
Cheltenham College, Cheltenham, Glos.
A physiotherapy aid indicating weight distribution between the feet of stroke patients and others with sporting injuries.

Richard Mead (18)
Cheltenham College, Cheltenham, Glos.
A built-in device to discourage the theft of electrical appliances.

New PCMCIA card in Europe

IBM has extended its family of PCMCIA (Personal Computer Memory Card Interface Association) cards to the European market. These include Infrared Wireless Adapters, an audio adapter, a Data/Fax Modem card and an RS232 Adapter based on the PCMCIA standard. A PCMCIA card makes a PC more flexible. Originally designed for the mobile computing market where low power consumption and small size are critical, these credit card-sized cards can make it easier to change the configuration of any type of PC and use it as a communications, I/O, storage and multimedia device.

IBM offers three infrared wireless adapters, for PCMCIA and each card includes a transceiver giving mobile users the benefits of a wireless Local Area Network (LAN).

The adapters transfer data at 1Megabit/sec and can be used with new or existing wired LANs such as Ethernet or LANs running Novell's Open Data Link Interface.

It is a non-line of sight adapter, which means that transceivers do not have to be aimed at each other or at a fixed position and will have full communication in a 30' x 30' room.

Kevin Noakes (22)
The Royal Naval Engineering College, Plymouth
Medical Monitor which automatically records and displays patient data such as temperature and respiration rates.

Caroline Turner (20)
Southampton University, Southampton, Hampshire
A low cost Time Interval Analyser for measuring the quality of recording/replay systems in conjunction with oscilloscopes.

Andrew Sime (18)
Hampton School, Hampton, Middlesex
A microprocessor based digital aquarium controller.

INTERMEDIATE CATEGORY

Gareth Sylvester-Bradley (17)
Bryanston School, Blandford, Dorset
A hand-held digital anemometer.

Nichola Hirschmann (18)
Woldingham School, Woldingham, Surrey
A project to develop a hardware solution to sequential computing to facilitate faster data processing speeds.

Graham Reith (17)
Merchiston Castle School, Edinburgh
The Intelligent Drawer, a system for cashiers at check-out tills enabling the correct change to be dispensed involving the fewest number of coins.

Neil Gray (16) & **Roberto Tyley** (15)
Blyth Ridley County High School, Blyth, Northumberland
A hazard detection and indication system to help with the safe evacuation of a building in the event of fire or other emergency.

Stephen Sinfield, **Angela Assorto**, **Alasair Mitchell** & **Avi Bhattacharyya** (All 17)
1st Prize Winners
St Albans School, St Albans, Hertfordshire
'Centranal' - an electronic corrosion tester for central heating systems.

Anthony Powell (15)
Chigwell School, Chigwell, Essex
An electronic timer mechanism to open/close doors at pre-set intervals.

JUNIOR CATEGORY

Oliver Webster & Fraser Prudhoe (both 13)
Amble Middle School, Amble, Northumberland
An in-car alarm which is activated when a motorist exceeds a pre-set speed limit in fog.

Jonathan Moodie & **Jeremy Brettell** (both 14)
1st Prize Winners
Cheltenham College, Cheltenham, Glos.
'Hypofill', a device which recharges hyperdermic syringes with the required dose of insulin for diabetics with impaired vision.

Claire Athey (14)
Bancroft's School, Woodford Green, Essex
An electronic learning device involving visual association.

Louise Valentine & **Helen Berry** (both 14)
Cardinal Newman School, Luton, Bedfordshire
A pocket-sized carbon monoxide warning device for cyclists and other road users.

Alistair Phipps (15)
Merchiston Castle School, Edinburgh
JIVE (Junior Interactive Visual Educator) a computer based system to develop key learning skills in pre and primary school children.

Rebecca Salmen & **Alys Patterson** (both 14)
King Edward VII School, Sheffield
'Number Spin', a device which randomly selects and displays numbers for children with impaired vision.

1st prize winners in the junior category, Jeremy Brettell and Jonathan Moodie show their 'Hypofill' to the Duke of York

An infrared wireless LAN allows users to share resources such as printers, ad hoc networks and disk drives by infrared-enabled mobile computer networks. For those wanting to take electronic mail and files away from their offices, wireless LANs provide a convenient way of uploading and downloading files from the desktop or even LAN-attached servers. They also allow multiple mobile users to share the same desktop printers and LAN access point.

The IBM Infrared Wireless Adapter is priced at around US$150.

The Audio Adapter Card expands multimedia opportunities for portable computing by adding high-definition 16-bit stereo sound to many software applications.

The audio adapter consists of a PCMCIA Type II card connected to a small, detachable, tethered Audio Interface Module. The module has a built-in mono microphone, stereo line-internal stereo microphone in-jack and a stereo-out/headphone jack. The audio adapter features a high quality 16-bit stereo CODEC to record and play back audio signals. The card can record and play back upto a sampling frequency of 48KHz, using 8 or 16bit resolution in stereo or mono form.

The audio adapter will be shipped with drivers for DOS and OS/2 operating systems that are compliant with PCMCIA Standard Version 2. A driver for Windows 3.1 that consists of a point enabler for Intel 82365SL chipsets is available with the initial release of the product. IBM software is packaged with the adapter to add a full range of audio functions, including text to speech translation, internal MIDI synthesis capabilities and .WAV recording and playback.

In DOS, OS/2 and Windows environments, the audio adapter card allows a full range of programs to be run, including: IBM's Linkway Live V1.0 and StoryBoard Live V2.0 for DOS; Microsoft Video for Windows; Lotus Freelance Graphics for Windows

2.0 and WordPerfect Presentation for Windows 2.0 in the Windows environment.

The 14.4/14.4 Data/Fax Modem allows users to communicate at data rates up to 57.6Kbps (with V.42bis data compression) and fax rates upto 14.4Kbps. It also supports AT and Fax Group 3, Class 1 and 2 commands used by most third party fax and data software. A lowerspeed, lower-cost 2.4/9.6Kbps data/fax modem will be available in the second quarter of the year.

BT Brings Information Superhighways closer

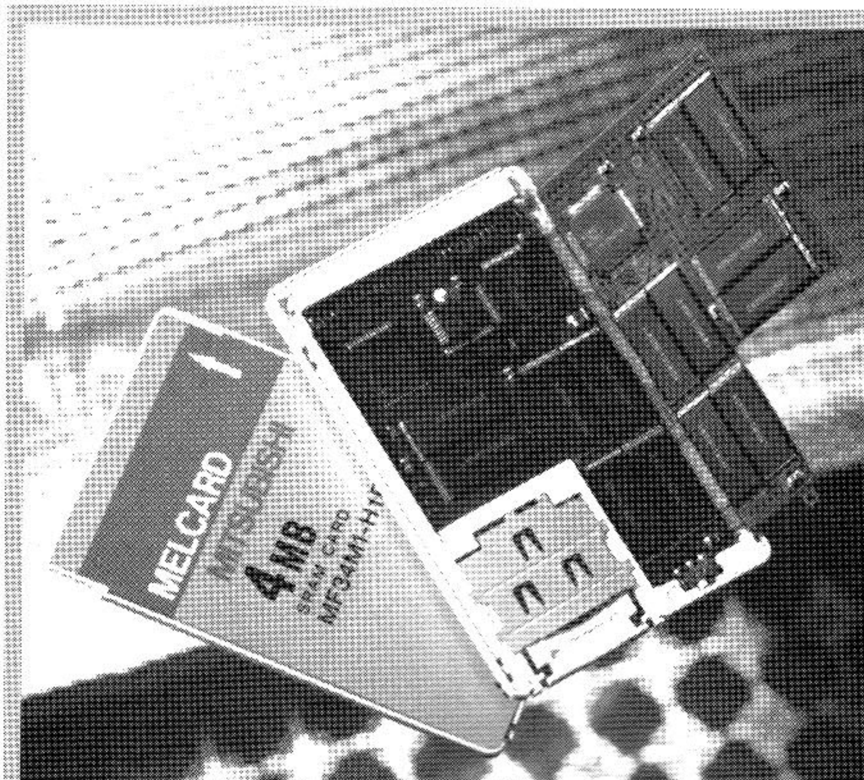
BT has taken another step forward in bringing the information superhighway nearer to reality. They have produced a device at its Martlesham Heath Laboratories, which will increase the capacity of current optical fibre links from around 1,900 simultaneous phone conversations to a high of 78,000. The technology also has the potential to handle 150,000 calls over 150km of continuous fibre optic cable.

The new product is expected to play a major part in tomorrow's information superhighways in allowing existing 1.3 micron fibre optics - of which there are more than three million kilometres in BT's UK network - to achieve far greater data rates over much longer transmission distances.

The superhighways could carry a mixture of analogue and both compressed and uncompressed digital signals to make it technically possible to deliver such services as three-dimensional television, interactive multimedia services (sometimes referred to as video on demand), high quality audio and computer data to the office or home.

Previously, BT researchers have been involved in the development of 1.5 micron amplifiers, which led to improvements in capacities in the longer wavelength 'window'. The current breakthrough increases the signal strength by 1,000 times at 1.3 microns in the region of the spectrum where pulse distortion is about 10 times lower.

The new optical amplifier is compatible with the existing optical fibre infrastructure and will allow the full bandwidth of BT's existing fibre optic investment to be used.



World's Thinnest IC Packaging Technology

The world's thinnest IC packaging technology is now in mass production. Dual Tape Carrier Packaging or DTP as it is known, is only 0.5mm thick and has been designed to make memory cards just 3.3mm thick with capacities greater than hard disk drives.

Mitsubishi has created DTP by making a variety of improvements particularly in moulding. The first memory cards produced incorporating the new process were the company's 4Mbyte SRAM cards called MelCards.

Hard disk drives that are currently being built into the latest notebook PCs in Japan have memory capacities up to 40Mbytes. DTP based memory cards, however can more than double that capacity and measure just 3.3 x 54 x 85.6mm. Furthermore, through the use of ICs, access times are considerably faster than can be achieved by magnetic media.

The benefits of solid state memory cards over conventional rotating media such as hard disk drives is that they provide long term reliability and have the major advantage of having no internal moving parts, and therefore nothing to wear out.

The full 20 bit treatment

Independent classical recording company, Modus Music has acquired two Nagra D recorders to further the company's commitment to 20 bit recording.

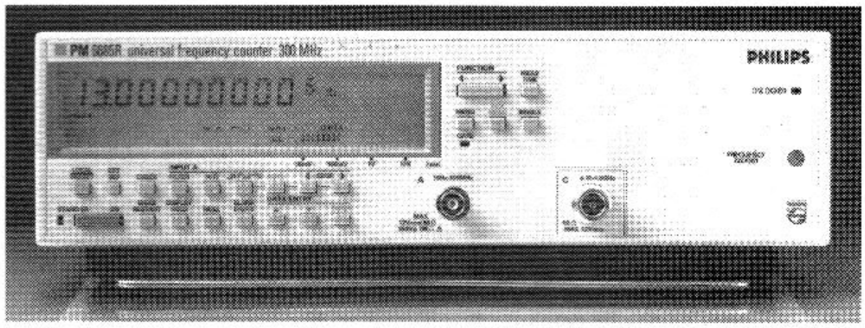
The company which produces recordings for labels as diverse as EMI, Chandos, Conifer, ASV, BMG and even Sony in Japan, has been able to offer a 20 bit recording and editing capability since November last year. With the purchase of the two Nagras, equipped with on-board 20 bit A to D converters, the company now has five 20 bit capable recorders in the field.

"The engineering quality is first class," confirms producer Tryggvi Tryggvason. "We often use two machines simultaneously for security, along with at least one time coded DAT so everything is in triplicate."

"Usually we record in stereo but having the facility for multitracking is very useful on a few occasions, where we have an organ that we require to overdub for instance. We have already used the four track facility in order to keep voices separate and not commit them to the mix until later. We can of course synchronize the two machines to provide us with an eight track system".

Modus operates four editing suites at its Hayes headquarters, with a choice of hard disk editing systems for 20 bit mastering. The label also maintains an almost permanent presence at All Saints Parish Church in Petersham, making extensive use of "one of the very best acoustics, anywhere".

Tryggvason sees the time when demand will dictate that all of Modus's work be in the 20 bit domain, as being in the near rather than the distant future. That of course will be good news for the CD listening public and good news for the Nagra D.



Atomic Accuracy

Fluke has increased the accuracy of its portable frequency counter/calibrators with the introduction of the PM6685 Rubidium. The built-in rubidium atomic clock gives a 10-digit display and an accuracy of 2.5 x 10-10 relative measuring resolution in a second, making this gadget the most accurate counter on the market.

The rubidium reference provides very high stability, even under severe operating conditions. The rubidium atomic resonance principle is intrinsically 100 times more stable than conventional electromechanical crystal resonators, which makes the use of a rubidium reference the right choice for field measurements and calibrations demanding accuracies better than one part in 10^{-8}, or where frequent calibration of the counter is not possible.

Warm-up time is within 6 minutes for a 1 x 10^{-9} accuracy.

Frequency options up to 4.5GHz.

Control Network Research

The Cybernetics Department of the University of Reading has launched a three year S.E.R.C. (Science and Engineering Research Council) funded project to demonstrate the benefits of applying Echelon's LONWORKS control network architecture to process control applications.

The project has been set up in collaboration with Quad Europe who manufacturer soldering machines, Toshiba who make the Neuron chip for Echelon, PAA Ltd, an independent consultancy and Echelon, developers of the LONWORKS architecture for linking Neuron-based control systems.

"The aim of the project is to show how distributed process control can bring benefits to a real life application," remarked Kevin Warwick, Head of Cybernetics at the University of Reading. " By distributing intelligence, wiring costs are decimated, the network stays up and running when one component fails and time-saving parallel processing becomes a reality," he said.

In three years, the University aims to develop a process control system suitable for commercial development.

Devices can be connected using any type of cable - twisted pair, power mains, optical fibre or RF. A mixture of these can be supported through routers.

The network centre is controlled by the Neuron chip, containing three microprocessors, on-board memory and a complete seven-layer open systems protocol. It is fully programmable and can be used in intrinsically safe applications by altering clock speed and power consumption. Neuron chips cost around $10 today and should be $5 within the year.

Musical Instrument Technology

Have you ever considered a career which combines both your interest in electronics and music? If you have then maybe you should be studying for a BTEC National Diploma in Musical Instrument Technology at Newark and Sherwood College, in Nottinghamshire. The College has for a number of years been one of the leading institutions in this specialized field and is able to offer one of only three such Courses in the country.

This exciting course was developed by course tutor Colin Boothman in conjunction with industry and BTEC It lasts for two years and is designed for anyone who wishes to follow a career in sound production or reproduction. Much of the course material is practically based and students get the opportunity to carry out projects such as the construction and testing of amplifiers, effects units and midi keyboards. Time is also spent looking at

electronics theory, the study of acoustics, microelectronics, musicianship and recording techniques. Although the course is -mainly concerned with musical instrument electronics, music

theory and keyboard tuition are also included and the college's 16 track recording studio is available for use. The Programme of study is unit based and because the course leads to a nationally recognised qualification which is equivalent to 'A' levels it's equally appropriate for those wishing to go straight into employment and those who wish to carry on their studies at degree level. Past students have progressed to BEng. and BSc. degree courses in electronics and music electronics.

To gain a place on the course Students are normally required to have 3-4 GCSEs (including maths and science) and a genuine interest in music. Mature students with relevant experience will also be welcomed.

Newark and Sherwood College is based in the historic town of Newark-on-Trent and has gained an international reputation for it's music based courses, attracting students of all ages from all over the world. As well as the National Diploma in Musical Instrument Technology the College also runs courses in subjects as diverse as Music Industry Studies, Piano Tuning and Violin Making.

Applications for the 1994/95 intake of students are currently being considered and if you would like further information about this course please telephone 0636 701411 for a brochure.

Research World

Technical Advances from around the Globe

Making the transition from electron beam X-rays lithography

A US company, Lockheed Corp., expects to be producing 0.15-micron microwave monolithic ICs by 1996 using X-ray lithography. It has already demonstrated 0.25-micron fabrication of MMIC amplifiers as it gears up for volume production of F-22 avionics modules.

Paul Hoff, director of MMIC circuit and module development at the defence contractor's Lockheed Sanders unit said Lockheed is making the transition from electron-beam to X-ray lithography as it moves toward affordable production of 0.25-micron MMICs for next-generation fighter aircraft.

Lockheed opened an automated module fabrication facility at the end of 1993 as the first step in the transition from e-beam to X-ray lithography. Hoff said the contractor expects to demonstrate production of 0.25-micron MMICs in a year.

The high cost of producing MMICs has been a major hurdle to broader military and commercial applications. Hoff said Lockheed's goal is to reduce the cost of high-performance MMICs for the F-22 from $10 to $2 per device.

Lockheed is the U.S. Air Force's prime contractor for the F-22 fighter and is also a member of the Texas Instrument Inc./Raytheon team for the second hardware-development phase of the Pentagon's Microwave/Millimeter-Wave Monolithic Integrated Circuits (Mimic) programme. That effort focuses on reducing MMIC production costs, conducting MMIC systems demonstrations and foundry support.

Using e-beam lithography, Lockheed has demonstrated MMIC technology by reducing the size of such equipment as a custom-made phase shifter to 200 equivalent phase-shifter chips in a 3-in wafer. While e-beam equipment is competitive with X-ray lithography in terms of definable feature size, it writes features directly, rather than patterning an area in a single exposure. The increased time involved makes the process unsuitable for mass production.

Since each F-22 avionics module will contain about 20 MMICs at 0.25-micron feature size, Lockheed has begun the transition to X-ray lithography to produce the high-performance parts affordably. Lockheed Sanders produced its first X-ray-defined MMIC last year and is under contract with the Advanced Research projects Agency (Arpa) for a MMIC manufacturing demonstration using X-ray lithography.

Lockheed currently uses an X-ray stepper developed by Hampshire Instruments, a start-up that has gone out of business since the purchase.

Along with the X-ray source, Lockheed's MMIC foundry includes gear for attaching components using epoxy and solder with 0.5-mil placement accuracy, a robotic wire bonder and a carrier prober. The equipment permits direct attachment of MMICs and ceramic substrates to module housings of aluminium and silicon carbide.

The line has the capacity to assemble similarly to the wideband receive module being developed by Lockheed Sanders for the F-22 electronic countermeasure suite.

"To realize the full potential of MMICs, the cost must drop," Hoff said during a technology briefing recently. "X-ray lithography can realize a 10-20 times throughput advantage over e-beam lithography, with an associated reduction in chip cost."

If the market takes off, high-volume MMIC production could provide a badly needed technology driver for X-ray lithography. The physics of Galliun Arsenide, a semi-insulating material that naturally isolates devices from one another, makes it easy to define small-geometry devices. For example, a metal semiconductor field-effect transistor (MESFET) can be defined entirely with a metal pattern over a doped epitaxial layer. That makes it easy to push galliun arsenide designs below 0.25 micron.

"Galliun arsenide reduces many of the processing steps required in silicon, cutting back on alignment problems," Heaton said. Another advantage is the confinement of the really fine detail to small areas of the circuit. Those considerations apply to the broadband power amplifiers the company plans to produce in volume.

Making photovoltaics more economical

A potentially low-cost solar energy technology which uses solar concentrators and a photovoltaic cell is currently generating 1KW at Georgia Power's Shenandoah Enviromment and Education Centre in Newman.

The systen, which has a measured, stabilized efficiency for converting sunlight in electricity without degradation of 15.5%, has been certified for participation in the Photovoltaics for Utility Scale Applications (PVUSA) project.

Called an integrated high-concentration photovoltaic (IRCPV) array, it consolidates structural, electrical, thermal and optical components into sinple building blocks.

The integrated array is intended to eliminate unnecessary components so construction is simplified, to reduce capital and labour costs.

Central to the IRCPV system is a high-concentration solar cell developed at Stanford University. Sunlight-concentration solar cell systems use smaller area silicon cells than systems without concentrators.

According to EPRI, system efficiency can be raised to 20% by the end of 1994.

Amonix recently demonstrated a stable individual cell efficiency of 24%. Current plans are to build a 20KW unit in 1994.

The company believes that by the end of the decade it will be able to deliver systems based on this technology for $2000/KW or provide power at 5 cents/KWh with a production run of 100MW.

At that price it should become competitive for a wide range of applications including utility scale generation in high sunlight areas.

the Centronic Sound Machine

Sample the better things in life with Dr. Pei An's PC compatible digital sound sampler

Computer technology has developed rapidly in recent years. Nowadays modern computers work at much greater speed than their old generations. Most of them are equipped with VGA (Video Graphic Array) cards and high-resolution colour monitors which allow the computers to produce impressive graphics on screen display. Some computers are installed with sophisticated sound cards which enable the computer to generate complicated

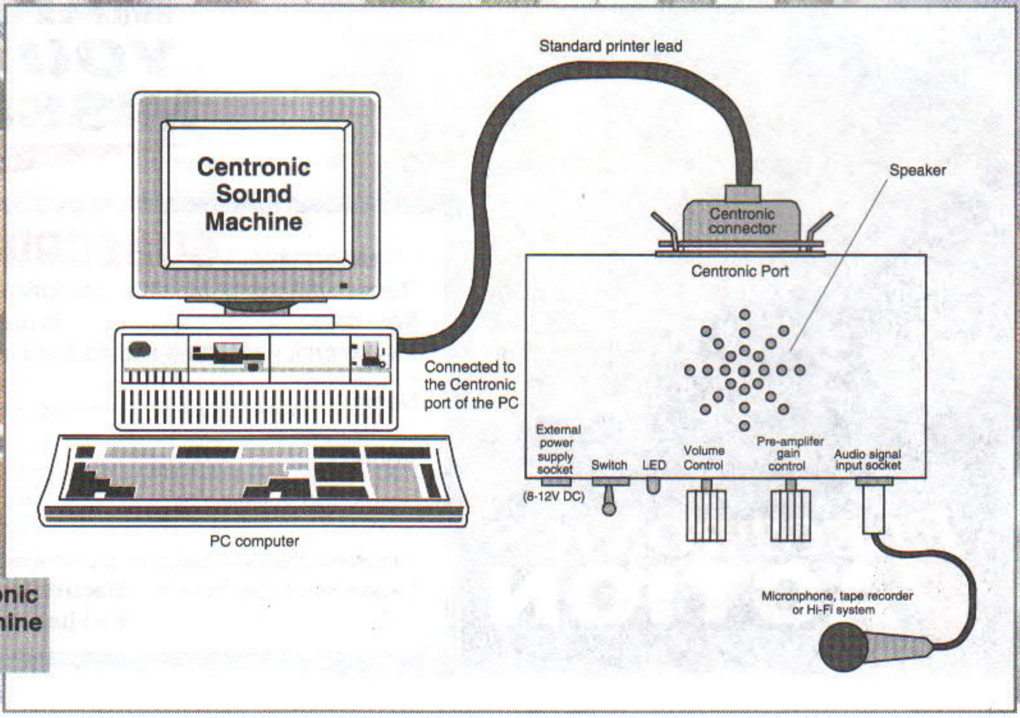


Fig.1 Centronic Sound machine system

sounds such as music and speech.

This is a computer project and is called, for want of a better term, 'the Centronic Sound Machine'. Briefly it is a sound sampling and playing-back device. It allows a computer to digitize sound samples into digital form and to convert the digitized data back to sound. Sounds to be digitized may come from a microphone, a tape recorder or a hi-fi system. This device is supported by several pieces of public domain musical software and is capable of producing high quality musical sounds. It is plugged directly into the PC's printer port and costs less than £30 to build. Figure 1 shows the simple arrangement of the Centronic Sound Machine system.

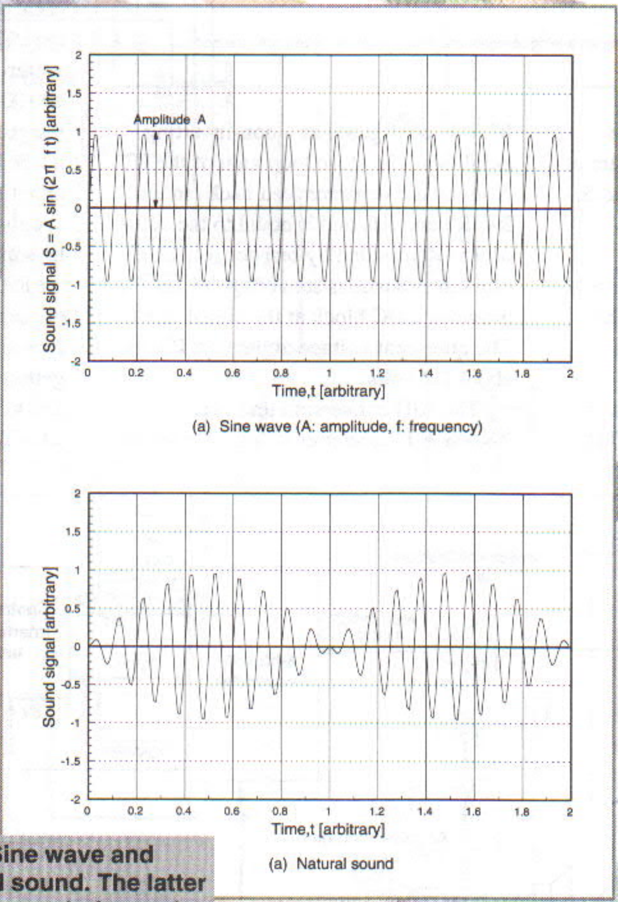
Computer Sound

The simplest form of sound wave is the sine wave. The amplitude (A) and frequency (f) are two parameters determining the characteristics of the sine wave (see Figure 2a). The amplitude represents the loudness of the sound and the frequency determines the pitch. Sounds from the real world, however, are far from sine waves. They are complicated and consist of a number of sine waves (Figure 2b). For example, when a piano playing a note of a basic frequency of 440Hz, it emits the basic sound wave of 440Hz. Meanwhile, it also emits other sounds, also termed as overtones, that occur as multiples of the sound's basic frequency e.g. 880Hz, 1320Hz, 1760Hz and other multiples of 440Hz with each having different amplitudes. The combination of overtones is unique for a particular musical instrument and determines the characteristic of the instrument.

Synthesising natural sounds digitally, mainly uses two methods. The first one is the Frequency Modulation (FM) method and the other one is the Wavetable method. The principle of the FM method is that a sine wave of a basic frequency is modulated by an envelope of a particular shape. The frequency of the sine wave determines the frequency of the sound and the envelope determines the progression of the sound's loudness. In most cases, the envelope is described by four parameters characterising the Attack, Decay, Sustain and the Release (Figure 3a). Sounds from various musical instruments can be synthesized by varying these parameters. The wavetable method is a new technique and has almost replaced the FM synthesis in the professional music industry. It uses actual digitized sound samples to reproduce sound of musical instruments and results in a very lifelike sound effect. The working of this Centronic Sound Machine is based on this method.

Sound digitizing and reproducing

Digitizing, also termed as sampling, is a process in which an analogue sound signal is converted into digital data. This is performed by a device known as the Analogue to Digital Converter (A/D converter). The sound is sampled at regular intervals (Figure 3b) determined by the sampling rate (or sampling frequency). The sampling rate is measured as the number of A/D conversions per second and has a unit of Hz. The quality of the digitized sound increases with the sampling rate. A high sampling rate will result in a digital sample which is truer to the original. However, a high sampling rate also generates massive data.



(a) Sine wave (A: amplitude, f: frequency)

(a) Natural sound

Fig.2 Sine wave and natural sound. The latter is composed of sound waves.

For example, a sampling rate of 4000Hz will generate 4000 digital values per second. With a sampling rate of 44100Hz, 44100 bytes of data are generated. This is obvious that the data will occupy a huge amount of memory space. Therefore, a compromise has to be made between the sampling rate and sample quality. To chose an optimum sampling rate, Shannon's formula is used. According to the formula, to retain all the information of a sound signal to be digitized, the minimum sampling rate should be equal to a frequency which doubles the highest frequency of the sound. Since the highest frequency that our ears can hear is around 20000Hz, a sampling frequency of at least 40000Hz is needed. This is why most audio CD systems utilize a sampling rate of 44100Hz.

Playing-back is a process in which digital sound data is used to reproduce the sound. Digital to Analogue converters are used in this case (Figure 3b). When a digitized sound is played back at the same rate as its sampling rate, It sounds the same as the original. If the playing back rate is lowered, the reproduced sound is slower and lower. On the contrary, if the sample is played back at a higher rate, the sound is faster and higher. This technique is used to create various sound effects. For example, if we sample a sound from a musical instrument playing Concert A (440Hz) with a sampling rate of 10KHz, and then we play back the sample at a rate of 20KHz, the sound of the sample will still have similar characteristics of that instrument but is one octave above the original.

The Works

The centronic sound machine consists of 6 units: the pre-amplification unit, A/D conversion unit, Centronic interfacing unit, D/A conversion unit, audio amplification unit and power supply unit. When digitizing sound, the sound signal from the microphone or from other sources, is amplified by the pre-amp unit. This amplified signal is fed into the A/D converter unit where it is converted into digital data. The data is then read into the computer via the Centronic interfacing unit. To play back digital sounds, the Centronic port sends data to the D/A conversion unit, where sound is reproduced by an audio ampli-

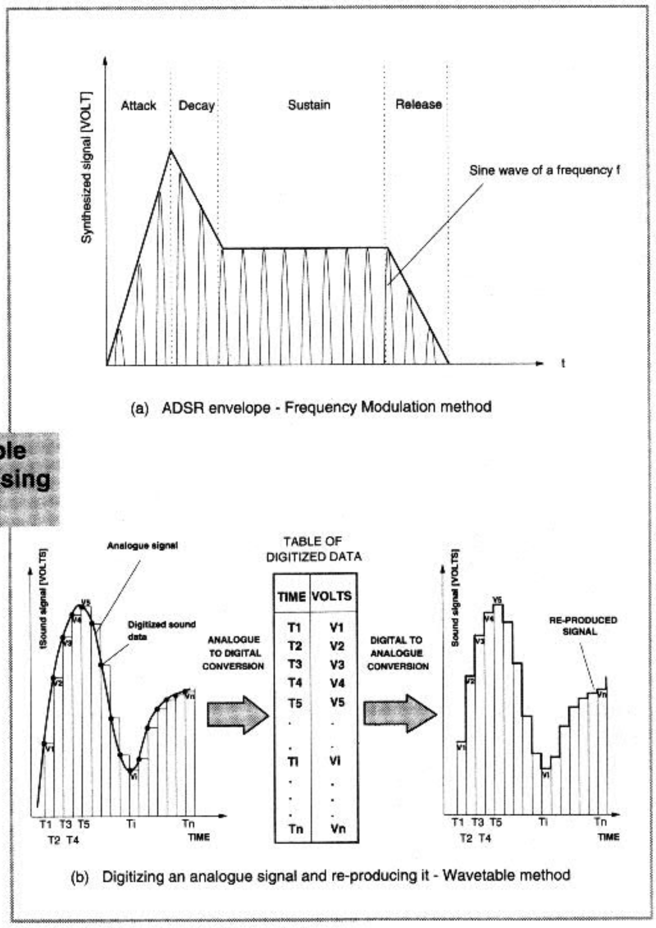


Fig.3 FM and wavetable methods for synthesising natural sounds

fier and then fed to the speaker. The block diagram and the circuit diagram of the device are given in Figures 4 and 5, respectively.

A detailed description of the Centronic port and the way it is controlled has been discussed in detail in the article "Mission Control" in April's issue of the Electronics in Action.

The pre-amplification unit is based on a popular operational amplifier 741 IC. It is configured as a non-inverting amplifier. The close loop gain of the IC is set by the negative feed back circuit R4, R5 and R6, and is equal to the value of R5+R6 divided by resistor R4. The input impedance is set at 27K by R3. C1 provides a DC block at the signal input. The quiescent voltage at pin 6 of IC1 is about 1.2 volts.

The A/D conversion unit uses a ZN448 A/D converter IC (IC2), which is an 8-bit successive approximation analogue to digital converter with a guaranteed accuracy of 0.5 LSB and minimum conversion time of 9μs. A clock generator and a bandgap voltage reference are included on the chip. When the -CONVERT input (pin 4) receives a low-going signal, the A/D converter is triggered to start A/D conversion and the -BUSY output (pin 1) becomes low. The -BUSY output will go high at the end of the conversion indicating that the conversion is completed. The -RD input (pin 2) is the data enable line which is taken low to enable the data on the output lines (pins 18 to 11) which otherwise are in impedance state. Refer to the circuit diagram (Figure 5), the -CONVERT output (pin 4) is connected to one of the control group lines of the Centronic port. To start a conversion, the computer will send a high-to-low-then-high signal to the -CONVERT line, then the computer reads the -BUSY output line and check whether it goes to high. When the high state is detected, the PC reads data from the A/D converter. A clock capacitor (C4) of a value of 100p is connected across pin 3 and the ground (pin 10) which enables the on-board clock generator to operate at about 800 KHz. A negative power supply ranging from -3V to -30V should be supplied to pin 5 (-V) via a tail resistor R8, the value of which has to be chosen according to the voltages. In the present circuit, the negative voltage is generated by a diode pump circuit, RT1, D1, C5 and R10, driven from the -BUSY output. Pin 8 of the IC is the

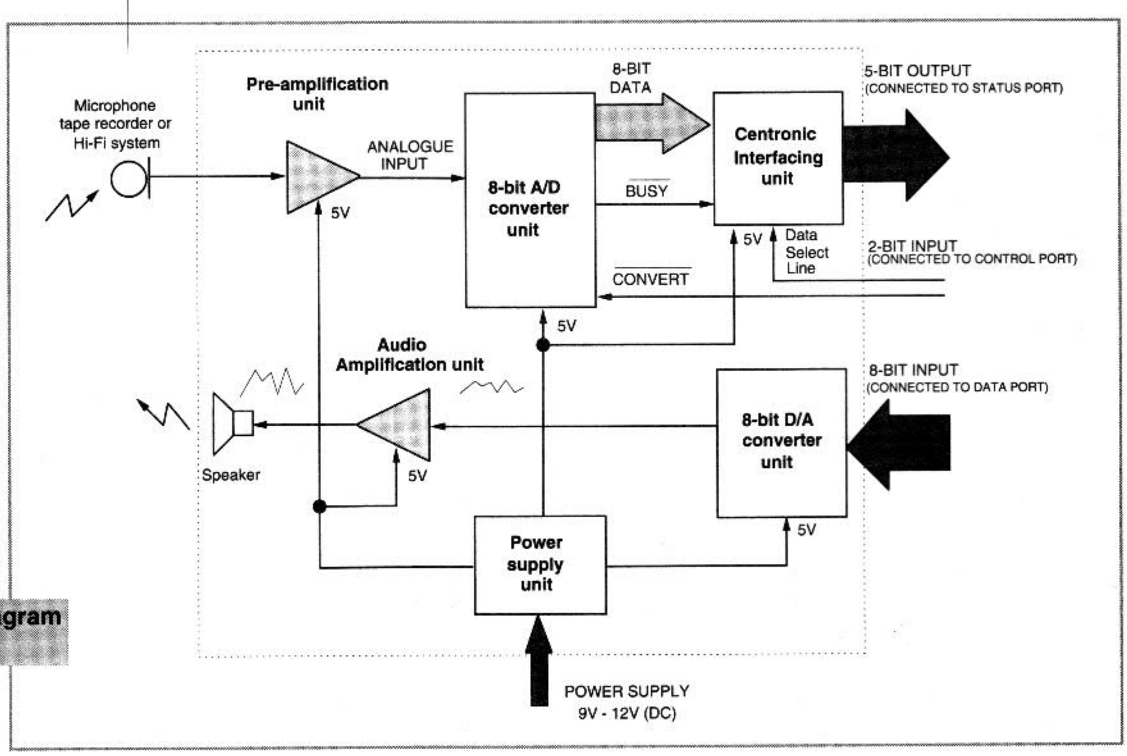


Fig.4 Block diagram of the system

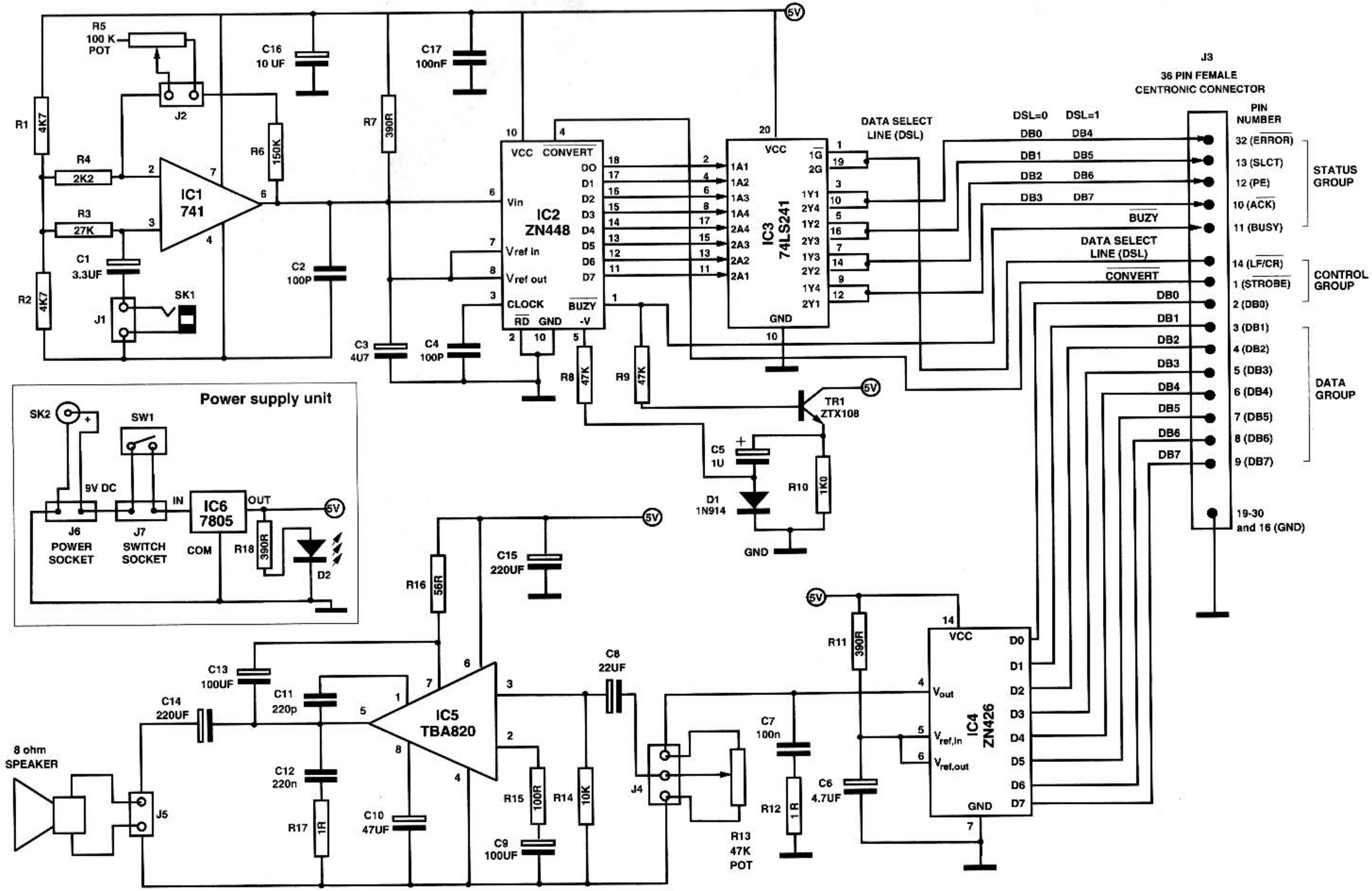


Fig.5 Circuit diagram

output of the 2.55V on-board reference $V_{ref,out}$. To use this reference, a resistor R7 and a decoupling capacitor C3 are required and $V_{ref,out}$ (pin 8) is connected to $V_{ref,in}$ (pin 7). The sound signal from the pre-amp is fed to V_{in} (pin 6). The digitized value for an input voltage is determined using the following equation:

$$\text{Decimal value} = \frac{\text{Input voltage}}{2.55} \times 255$$

The interfacing unit consists of a 74LS241 IC (IC3). Since the Centronic port only has 5 input lines and one of them has already been used for monitoring the -BUSY output of ZN448, only four lines are left for reading the data. To read an 8-bit byte, 74LS241 IC, the 3 state octal buffer, is used. When pin 1 (the 1st enable input) is taken low, the four left hand side buffers work (ie. the outputs will follow the status of the inputs). When pin 19 (the 2nd enable input) goes high the 4 right hand side buffers will work. If pin 1 and pin 19 are connected together to form the Data Selection Line (DSL), by putting the line low and then high, we can read 4 bits connected to the left hand buffers and the other 4 bits connected to the right hand buffers in turns. Operating in such a manner, the 8-bit data from the A/D converter can be read into the computer.

The digital to analogue unit consists of an 8-bit D/A converter ZN426E (IC4) which consumes a typical supply current of 5 milliamps and has a setting time of typically 9µs. It is contained in a standard 14 pin DIL encapsulation. The device has an on-board 2.55V voltage reference with its output at pin 6 ($V_{ref,out}$). To use this reference, an external load resister R11 (390 ohm) and a decoupling capacitor C6 (4.7µF) are used. The voltage reference output is directly connected to the reference voltage input at pin 5($V_{ref,in}$). The analogue output voltage is taken from pin 4 (V_{out}) and is in the range 0 to 2.55 volts. The relationship between the output voltage and the value of the input data is shown below:

$$\text{Output Voltage} = \frac{\text{Input data}}{255} \times 2.55[\text{V}]$$

This signal is fed into the audio amplification unit which is based on the TBA820 IC (IC5). This IC requires a power supply from 3 volts to 12 volts and is able to provide an output power of up to 2 watts RMS using an 8 ohm speaker. Pin 2 on IC5 is the inverting input and there is an internal 6K resistor between the input and output of the IC. This allows the voltage gain of the amplified to be adjusted by an external resistor (R15) and the voltage gain is approximately equal to the value of the internal resistor divided by resistor R15. R15 is recommended to be in the range from 22 to 220R, C8 and C9 provides DC blocking. C10 decouples the supply to the pre-amplified stages of IC5, which helps to give the circuit good ripple rejection. C11, C12 and R17 are components to prevent high frequency instability. R16 and C13 are boot-strapping components which give the circuit good efficiency. C15 is the power supply decoupling capacitors and C14 is the output coupling capacitor. The input sound signal from the D/A converter is attenuated by a variable resistor R13.

The power supply unit uses a 7805 5V regulator (IC6) and requires an 8-12V DC power supply. The current requried is below 500mA.

Testing

After soldering, a careful inspection must be conducted to check all the joints and connectors to make sure there are **no shorts**. After this, power can be connected to the device. An oscilloscope can be used to check the waveform at pin 3 (CLOCK) of ZN448. It should be a 800KHz square wave. Plug a microphone into the audio signal input socket, the waveform of the sounds picked up by the microphone can be seen at pin 6 of 741 IC. **Unplug ZN426 IC from the socket** and connect pin 6 of 741 IC to pin 4 (V_{out}) of ZN426, if the audio system is OK, sound received by the microphone can be heard from the speaker. At this stage, the device can be connected to the computer. First of all, switch off the computer and the sound machine, then plug the device to the printer port using the printer lead. Next switch on the sound machine and the computer. If the computer does not boot up properly turn off the computer, unplug the device and check the board again. If everything is OK, you can now run programs. Firstly, use a musical software (such as the one introduced later) to check the D/A circuit, then use the control program provided to check the whole circuit. Please follow the instructions given by the software and adjust the volume control and pre-amplification gain to achieve the best result.

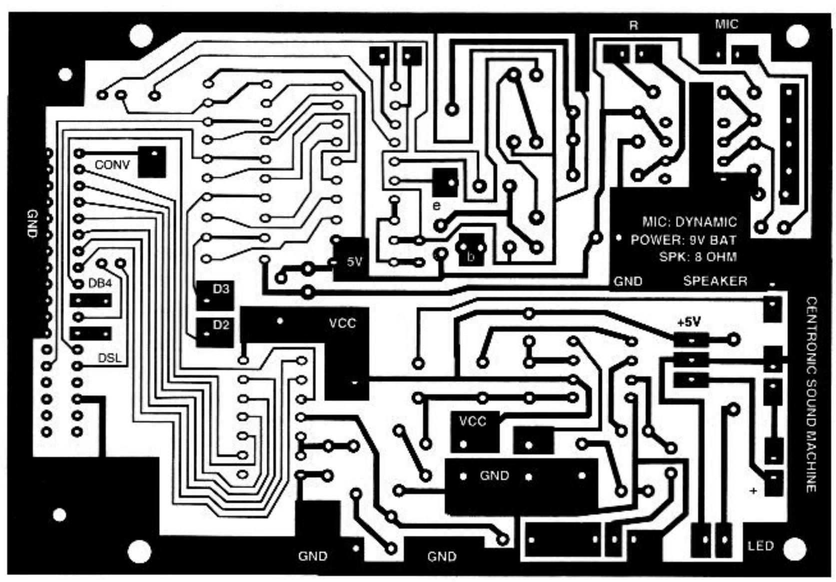


Fig.6 Copper foil

Construction

The Centronic sound machine is constructed on a single-sided print circuit board. The full size copper foil pattern and component layout are shown in Figures 6 and 7. The printed circuit board is available from the EIA PCB service (see page 61).

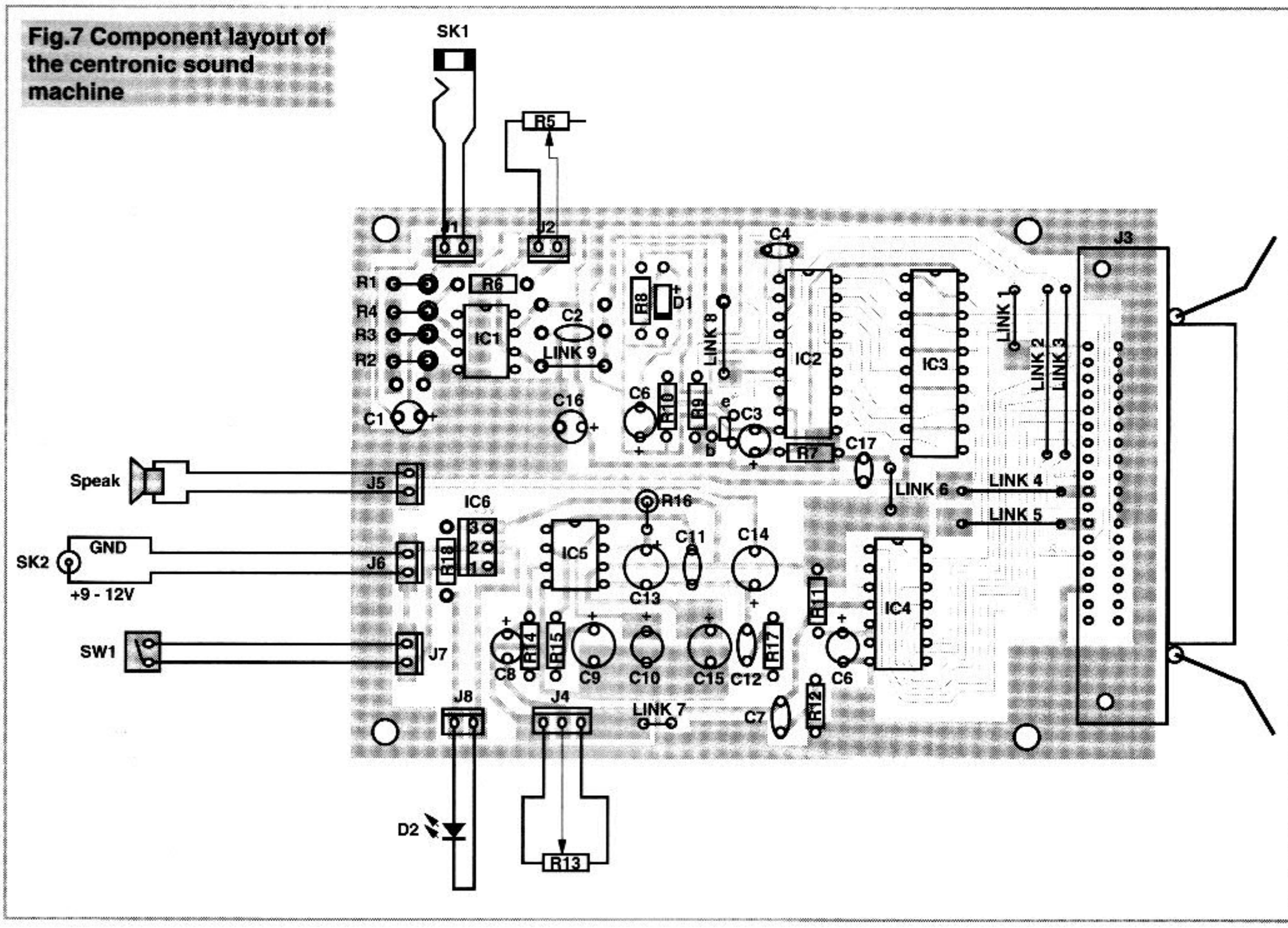
Components may be mounted in the following order: links, resistors, diodes, DIL IC sockets, capacitors, electrolytic capacitors, PCB connectors, the voltage regulator, Centronix connector and finally the ICs. It is suggested that IC sockets are used for all ICs.

Programming

1 Control program

A program for controlling the sound sampling and play back has been written in TURBO PASCAL 6 and is listed below. The flow chart of the program is

Fig.7 Component layout of the centronic sound machine



shown in Figure 8. This program is written as a demo to show how to use the device to sample sound and play. The program will sample the sound for about 10 seconds, then play the sample back over and over again. Readers are encouraged to develop their own software to include more functions and achieve various sound effects.

To start the sound sampling, a short high-to-low-then-high pulse is sent to the convert input. After this the computer starts to check the BUSY outline continuously until it becomes high. Then the computer sets the data selection line high and reads the 4-high bits from the converter. Next the data selection line is taken low and the computer reads the 4 low bits. The two readings are rearranged and combined to form a signal 8-bit data. To play back the digitized sound, the computer sends data to the D/A converter and the sound is produced by the D/A converter.

The complete program list is given at the end of this article. Readers can either type the program into the computer or contact the EIA office (details are at the end of this article) for the software on floppy disks. Messages in { } are the explanations and can be omitted when inputting the program into the computer. It is noted that the variable DELAY NUMBER in the program has to be experimented by the readers to find the best value, since different computers may run at different speeds and the time delay must be changed accordingly.

Public domain software

The are several public domain software packages available from various PD software libraries which support the D/A converter module connected to the printer port. One of them is **ModEdit** which allows users to create sound files (MOD files). This program combines a memory-resident program **ModRes** which is able to play back the music created from within the editor environment. It is easy to compose with ModEdit, which is pull-down menu driven. You just enter the notes one by one into the computer under the edit environment. To hear the music you have entered, you select the correct option from the menu.

Applications

The idea of converting an analogue sound into a digital form, processing it and converting it back to the analogue form is very useful for the purpose of producing various sound effects and studying characteristics of the sound. The following is a brief introduction of what we can do with this device.

1 Simple Effects

The simplest way of experimenting digital sound effects is that you store a long sequence of sampled sound data, such as your speech, in the memory. Then you program the computer to play them back in different ways. If the playback rate is higher than the sampling rate, you will be surprised to find that you sound like a little baby. If it is lower, you sound like an old man. Another sound effect is to send the sampled sound data in the reverse order. What a sound that will be!

2 Delay effect

This effect will delay a sound sample (played in the same order as it was received) . The effect of delay can be achieved using the following procedures. The computer samples the sound and stores the digitized data in the memory. In between each sample, the computer sends a previously stored value to the D/A converter, to produce a delay effect.

3 Echo and reverberation

Samples are taken and stored in the memory. Each sample is sent out to the D/A converter. It is immediately followed by sending data which are a fraction of an earlier sample stored in

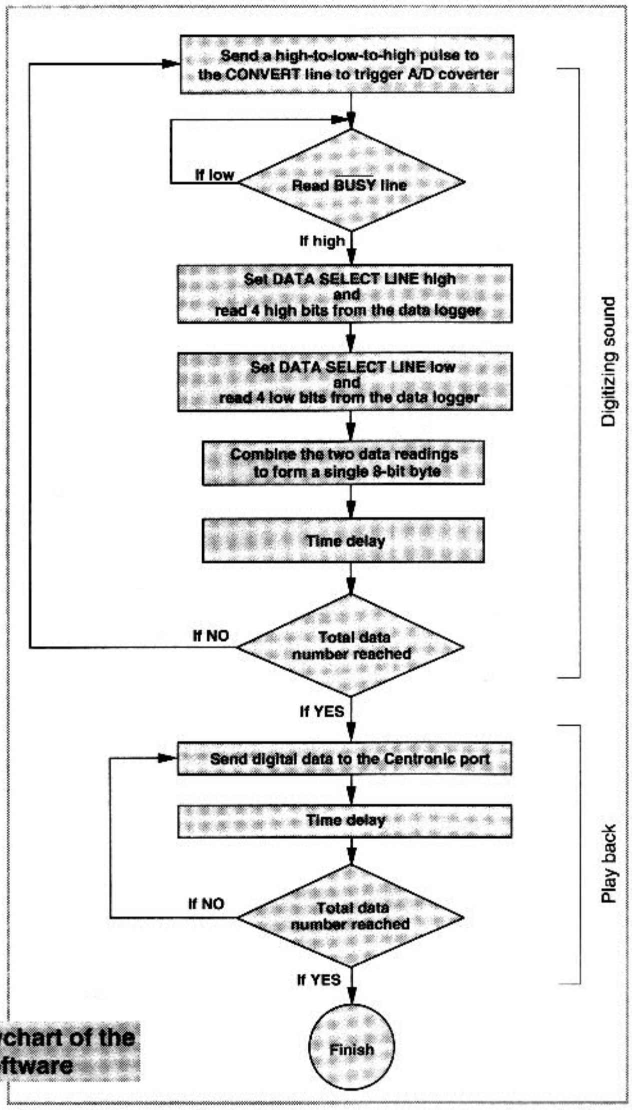


Fig.8 Flowchart of the control software

If the sign of the previous value is reversed before adding a small fraction of it to a large fraction of the current signal, the reversed effect is produced (i.e. aY - bX, a+b=1 and a>b). The resulting signal is composed mainly of the most recent sample and the influence of earlier samples is removed by subtraction. Low-frequency trends die out quickly and high frequency changes are retained. We therefore have a high-pass filter. To combine the low-pass and high pass filter, we obtain a band-pass filter. Different filter combinations can be achieved to create various sound effects.

5 Fourier analysis

The Fourier analysis, also known as Fourier Transformation named after the mathematican J.B.Fourier, is a process which divides a sound into a finite number of different sine waves. This analysis will tell you how a sound is composed and what are the dominant frequencies. Numerical procedures of Fourier transformation are available from various software libraries and textbooks. You can develop a program to record the sounds of birds, musical, instruments and voices of your friends, carry out the Fourier Analysis and find out the characteristics of these sounds. This information can be used as a data base to recognise birds, musical instruments and you friends by analysing their sounds.

the memory. We therefore hear the sound plus a faint echo of the sound already heard earlier. Processing in this way gives just a signal echo. The computer can be also programmed to play the echo many times with its amplitude gradually decreased. We thus get a repeated echo, gradually dying away.

Previous data (X)	Current data (Y)	Weighted data aX + bY (a=b=0.5)
100	120	110
110	130	120
120	140	130
130	120	125
...	...	...

4 Filtering

Filtering is achieved by summing a fraction of the current sample with a fraction of the previous sample. The summed value is sent to the D/A converter. An example is shown above.

At each sample, the resulting value sent out becomes the previous sample for the next stage. The fraction weights for the two values, a and b, can be changed as long as the sum of the two is equal to 1 (e.g. 1/4X + 3/4Y, or 1/20X + 19/20Y). Processing in this was, rapid changes in the signal do not have a significant effects on the output signal, since they are averaged out with the previous values. Only low frequency changes are effectively in changing the output value. As a result, the program acts as a low-pass filter.

6 A BIG Program

Combining these ideas and others together, you can write a BIG program that not only plays the preprogrammed music and speech but also composes music itself and understands what you say.

(0.25W, Metal film 1%)		C1	3.3µF/10V electrolytic
R1,R2	4K7	C2,12	220nF Mylar film
R3	27K	C3,6	4.7µF/10V electrolytics
R4	2K2	C4	100pF Ceramic disc
R5	100K potentiometer	C5	1µF/10V electrolytic
R6	150K	C7	100nF ceramic disc
R7,11,18	390R	C8	22µF/10V electrolytic
R8,R9	47K	C9,13	100µF/10V electrolytics
R10	1K0	C10	47µF/10V electrolytic
R12,17	1R	C11	220pF ceramic disc
R13	47K potentiometer		
R14	10K	C14,15	220µF electrolytics
R15	100R	C16	10µF electrolytic
R16	56R	C17	100nF ceramic disc

List of the control program

```
program Centronic_Sound_Machine;
{Program written by Pei An for controlling the CENTRONIC SOUND MACHINE}
{Program is written for 12MHz 80286 machine and LPT1 Centronic port}
uses
    crt;
var
   byte1,byte2,truebyte:byte;
   i,j,delaynumber:integer;
   ii:longint;
   soundbyte:array [1..63000] of byte; {Total number of digitized data is 63000}

procedure delaytime;
{Delay a short while}
begin
      for i:=1 to delaynumber do j:=10;
end;

Procedure sample;
{To digitize sounds}
begin
for ii:=1 to 63000 do begin
   begin
       port[890]:=1;              {CONVERT=0, DSL=1}
       port[890]:=0;              {CONVERT=1, DSL=1}
             {To start A/D converter}
repeat
        byte1:=port[889];         {To check the BUZY line and read byte1, byte1=xxxhhhhx}
until byte1<128;
      port[890]:=2; {DSL=0}
      byte2:=port[889];    {To read byte2, byte2=xxxllllx}
      byte1:=byte1 and 120;      {byte1 and 00011110 = 000hhhh0}
      byte1:=byte1 shl 1; {shift 1 bit left,   byte1=0000hhhh}
      byte2:=byte2 and 120;      {byte2 and 00011110 = 00011110}
      byte2:=byte2 shr 3; {shift 3 bits right, byte2=11110000}
      truebyte:=byte1 or byte2;  {byte1 or byte2 =11110000 or 0000hhhh = 1111hhhh}
      port[888]:=truebyte;{To send data to D/A converter}
      soundbyte[ii]:=truebyte;
      delaynumber:=15;    {To set delaynumber=25. This number is for 12MHz 286}
       {For other machines, this number needs to be changed}
      delaytime;    {To delay a short time}
end;
end;
end;

Procedure playsound;
{To play back the sound via D/A converter}
begin
repeat
      delaynumber:=40;            {Delaynumber for play back the sound}
      for ii:=1 to 63000 do begin{Loop to play back the sound}
          delaytime;              {Time delay}
          port[888]:=soundbyte[ii];     {Sent data to D/A coverter from port[888]}
                             end;
      delay(1000)
until ii<0
end;

{================main program=================}
begin
       clrscr;
       writeln('Connect the microphone to the device');
       writeln('Use pre-amplification gain and volume control to obtain the best sound quality');
       writeln('Start sampling. Press RETURN to start sampling');
       Writeln('At the end of the sampling, a beep will sound'); readln;
       sample;
       sound(440); delay(500); nosound;
       writeln('Finished sampling. Press RETURN to start replaying');
writeln('The digitized sound will be played back over and over again');
readln;
       playsound;
end.
```

Anyone who would like a copy of this program on a 3.5" disk should send a cheque for £3.50 made payable to Dr. P. An to the EIA office along with your name and address

Parts

Semiconductors

IC1	741 (MC1741) Op-Amp IC
IC2	ZN448E A/D converter IC
IC3	74LS241 octal buffer IC
IC4	ZN426E-8 D/A converter IC
IC5	TBA820M audio amplifier IC
IC6	7805 (M78M05CV) 5V voltage regulator IC
TR1	ZTX108C
D1	1N914 or 1N4148
D2	Red 5mm LED

Additional items

PCB	See page 61
IC sockets	8-pin (2 off), 18-pin (1 off) and 20-pin (1 off)
J1,J2,J5,J6,J7	2 way PCB connector
J4	3 way PCB connector
J3	36 pin female Centronix-type connector
SK1	1/4 in Mono moulded jack socket
SK2	2.5mm power socket
SW1	Toggle switch
Knobs	Knobs for R5 and R13 POTs
LINK1-LINK9	0.6mm copper core wire
Microphone	Unidirectional dynamic dual impedance (600 ohm and 50 Kohm) microphone
Speaker	8 ohm impedence
Spacers	6BA spacers (4 off) for PCB mounting
Screws	6BA screws for mounting the PCB

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SWITCHED MODE PSU ex equip, 60w +5v @5A, -5v@.5A, +12v@2A,-12v@.5A 120/220v cased 245x88x55mm IECinput socket £6.99 REF MAG7P1
PLUG IN PSU 9V 200mA DC £2.99 each REF MAG3P9
PLUG IN ACORN PSU 19v AC 14w , £2.99 REF MAG3P10
POWER SUPPLY fully cased with mains and o/p leads 17v DC 900mA output. Bargain price £5.99 ref MAG6P9
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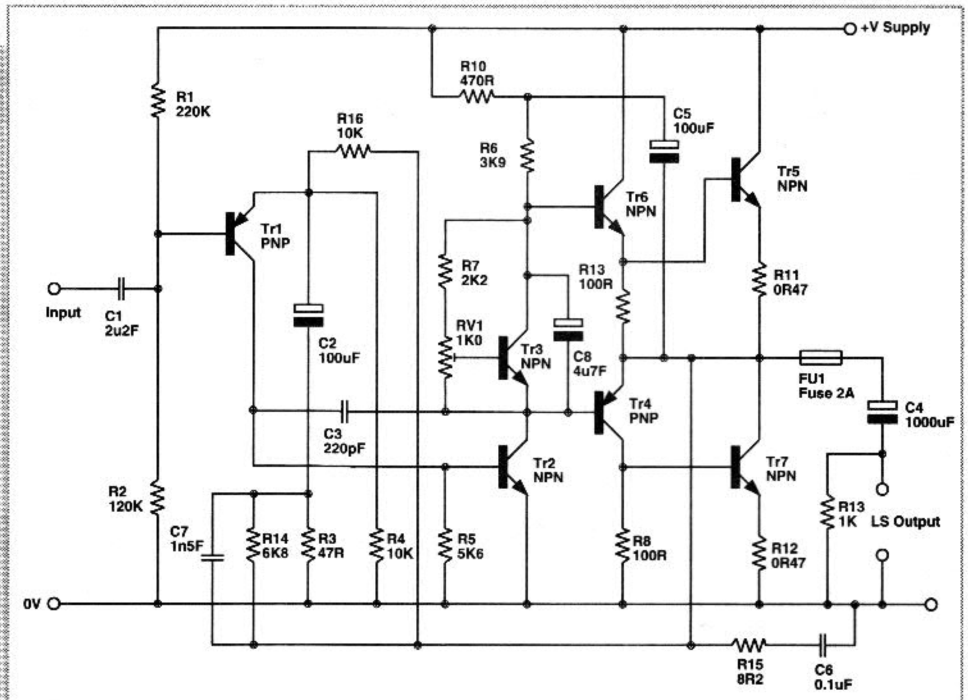
Signal to NOISE

A Selection of your views and comments raided from the post bag



Evolving Evolution

Your new magazine continues to find interesting topics to beguile the reader. Long may it continue at this standard.

I regret that there is a bit of a snarl up in the correlation between the text/illustrations of the 'Evolution of Audio Amps' Part 3, in relation to Figure 3. The correct drawing is shown here. I don't think anyone would wish to build this circuit, but please print the correction to keep the record straight, with my apologies.

I apologise to your readers for the errors which occurred in this figure of the April 1994 article, in which TR2 and TR3 had unfortunately got their numbering crossed, and C3 had become interposed between TR1 collector and TR2 base, instead of being connected between TR2 base and collector.

John Linsley-Hood
Taunton
Somerset

Ideas, Criticisms and back to Ideas

In the Ideas Forum of the April issue of Electronics in Action you asked for a simple solution to the problem of detecting the level of a non-ionic fluid. There are in fact a great many simple solutions. Off the top of my head I could suggest the following all of which have been used commercially:

1. A float containing a magnet that operates a reed switch sealed in a tube.
2. Detection of the reduction in temperature of a self-heated probe such as a thermistor when it is immersed in the liquid. National Semiconductor produced (and may still do) an IC, the LM903, which utilised this principle.
3. Use of a transparent rod with the bottom face angled so that light is normally totally internally reflected at the face. However when immersed in a liquid of reasonable refractive index the light is no longer reflected. This could be detected by a LED/photodetector combination at the end of the rod. The length of the rod determines the detection level.

However my main reason for writing was to comment on the general standard of the construction projects in EIA. After a good start, with some innovative and well thought out designs, the level seems to be falling back to that of other hobby magazines.

The A-B guitar switching box that appeared in the April issue is a good example of what I mean. The author does not fully understand the operation of the 4066 analogue switch. After raising the DC level of the signal to half the supply by means of IC1 he then removes the DC component before passing the signal through the 4066. The signal is therefore referenced to the 4066 negative supply and will be prone to unnecessary distortion. In addition only one 4066 is in fact required. SW1 could quite easily switch the LEDs directly and the use of the other half of two sections of IC2 in parallel makes no significant improvement in performance.

Similar muddle-headed design was evident in the circuit for an egg incubator which appeared in the February issue. Temperature control is a simple matter that nowadays can be achieved with one IC as even the author admits! If he insists on safety then why not isolate or insulate the probe instead of the whole circuit? Even more thoughtless is to generate a relatively stable reference voltage using IC5c to feed the setpoint control and then to have a separate reference, D1, to produce the zero offset current.

Perhaps I could make three suggestions that I believe would maintain or even improve on the early high standard of EIA.

Firstly as many readers are primarily interested in the design aspect of the circuits, why not extend this part of project articles at the expense of construction details. These are usually the same anyway and could even be repeated every month in a separate section at the back of the magazine with only special points included in the project description. The article could then concentrate in far more detail on why the author had chosen a particular design and, more importantly, what other options he had considered. Generating and evaluating ideas is fundamental to the innovative process, something EIA is supposed to be nurturing.

Alternatively why not actively encourage criticism of published designs and seek suggestions for improvements. Having a practical starting point might tempt readers to think and experiment more than maybe Ideas Forum is doing. I noticed you did take a small step in this direction in publishing a letter commenting on Mike Meechan's mixer project.

The final point concerns information about the range of ICs now available. Much inelegant design is due to either being unaware of an IC which will perform a function simply and effectively or of then using it incorrectly due to lack of understanding or inability to read a data sheet. Although catalogues do contain information on a range of ICs, this is not comprehensive or up-to-date. I remember a regular column in the old Practical Electronics called Semiconductor Update which each month featured details and applications for a handful of new ICs which could be relevant to the hobbyist. It used to be one of the first pages I turned to! How about something similar for EIA? I'm certain it would be of more value to innovative readers than rehashing tired old circuits. I'll even volunteer to write it for you!

Andrew Chadwick
Hull

Thank you for your comments and valued criticism. I have to say that one of the main objectives of the magazine is to get people to look at the circuits we publish and to update or improve them if needs be or to use them for a different application. We have to remember these are all working prototypes and not well established designs. Yes I agree, part of the emphasis should be on why the author chose an idea or design and is in our brief to potential authors. Regarding the latest news on ICs, we already have a couple of articles prepared. However, I will take you up on your suggested designs, please forward them for possible publication.

Finally - No doubt Daniel Coggins and David Silvester, the authors of the projects you mention, will provide suitable return comments. - Ed.

Stereo Image

I find the projects by Mike Meechan most interesting as I am in the music business myself and into electronics as a hobby.

I have a problem which may or may not be able to help me with. I came across an article dated back about 1980 on a Stereo imager which used a VCA chip namely a 1537A. At the time it was available from a company called Aphex audio systems in London. However they no longer exist. Can you give me any idea of where I might find such a chip?

M Coutts
Aberdeen

PS An idea for a project - A stereo imager/panner for studio use.

Any suggestions from any music people out there? - Ed.

If you have any views, comments, criticism or ideas please address them to:
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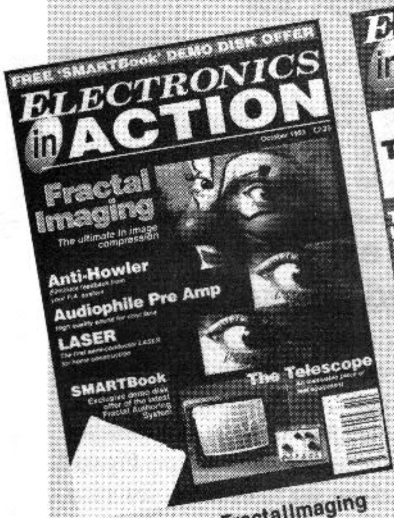
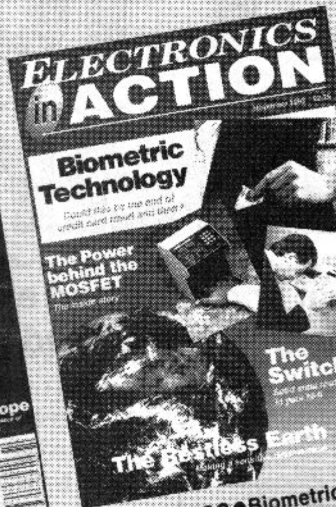


October 93 ●Fractal Imaging ●The Telescope Pt1 ●Anti-Howler ●Audiophile Pre-amp ●Sine wave generator



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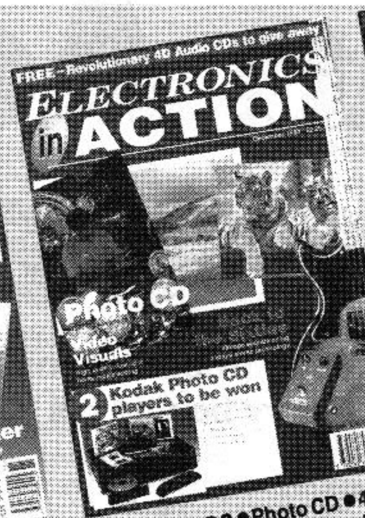
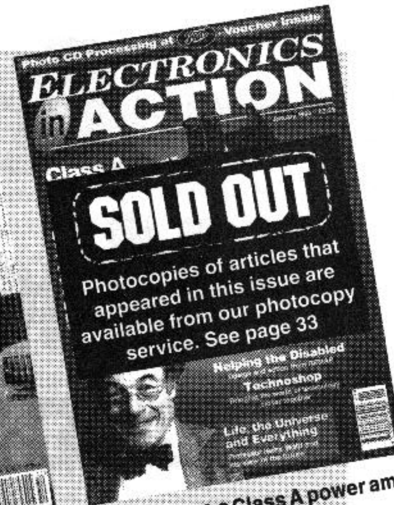
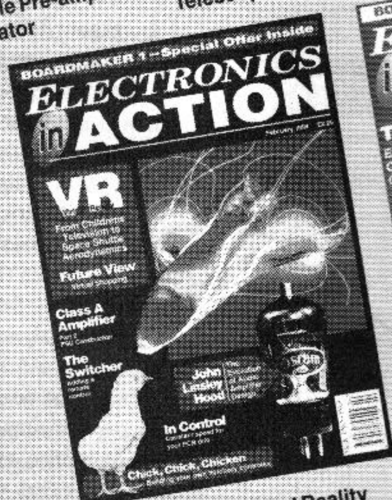


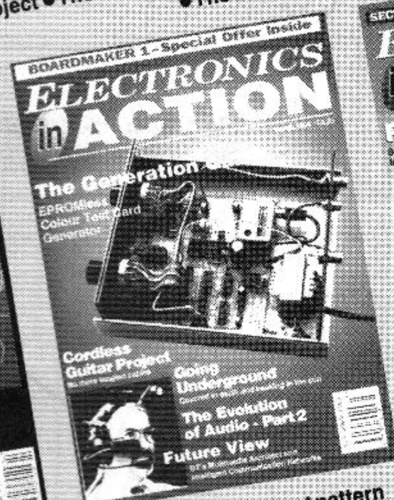
December 93 ●Photo CD ●4D Recording ●Video editing unit ●Stereo tremolo unit ●The Switcher Pt2



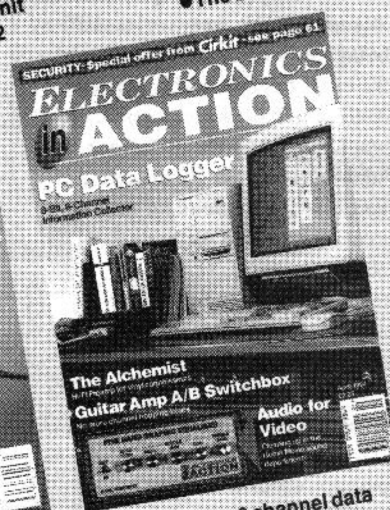
January 94 ●Class A power amp ●The Harmoniser ●Remote control extender ●The Switcher Pt3



February 94 ●Virtual Reality ●Class A amp Pt2 ●Hatchery Controller ●The Switcher Pt4



March 94 ●Colour test pattern generator ●Cordless guitar project ●Plug-in mains wiring tester ●Intelligent Communication Networks ●The Switcher Pt5



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An Act for the Cabaret

Part 1 Peter Roberts dons hat and cane and explains the theory behind his 'Cabaret' active loudspeaker design.

Ever looked at the distortion versus frequency curve of a loudspeaker? It gets incredibly high at low frequencies, 10% being fairly common. Fortunately the ear is tolerant of low frequency distortion. Note I said tolerant, you would recognise the difference between 10% and 1% distortion if you heard it. Reasons for this tolerance include the brains' ability to synthesise missing frequencies using harmonics as cues. Noticeable, acceptable and tolerable distortion levels vary with frequency, level and programme material and are the realms of psycho- acoustics.

There is a way to reduce low frequency distortion (dating back to 1950) which I came across by accident when I was designing an earlier active speaker.

The distortion is caused by the increased cone movement necessary at low frequencies. The lower the frequency the more cone excursion required. There will be non linearity in the cone suspension and the magnetic circuit, over the distance travelled, and this will cause distortion. There is usually a fair amount of non linearity especially at the extremes of travel.

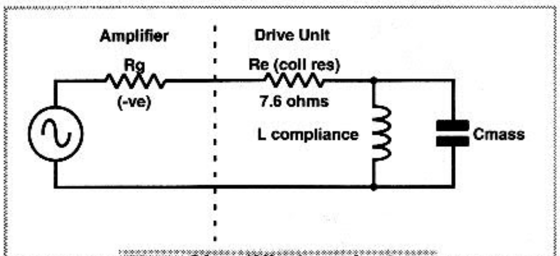


Fig.1 Simplified equivalent circuit of amplifier

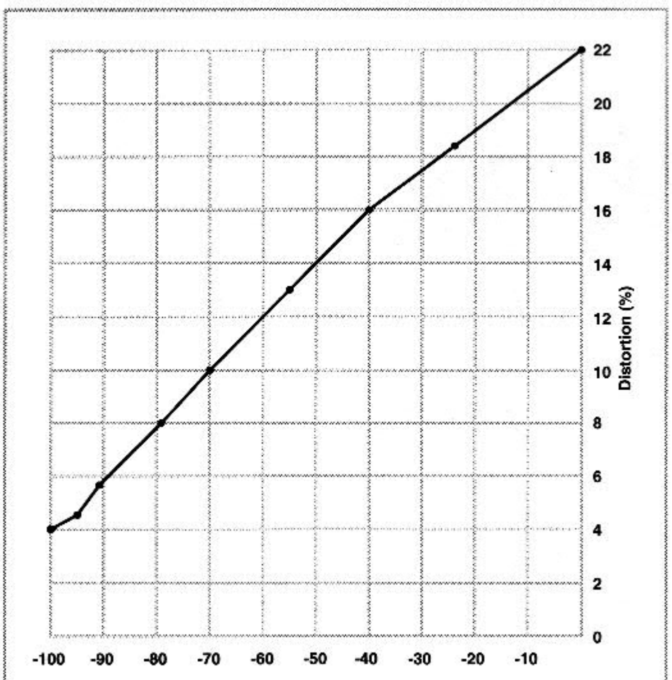


Fig.2 Negative resistance as % of voice coil

The very simplified equivalent circuit of a drive unit is shown in Figure 1. The coil represents the mechanical compliance (opposite of stiffness) of the cone surround. The capacitor represents the moving mass of the cone and coil. The L-C resonant circuit so formed gives the resonant frequency of the speaker.

If you are having trouble visualising it, consider this. The mechanical load of an electric motor is reflected as a change in electrical load on its supply. Similarly, the mechanical properties of the speaker have been reflected into the electrical circuit. This is a time honoured way of electrically modelling speaker performance. If we could short the coil and capacitor in the LC resonant circuit, we could damp the tuned circuit and damp out the speaker resonance. Unfortunately the voice coil resistance stops us directly shorting the tuned circuit. What we can do through is generate a negative resistance in the drive amplifier. This lowers the net resistance in circuit and damps the speaker. Since we are also shorting the compliance we are also reducing the non-linear distortion. Figure 2 shows the reduction in distortion of an actual speaker as the negative resistance is increased, while Figure 3 shows the improvement in damping for a negative resistance of 85% of the voice coil resistance.

This technique was first described in 1950 by an American, by the name of Warner Clements. It was taken further in 1981 by Karl Erik Stahl of Sweden who designed an amplifier which could independently control the output inductance, capacitance and negative resistance, and thus electronically tweak the compliance, moving mass and damping of the drive unit connected to it. I should point out that there is a patent on this particular technique.

Funnily enough, the lower distortion etc. is a beneficial side effect of the main reason for using negative resistance. This is to tune the Qts of the drive unit to a suitable value for a particular cabinet size. If the Qts is too high you get an underdamped response, with a muddy boomy bass (see Figure 4) and conversely if the Qts is too low you get an overdamped response with a loss of bass. Changing the Qts is a different, more precise way of saying we are changing the damping. The only drawback is that the amplifier driving the speaker has been matched to it and will not operate correctly with other speakers.

The Wizard of Oz

The full equivalent circuit of a closed box bass system is a second order high

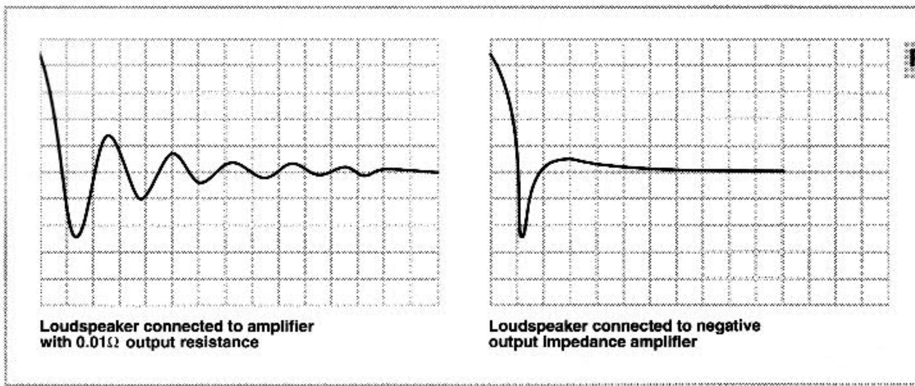


Fig.3 Response to step function

pass filter. The port in a bass reflex design introduces another second order filter. It was understanding these filters that led an Australian TV design engineer, Thiele, to formulate a method of designing bass reflex systems, back in 1961. He also showed that one can put in an extra second order filter, making a sixth order design.

This extra filter not only removes any rumble, it also reduces the cone excursion. A sixth order bass reflex gives a cone excursion one quarter the size of a closed box for the same output and -3dB bass down point (see Figure 5).

A bass response to down under

There are only three drive unit parameters needed to design a bass reflex speaker, the Q_ts, the volume equivalent of compliance (V_{as}) and the resonant frequency. Even these are not always given (in some catalogues, all three values are not presented) However given some test gear they can be measured fairly easily, this is described later. Using graphs or tables one can then design an enclosure. Some authorities say you need a computer and a speaker design program, but these are not really necessary. It is when you stray off the beaten track of the classic alignments, that you need a computer. Indeed working with graphs often gives you more feel about the design. See Figure 6, which shows the Q_t versus volume multiplier for a standard bass reflex speaker. Look along the bottom axis and find the Q_t of the drive unit, extend the point vertically upwards until it intersects the curve. Then project horizontally across to find the volume multiplier. This is the factor by which you multiply the V_{as}, to give the enclosure volume.

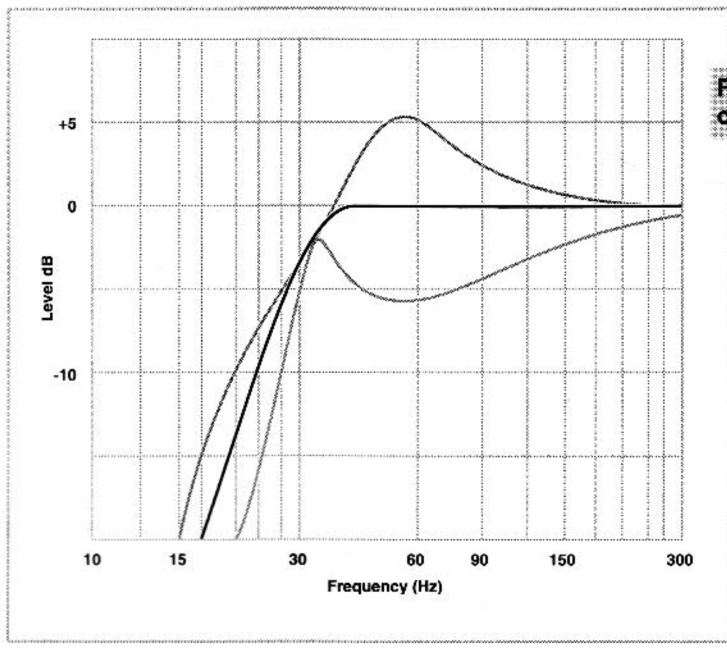
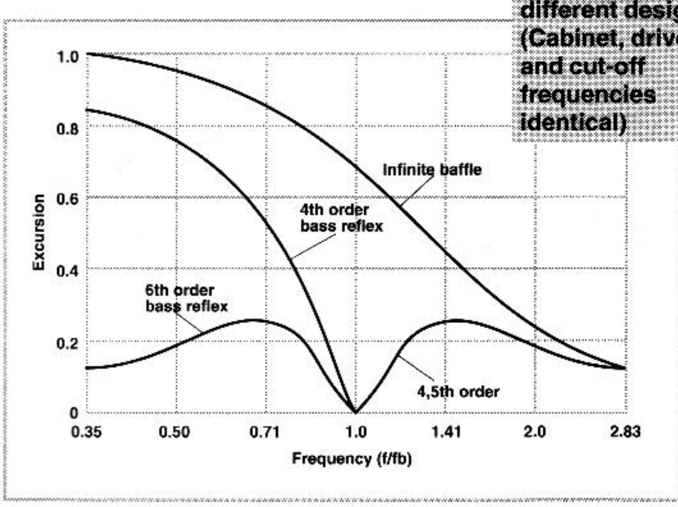


Fig.4 The effect of varying Qts

Similarly, to find the port tuning frequency see Figure 7. Multiply the tuning frequency factor by the drive unit resonant frequency. This gives you the box tuning frequency.

To determine the length of the port, use the nomogram in Figure 8, which is due to Small of Thiele/Small fame. Locate the box volume you require on the V_b column, for the example given, 2 cubic ft. Also find the box tuning frequency, example 40Hz, on the F_b column. With a ruler linking the two points extend the line to the L_v/S_v column. From here project a horizontal line across until it intersects one of the curves. In this case it intersects two curves. If the larger diameter one is a reasonable length (ie. less than 25 cm) it is best to go for that. Extend the intersect point vertically upwards to give the port length. The larger diameter port will give a lower port air velocity.

Fig.5 Cone excursion against frequency for different designs (Cabinet, drive unit and cut-off frequencies identical)



My Kef B200A has a Qt of 0.6, a resonant frequency of 29Hz and a V_{as} of 100 litres. Multiplying the V_{as} by the 3.5 factor given in Figure 6, corresponding to the Q_t of 0.6, gives a box size of 350 litres or 12 cubic feet, which might impress one's fellow hi-fi addicts but would make you unpopular with anyone else. This also shows that a low compliance woofer is required for a bass reflex design, to achieve a small cabinet size. For a closed box design a high compliance driver is needed so that the air in the box can be the dominant compliance.

As can be seen from Figure 6 low Q_t values will give a small enclosure size.

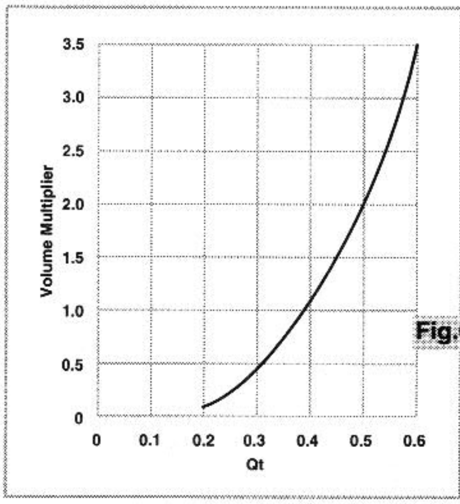


Fig.6

Since the Q_t is inversely related to magnet size, a large speaker magnet is needed to give a low Q_t and large magnets are expensive. An alternative way to vary the Q_t is with a negative or positive output resistance from the amplifier driving the speaker. It is then possible to vary the Q_t at will, so that a given speaker can be matched to a given enclosure. Given the Kef B200A above, using negative resistance, one could easily modify the Q_t to 0.3, giving a volume multiplier of 0.47, hence a much more suitable cabinet size of 47 litres. The distortion and damping will have improved as well.

Figure 6 is a plot of the function

$$\frac{V_b}{V_{as}} = Q_t^{2.87} \times 15$$

where V_b is the box volume and V_{as} is the volume equivalent of compliance. Figure 7 is a plot of

$$\frac{F_b}{F_s} = Q_t^{-0.9} \times 0.42$$

where F_b is the box resonant frequency and F_s is the drive unit resonant frequency.

The cutoff frequency can be computed from

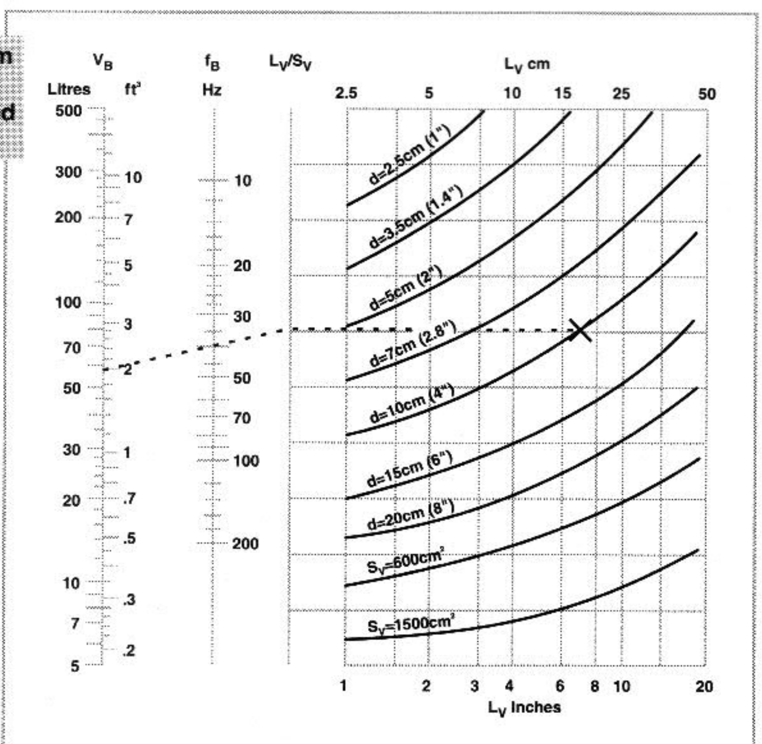
$$F_{co} = Q_t^{-1.4} \times 0.26 \times F_s$$

For the B200A example, the -3dB point would thus be 40.7Hz.

I use the sixth order designs for which you also need to know the Q and the frequency cut-off of the extra electronic filter. Note that you cannot "bolt-on" the electronic filter to a standard reflex design, the system has to be treated as an integrated whole. Thus the sixth order box size and port tuning will be different to a standard forth order design. The sixth order alignments can give a considerable bass extension for a given box size than the standard reflex design. For the B200A example the sixth order cut-off would be 29Hz for the same cabinet size as the standard forth order reflex.

The sixth order parameters are presented in tabular form in Bywater and Weibulls paper, Alignment of filter assisted vented box loudspeaker systems with enclosure losses. JAES May 1982. Another paper from the considerable body of work by Australian speaker engineers, presumably prompted by Thieles work. Some of the worlds leading speaker engineers have Australian connections. Once you have heard a speaker designed according to their principles, you wouldn't give a XXXX for anything else.

Fig.8 Nomogram and chart for design of ducted vents



Measuring the manifold

To measure the drive unit parameters firstly, if the speaker is brand new, you need to run it in. Believe me, the resonant frequency and especially the compliance can change significantly as the suspension loosens up. I run units in for 8 hours minimum at 20Hz and 10watts. Measure the resistance of the voice coil,

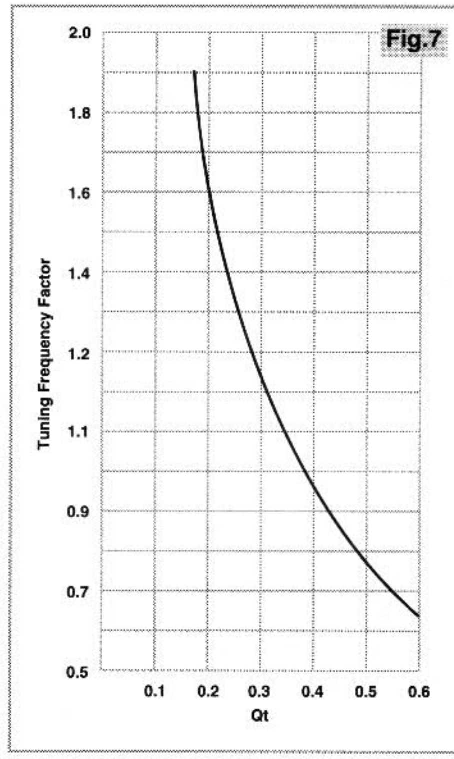


Fig.7

Re. Then create the set-up shown in Figure 9. You need a audio signal generator with a high output or an amplifier to boost the output of a low output unit. The 680 ohm resistor simulates a constant current source. The frequency counter and AC voltmeter are optional to improve accuracy.

With the oscillator frequency set to about 100Hz, Set the oscillator output level so that 100mV RMS (or 100mV peak to peak if using the oscilloscope) is developed across the 10ohm resistor. Then each 10mV represents 1 ohm, so that if you read 76mV the resistance is 7.6 ohm. Switch the output to the speaker, lying face up on the bench. Adjust the frequency until the output is a peak as observed on the 'scope. Note this frequency and the voltage. You now have the resonant frequency Fr and the resistance at resonance, Rs. Divide Rs by the Dc resistance Re to give ro. Compute ro * Re. You then need to find the two frequencies f1 and f2 either side of resonance which have this resistance.

Then $Q_{ts} = \frac{f_r \left(\frac{Re}{Rs}\right)}{f2\text{-}f1}$

$$Q_m = f_r \frac{ro}{f2-f1}$$

$$Q_e = \frac{Q_m}{ro-1}$$

The total Q, Q_t is made up of the Q_m, mechanical Q and the Q_e, the electrical Q. We can modify the electrical Q with the negative resistance technique to lower the Q_t to that required for one's design then

$$\frac{1}{Q_{t(required)}} = \frac{1}{Q_m} + \frac{1}{Q_e}\left(\frac{1}{1+\frac{Rg}{Re}}\right)$$

where Rg is the negative resistance.

To measure the compliance, V_{as}, put the drive unit in a ported unstuffed cabinet with a known volume V_b litres. In the cabinet, the speaker will have 2 resonant frequencies f_h and f_l. Use the test set-up in Figure 8 to find these frequencies. Now tape some stiff card over the port hole to make a closed box. Measure the one resonant frequency, f_b

Then $$\frac{V_{as}}{V_b} = \frac{(f_h^2 - f_b^2)(f_b^2 - f_l^2)}{f_h^2 f_l^2}$$

This is also a useful test to run on the completed speaker.

Come on down

To generate high levels of low frequency sound the speaker cone has to move air. The lower the frequency the

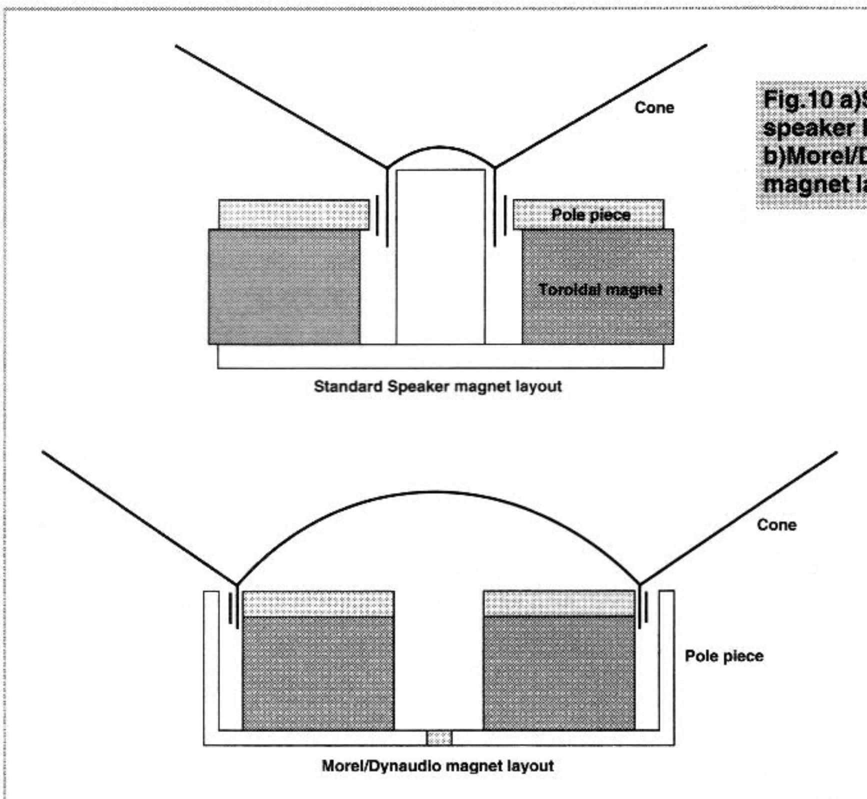


Fig.10 a)Standard speaker layout b)Morel/Dynaudio magnet layout

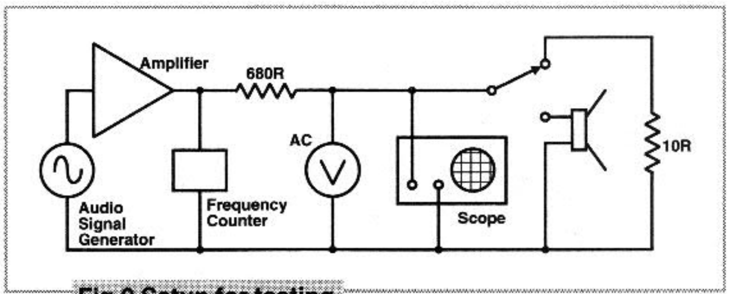


Fig.9 Setup for testing loudspeakers

more air the cone has to move. This means large cone excursions for the bass driver. For instance to generate 1 acoustic watt at 50Hz, the closed box cone excursion of an 200mm diameter driver would be 45mm, a 300mm driver would have 20mm and a 375mm unit 13mm. The 45mm throw is extremely difficult to accomplish. Exceptional designs like ATC's can achieve about 38mm. If those distances look large, it is because 1 acoustic watt/m is a sound level of 120dB which is the threshold of pain.

There are 4 rules about bass frequencies here, for a given output level;

1. If the cone area is doubled, the cone excursion is halved and vice versa.
2. If the frequency is halved, say 50 to 25Hz, the cone excursion is in creased by a factor of four.
3. If the cone excursion is halved the acoustic power is halved.
4. If the cone area is doubled, for the same excursion the acoustic power increases by 4 times.

Design choices

The question is how much acoustic power does one need? Surprisingly little if you listen at normal domestic listening levels. However, if you trying to reproduce orchestral levels in your living room or you are merely a head-banger, then to achieve bass transients like the cannons in the 1812 overture with adequate dynamic range you need something better than your average 165-200mm bass driver.

Bigger is better

So for low frequency bass the bigger the driver the better. However Usually the bigger the driver the greater the compliance so to achieve a reasonable box size, a compromise has been set with a 250mm driver.

This size also will be less troubled by the break-up modes of the cone, which are problematical with large cones at high frequencies. Further it allows a 2 way design, which is easier to integrate than a 3 way. The more is better approach does not work here, a good 2 way design is far superior to a 3 way design costing the same price.

The Cabaret is a tower of power, namely a tower type active speaker design. Despite being 60 litres capacity, its footprint area is small and it does not need stands. It's 3dB down point is 30Hz. To achieve substantial outputs the amplifiers used are 40 watts. A Morel drive unit is used which has a different magnet topology to a standard speaker, see Figure 10. The massive coil assembly gives a low resistance change with a loud programme and optimises the negative resistance operation.

NEXT MONTH

Peter Roberts begins building the 'Cabaret' Loudspeaker

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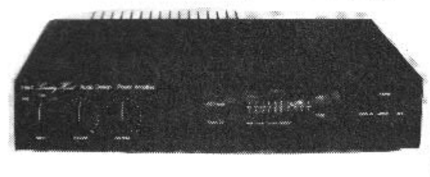

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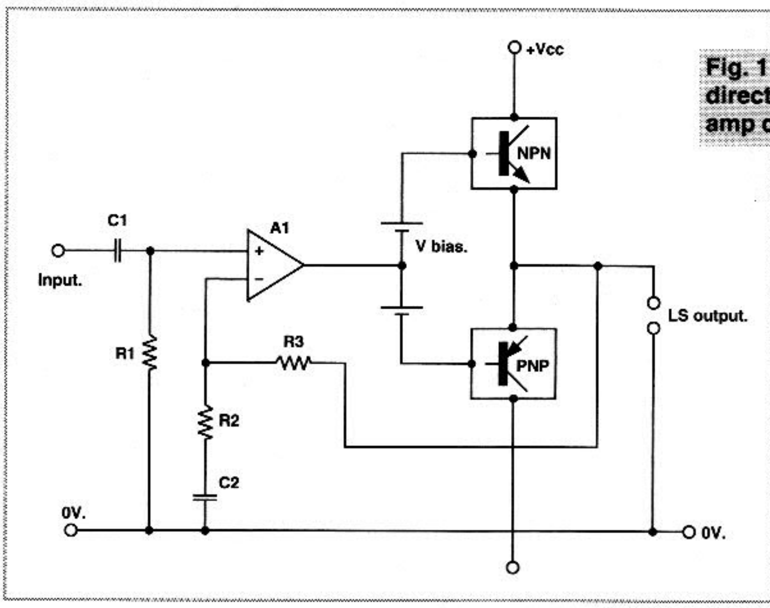


Fig. 1 Basic direct coupled amp circuit

In the last part of this article I looked at the developments in audio amplifier circuitry up to the early 1970s, and I think that this was about the time when transistor operated power amps. began to meet the standards of quality achieved by valve designs some 15-20 or more years before. There were, of course, residual advantages and relative disadvantages in both of these systems, and, in general, these remain to this day.

Valves vs. Transistors

On the side of the valve amps., they used old and trusted technology, they were overload and output short-circuit proof, though they did **not** like operating into an open circuit load, which could cause the output valves to spark over from anode to grids inside the glass envelope. They normally only needed to, or indeed could, use a moderate degree of negative feedback, usually about 26dB (20x) or so, and this usually allowed a good loop stability margin and a thoroughly house-trained approach to awkward LS load characteristics, and, provided that they had a decently designed output transformer, they would have a frequency response which covered from a few Hz, to say, 100KHz.

On behalf of the transistor systems, one could claim that, by about 1975, almost all of the amplifiers from the better manufacturers had thoroughly respectable harmonic and intermodulation distortion figures - which did not worsen greatly at very low power levels - an adequate degree of load stability, and a reasonable reactive load transient response. (I know that I tend to be a bit unenthusiastic about the average solid state amp. transient response, but then, if judged by the same standards, valve amps. were usually a whole lot worse. For excellence in this quality at that time one needed to turn to the better DIY designs.)

What transistor operated power amps. *did* offer that none of the valve amps. could match, was freedom from the need to use an output transformer. In particular, where the amp. was powered from a pair of +ve and -ve supply lines, its output could be coupled directly to the LS load; a layout which I have shown in schematic form in Figure 1.

Although it is 'direct coupled', the amplifier will not work at full gain right down to DC since it is usually necessary to include a capacitor, C2, in the negative feedback loop (R3,R2,C2) in order to stop long-term DC level drifts in the voltage amplifier (A1) causing unwanted DC offsets at the LS output terminals, and that capacitor sets a lower limit to the LF response of the amplifier - given by the formula $f_T = 1/(2\pi C2R2)$. For an extended bass response a large capacitor (usually electrolytic) was needed for C2, and this led to further problems.

The other enormous advantage of solid-state amp. designs was the amount of power they could provide. When this advantage was first realised, LS system designers - who mostly had tunnel vision where their own speciality was concerned - began to design speaker units whose improved sound quality was gained at the cost of very low acoustic efficiency. After all, why worry? A few hundred watts is easy enough to obtain. Well, so it is, but not with valve designs, which operate, almost always, in class 'A', at a maximum efficiency which will not exceed 25%.

So, for a stereo pair of 100 watt valve amplifiers, one is going to use and dissipate somewhere in the region of 0.8 - 1KW of power. This kind of kit is very expensive to buy, hot in use and too large to tuck away out of sight on a bookshelf. A further advantage of the solid-state approach is that if it is designed competently, and treated with reasonable care, it will have an exceedingly long service life - unlike the hot-glass devices, which will deteriorate continuously in use, and whose valves will need replacing after five years, and the electrolytics after ten to fifteen, if one wishes to retain the original performance of the system.

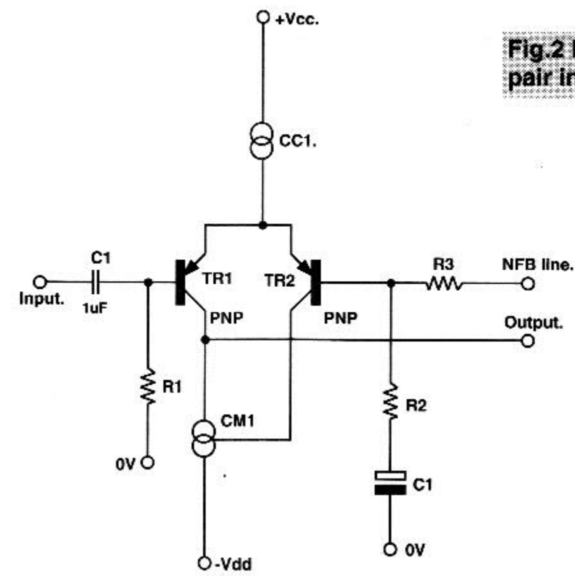


Fig.2 Long tailed pair input stage

But there are snags

Aluminium electrolytic capacitors, of the kind generally available in the 1960s, did not work well if they were operated (as in Figure 1) with zero voltage between their terminals. Under this condition they could, in due course, become both leaky and noisy. Also, they would go short-circuit if reverse biased. Tantalum capacitors were much better in this respect, and could tolerate both long duration zero voltage operation, in addition to occasional small (up to, say, 1V) reverse voltages, without breakdown, but they were expensive. So, what really gave the boost to the 'direct coupled' scheme was the development of the (then quite cheap) epoxy resin encapsulated Tantalum 'bead' capacitors, in which form a 100µF/3V device would be about the size of a pea, and cost in those days about 10p.

[However, a point which must be remembered here is that the components used in the feedback circuit (R3,R2,C2) are critical in determining the amplifier performance, and C2, in particular, must have the highest practicable electrical performance. Some ultra-Fi systems put up with the operating problems which arise from this decision, and delete C2 altogether.]

Another problem which cropped up, immediately the direct-coupled layout was adopted, (in which the large capacity output coupling capacitor was no longer used), was that of LS protection. Good quality LS units are expensive, and if one of the output transistors went 'short-circuit' the whole output of one or other of the power supply rails would pour through the LS speech coil, causing a loud noise followed by some expensive smoke. The simple answer usually adopted is to use a 'slow blow' fuse, or a DC offset-level sensitive relay 'cut-out' in the LS output circuit.

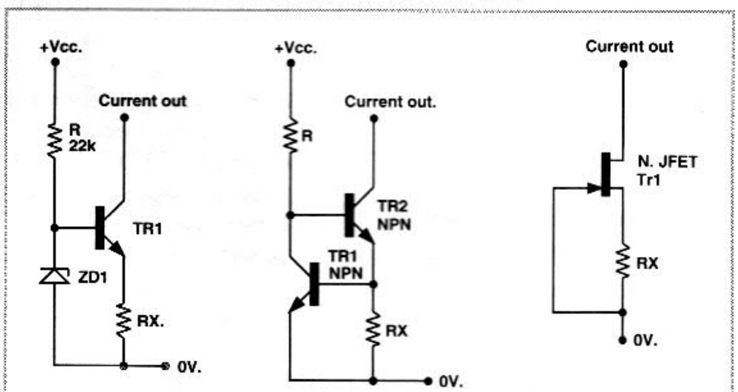


Fig. 3 Current source circuits

Since the mechanical contacts in the fuse holder or relay sockets may well become tarnished or corroded with time, better practice is to have some purely electronic current limiting circuitry to prevent possible LS damage, but even this is not always a good idea, since it can cause peak-clipping of the output signal - which is unpleasant to the listener. It is because such overload-protected solid state amps. will hard-clip on any over-voltage swing - however brief - which makes the average valve amp., with its gentle overload characteristics, seem, in use, to be more powerful than the theoretically more beefy transistor design.

Still, this is a quality which will mostly appeal to the 'heavy metal' and 'hard rock' brigade - especially if they like their music loud. Since they don't make very comfortable neighbours, it is lucky that there aren't too many of them.

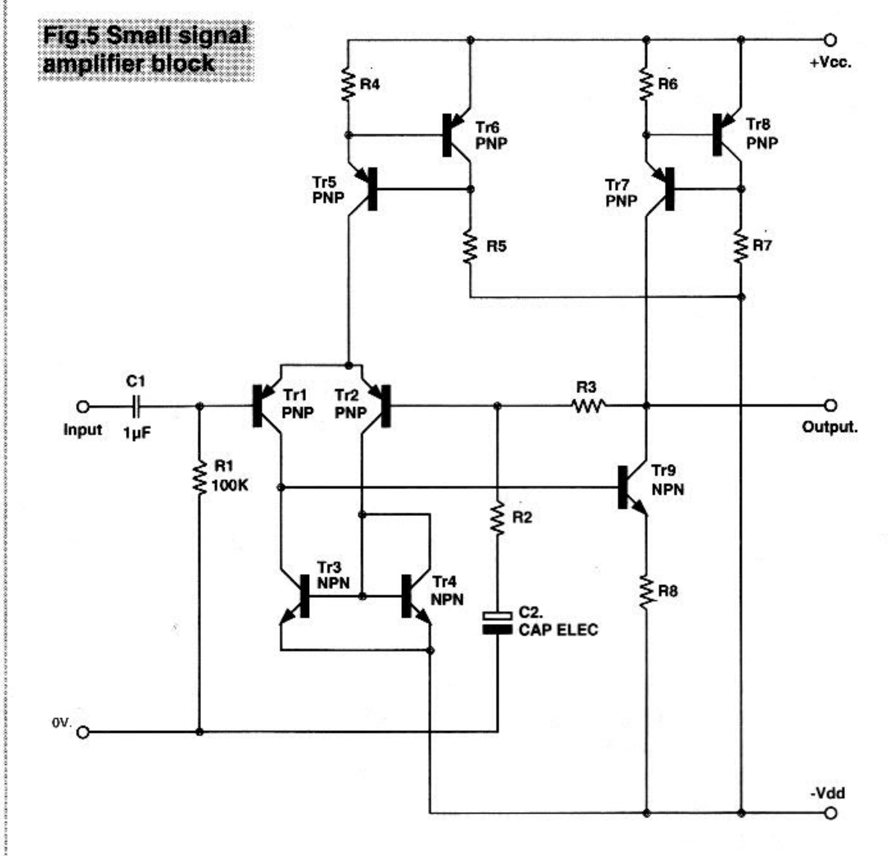
Circuit design techniques

There are, I suppose, four main objectives for the audio amplifier designer - reliability, pleasing sound quality, good technical specification and simplicity of design. Luckily these requirements are not incompatible, provided one doesn't go 'over the top' on specifications. The evolution of circuit design over the years has given the amplifier designer a wide range of circuit layouts from which to choose, which help improve circuit performance in various ways.

Long-tailed pair input stages

In general, small-signal (preamp. type) circuitry encounters few problems and is easy to design. Since the circuit layouts used in this, where discrete components rather than ICs are used, are very similar to those used in power amps. we can consider these together. As a rule, all modern designs, both for pre. and power amps, now use split rail (+/-V) power supplies, with both the input and output signals referred to a '0V' line, and this is most easily arranged by the use of an input 'long--tailed pair circuit' of the kind I have sketched in Figure 2. This layout has a much lower THD than a single transistor amplifier stage, and is balanced about the '0V' line - a feature needed in a direct coupled system. I have shown Tr1 and Tr2 as PNP transistors in this circuit, but it could equally well be turned upside down and built with NPN ones.

Fig.5 Small signal amplifier block



Constant-current sources and 'Current Mirrors'

The use of a high dynamic impedance 'constant-current source' (CC1) as the 'tail' is a good idea because it improves the integrity of signal transfer from Tr2 to Tr1, and it also helps to reject distortion, noise and 'hum' signals from the +ve line. The use of a 'current-mirror' as its output load combines the outputs from Tr1 and Tr2, doubles the circuit gain, and also helps exclude unwanted rubbish from the -ve line.

I have used conventional 'short-hand' symbols for the current source and current mirror parts of the circuit. These will usually be circuit arrangements of the kind shown in Figure 3 and 4. The circuit used in Fig. 3a is often used in commercial systems, but for a high dynamic impedance Rx must be moderately large, and this requires that ZD1 should be a zener diode, and zeners are noisy. The layout of Figure 3b is better, and is my own preference for this job. The junction FET circuit of Figure 3c, has an excellent performance, within the voltage limits imposed by the FET breakdown characteristics - usually 20-30V.

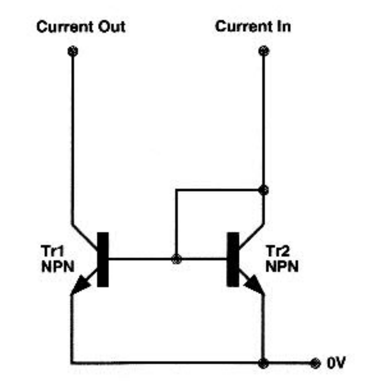
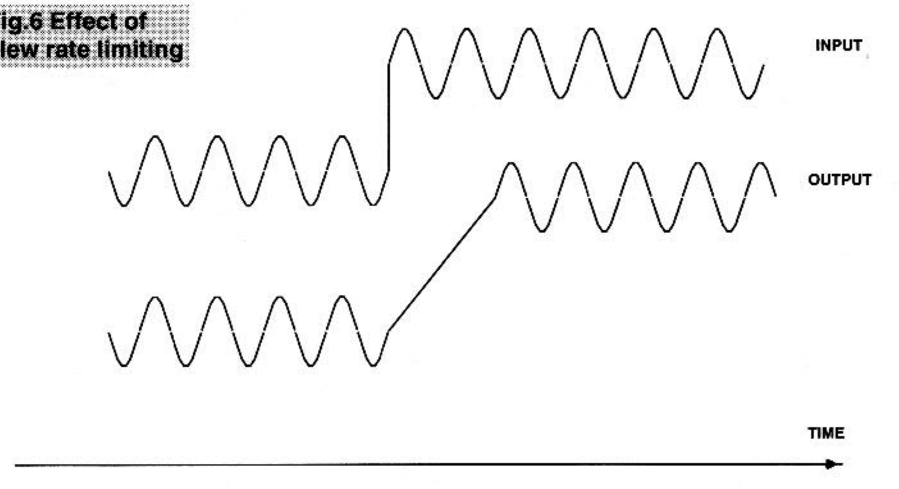


Fig.4 Simple current-mirror layout

The circuits of both 3a and 3b can be turned upside down, using PNP transistors, to operate towards a -ve voltage line. The circuit of 3c, since it is a two-terminal layout, works happily in either direction, provided the polarities are observed. In all these circuits, Rx determines the output current. Two-terminal constant current diodes, for fixed output currents, using similar internal circuitry to that of Figure 3c, are commercially available, with working voltages up to 100V, and with dynamic impedances of up to 10M or more, but they are dear.

The other useful circuit block is the 'current mirror', for which I have shown a simple circuit layout in Figure 4: an arrangement which acts to 'mirror' the input current at its output. Once again this circuit is available as a commercial device, offering various input:output 'mirror' ratios. This circuit makes a high dynamic impedance and

Fig.6 Effect of slew rate limiting



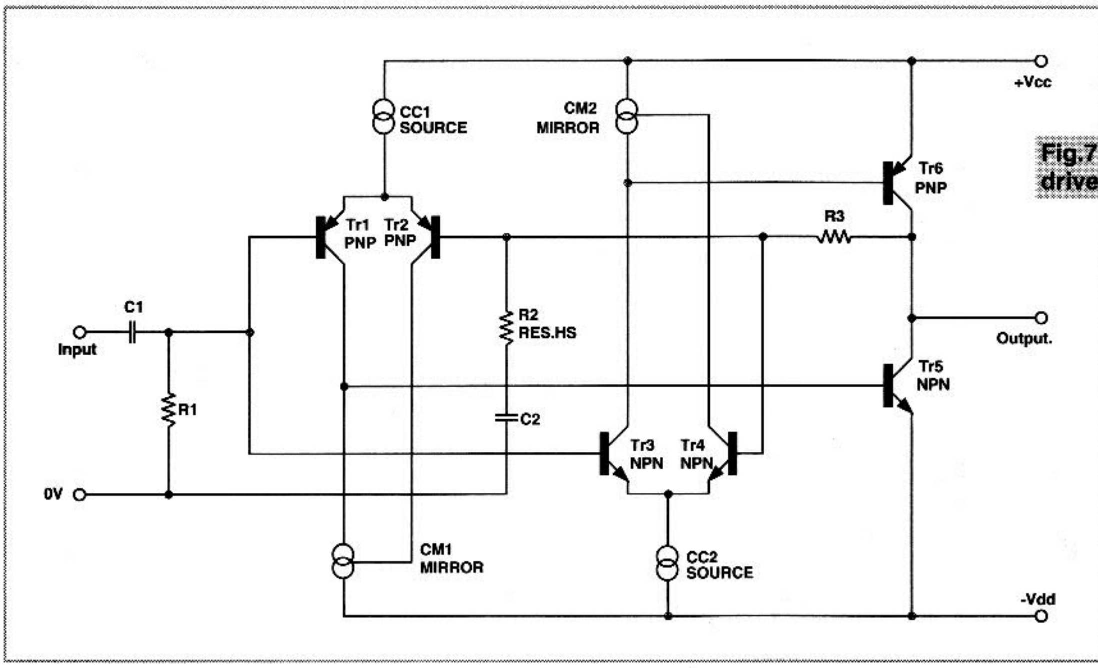


Fig.7 Symmetrical drive circuit

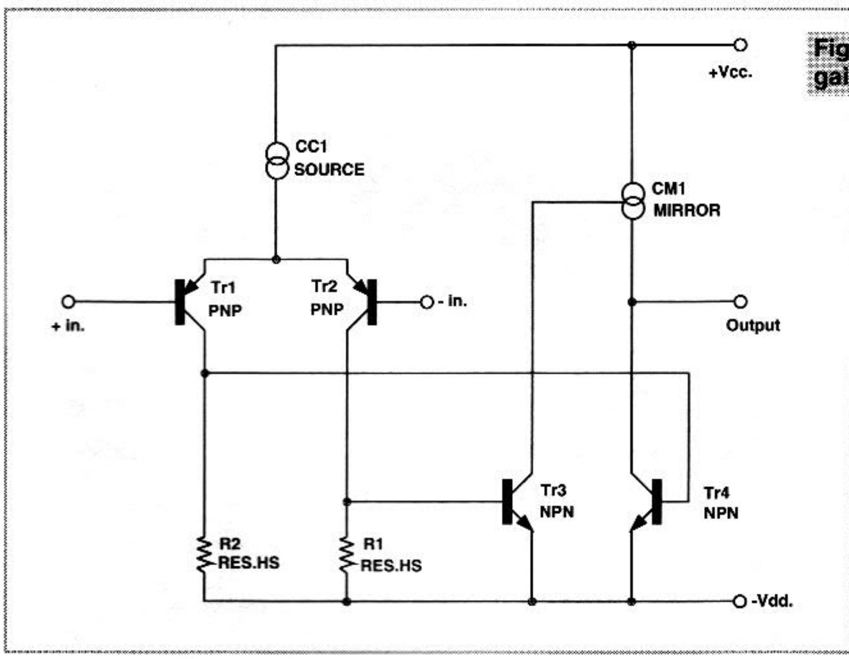


Fig.8 NS/Hitachi gain block circuit

high gain `active' load, particularly effective in improving the performance of a long-tailed pair, leading to the kind of circuit I have shown in Figure 5. I have elaborated this by the addition of Tr9, and a further constant current load, to make a complete high performance gain block, with an open-loop (i.e., before negative feedback is applied) gain of some 100,000x or more, and a very low distortion.

Once again, this whole circuit can be turned upside down with N instead of P devices, and vice versa. Anyone needing a high performance small-signal amplifier stage and not wishing to use an IC, (perhaps because the output voltage swing is too large for a normal IC to handle), could use this circuit without reservations.

Symmetry

A potential problem with all amplifier systems is that of 'slew rate limiting' - or Transient Intermodulation Distortion' as it is more romantically called - in which a large input signal can drive the amplifier into a condition, shown in Figure 6, when any smaller signal applied to the amplifier at the same time could be blotted out.

This defect could occur with any unsymmetrical circuit, such as, for example, the circuit of Figure 5 if there was a significant amount of capacitance connected across its output, since although this could be discharged rapidly when Tr9 was turned hard 'on', it could take a longer time to charge up again through its constant current load (Tr7/Tr8). In this case, the most straightforward answer would be to simply double-up the circuit with its mirror image, shown schematically in Figure 7, which is as fast to respond in the +ve going direction as in the -ve one.

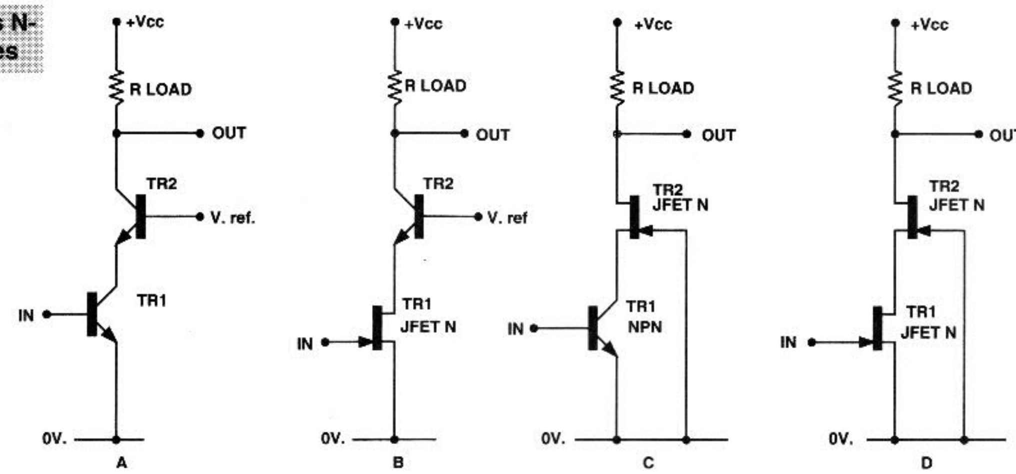
This layout has been used in some very good preamplifier systems, as well as in some excellent audio power amplifiers, with circuits developed by engineers, mainly working in the USA, such as Hafler, Borbeley and Bongiorno. However, the same symmetrical drive capability can be obtained more simply by the layout shown in Figure 8, first shown by National Semiconductors as an IC design, but used commercially by Hitachi in an audio power amplifier,

'Cascode' circuits

The other circuit tool available to the amplifier designer to increase bandwidth, reduce distortion, and isolate input from output, is the 'cascode' connection, shown for various bipolar transistor and FET combinations in Figure 9, and, once again, they can be inverted to use P-type devices if more convenient for the circuit. All these arrangements have the common advantage that the amplifier transistor (Tr1) operates into a low impedance, fixed voltage load, so there is virtually no feedback through the transistor. Also, since the cascode transistor (Tr2) is just a current amplifier, and feedback free, the stage will give a very high voltage gain, and a very low distortion if used with a high impedance load.

All the circuit layouts shown above are used by circuit designers, in various combinations, to produce very high

Fig.9 Various N-type cascodes



gain, low distortion gain blocks, and these have been used (in the position of 'A1' in Figure 1), as the basis for power amplifier designs, with some kind of impedance converting stage between the amplifier block and the LS output. Much patience is required to get these parts to live together.

Power amplifier design

As I have suggested above, the design of pre-amplifier stages, which are mainly just voltage amplifiers operating into resistive loads, is a pretty easy task, and a power amplifier - according to Figure 1 - is just a voltage amplifier with a pair of push-pull emitter followers hung on the end to reduce the output impedance to a low enough level for the system to be able to drive a loudspeaker adequately loudly. Unfortunately, this is where the real problems begin, and I'll look at these next month.

NEXT MONTH PROBLEMS,PROBLEMS,PROBLEMS.

Under Pressure

What's compact, easy to build and flattens you before giving you a boost? An Audio compressor by Daniel Coggins

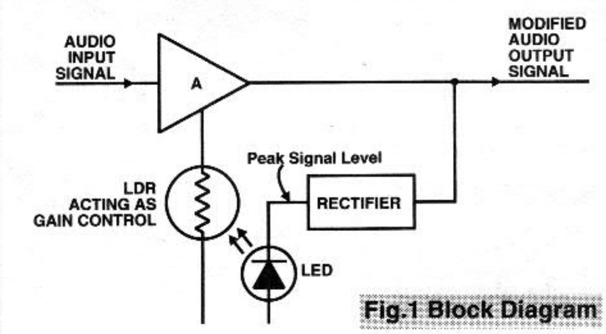


Fig.1 Block Diagram

Most readers with an interest in audio or electronic music will already be familiar with compressors. Basically a compressor (or 'sustain' unit as it sometimes gets called in the context of musical instruments) is used to flatten out signal peaks, as well as boosting the amplitude of low level signals. This effectively reduces the signals' dynamic range and gives the effect of making the signal sound subjectively louder, due to its higher average level.

This effect is frequently used in broadcasting for making advertisements 'stand out' and also as general signal conditioning to make certain daytime radio stations appear louder than others. It is also partly responsible for giving disc jockey's their 'magic' voices! The disadvantage of this is to reduce the 'light and shade' of the sound to the extent that one could argue that it becomes 'flat' and synthetic.

However, in broadcasting it is usually necessary to employ compression to protect transmitters from large signal peaks, though BBC Radio 3 uses much less compression than say, Radio One or Capital radio. When recording, compression can be used to get 'optimum' recording levels (as on some cassette players) or to 'tighten up' the sound of percussive or stringed instruments.

Design Considerations

This design was a result of some experimentation to produce a compressor with a suitable response that could be used with stringed amplified instruments in

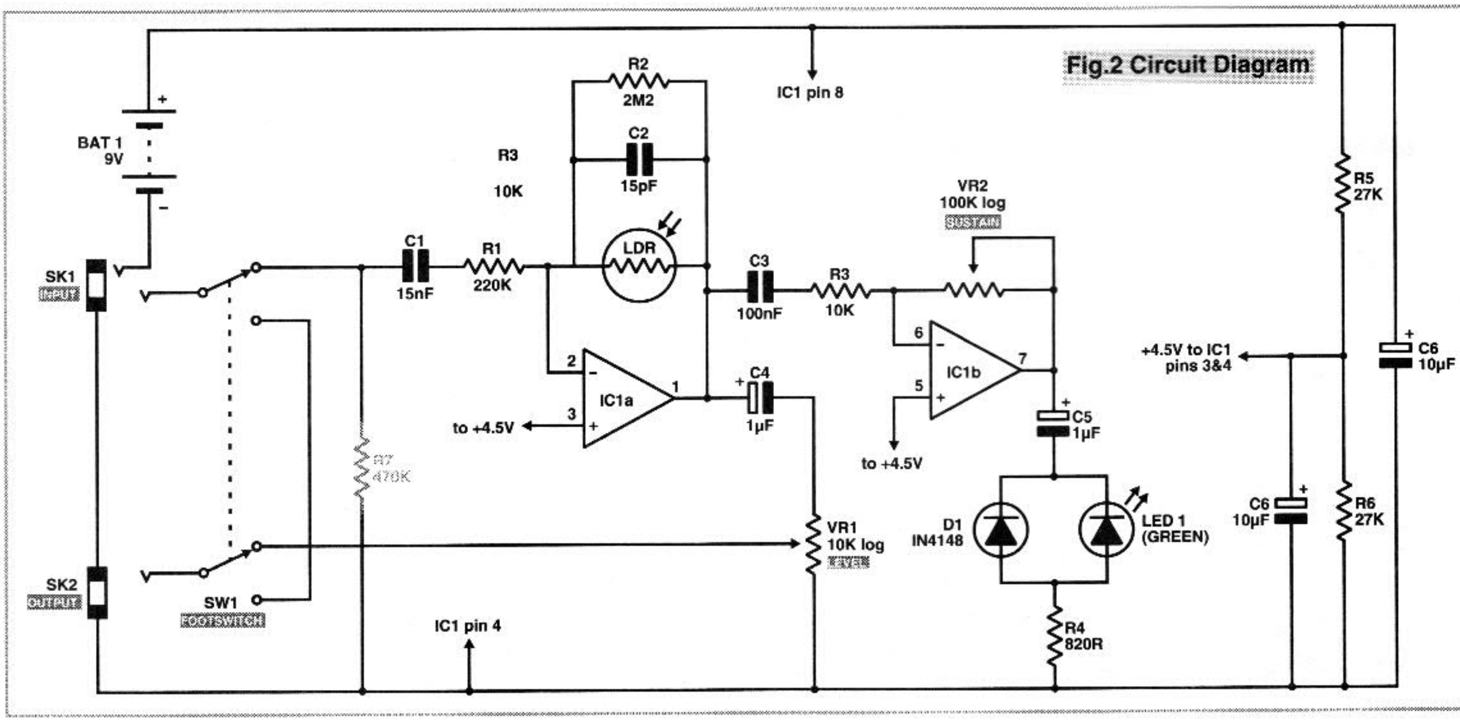


Fig.2 Circuit Diagram

The Works

Let us first look at what a compressor basically does, electronically speaking.

An audio signal (see Figure 1) is fed into an amplifier and the output of this amplifier is fed to a 'side chain' where the average peak level is detected (rectified) and used to control the gain of the aforementioned amplifier.

The final signal is then taken from this amplifiers' output. Various methods of controlling the amplifier exist: by using isolating transformers, or FET's as voltage controlled resistors - or by using an optical coupler, a nice solution which is used in this design. The great advantage of it is the minimal noise due to isolating the DC control voltage from the audio path.

So, if we feed a low level signal into the amplifier, the average peak level will be very low so the LED will not shed much light onto the LDR (light dependent resistor). Consequently, the amplifier will have high gain and will boost the low level to a much higher one, that is, until the peak level is sufficient to light the LED and effectively reduce the amplifiers gain, and so maintain a constant output level. The opposite process takes place for very loud signals.

It is possible to adjust the threshold at which these changes occur, simply by adjusting the amount of voltage to the LED. Because there is plenty of available gain, a level control/attenuator at the output is provided to enable the effect to be set to a wide range of levels. The attack and decay times are somewhat a function of the threshold in this circuit, and so it is best to call the control a 'sustain' control.

'Sustain' is often associated with the compression of notes from musical instruments, because with a guitar for instance, the note reaches a peak amplitude very quickly and then rapidly dies away to a much lower level. Using a compressor can make the note have a constant amplitude by inherently compensating for this. However, the note does die out eventually, as the compressor will only sustain a signal as long as it is there - and a guitar string does not provide a signal when it has ceased to vibrate. If infinite sustain is what you want - then use acoustic feedback in conjunction with a compressor.

order to 'thicken up' the sound and still retain the natural attack of the notes. Of equal importance was the release time which was tailored to be near-instantaneous in order to facilitate fast passages of notes or quick rhythms.

Other applications of this simple but effective design will be considered later, but as it stands it is intended for use with electric guitars, or electric basses - where it works very well. Using just one dual op-amp with an optical link and a few other parts, this has to be one of the simplest designs ever, but the noise spec. is quite reasonable, as is the current consumption.

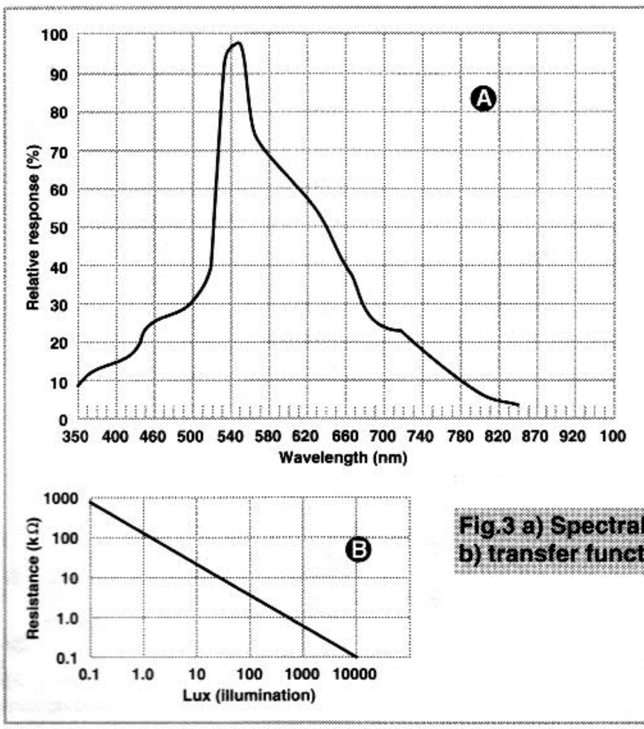


Fig.3 a) Spectral response b) transfer function of LDR

Circuit Description

SK1 is a stereo 1/4" socket configured to switch the 9V battery into the circuit when a jack is inserted. The supply goes to IC1 - a dual JFET low noise op-amp - and is decoupled by C7. This 9V rail is potential divided to 4.5V by equal value resistors R5 and R6, decoupled by C6 and provides an operating point for both op-amps non-inverting terminals.

The footswitch SW1 is used to bypass the circuit if so desired and does so by either connecting both 'in' and 'out' sockets together or by routing the signals through the circuit.

C1 couples the input signal via R1 to IC1a. This gives the circuit an input impedance of 220K - fine for the majority of guitar pickups. R2 assigns a maximum gain of 10 (20db) to this inverting amplifier stage. LDR1 is a miniature light dependant resistor whose resistance is extremely high in dark conditions (several megohms), and this resistance will decrease linearly as light levels increase (see Figure 3b). This varying resistance modifies the overall gain of the amplifier stage formed by IC1 - by connecting it in parallel with feedback resistor R2. C2 is used here to prevent RF breakthrough at high gain (which prevents the circuit from functioning properly - it is also very annoying!), and the chosen value gives high frequency roll-off progressively with increasing gain. This is an important consideration, as the noise level obviously increases as the gain of IC1a increases for low input levels (and zero level) and is predominantly heard as hiss, which is largely eradicated by this capacitors' inclusion.

The output of this stage is coupled to the footswitch via C4 and VR1 - the level control - and also through C3 and R3 to IC1b - another inverting amplifier with a variable gain of up to 20dB. However, the purpose of this stage is to raise the signal to a large enough level to drive the LED. C5 couples the audio signal to LED1 but (most importantly) is half-wave rectified by D1. Current limiting is provided by R4. The LED glows with proportion to the average

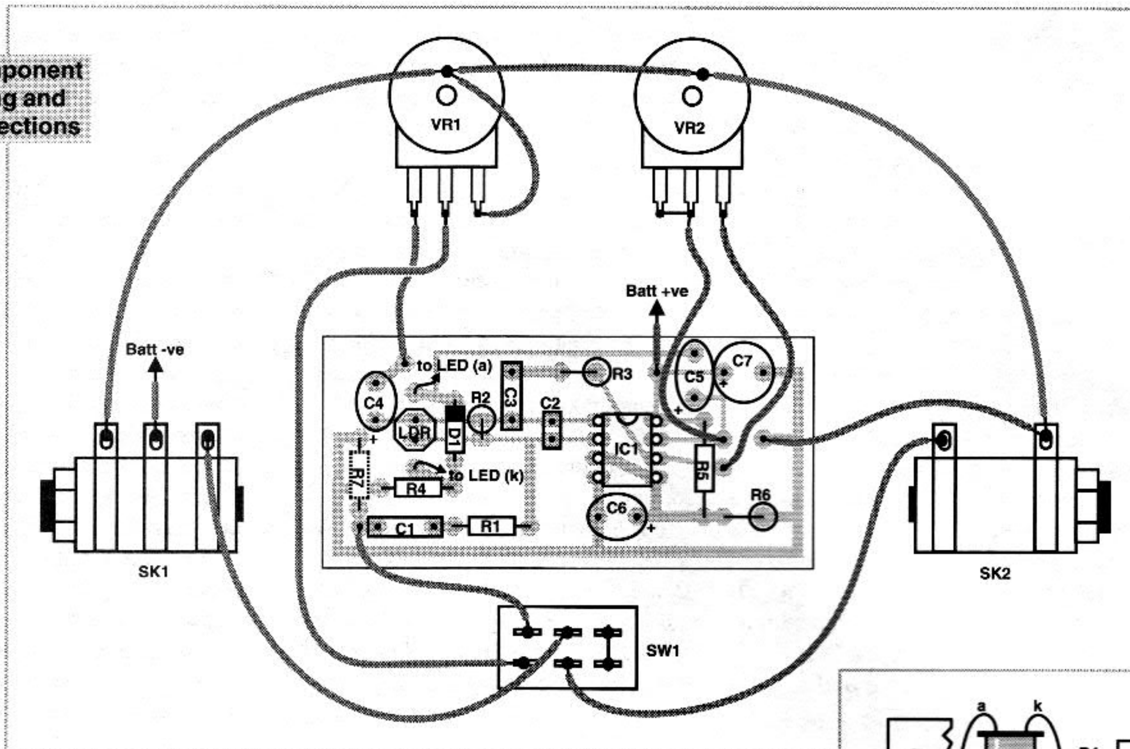


Fig.4 Component positioning and interconnections

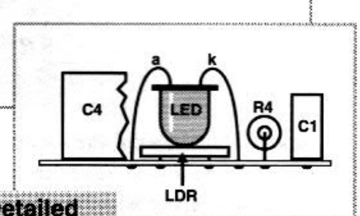


Fig.4b Detailed positioning of the LED and LDR

peak signal level fed to it, and just how much of this level is determined by the setting of VR2.

By placing the LED so that its' lens is illuminating the LDR's active face, the circuit will now operate as a compressor.

Why a GREEN LED? Because if we look at Figure 3a. we will see that the best spectral response for this particular LDR is about 550nm - which is approximately the wavelength of green light. Which goes some way to suggesting that red and yellow LED's will not give suitable results here, as their wavelengths are far outside of the LDR's peak spectral response. If only I'd known this when I threw away all those 'faulty' ORP 12's when I couldn't get them to work with red LED's!

Construction Details

Using the PCB available from EIA, check that it fits into the guide slots in the specified case. Then, in accordance with Figure 7a solder all the resistors and capacitors and D1 onto it, being careful to check polarities. Solder in IC1 and then the LDR - take great care not to overheat this sensitive component.

Figure 7b shows how the LED is mounted upside down so that it looks directly at the LDR. Take care when bending the leads, leaving about 5mm clearance from the LED case. Again, be sure to get the LED polarity correct!

Once satisfied that the PCB is complete, connect it to the remaining components as shown in Figure 4 - use some screened cable for connection 6 and connect one end of the screen to the case, to keep the noise down.

Drill out the case as shown in Figure 5 and use rub-down letters to label the controls/switch/sockets.

Several coats of clear lacquer will ensure that these words remain visible after rough handling - this is a foot-operated device!

Assemble the finished unit by putting the 'guts' into the case (neatly, of course!) and furnishing it with a fresh PP3 and a little foam rubber to prevent any internal movement in use.

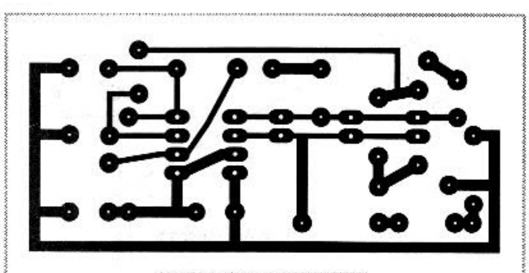


Fig.5 Foil Pattern

Testing and Use

Connect an ammeter between battery negative and case - it should read around 4-5mA. Gross deviations from this figure suggest that you should go back and check for any mistakes....

It might be worth pointing out that as this unit features a light dependant component that a light - proof case is essential for proper operation, but before you put on the lid plug your instrument into the input socket, turn up VR2 fully, dim the lights and make a few sounds - you should see the LED glow in harmony with you! If not, then press SW1 or check the LED's polarity again. Make sure it is at 90 degrees to the PCB. Assuming all is well, fasten the lid to the case, connect up to your amplifier and pluck a single note. It should be possible to hear the compression effect, but bear in mind that on low settings it will be quite subtle - and if SW1 is in the wrong position the thing will sound as normal, because it is!

Try playing soft and loud alternately - and slowly and quickly to appreciate the effects of compression. Higher settings of the 'sustain' control will give more dramatic compression and will require a corresponding high setting of the 'level' control. Low settings of sustain (up to around mid-way) will give a limiting effect, 'squashing' signal peaks, but not affecting the quieter signals as much. The 'squashed' sound can be quite expressive, especially as the LDR responds sluggishly at low signal levels and quite quickly at higher light levels. This is how it preserves the attack of the notes. This non-linearity gives the unit a sound not dissimilar to the old valve compressor/limiters of the 60s, and as such is quite subjectively pleasing.

As with any effect that can produce

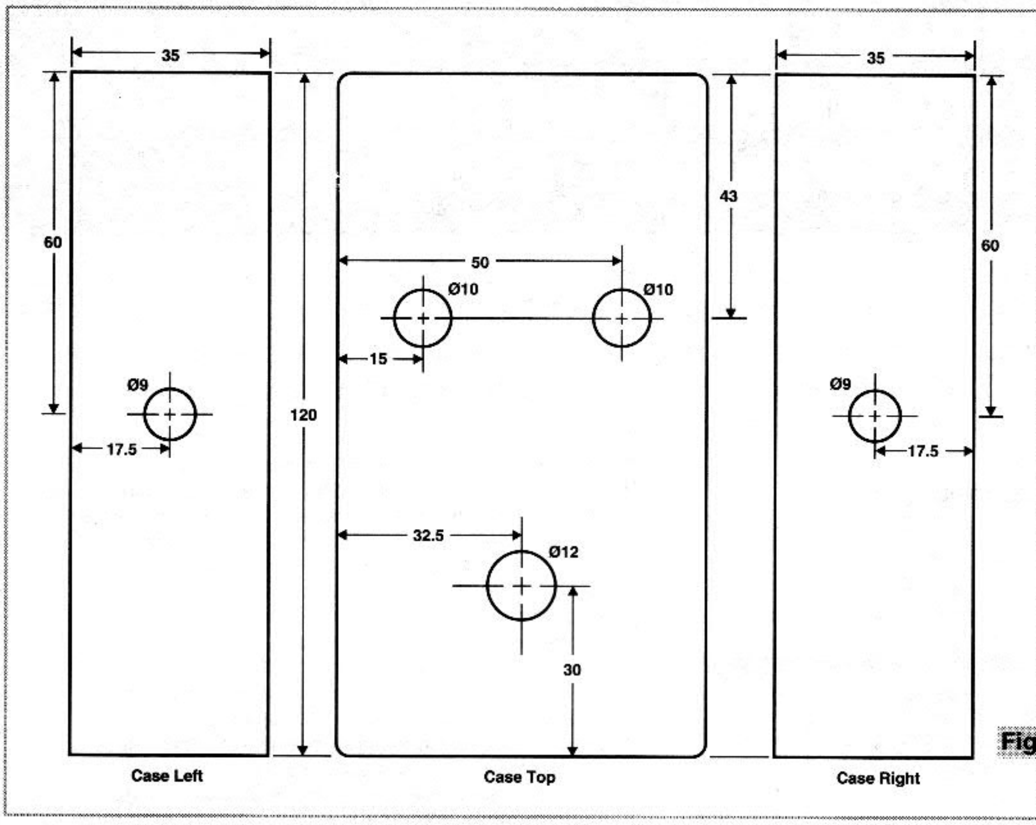


Fig.6 Drilling Details

high gain levels it is best to put it as close to the beginning of the signal chain as possible (i.e. the guitar), in order to avoid excessive noise. Remember that a compressor always seeks to amplify low level signals and will interpret noise as a signal, too.

Afterthoughts

It may be of interest to note that this design was largely the result of trial and error and the component values were chosen as a result of subjective listening tests. For instance, the value of C3 in conjunction with R3 gives significant bass roll-off as far as the side chain is concerned, resulting in greater compression at mid to high frequencies. The net result of this, particularly with a bass guitar is to reinforce the strength of lower notes, with slightly greater dynamics than for higher notes or harmonics. If you wish to try different values here or, for instance C5 / R4 / R2 / C2 / R1 / C1, feel free to do so - it is in this way that you can make the sound work for YOU. Audio *is* subjective and of course so is music, but technology and all the rules and equations that it carries with it must only be a means to an end. Avoid 'paralysis by analysis' and experiment!

In a simple circuit such as this, there is a lot going on interactively, which can lead to more complex and interesting results. Should the footswitch in this design produce any 'clicks' or 'thumps', connect a 470K resistor (R7) shown as dotted on the overlay, between C1 (switch side) and ground - this should minimise it.

By combining this design with that of last months' A-B BOX you can make a compressor with greater reliability, quieter and with LED status indicators.

You may wish to modify the side-chain by simply connecting a large capacitor across LED1 to modify the release time for putting say, speech or music through this unit - but that is somewhat crude and would best be expanded upon..... ....over to you!

Reference

RS data sheet F14188.

Parts

Resistors

(all 1/4W 5% carbon)

R1	220K
R2	2M2
R3	10K
R4	820R
R5,6	27K
R7	470K optional
VR1	10K log potentiometer
VR2	100K log potentiometer

Capacitors

C1	15n mylar/polyester
C2	15p ceramic plate
C3	100n mylar/polyester
C4,5	1µ/16V radial electrolytic
C6	10µ/16V radial electrolytic
C7	47µ/16V radial electrolytic

Semiconductors

IC1	TL072 Dual J/FET OP AMP
D1	1N4148 silicon diode
LED1	Green Low Current 5mm LED
LDR1	Light Dependent Resistor type NSL 19 - M51 (RS stock No.596 - 141)

Additional Components

B1	PP3 9V battery and clip
SK1	stereo 1/4" skeleton jack socket
SK2	mono 1/4" skeleton jack socket
SW1	heavy-duty DPDT footswitch

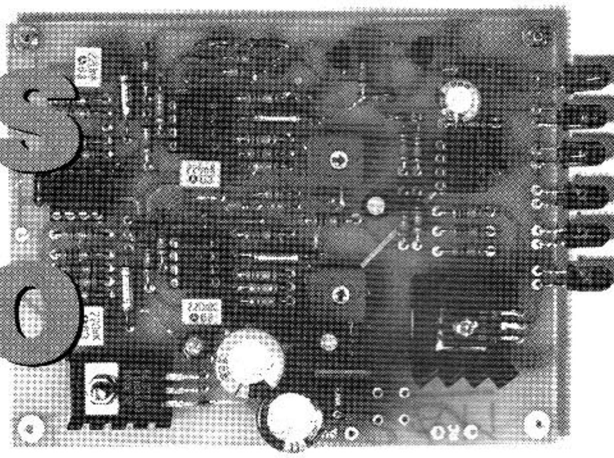
Die cast box - 120 x 65 x 40 mm (bimbox type 5004)
PCB (see page 61)
Solder/wire/screened cable
rub-down letters/spray lacquer
Knobs to suit VR1
Foam rubber anti-slip base

Buying Tips

The LDR can be obtained from ELECTROMAIL (RADIOSPARES mail order subsidiary - Tel. 0536 204555. The footswitch and diecast case can be obtained from Maplin (order codes FH93 and LH71 respectively)

Cordless Audio

Part 1



Andrew Armstrong helps us keep our neighbours friendly with his Infra red stereo audio transmitter

As Britain becomes more crowded, and newer housing is more cramped, the sound of heavy metal played through loudspeakers at high volume becomes less acceptable. Listening on headphones is a good answer, but the cable tends to get snagged and people trip over.

The answer is to use an infra red headphone link. Unfortunately the ones I have seen on sale are only mono, and intended for use with a television, so I have designed a stereo transmitter to use with a hi-fi system. Obviously a direct cable connection will offer better sound, but the design here is intended to provide a good quality link for use when a cable is impractical.

Modulation

There are several ways to send sound information via infra red. The simplest is to modulate the brightness of the led directly with the sound signal. The most obvious disadvantages of this approach are that LEDs are not perfectly linear, that infra red in the environment will interfere directly with the signal, and that this method makes it difficult to provide stereo modulation.

The linearity problem can be solved by using pulse width modulation, but a much better answer is to learn from radio transmission standards and to frequency modulate a high frequency carrier. This can be reasonably linear, and it avoids most interference problems. One example is infra red from a light source. It will contain 100Hz and near harmonics thereof which are far below any reasonable carrier frequency.

Stereo

A number of years ago I read that there was a Standard for infra-red stereo headphones which used 95KHz for one channel, and 250KHz for the other. I do not know if there are any infra-red headphones to this standard manufactured now, but in case there are I will use these frequencies in order to be compatible.

The frequency modulated carriers are provided by two voltage controlled oscillators. The LM566 is a reasonably

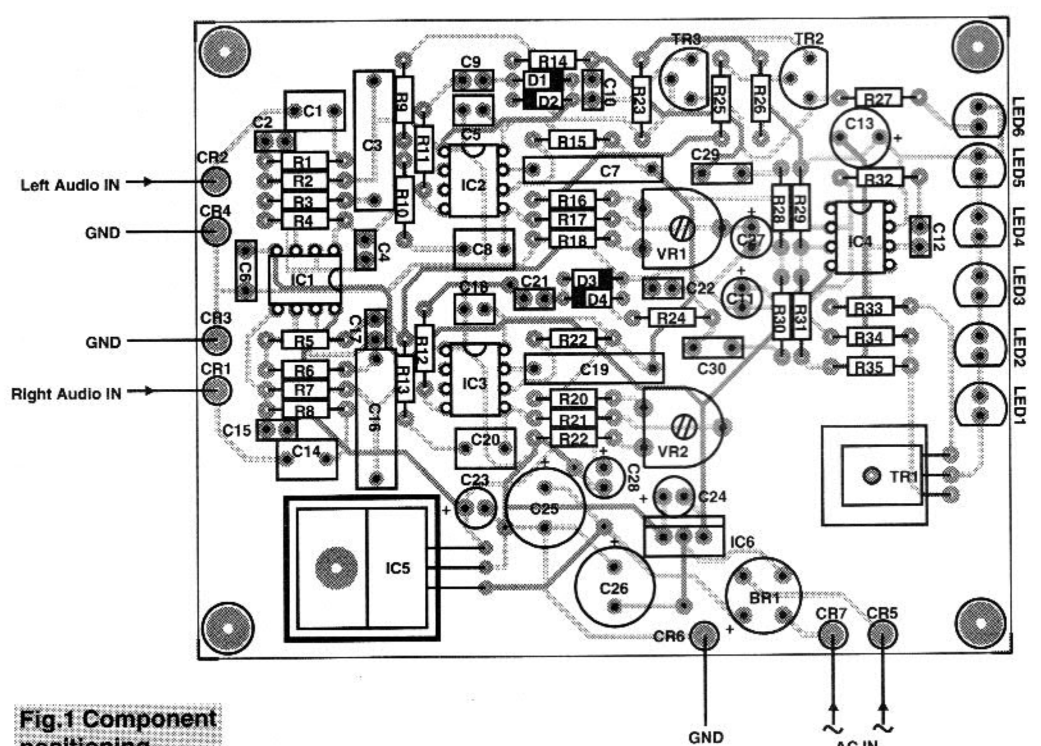


Fig.1 Component positioning

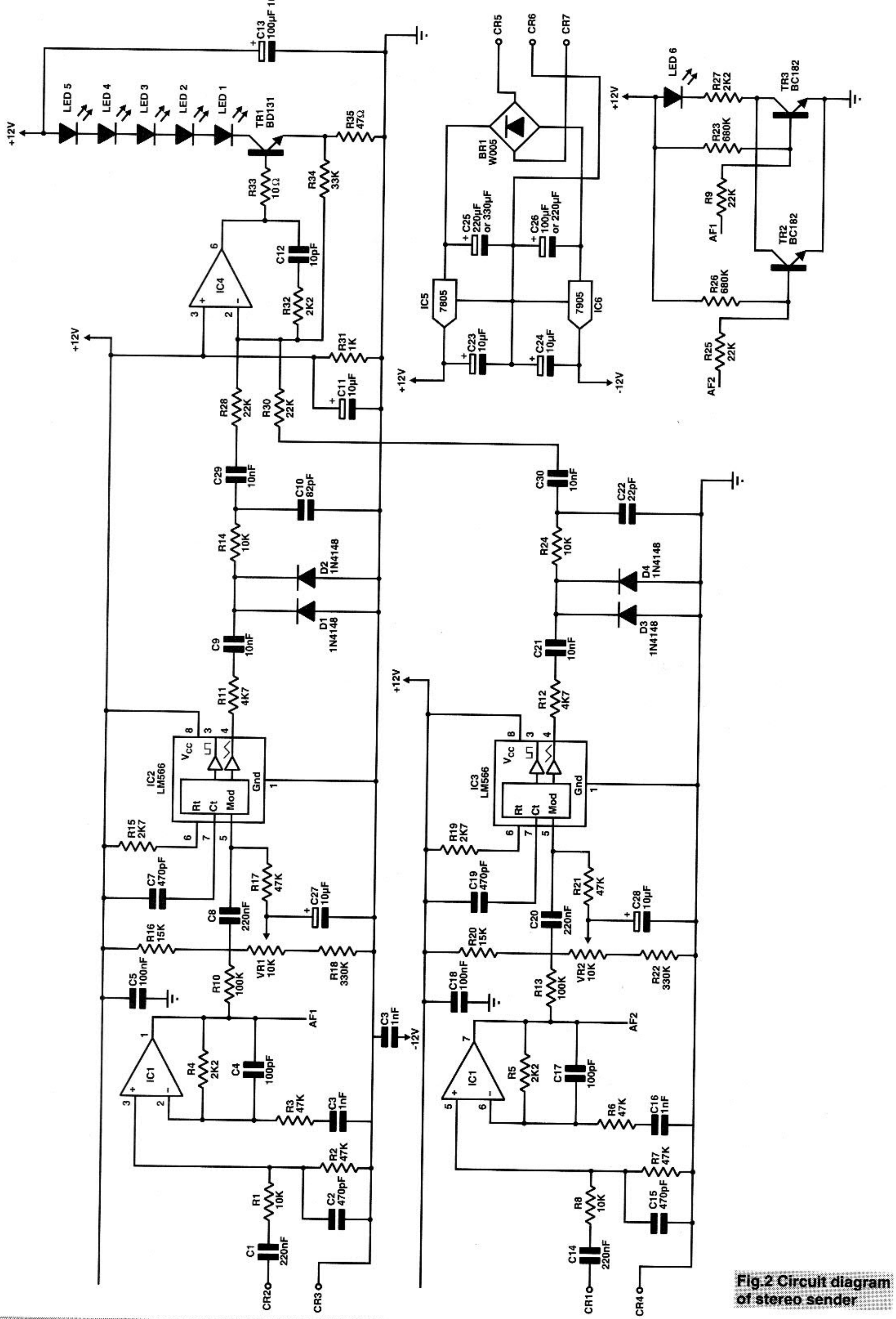


Fig.2 Circuit diagram of stereo sender

priced VCO which covers the required frequency range, and is available from Cirkit, so I chose that for this project.

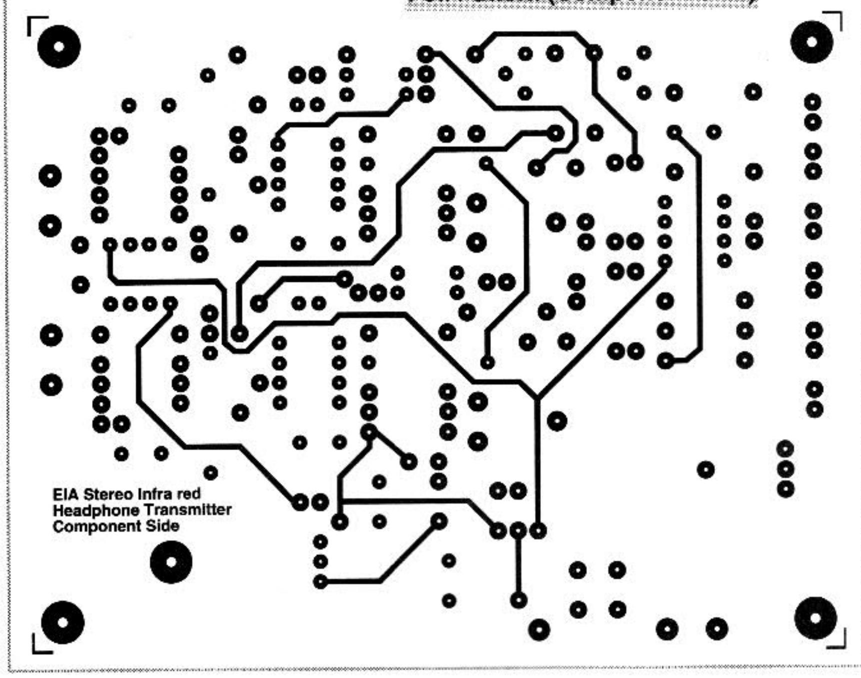
One way to transmit the left and right signals would be to use two separate strings of LEDs, but economy and elegance suggested that a single string should be used. Simple square wave modulation was then ruled out because, if one modulating signal is on and the other off, should the LEDs be on or off? Linear addition of the squarewave drives might work, but experience has taught me that in a potentially non-linear system, in any situation where beats between frequencies or harmonics can occur, they probably will, and in so doing will spoil the performance of the system. Accordingly, I decided to generate sine-wave drive signals if it could be done without too much complexity.

The Signal Path

Starting at the input, for the left-hand channel, the audio signal is filtered to remove any RF which may be present, and fed to a low-noise op-amp which buffers the signal. The op-amp circuit also provides pre-emphasis to bring the high frequency content of the modulation more in line with the middle frequency content. De-emphasis can then be applied at the receiver, resulting in a lowering of the noise level.

This buffered output is fed to the modulation input of the VCO, via an attenuating network comprising R10 and R17. Because the left-hand side of R17 is decoupled to ground, it works as a potential divider with R10 at audio frequencies. The buffered signal is AC coupled rather than DC coupled to the modulation input of the VCO, because the bias voltage on the modulation input is used to set the exact VCO frequency.

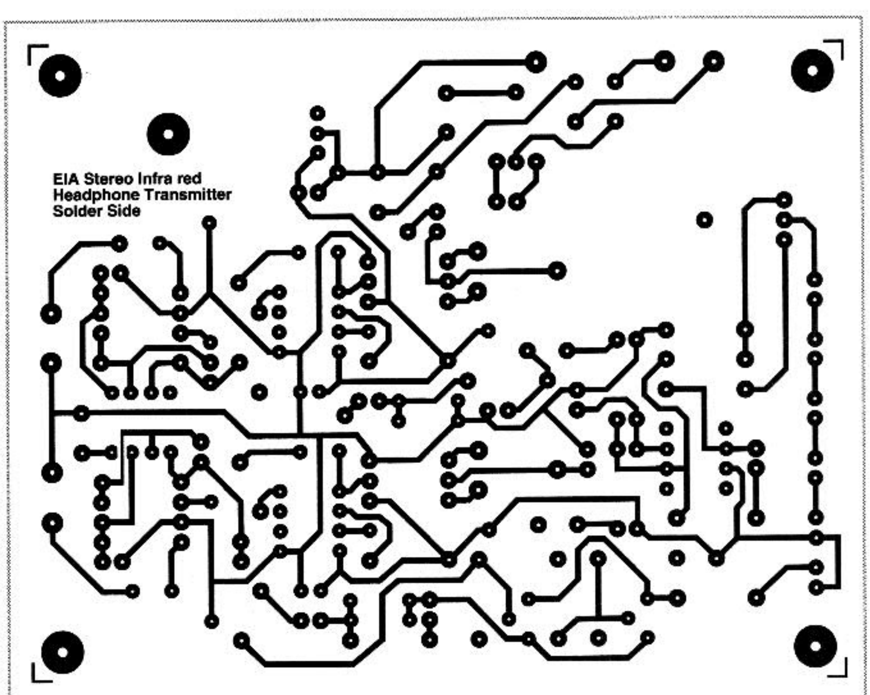
Foil Pattern (Component Side)



The triangle wave output from the VCO is fed via R11 and C9 to a pair of back-to-back diodes. The resistor value is chosen so that the diodes will gently round off the peaks of the triangle wave, and it is AC-coupled to make this work symmetrically. Some of the remaining harmonics are removed by the RC filter comprising R14 and C10. The resultant signal when displayed on an oscilloscope appears to be a good sine wave.

Signals from the two channels are added in the high frequency op-amp which drives the output transistor. The function of this stage is to vary the current through the light-emitting diodes linearly with the input signals, and tests show that it does this effectively. The op-amp chosen for the job is unity-gain stable, but it cannot be guaranteed to be stable if its output is fed through a further active device before feedback is applied. Any instability will only manifest itself at frequencies of several megahertz, so the gain of the op-amp stage itself is reduced at frequencies well outside the operating range of the circuit by C12 and R32. As a further precaution to keep the circuit stable, a 10 ohm resistor is placed in series with the base of Tr1, to prevent oscillation caused by the strange and complex impedances which transistors can sometimes exhibit on their terminals.

This stage draws a significant current, so in order to avoid unnecessary disruption of the power supply voltage to the rest of the unit, local decoupling is provided by C13. The ground connection of C13 is directly routed to R35 on the PCB layout to avoid coupling output signals into an earlier part of the circuit via common ground resistance.



Foil Pattern (Solder Side)

Ancillary Bits

On-board voltage regulators are provided in order to guarantee a clean power supply. Since the regulators are there anyway, the rectifier and smoothing capacitors are also fitted to the board, so the only extra component required is a mains transformer.

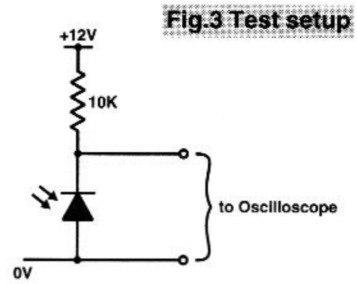
An indication of the correct signal level for a reasonable modulation index is provided by the circuitry comprising Tr2 and Tr3. This circuit flashes the LED if either of the buffer amplifier outputs swings positive enough to bias the transistor On. Considering the left hand channel, Tr3 has some bias provided already by R23, so that a 200 millivolt positive excursion on IC1 pin 1 will serve to light the LED. Initial tests showed that this level of signal gave a reasonable amount of modulation, but should the indicator need to be changed, the value of R23 can be changed.

Assembly and Testing

The PCB for this project is double sided, and the first thing to do is to insert and solder the links between top and bottom of the PCB. Track pins are ideal, but bits of clipped component lead will do. Separate pin throughs are provided for components which obscure their top connections, but ones whose top connections are accessible should be top soldered.

Several IC connections are top soldered, which is not clear on the photograph of the prototype because I used sockets in developing this design. IC sockets are not easy to top solder, so I used the Multicore Copperset through hole plating system to make these connections.

Fig.3 Test setup



If all the top soldered connections are made, the infra red LEDs are the only other item needing special attention. Their polarity is not obvious, and their illumination or otherwise cannot be seen. My solution to the problem was to set the lab power supply to 4V, and connect the infra red LED to the supply via a 1K resistor. Measuring the voltage across the LED indicates polarity, as the LED conducts at about 1.1V, but is capable of blocking 4V with no trouble.

When assembly is complete, test the power supply first, initially using a current limited lab power supply to feed +/-15V into the reservoir capacitors in case all is not well. Assuming that you measure +/- 12V on IC1, the next step is to set the frequency of the voltage controlled oscillators. Connect a frequency meter to pin 3 of IC2 and adjust RV1 to set the frequency to 95KHz. Then measure on pin 3 of IC3 and adjust VR2 for a frequency of 250KHz.

Without a receiver, further testing is limited. If you purchase the infra red diode which the receiver will use, then you can test that the transmitter is transmitting by using the circuit of Figure 2 to detect the modulated waveform. This will only work close to the transmitting LEDs - the signal from the receiving diode will need a lot of amplification to work in the receiver.

Finally, if you connect the oscilloscope across R35, and apply a signal to each input in turn, you should be able to see the frequency of the waveform change. Then you will have got as far as is possible until the receiver is built.

Parts

Resistors

R1,8,14,24	10K
R2,4,5,7,17,21	47K
R3,6,27,32	2K2
R9,25,28,29,30	22K
R10,13	100K
R11,12	4K7
R15,19	2K7
R16,20	15K
R18	390K
R22	150K
R23,26	680K
R31	1K
R33	10R
R34	33K
R35	47R
VR1,2	10K preset
VR3	10K log stereo potentiometer

Additional Items

Heatsinks for TR1 and IC5
M3 nuts and bolts for same.
Receiver diode for testing (and later use in the receiver) Electromail 195-574

Capacitors

C1,4,8,14,20	220n polyester
C2,15	470p ceramic
C3,16	1n polystyrene
C4,17	100p ceramic
C5,6,18	100n ceramic or polyester
C7,19	470p polystyrene 2%
C9,21,29,30	10n ceramic or polyester
C10	82p ceram
C11,23,24	10µ radial electrolytic
C12	10p ceramic
C13	100µ 16V low esr electrolytic (eg Electromail part 105-903)
C22	22p ceramic
C25	220µ or 330µ 25V radial electrolytic
C26	100µ or 220µ 25V radial electrolytic

Semiconductors

D1,2,3,4	1N4148
BR1	W005
LED1,2,3,4,5	Infra red LEDs eg Electromail 635-296
LED6	red LED
TR1	BD131
TR2, 3	BC182 or BC108 or similar
IC1	NE5532
IC2, 3	LM566 (available from Cirkit)
IC4	LM318
IC5	7812
IC6	7912

The Receiver

As it stands, this is no use without a receiver. The receiver design will be published in Part Two. Currently I am torn between using a pair of phased-locked loops to demodulate the signals, or using a conventional FM detector, possibly using an IC intended for an FM radio. It may take more than a month to choose and design a good-enough system, and if the receiver design does miss the next issue, I apologise to those who have to wait for the following month.

the Alchemist

Part 2

Mike Meechan now gets to grips with this hi-fi pre-amp project

Last month, we looked at the mechanics involved in getting sound off the vinyl disc via the moving coil-type cartridge. This month, before unravelling the secret of The Alchemist, let us look at an enemy which is ever-present in all audio systems, that of noise.

Noise

Since any RIAA equalisation is done post-front end, as far as moving coil signal conditioning is concerned, it is a matter simply of straight amplification. Or is it..? Let us look closely at what is involved.

In any electrical system, and before we even think about man-made noise or interference, naturally-occurring noise lurks in there somewhere, even if it's below audible levels or above the audible upper frequency ceiling. We must appreciate a basic understanding of noise before any efforts can be made to minimise effects which it might have on an electronic system intended to amplify, with a great deal of integrity, a low-level signal which may indeed originate from a vinyl record groove. Many forms of noise are naturally-occurring, whilst others are man-made. Good design techniques, with judicious input filtering, shielding of sensitive areas of the circuit etc. can just about eliminate the effects of man-made noise. Naturally occurring noise, however, is another matter.

Where very low level signals, high amplification factors, and specific working impedances for optimum performance are required, the problems caused by noise worsen by a large order of magnitude. We're now going to look at the mechanics of noise, and what can be done about it before we put the theories into practise.

The Physics of Noise

Noise Figure and EIN

It can be illustrated using statistical mechanics and the laws of thermodynamics that resistors will contribute noise because of thermal activity. It is a naturally-occurring phenomena, and is generated irrespective of how perfect the

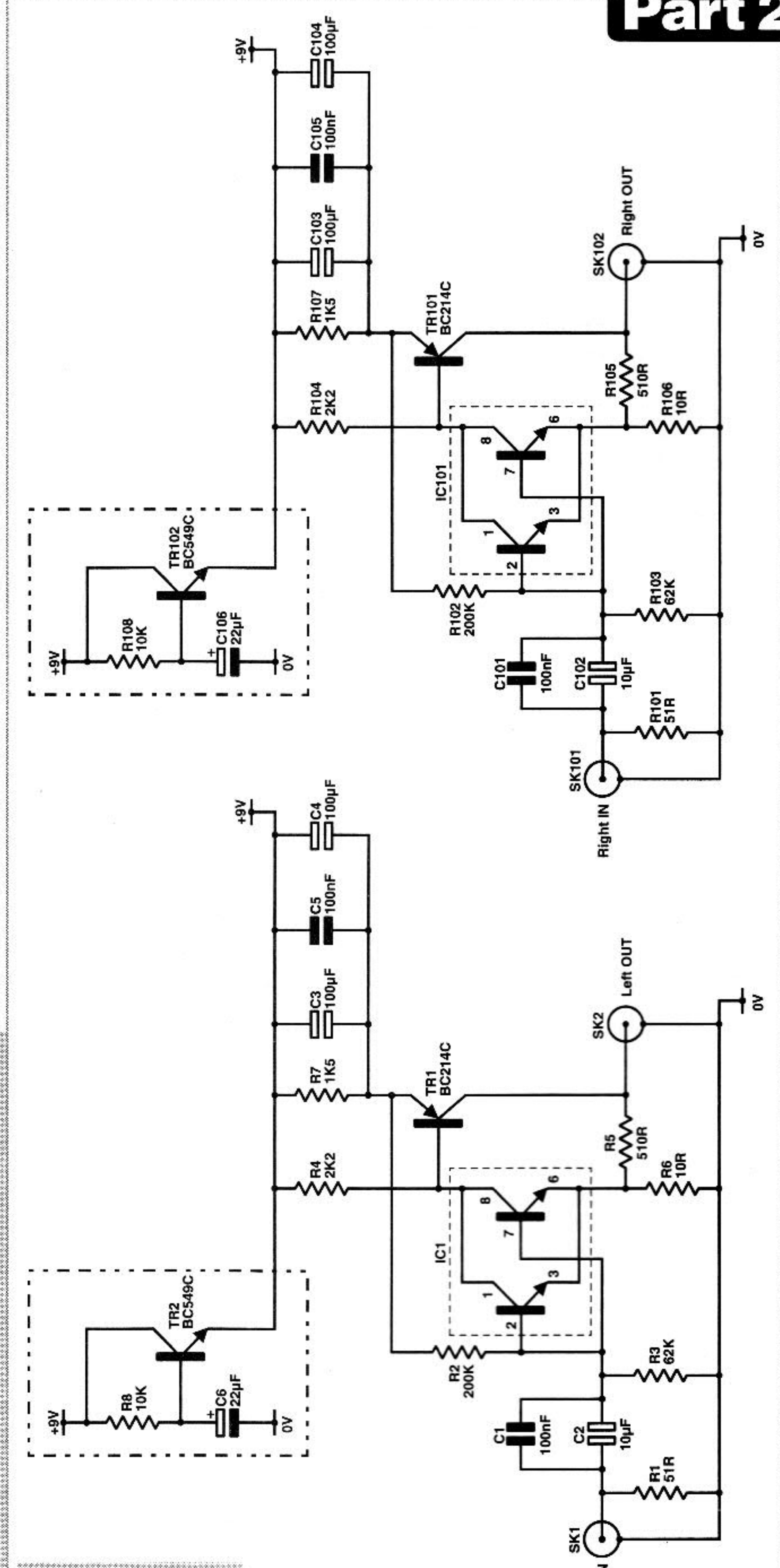


Fig.1 Circuit diagram of Alchemist

resistor is. Noise must be added to any signal dropped across a resistor, irrespective of the quality of the resistor. Since they are an integral part of amplifier design and application, it is as well to understand, from the onset, what physical limitations they place upon the absolute lowest value of noise which can be achieved from a given circuit.

Noise from resistance in series with the amplifier input is generated according to the following equation;

$$e_n=\sqrt{(4KTBR)}$$

where

K is Boltzmann's constant (1.38 x 10^{23} J/K^{-1})
T is the absolute temperature in **K**, and is typically quoted at room temp (290K)
B is the noise bandwidth in Hertz
e_n is the amplifier noise voltage V/√Hz
R is the resistance in ohms

The coil resistance of a moving magnet cartridge is around 500R, (although it can be as much as 2K) while the moving coil's is around a tenth of that, although some have resistances as low as 5R. The noise voltage is normally quoted in nV/√Hz. The value of a 50R resistor is calculated as shown below;

$$E_n = \sqrt{4KTBR}$$
$$= 4 \times 1.38\times10^{-23} \times 298 \times 50 \times 20 \times 10^{23}$$
$$=128nV$$

This figure is known as the thermal noise or so-called Johnson Noise, named after J.B. Johnson who first explained it in 1928 when he presented his paper, 'Thermal Agitation of Electricity in Conductors'. The significance of Johnson noise is that it sets a lower limit on the noise voltage on any signal source or amplifier having resistance.

Another naturally-occurring, irreducible form of noise is 'shot noise', which is caused by statistical fluctuations in the current. When referred to the standard reference level of 0.775V (0dBu), noise from a 500R resistor (moving magnet cartridge) is as follows;

$$20\log\frac{0.775}{128 \times 10^{-23}}$$
$$= -135.6dBu$$

Unfortunately, when the input signal is amplified, so is the noise, and if the ratio of signal power to noise power is the same at the output of the amplifier as at the input, it is said to be 'noiseless'. With a typical output of around 0.5mV from the MC cartridge and a gain of around 64dB from the amplifier, signal-to-noise is typically 72dB, dropping to about 52dB with very low output cartridges (50µV). This is the best which can be achieved, and assumes an amplifier which adds no noise of its own, which is a practical impossibility. Real amplifiers degrade this ratio by injecting noise of their own, and this is in addition to any present on generated by the load resistor at the amplifier input.

The degree of this noise impairment is known as the noise figure of the amplifier and is expressed as a ratio in the form;

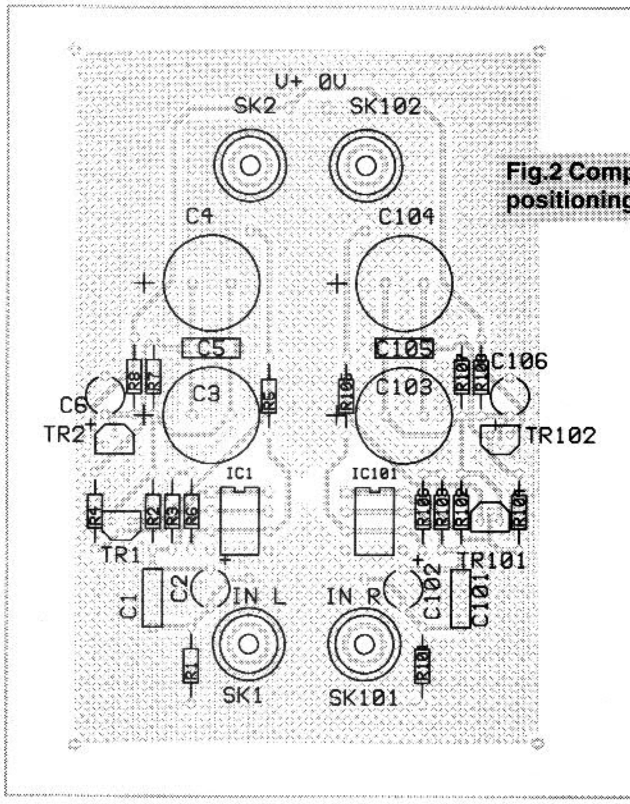


Fig.2 Component positioning

$$NF=SIG_{out-} \times N/SIG_e \times N$$

and NF dB = 10 log (NF of the power ratio)

These other noise sources are caused by 'additional' noise sources such as 1/f or 'flicker noise', which we'll discuss later. In an MC preamplifier application, keeping the noise figure as near the theoretical minimum is of paramount importance, since the noise figure is the amount by which the Equivalent Input Noise (EIN) is higher than the thermal noise of a resistor of specified value - or the cartridge - which would normally be connected to the amplifier input. Noise

The Works

It has been designed for source impedances below 250R so the collector current for the SSM2210 should be around (or higher than) the 2.5mA mark. Regrettably, as we've already shown, rbb, even at 28R, is the limiting factor on noise at current levels such as these. In order to achieve better performance in this respect, the two halves of the SSM2210 are paralleled to reduce rbb by a factor of 2 to around 14R.

Total input voltage noise for this design can be calculated as follows: -

$$e_n = \sqrt{4KT(r_{bb}+R3)} + e_{n^2}$$

Numerically, this equates to:-

0.623nV/√Hz

The current noise is 5pA/√Hz which flows through a son (typical) cartridge source resistance and causes an additional 0.25nV/√Hz. Since the Johnson noise of a 50R resistor is 0.9nV/√Hz, the noise figure is:

NF = 1.9dB
0.9nV

The circuit has only one internal capacitor, C1. This functions as an AC bypass for both stages. There is also no input stage load resistor bypassing, yet there is 56dB or better of supply rejection referred to the input. In spite of this, the constructor can add the optional supply filter of C3 and R7 (shown inside the dashed section of the circuit diagram) which improves matters in this respect by 50dB. Noise injected from power supply lines is an often-overlooked aspect of low noise designs, probably due to the inherent high rejection qualiies of operational amplifiers in this respect (attributed, in the main, to good PSRR and differential inputs), and their widespread use in this field.

The single-ended nature of many low-noise designs means that they do not enjoy the inherent supply rejection of differential designs. Where the load resistor is tied directly to the power supply, noise from this supply must be no higher than $(R_L I_c V_n)/(3kT/q)$ or the overall noise performance of the system will suffer.

Normally, AC coupling of the 10R feedback resistor is a problem. It is eliminated here because a DC biasing scheme is used. This biases both stages simultaneously without relying on feedback from the output.

figures of 3dB or less can be assumed to be good, and those of 1dB or less, excellent. EIN can also be specified as the noise measured at the amplifier output plus the amplifier gain when the input is terminated with a resistor of the nominal value of cartridge impedance.

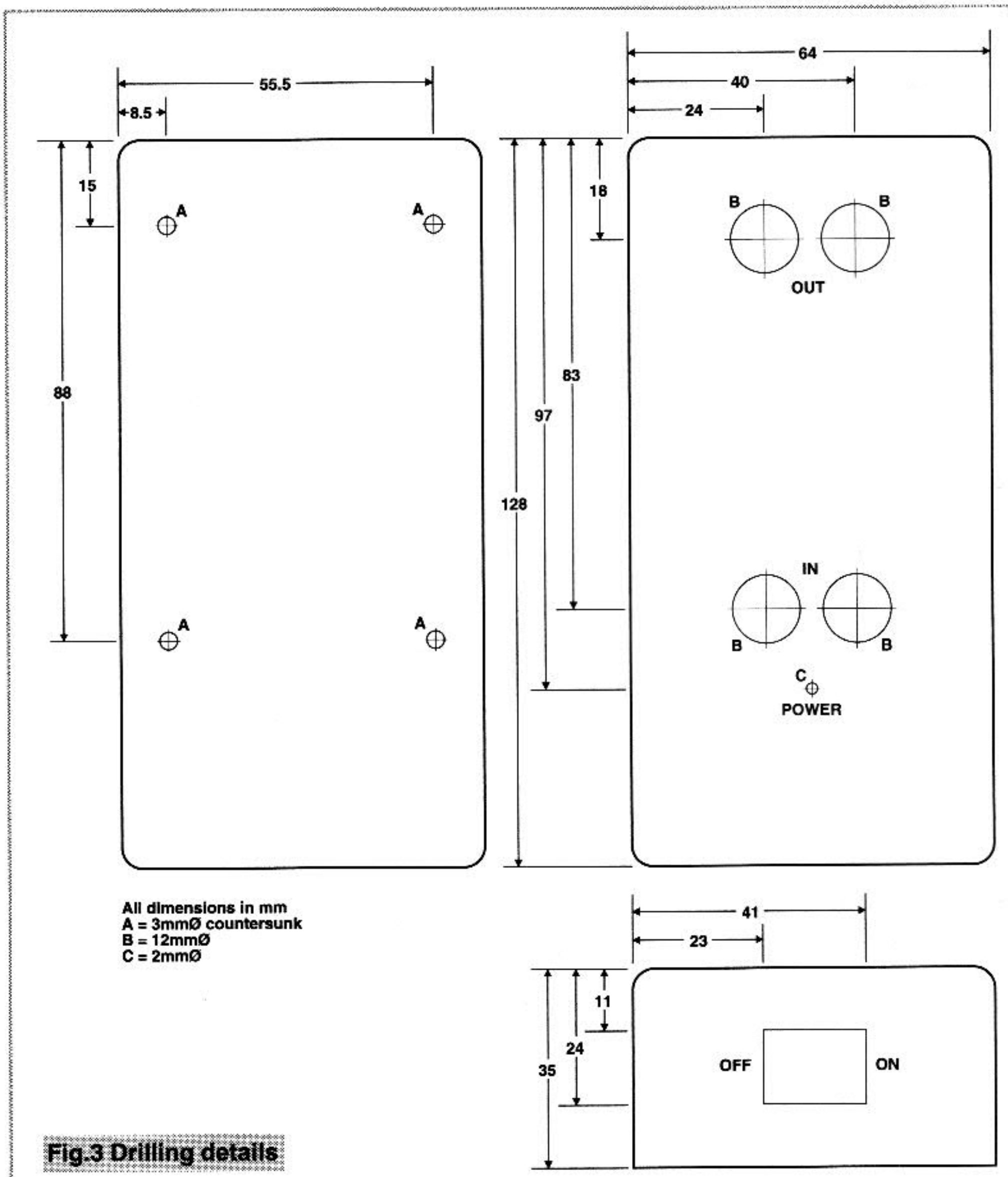


Fig.3 Drilling details

The Use of Transformers in Low Noise Design

Obviously, cartridges have an impedance which can't be altered. While the impedance of the moving magnet type is high enough that, with the right choice of op-amp, it can be amplified directly, with little or no noise degradation, the MC-type cannot. Consequently, noise takes on special significance in the design of MC preamplifiers, since the much lower impedance of the MC type means that the direct op-amp approach can't be taken if adequate signal-to-noise ratio is to be maintained. This is because we can't adequately match the higher optimum source impedance or noise resistance of most op-amps - more of this later - to the much lower source resistance of the cartridge. There are three accepted solutions to the problem. With the first, and earliest method employed, optimum performance may be obtained by using a transformer with a turns ratio selected to transform the actual source impedance of the cartridge to the noise resistance of the amplifier. However, the transformer has several shortcomings which include cost, distortion at low frequencies, (caused by magnetic non-linearities in the core which cause saturation), and the inability to handle transients because of leakage inductance and stray capacitance, which also affects performance at HF. There are further losses from the magnetic properties of the core, and from winding resistance, the latter worsening noise performance because it is, in itself, a source of Johnson noise. Also, the very low level signals present both on the primary and secondary windings of the transformer mean that hum pick up can be a problem, and special, purpose-designed shielding, manufactured from materials such as mumetal, is sometimes used so that noise performance isn't impaired. In spite of these shortcomings, transformer coupling is used extensively in many high quality moving-coil preamplifier designs. These match the very low impedance of the MC cartridge to the much higher impedance needed to load the op-amp optimally for noise, and also produce the necessary 20-30dB of gain. Ortofon, one of the companies who pioneered the manufacture of moving coil type cartridges, use the transformer approach extensively in their head amp designs. To summarise, the transformer must match the source resistance to the characteristic noise resistance of the op-amp or transistor so that the best noise performance is attained (when $R_s = R_w$), match the impedance of the cartridge (non-critical), and provide some degree of voltage gain for the cartridge input signal.

Transistor Preamp Stage

The need for a very low value of circuit input impedance, in order to minimise thermal noise effects, can be met because of the much lower characteristic OSI (Optimum Source Impedance or noise resistance) of the transistor, so that it can be used directly to amplify the signal. With transistors, noise voltages and noise currents alter in magnitude and in ratio to one another. Lower collector current gives rise to lower noise current - not inconceivably, since the current noise is due to minor random discontinuities in the device currents (i.e. shot noise) which adds to the fluctuations caused by noise. The ratio between the two - noise impedance - can therefore be altered.

The resistance value at which the device is optimally quiet for audio purposes is also the value which coincides with that required for optimum device transfer characteristics. In other words, there is good frequency versus phase linearity response, and so the device will be stable at high frequency, and in high feedback-low gain configurations.

With bipolar transistors, the theoretical value for emitter-base voltage noise is a function only of absolute temperature and collector current

$$en= kT\sqrt{\frac{2}{qI_c}}$$

This formula indicates that a reduction in voltage noise, e_n, can be made low in value by increasing collector current. This is bourne out in practise as I_c is increased until a level is reached where parasitic transistor noise limits any further reduction. This noise floor is usually created by and modelled as an equivalent resistor (r_{bb}) - the so-called 'base-spreading resistance' - in series with the base of the transistor. This is, in fact, the real part of h_{ie}. Low parasitic transistor noise is therefore an important factor in ultra-low noise applications. For finite source impedance, current noise must be considered as a quadrature addition to voltage noise. Shot noise is white noise - equal energies at all frequencies - which worsens with increasing emitter current. Base spreading resistance decreases with increasing emitter current, so thermal noise because of this parasitic resistance decreases also. This gives rise to something of a conflict of interest - low emitter current lessens shot noise but increases thermal noise because increases, while higher current lessens the thermal noise effect but worsens shot noise.

To find the collector current which yields the minimum overall equivalent input noise with a given R_s, i.e. a compromise between thermal noise and shot noise, the total noise formula can be differentiated with respect to Ic and set to zero for finding a minimum. For very low R_s - as in the case of a moving coil cartridge - the rbb of the transistor must be added to R_s in the calculation.

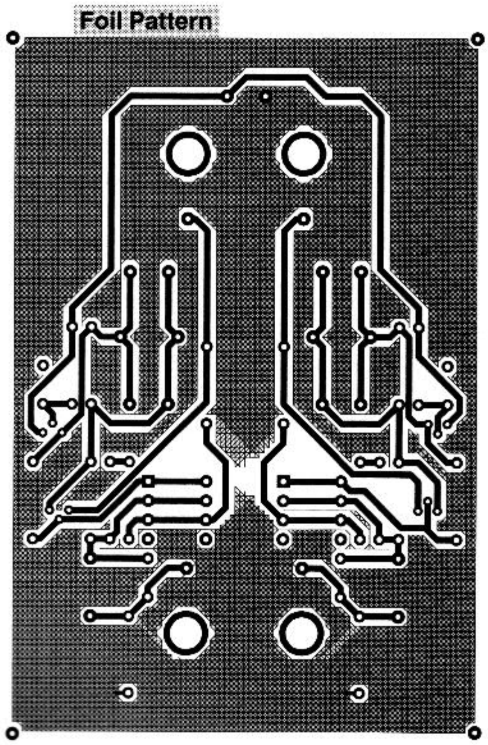
Even a noiseless source has an irreducible source of Johnson noise from the value of its source resistance and, in any case, the amplifier adds its own noise. We've already shown that the amplifier's noise voltage is added to the input signal while its noise current generates a noise voltage across the source impedance. With a low source resistance - 50R cartridge - a low e_n is more important, so in practise we run the transistor at a much higher value of collector current. This also has the benefit of widening the bandwidth of the system.

We can bring the OSI down by reducing the ratio of inherent noise voltages and currents. Lower OSI can be achieved by paralleling identical, low noise, input devices. This maintains the same noise voltage but alters the noise current. The noise impedance changes proportionately because the base spreading resistance of the transistor - which, as far as noise is concerned, adds itself to the source resistance of the cartridge - is reduced by a factor dependent upon the number of devices placed in parallel. In the Alchemist design, noise is reduced by the square root of this paralleling factor. Shot noise (or Schottky noise) contribution can be reduced by maintaining a high collector current (which reduces dynamic emitter resistance), while voltage noise is inversely proportional to the square root of stage current. Conversely, current noise increases proportionally to the square root of stage current. Fortunately, high current

noise is of less importance when dealing with low impedance sources such as MC cartridges, and optimisation of impedances is not necessary. Designs of this type are marketed by companies such as Braithwaite, Linn/Naim and Ortofon.

From a commercial viewpoint, careful selection and matching of components is a time-consuming - and therefore costly - process. To even out the spread in characteristics of electrically unmatched, but ostensibly identical devices, where there may be differences in the base-emitter turn-on voltage, for example, resistors are sometimes used to create a biasing network. Regrettably, resistors in the collector arm of the circuit mean a loss of some usable signal. Conversely, circuit topologies based around a common base connection between all transistors use individual emitter resistors. This swamps any differences in device characteristics, at the expense of a small but finite addition to the base-spreading resistance of the transistor in question which, in turn, worsens noise performance.

Foil Pattern



Supermatched Pairs

An ideal way to avoid using either a transformer, or having to closely match several discrete transistors in a head amp design is to make use of what is known as a super-matched pair. This exploits the low-noise-at-low-source-impedance characteristics of the discrete transistor, but without the attendant selection/matching problem, or the introduction of superfluous and performance-damaging passive component arrays.

Monolithic transistor arrays have been around for a long time, but ultimate performance was being compromised by statistical fluctuations in the material itself, and in the processing environment. The so-called 'super-matched' transistor differs in that many individual transistors are physically located in a way which tends to average out any residual process or material gradients. This yields a pair which offers order-of-magnitude improvements both in matching properties, and in parasitic base and emitter resistance specifications, since the paralleling of many devices reduces overall rbb and ree, which are around 20R and 0.4R respectively for packages such as the SSM2220 and the LM394. Such values are considerably smaller than for other small signal transistors. Typical h_{FE} mismatch is 2%, and is valued at 500 minimum when Ic is equal to 1mA. Broadband noise is very low, and the devices have no excess noise at lower current levels, although the large geometry transistor design, used to minimise base-spreading resistance, results in relatively higher collector-to-base capacitance (C_{OB}) than ordinary transistors. The Miller effect from these can limit voltage-gain bandwidth, especially for high gain applications.

Construction

The circuit is simple but effective. There are no lineup procedures, no presets to trim, and no matching of components. For this reason, all of the components there are can be considered to be critical to the well-being of the design, and tolerances and types specified in the parts list should be strictly adhered to if noise and distortion performance is not to be compromised or impaired.

The first job which must be carried out is a drilling operation. The four mounting holes must be opened out to 3mm, while the phono socket mounting holes (if intended for use with those specified in the Parts List) should be drilled using a 8mm drill. Use new, sharp drills and speeds around the 3000rpm mark or higher. The PCB must also be trimmed slightly if it is to fit in the box specified. This amounts to about 1.5-2mm on each side.

Since the ground plane was extended around to the far side of the board using PCB material which has just been removed, a wire link must be soldered to the underside of the board to join both halves of the ground plane together. This should be done on the central part of the board. Note also that whilst R3 and R5, R103, R105 are shown earthed in the circuit diagram, various gremlins in the PCB design seem to have thwarted this. The node connecting these two resistors (adjacent to and just in front of pin 1 IC1, pin 8 of IC101, should be joined to the ground plane with a wire link/ large blob of solder. C4, C104 optional supply decoupling capacitors seemed to have suffered in this respect too, with the earth connection to the - pin missed. Bridge across from this to the ground plane. Once this is complete, the IC sockets should be soldered in first, followed by resistors and small capacitors. R1,101 should be chosen, in accordance with the cartridge manufacturer's specifications, to be a proper impedance match for the device concerned. Transistors come next, then Veropins and the larger non-polarised electrolytics, before finishing with the two gold-plated phono sockets. These are optional but make for a tidy and uncluttered interior. They also eliminate the possibility of any infiltration of the signal by hum or other stray pick-up. Should off-board connectors be more attractive for whatever reason, ensure that the connecting leads are adequately shielded and screened and that good earthing arrangements are strictly adhered to. The sockets are mounted on the reverse (foil) side of the PCB. Use the washers supplied and tighten the nuts securely. The earthing tag can be soldered to the surrounding earthed trackwork while the centre, signal pin is joined to its associated Veropin using a short length of wire. The connections to the centre pins should be shrouded. Throughout construction be aware of solder bridges across tracks.

Rather than have one set of 'IN' and one set of 'OUT' sockets, the output of the circuit can, instead, be terminated in a phono plug/screened lead combination. In this way, one set of potentially-unreliable, potentially hum-inducing connections are eliminated. It goes without saying, of course, that both the screened lead and the phono plugs should be of a quality commensurate with the performance expected of the unit.

The last connections to be made are the output phono leads/plugs (if you've omitted the output sockets SK2 and SK102), power connections to and from the on/off switch and finally, the battery clip connection. The diecast box is drilled with reference to the diagrams shown in Figure 3. The PCB is mounted inside the box using M3 tapped spacers or PCB standoffs. Spacers are bolted to the lid part of the box. Where metal spacers are used, all but one of the screws should be insulated using plastic washers. This avoids the PCB being earthed in four different places and the hum loops it might cause. Deviate from these at your peril - when I was measuring noise and distortion performance, non optimised ground paths in the input and output leads used to connect the measuring apparatus caused performance to be worsened in both respects by at least two orders of magnitude.

Current consumption is low enough (about 10mA) that the unit can be powered from a 9V PP3 battery. This eliminates the possibility of hum induction from the power supply, but a mains PSU could be used, as long as it's adequately screened (ideally in a separate box) and well-regulated. The preamp should, in any case, be housed in its own metal box - the diecast example specified in the parts list is ideal. The battery can be wrapped in some foam and sealed inside this box and changed once a year or thereabouts. The mounting position for the switch has to satisfy two, somewhat conflicting, criteria. It must be positioned where it is easily accessible but not able to be inadvertently knocked on or off. A DPDT rocker switch from RS was used in the prototype and seemed ideal. A worthwhile addition is the LED across the switched supply. I used a high efficiency type fed through an 8K2 resistor.

When interfacing between turntable and preamp, The Alchemist should be sited as near to the turntable as possible, connected to the cartridge with short, high quality screened lead. Connections between it and the preamp should also be as short as practical. We've mentioned already that the moving-coil cartridge, because of its low impedance/ low inductance nature, is able to drive long and or capacitive cables. However, longer cables are more prone to hum pick-up and other forms of interference injection and so should be avoided where possible. All that now remains is to connect the battery, turntable, amplifier, select a favourite track, turn on The Alchemist, and listen to it work its magic on vinyl.

Parts

Resistors

(all 1/4 watt 1% metal film)

R1,101	51R (or in accordance with cartridge manufacturer's specifications)
R2,102	200K
R3,103	62K
R4,104	2K2
R5,105	510R
R6,106	10R
R7,107	510R (see text)
R8,108	10K (optional supply smoothing)

Capacitors

C1,101	10µF non-polarised radial electrolytic
C2,102	100µF non-polarised radial electrolytic
C3 103 / **C4,104**	25µF/16V radial electrolytic

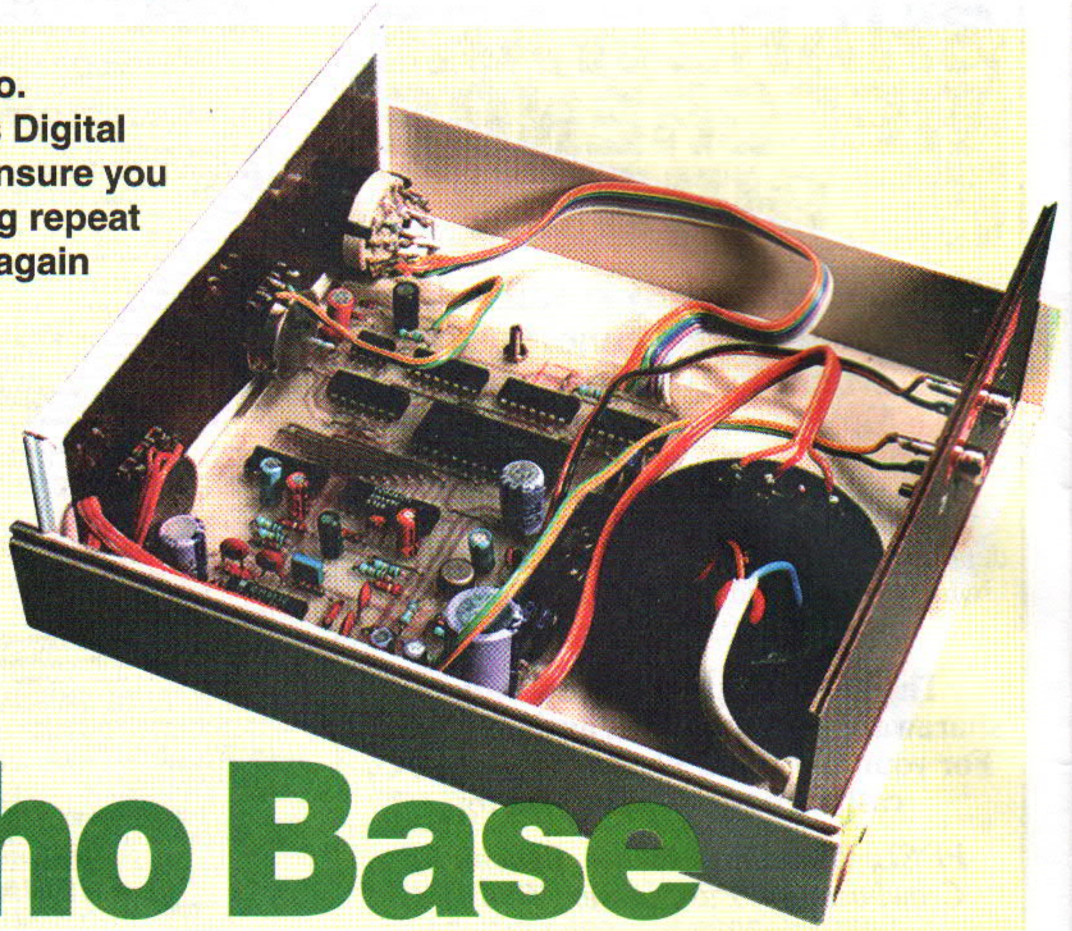
Semiconductors

IC1,101	SSM2210
Tr1,101	BC214C
Tr2,102	BC549C (optional supply smoothing)

Additional Items

PCB, Veropins, PCB mounting gold-plated phono sockets (Maplin JZ05F, JZ05G), 8 pin DIL IC sockets, type M5004 diecast box (Maplin LH71N), on/off switch, PP3 battery clip and battery (both optional), screened lead, phono plugs, LED

Echo Echo Echo. Paul Stenning's Digital echo unit will ensure you have astounding repeat performances, again and again.



Echo Base

This digital echo unit resulted from a discussion with a friend who is a member of a local amateur dramatic group. Until now they had been producing echo sound manually by tape recording the same sound several times at lower recording levels - I admire their patience! Although this method worked OK for longer echoes on single sounds, it could not be used to add a short reverberation on speech.

An exact specification was not offered, they felt that almost anything would be better than the existing system. We decided that about half a second between echoes would be more than adequate for a maximum delay range. The shortest delay is so fast that sounds resembling microphone feedback can be obtained. Metallic alien reverberations similar to those on the "Smash" instant

The Works

The complete circuit diagram appears in Figures 1, 2 and 3.

Starting with the digital section. IC1 is the clock generator, which runs at four times the sampling frequency. The frequency is adjustable over a limited range by VR1, the Delay Fine control. This slightly unusual configuration gives an output with an approximately equal mark-space ratio.

The A-D convertor IC7, requires a negative bias on pin five. Since the current required in minimal, this negative voltage is obtained by rectifying the clock signal from IC1, giving approximately -4V.

IC2 produces the four timing pulses required, three of which are inverted by gates in IC3. When Q1 goes high, the A-D convertor IC7 starts a conversion. When Q2 is high, the data in the RAM is sent to the D-A convertor IC8, which produces the appropriate voltage. When Q3 is high, the data from the A-D is stored in RAM. Finally when Q4 pulses high, the address counters IC4 and IC5 are incremented.

SW2 (Delay Coarse) sets the count reached by the address counters before they are reset. This sets the amount of the RAM chip to be used.

Ideally I would have had just one continuously variable delay pot, on the 555 clock circuit. However, it was not possible to achieve the required range with a respectable frequency response due to the A-D conversion time.

The ZN448E (IC7) is an 8 bit successive approximation A-D convertor. Nine clock cycles are required for each conversion. The eight data lines are tri-state, and controlled by the OE- pin.

The device contains a built in clock circuit, the frequency of which is set by C13. The data sheet gives the maximum clock frequency as 1MHz, and a graph shows that a capacitor value of 100pF will give this. However tests with several devices showed that 100pF gave a

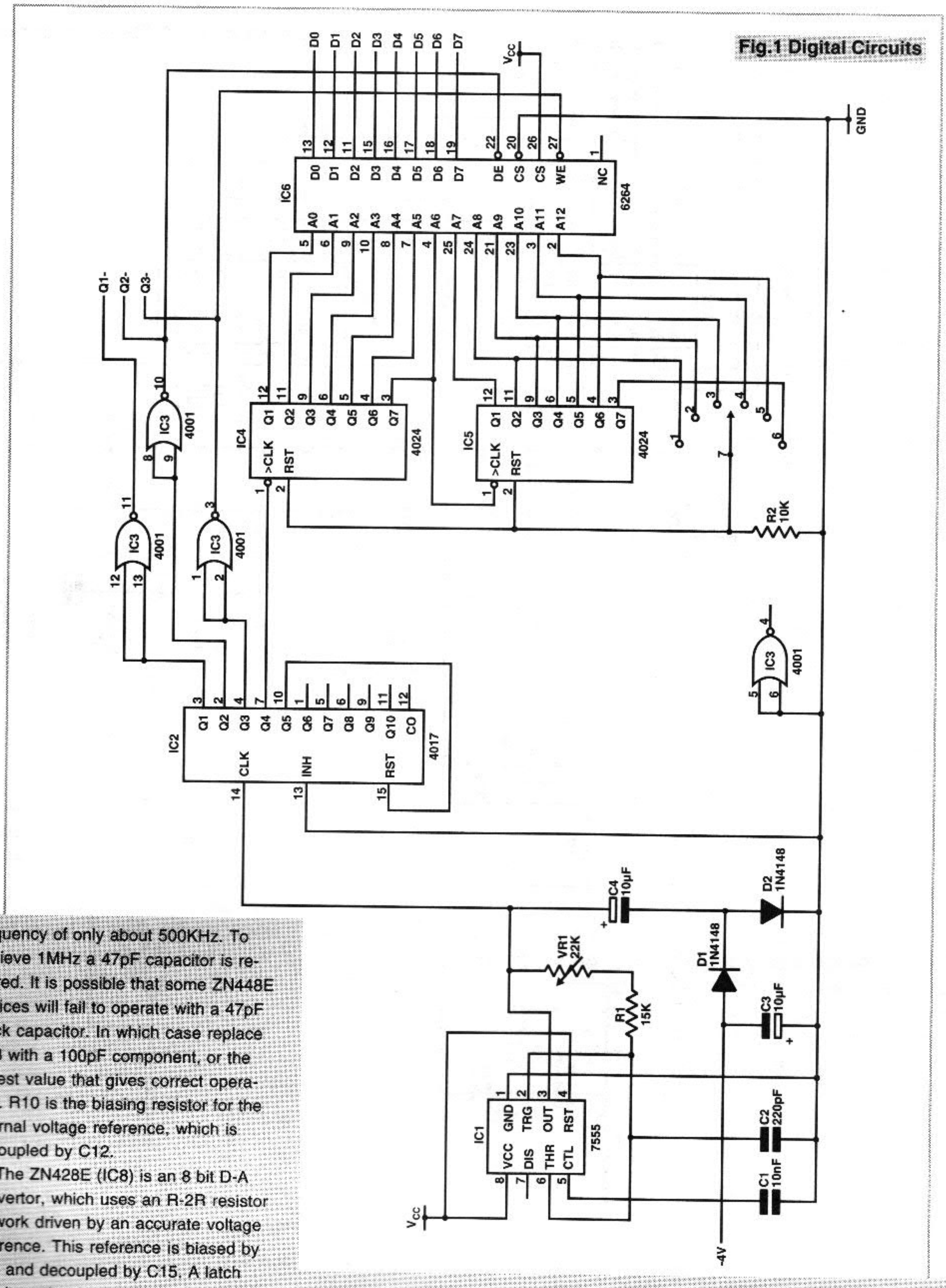


Fig.1 Digital Circuits

frequency of only about 500KHz. To achieve 1MHz a 47pF capacitor is required. It is possible that some ZN448E devices will fail to operate with a 47pF clock capacitor. In which case replace C13 with a 100pF component, or the lowest value that gives correct operation. R10 is the biasing resistor for the internal voltage reference, which is decoupled by C12.

The ZN428E (IC8) is an 8 bit D-A convertor, which uses an R-2R resistor network driven by an accurate voltage reference. This reference is biased by R14 and decoupled by C15. A latch circuit on the data input lines is controlled by the EN- pin. When EN- is inactive, the previous output voltage remains.

The audio signals to the A-D convertor and from the D-A convertor pass through identical low pass filter circuits. These have a slope of 24dB per octave and a -3dB point at about 8KHz. The frequency response of these filter circuits is shown in Figure 4. The variation of +/- 1.3dB before the cutoff point is not important in this application.

The filter on the input to the A-D prevents any signals with a frequency too high to be accurately sampled from reaching the convertor. Allowing these signals through would result in severe aliasing distortion, leading to a poor quality output. The filter on the D-A output removes noise due to the sampling frequency, and smooths the stepped output to some extent.

The audio output from the delay circuit is mixed with the input signal by the circuitry around IC10. VR2 sets the amount of signal that is fed back around the system.

The circuit is powered by a 12V 250mA transformer. The required +5V (VCC) and +12V supplies are regulated by standard three pin linear regulator IC's.

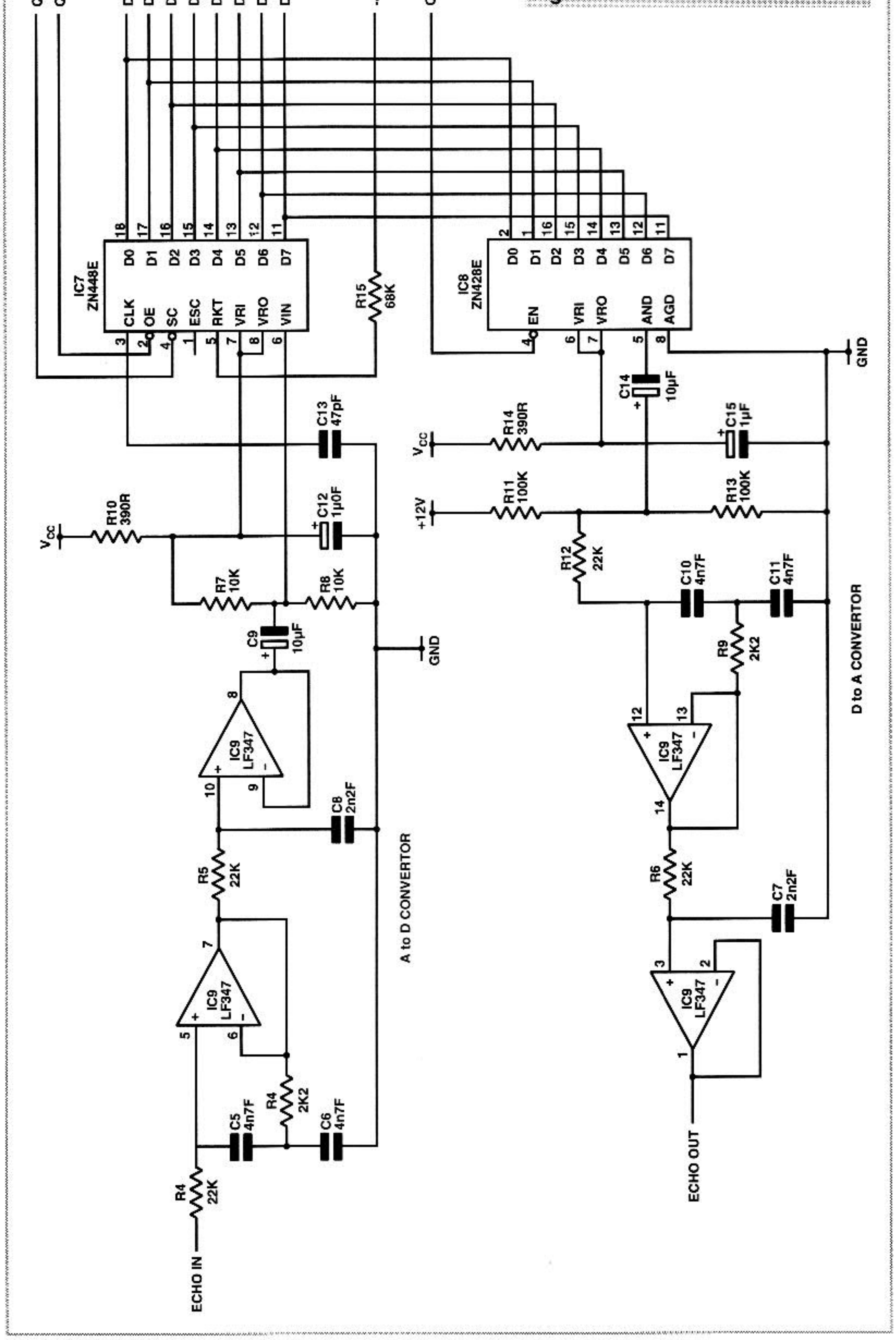


Fig.2 A to D and D to A convertors

potato television adverts are also possible.

Simplicity of operation is essential for the intended use. Consequently, there are only three controls on the front panel. Two of these set the delay required (coarse and fine). The third is the Repeat control, which sets the amount of echo signal fed back around the loop. This ranges from 0% to almost 100%.

My original plan was to use bucket brigade delay line chips, however several devices would have been necessary to cover the required delay at a reasonable frequency bandwidth. Significant distortion can also result as the voltage samples are passed down the chain.

The design presented here uses an 8K RAM chip to store the data samples. The delay of half a second is achievable, with a bandwidth of 8KHz. The incoming signal passes through with no bandwidth limiting. Distortion on the echo signals is generally less than 5%, although this will build up each time a sound is echoed.

The upper frequencies would normally be lost on natural echos, so the bandwidth limitation is not a problem in practice. Similarly the distortion on the echo signals will generally pass unnoticed.

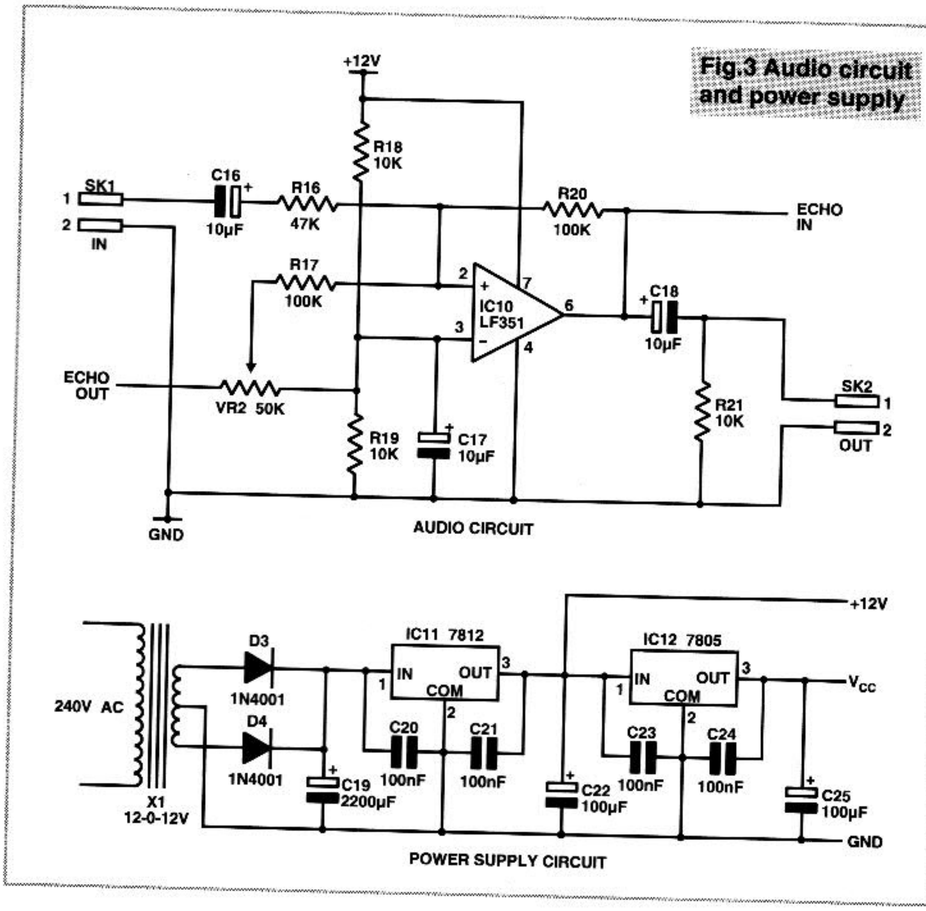


Fig.3 Audio circuit and power supply

Construction.

The circuit is constructed on a single sided PCB, which is available from Electronics in Action (see page 61). The track layout and component positioning are shown in Figure 5.

There are 30 links that should be made first, using thin (approx. 26SWG) tinned copper wire. The resistors, diodes and capacitors can then be fitted in size order. Sockets may be used for the IC's if required - since all the devices are static sensitive this may be a good idea. Do not insert the IC's into the sockets until the remainder of the PCB construction is complete.

IC11 and IC12 will become warm in operation and should be mounted on a small heatsink. The mounting tabs of both devices are connected to the 0V rail so no insulation washers are required. A small amount of silicone grease or heat transfer paste should be placed between the devices and the heatsink.

Terminal pins may be fitted in the holes for the off-board wiring, so that the connections can be made after the PCB is fitted into the case.

The prototype was constructed in a plastic case, 190mm x 165mm x 68mm, see parts list for details. A suitable overlay for the front panel is shown in Figure 5. Two photocopies may be taken (enlarge to 162mm * 64mm), one can then be used as a drilling template while the other may be fixed to the front panel with clear self-adhesive vinyl. No actual values are shown for the delay controls, since they might confuse a nontechnical user.

Set the stop on SW1 to position six. Fit the pots and switches to the front panel, and then position the PCB and transformer in the base of the case. The PCB should be mounted with three M3 screws, with extra nuts to act as spacers. The transformer is also mounted with M3 hardware, which should ideally be nylon to insulate the mounting frame from the outside world. A 10VA toroidal transformer was used in the prototype because it was to hand, however a 250mA conventional type is suitable.

The rear panel needs to be drilled for the audio input and output sockets, and the mains flex. The hole for the latter should be fitted with a cable gland or grommet, and the cable must be secured to prevent it being pulled out.

The interwiring is shown in figure 5, the pots are shown viewed from the rear. Audio coax may be used for the connections to the rear panel sockets if desired, although the prototype worked fine with ribbon cable. The remaining connections may be made with ribbon cable or whatever is available.

The mains flex should be connected to the flying leads on the recommended transformer with a chock-block type connector. No mains switch or fuse were fitted on the prototype for simplicity. A 3A (or lower) fuse must be fitted in the mains plug.

Testing and Using.

Ensure that the internal mains connections are adequately insulated. Connect the unit to the mains and switch on. If a

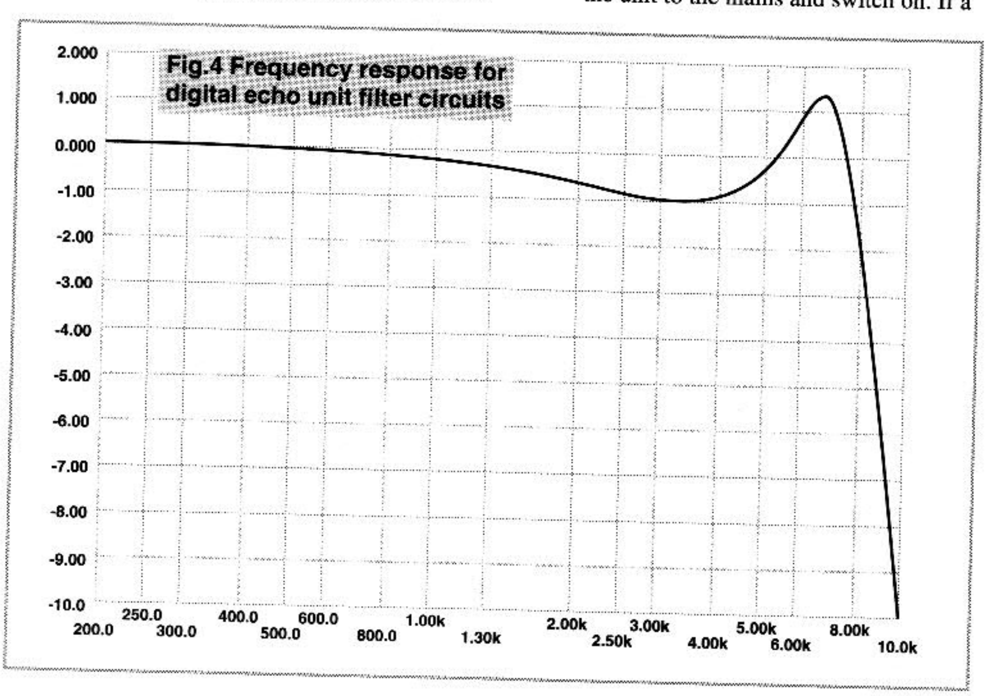


Fig.4 Frequency response for digital echo unit filter circuits

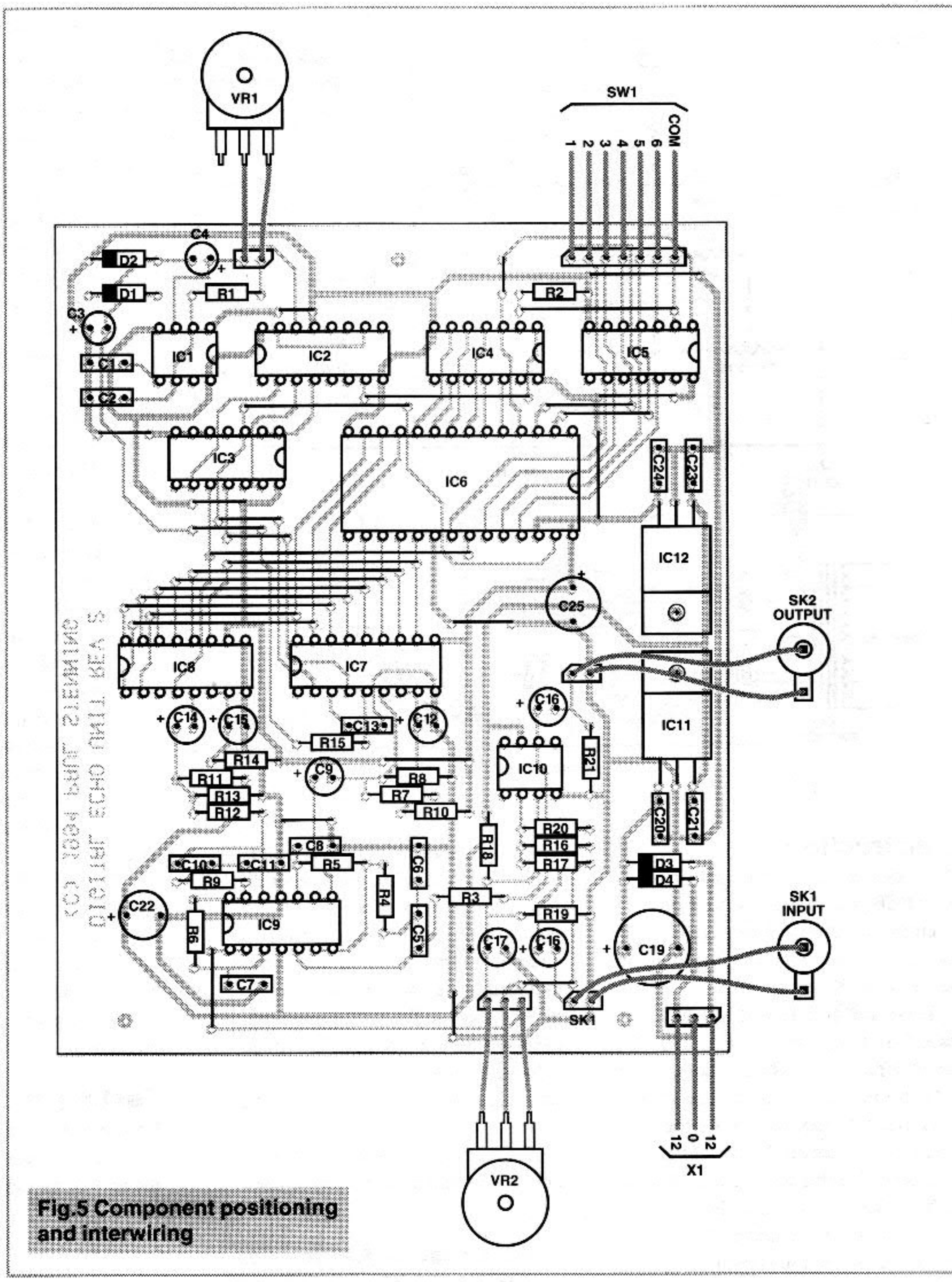


Fig.5 Component positioning and interwiring

test meter is available, check the voltage outputs from IC11 and IC12 on the appropriate pins. The voltages required should be 12V +/-0.5V and 5V +/- 0.25V.

Unless an audio pulse generator and oscilloscope are available, the remainder of the unit is probably best tested by connecting it to suitable audio equipment and trying it. The effects can be heard readily on male speech.

The audio input signal should be between 0.5 and 1V RMS for optimum performance. Lower levels will give greatly increased distortion on the echo signals due to the lower number of sampling points available. Larger signals (over 2.5V pk-pk) will be distorted due to clipping at the A-D convertor. If the signal level is likely to vary significantly, this unit should be preceded by an automatic level control or compressor circuit.

Initially set both delay controls fully clockwise. As the repeat control is advanced, the echo should be heard more times. You may notice that if a sound is repeated several times it becomes a little more distorted each time. With a more normal decay, the distortion will tend to occur as the sounds are getting quieter, and will generally pass unnoticed. When the control is at maximum, all the signal is recycled which can cause some awful sounds!

By operating the delay controls, the period between each echo can be altered. You may notice an apparent pitch shift as the Fine Delay control is operated, this is due to sounds being replayed at a different rate to the one they were recorded at. Momentary bursts of noise may also be heard as the Coarse Delay control is set to longer periods, this is due to old data stored in the RAM being replayed. Neither of these effects is relevant in practice, since the controls would not be altered when the unit is being used.

The best way to understand the unit's capabilities is to play with it for a while. Have fun . . . fun . . . fun! EIA

Parts

Resistors
(0.25W 5% or better)

R1,2,7,8 R18,19,21	10K
R3,5,6,12	22K
R4,9	2K2
R10,14	390R
R11,13,17,20	100K
R15	68K
R16	47K
VR1	20K or 22K Lin Pot
VR2	47K or 50K Log Pot

Capacitors

C1	10nF
C2	220pF
C3,4,9,14 C16,17,18	10µF
C5,6,10,11	4n7
C7,8	2n2
C12,15	1µ0
C13	47pF
C19	2200µF/25V
C20,21,23,24	100nF
C22,25	100µF/16V

Semiconductors

IC1	7555
IC2	4017
IC3	4001
IC4,5	4024
IC6	6264
IC7	ZN448E
IC8	ZN428E
IC9	LF347
IC10	LF351
IC11	7812
IC12	7805
D1,2	1N4148
D3,4	1N4001

Additional Items

SK1,2	Phono Socket
SW1	1 Pole 12 Way Rotary
X1	12-0-12V 250mA

PCB, Knobs, Case, Wire
2 Core Mains Flex,
13A Plug with 3A Fuse, IC Sockets
M3 Screws and Nuts.
The plastic case used for the prototype is made by Bafbox, and is available from RS/Electromail, stock no 506-788.

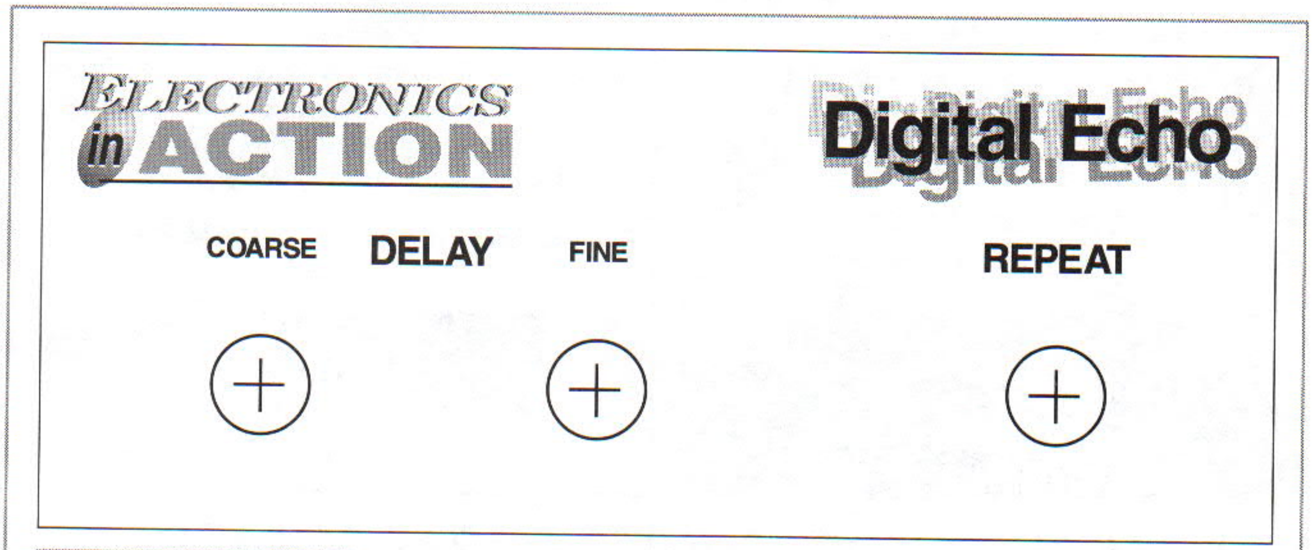
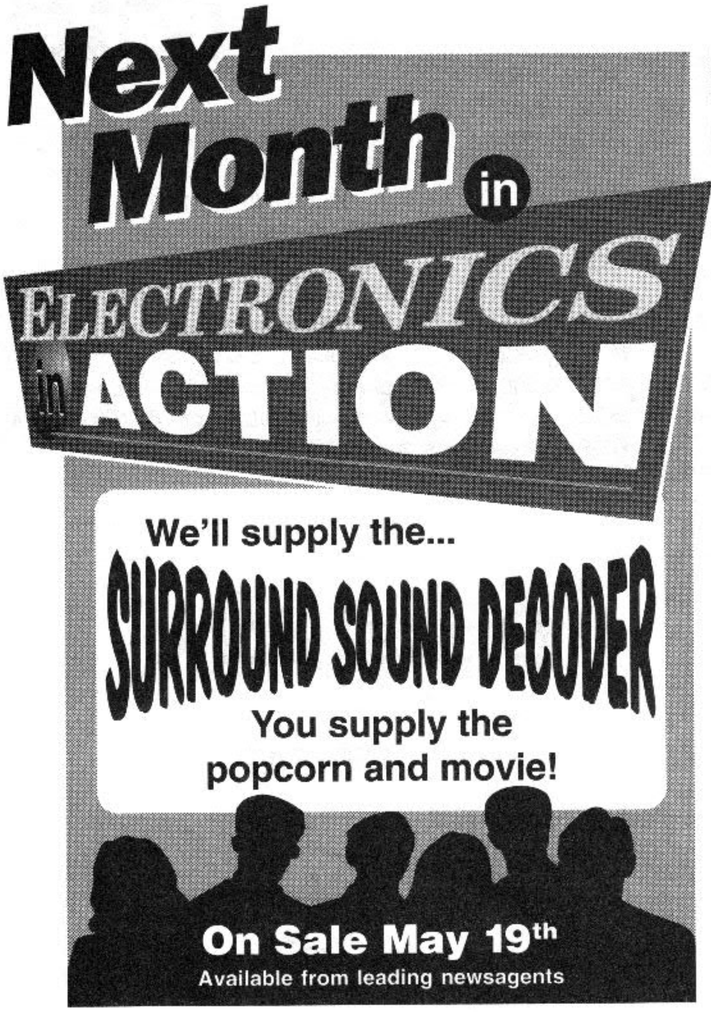


Fig.6 Front panel template



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MAY

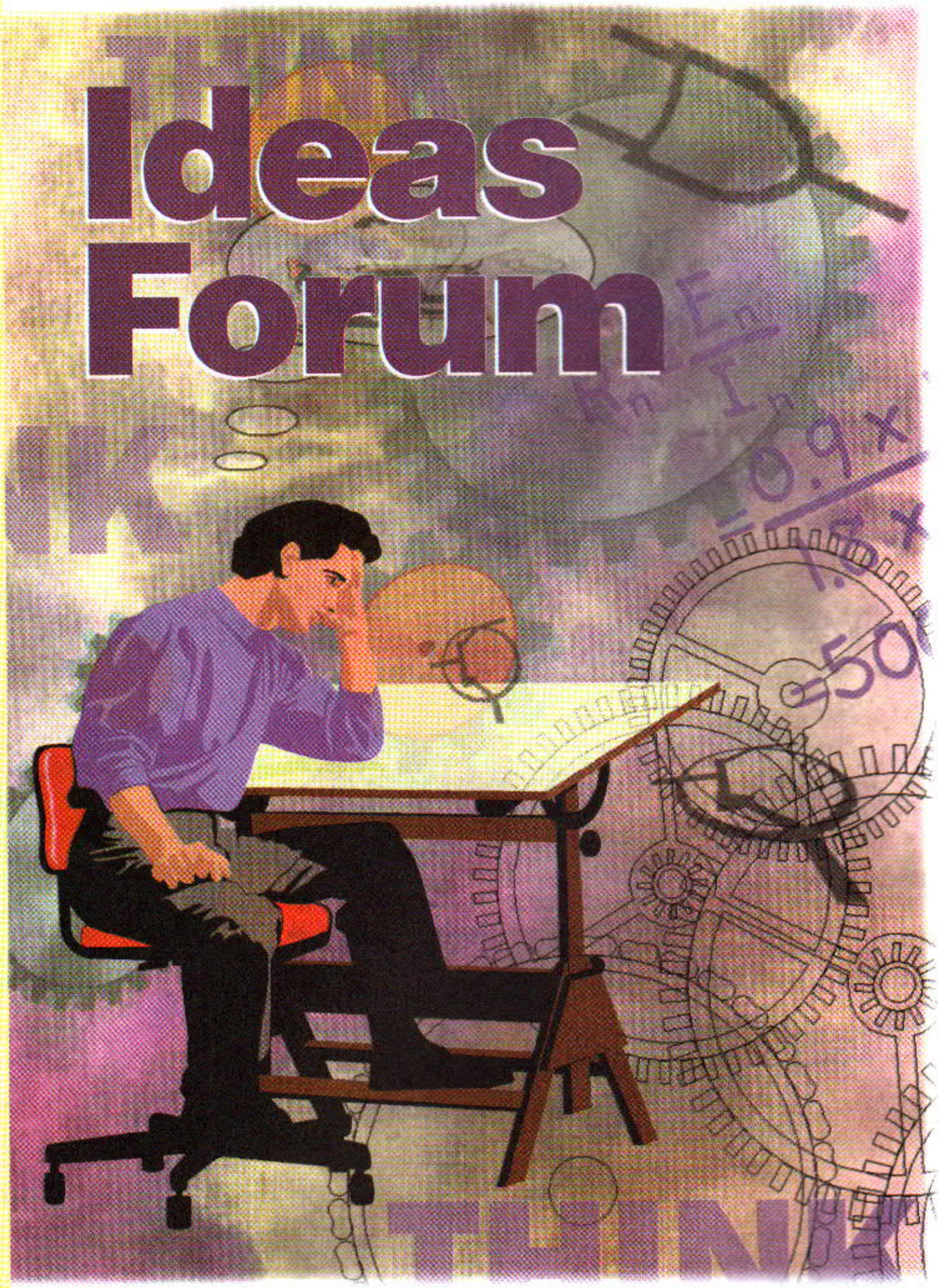
The Sniff continues
That ultra sensitive nose on a chip announced by IBM might well detect certain chemical reactions at molecular level and could provide remote electronic signals say from the upper atmosphere. One wonders whether it is beyond the bounds of possibility to generate smells (chemicals) from electrical signals?

Car Security
Change the simple system of a mechanical key to unlock the door and start the engine by an electronic key. This would not only open the door but it could start the engine through unlocking an encrypted code for the electronic ignition system. This way no thief could get the car moving as the spark plugs would not fire. This makes you think how primitive the system is. The under-bonnet electronics could be potted in black epoxy resin with a minimum of wires and information externally. This would give added security. The encrypted code for this vehicle, apart from being unique, could also be the registration number and if in the event the car does somehow move without the electronic key it would automatically trigger off an alert beacon.

Car Parking
Proximity detector on front and rear of car for parking in small spaces giving alert of distances with a change of audible pitch.

Mains Failure
Audible warning for failure of deep freeze units.

Ideas Forum



Work those brains, exercise, exercise, exercise!

Ideas never come easy do they? Some say that talking to a like minded person will bring out the best in you. The more you talk, the greater the chance that an idea will come out in conversation. Also to some it depends on what the weather is like or how much sleep you have had the previous night.

All great inventions start as little notions that get scribbled down and expanded upon, and that's what the Electronics in Action Ideas Forum is all about. Now you are the inventors.

Even though great inventions come to mind, the tragedy is you may dismiss the idea as worthless. So if you think of a good idea discuss it with a friend to see if they agree with you that what you have to tell the world is brilliant. Then ask yourself, do lots of other people want to hear about it and would you like them to benefit from your idea? Is there a market for it? If there is, who will mass manufacture it? These are some of the questions that have to be thought about.

If you have any suggestions or have developed any of the ideas that have been appearing in this column we would love to hear from you, feel free to drop us a line. It could be the first step along the road to fame.

Electronics in Action, PO Box 600, Berkhamsted Herts. HP4 1NL

At Your Service

So you don't fancy messing up your best measuring jug with Ferric Chloride, but you still want to make some of our projects properly - with a PCB? This is where we step in. At Electronics in Action we can offer, at very reasonable prices, PCBs for all of our featured projects.

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QH0194-3
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December issue

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QH1293-3
Switcher Front panel £6.00
QH1293-4
Video Editing Unit £8.00

November Issue

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Power Supply (double sided) ... £6.00
QH1193-2
Telescope Input board £5.50
QH1193-3
Drainpipe Seismometer £5.50

October Issue

QH1093-1
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☐	QH0494-5	Simple Intercom	£5.00
March Issue			
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☐	QH0394-2	Guitar Transmitter (double sided)	£8.00
☐	QH0394-3	Mains Checker	£5.00
February Issue			
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☐	QH0294-3	The Switcher IR transmitter	£5.50
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ADVERTISERS' INDEX

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Electronics in Action in co-operation with The Technology Exchange Ltd., the international technology matchmaking service based in the UK, brings you each month a selection of technology partnership opportunities to which you are invited to respond.

The Technology Exchange, which was formed in 1985 as a not-for-profit technology sourcing service to industry, holds a biannual 'Technoshop' Technology Transfer Fair at Heathrow Airport and several 'Techmart' Fairs overseas for the United Nations (UNIDO).

For this issue of Electronics in Action, we are presenting a series of offers of licence, joint venture and patents rights for sale from organisations in 34 countries.

If you would like to have an introduction to any of the sources of the offers describes in these profiles, please write to The Technology Exchange quoting the reference number at the head of the entry and giving full contact details for the contact person in your own organisation and your requirements for a new product or process development.

The only cost associated with this process is a simple £10 plus VAT introduction fee for each entry to which you respond. For this we will send you full contact details for the source of the offer and invite them to send you more detailed information about their offer.

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KEY

Example: SPOLSU (Status, Offer, Source)

Status: Indicates the stage of development.

P Patent or design only	M Laboratory model	W Working prototype
U Pre-production units available	F In current production	C Commercialised

Offer: Indicates the type of agreement sought.

L Manufacture under licence	S Design and/or patent for sale	J Joint venture offer

Source: Indicates the type of organisation making the offer.

C Limited Company	C Educational Institution	R Independant Research Organisation
G Government Research	P Private Individual	

408737
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108733
KNOW-HOW FOR COMPOSITION RESISTOR PRODUCTION
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708750
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507434
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New type SOS (silicon on sapphire) pressure sensors, using good dielectric strength sapphire to replace P-N junction isolation, have higher working temperature, high radiation resistant property, high accuracy, sensitivity and frequency response. It can be used in many areas such as aerospace, nuclear engineering, energy, oil and chemical engineering etc. Main specifications: pressure range 0-50.0kg f/sq cm; operating temperature: -40 degrees C to +200 degrees C; fall output: 100mV; Accuracy: 0.2%, 0.3%, 0.5%, 1.0% of full scale.

507443
THE EQUIPMENT AND TECHNOLOGY FOR MANUFACTURE OF CERAMIC TRIMMING CAPACITOR
The main application of Phi 7 ceramic trimmer is for frequency adjusting in the circuits of small sized radio receivers, walk-talkers, receive-recorders and color TV. The ceramic trimmers have advantages of small volume, easy frequency modulation and excellent electrical properties. The production line has mostly realised automisation for the equipment, and jigs have also been automised.

207431
LONG WAVELENGTH INGAAS PHOTODIODES
Long wavelength InGaAs photodiodes are key devices for optical communication and new products of high technology. The structures include PIN photodiode and avalanche photodiode (APD). The dark current for PIN photodiodes is less than 1nA, responsibility > 0.7 mA/mW. The multiplity factor for APD is larger than 30, responsibility >0.5 mA/mW.

607452
MULTI-FUNCTIONAL CONTROLLER FOR BACKDROP LIGHT
This equipment is used to control and adjust backdrop light. It features large in unit capacity and output, small size, full-standard elements and devices, full-electronic circuits, easy maintenance etc. It is capable of adjust in the light either

manually or automatically with program control or audio control, and matrix control for all the light is also available.

707442

ASSEMBLING AND MANUFACTURING TECHNIQUE FOR HIGH-FREQUENCY LOW-NOISE TRANSISTORS

The transfer of the exclusive techniques includes: Product research development, manufacturing method and formulation, quality control used to fabricate cased TO-92 B37 (Model M), R246 (3P cross and 4P cross), SOT-23 (3Pmini) and SOT-143 (4Pmini), as well as high-frequency, low-noise transistors for black/white TV tuners and color TV tuners.

307430

RESISTANCE-OXIDE-SEMICONDUCTOR FIELD-EFFECT TRANSISTOR (ROST)

The present invention (ROST) relates to an IGFET in which two gate terminals are set up. The device provides speciality of its transfer curve with non-cut-off, remote cut-off and sharp cut-off characteristics. When necessary, the values of cutt-off voltage are obtained by means of gate bars.

307444

THE FABRICATION TECHNIQUE FOR LC FILTER

The transfer of the exclusive techniques includes: product research development, manufacturing method and formulation, as well as quality control used to fabricate high-pass LC filter for TV tuner, band-pass LC filter for FM receiver. LC filter for CATV system and isolator.

307435

PTC CERAMIC THERMAL SENSOR

The PTC sensor is a kind of semiconducting thermal sensor with a main composition of Barium Titanate. It is a good semiconductor below Curie temperature while its resistivity increases rapidly by 3-5 orders of magnitude within a narrow range of temperature above Curie point. Because of this characteristic, PTC sensors have wide applications in electrical utensil, electronic industry and medical treatment, such as the electrical heater with constant temperature, motor starter, over current protector etc.

208217 C L/J G

BATTERY SEPARATOR

Glass fibre paper type of battery separator has features of quick electrolyte absorption, high electrolyte retention, acid resistance, low internal resistance, fine pore size, no gas generation other than 02 and H2 etc. The separator paper is used for sealed lead acid battery.

908210 C L/J G

ELECTRIC FAN PRODUCTION

MAJRO series of electric fans of our factory or of advanced design. They have high air flow, small power consumption, low temperature rise, and negligible noise. We would export items of electric fan technology as follows: 1) fan assembly (involving equipment, training, inspection testing and repairing); 2) technology for manufacturing fan parts; 3) technology for manufacturing motor and complete set of equipment.

108118 C L/J G

PLATED TYPE CYLINDRICAL NICKEL CADMIUM STORAGE BATTERY

Plant and technologies for producing plated nickel cadmium storage batteries with the annual production capacity of 300,000 cells. By additional improvements the plated type of cylindrical nickel cadmium storage batteries to other specification can be produced.

108117 C L/J G

SINTERED TYPE PRISMATIC NICKEL CADMIUM STORAGE BATTERY

Plant and technologies for producing sintered prismatic nickel cadmium storage batteries with the production capacity of 100,000 cells (20 Ah each) per year. Batteries to other specifications can also be produced.

108218 F L/J G

ELECTRONIC ENERGY-SAVING LAMP

This electronic energy-saving lamp is produced by means of the latest patented technology. The product has the unique temperature and voltage protective device. Its circuit design can raise lighting efficiency. So it has solved the problem of service life of the energy-saving lamp.

7445

24 SERIES VIGURESE, CHINESE, ENGLISH AND RUSSIAN PRINTER

The new product is based on the STONE-2401 electric printer while its software and hardware are redesigned. It can be used as a Vigurese printer, a Chinese printer, an English printer and a Russian printer. There are three subtypes: Vigurese-Chinese-English printer, Russian-Chinese-English printer and Russian-English printer. This product can be utilised in countries and regions where Vigurese, Chinese, English, Russian are used.

108116 C L/J G

SINTERED PLATE CYLINDRICAL SEALED NICKEL CADMIUM STORAGE BATTERY

The product is in accordance with the IEC standards using all sintered electrodes. The battery capacities of 0.5, 1.5, 3.5 Ah, can be discharged at high, medium or low rates. The products are characterised by good performance at low temperature, long service life, shock and vibration tolerance, easy maintenance.

907400

TECHNOLOGY AND EQUIPMENT FOR PRODUCTION OF CIVIL POLYCRYSTALLINE SILICON SOLAR BATTERY

The technology and special equipment for producing civil polycrystalline silicone solar cells include: The process of casting polycrystalline silicon ingots. The process of producing polycrystalline silicon solar cells adopts the world advanced technique, such as screen printing, plasma etching, antireflection spray film coating etc. The process of sealing and packing modules uses some special equipment.

107399

SILICON SOLAR CELL MANUFACTURE TECHNIQUE

AMI 5, 25 degrees C, conversion efficiency of single solar cell is 12-13% AMO 25 degrees C, TJDG BSR, BSE conversion efficiency is 13-13.5%. It was awarded the silver medal in 1980. Using for the terrestrial solar array and satellite solar array.

8115 C L/J G

METALLISED FILM CAPACITOR

can be used in single phase electric machines such as refrigerator, air conditioner, washing machine, electric fan, ventilator and water pump. It is also applicable in power factor correction eg in fluorescent light.

MORE TECHNOSHOP NEXT MONTH

Future View

Alan McKeon, International Vice President of Iterated Systems, considers still and moving images: how the human eye sees them, how computers handle them and how they will be delivered across the digital information super-highway.

The human eye sees the world as a set of attractive pictures, continuous images whose elements flow one into the other and are understood as a whole - basically in analogue form. When we take pictures into the world of computers, on the other hand, they are approximated into a digital format, sampled at a given resolution - rather like looking through a mesh - and then encoded and stored digitally as a series of ones and zeros. These closed sets of information are then used to reproduce the image on screen or in print. The challenge to computer hardware and software designers has always been to encode images in the same natural way that the eye sees them - and reproduce them in the same way that the brain decodes them.

During the seventies and eighties, mathematical researchers investigated this challenge, exploring Chaos Theory and Fractals. Fractal images were first identified by the Swedish mathematician Helge Von Koch in the early 1900s, although it was only in 1978 that Benoit Mandelbrot first used the term fractal to mean 'a broken structure possessing similar looking forms at many different sizes'. Mandelbrot's research in pure mathematics enabled him to prove that natural structures and forms could be described and imitated using recursive algorithms (sets of mathematical instructions). He demonstrated that apparently disparate collections can be thought of as single geometric structures, and pictures of them created.

Mathematically, it is the infinite complexity of fractals combined with low information content that makes them fascinating. Most people are now familiar with fractally-produced 'paisley' patterns and computer 'landscape generators' which can create entire terrains from small sets of data. Using fractals to create psychedelic patterns or imaginary landscapes is, however, of fairly limited use. It was British-born Dr Michael Barnsley and his research team at Georgia Tech who first began seriously to examine how the process might be reversed, and a practical and valuable tool created.

Rather than 'growing' an image from a small amount of data they set out to discover ways in which real-world images could be described in terms of fractal mathematics. Barnsley's Collage Theorem proposed that all images could be reduced, using fractal analysis, and defined by simple formulae. A highly complex and information-rich picture could therefore be described in very brief terms.

The process of identifying the fractal codes was extremely time consuming until 1988, when Barnsley discovered the Fractal Transform process. This automated the identification of fractal codes within real-world images so that computers could take over the task. Very quickly Barnsley and his team were using this discovery to provide a completely automatic compression/decompression process.

Fractal images share many of the same qualities of images in nature; unlike other digitised pictures they are scalable, stretchable and can be zoomed or enhanced at will. The iterative nature of the mathematics simply creates 'new' detail.

Asymmetric processing

So what does this mean to the delivery of services on the digital information super-highway? Look at the parallels in standard publishing delivery - printing a magazine is time and cost intensive, but buying a magazine is easy and cheap. Fractal compression is similarly 'asymmetrical' - whilst the compression process is intensive, decompression is low cost and very fast. File sizes are small (important for bandwidth), and because information is stored in a descriptive rather than representative language (rather like Postscript) decompressed images can be reproduced at resolutions and bit-depths independent of the original input.

Resolution independence, as this aspect of the technology is known, is one of the key components and advantages of fractal image handling. Without a fixed resolution, for instance, one compressed image can serve as both thumbnail portrait and full screen picture. More traditional image compression techniques have their qualities defined at the time of compression, which can make them more than a little inflexible if at some later date one needs to use them in a way not foreseen at the time.

Using fractals, images can also be zoomed to reveal details without the pixellation and blockiness associated with the techniques available to date.

The data is simply 'reiterated' to fill in the gaps. Fractally compressed images are in any case easier for the eye to read, since they mimic the natural process. Even at extremely high compression ratios, or at high zoom levels, where the image might be expected to deteriorate beyond recognition, fractal images tend to appear as slightly blurry images rather than as a mass of barely indistinguishable pixels.

In terms of where we are now, the vision of the super-highway is in many ways just that, a future horizon, and the idea of fully 'wired' areas, a thousand channels of interactive TV, video on demand and home shopping have yet to hit the mainstream. However, given the upgrading of current linking technologies that carry voice and data, of bulletin board services and gateways into and out of the all consuming InterNet, we can foresee sporadic growth in an unregulated way - with the InterNet as the backbone of the first incarnation of the digital super-highway.

Most immediately, the implementation of Graphical User Interfaces (GUIs) (and the way in which fractal image compression will make it easier to include images as well as text) will make these services more and more usable and

Image files take up huge amounts of disk space. This original uncompressed bitmap represents 1,181KB of storage...

accessible, and much more likely to become a part of every day life.

The services available on the InterNet now and accessible to an ever growing pool of users will inevitably be reflected in the services delivered on the full highway. On-line education, for instance, is available now, as is teleshopping. On an InterNet super-highway these services will be interactive and much more visual than they are at present. Teleshoppers will be able to see and stroll around a store; video previews and reviews will be instantly available and Video On Demand services will enable you to browse through listings, look at a clip and then download the video of your choice. And even if you are going out you will probably be able first to call on pictures of the theatre or restaurant or museum you want to visit, as well as book your tickets.

Hold on - We are connecting you

It has to be said in passing that this is a marketing person's dream - to be able to target, precisely and according to demographic

...whereas this fractally compressed image takes up only 40KB...

profile and a host of other criteria, the customers you want without requiring them to step outside of their doors.

As time goes by the 'front end' of all these on-line services will continue to be the personal computer, at least in some form. They will form the basis of all interactive multimedia systems, while other devices and systems gradually migrate and become 'pluggable' to PCs. Both CD-I and 3DO have already announced plans to do just that. With the various communications technologies we have today gradually converging it is possible, even likely, that in the future some hybrid machine combining the qualities of television, videophone and general purpose computer will offer access to the super-highway via one control unit.

In essence the vast installed base of PCs will drive down component costs and drive technological advances very much faster than static technologies such as CD-I and 3DO on their own, and software advances will make 'surfing the net' immensely simpler.

...Reduced even further to 20KB, fractal images can show insignificant loss of detail. (Despite a transfer to print, virtually no difference can be seen even in the original photograph)

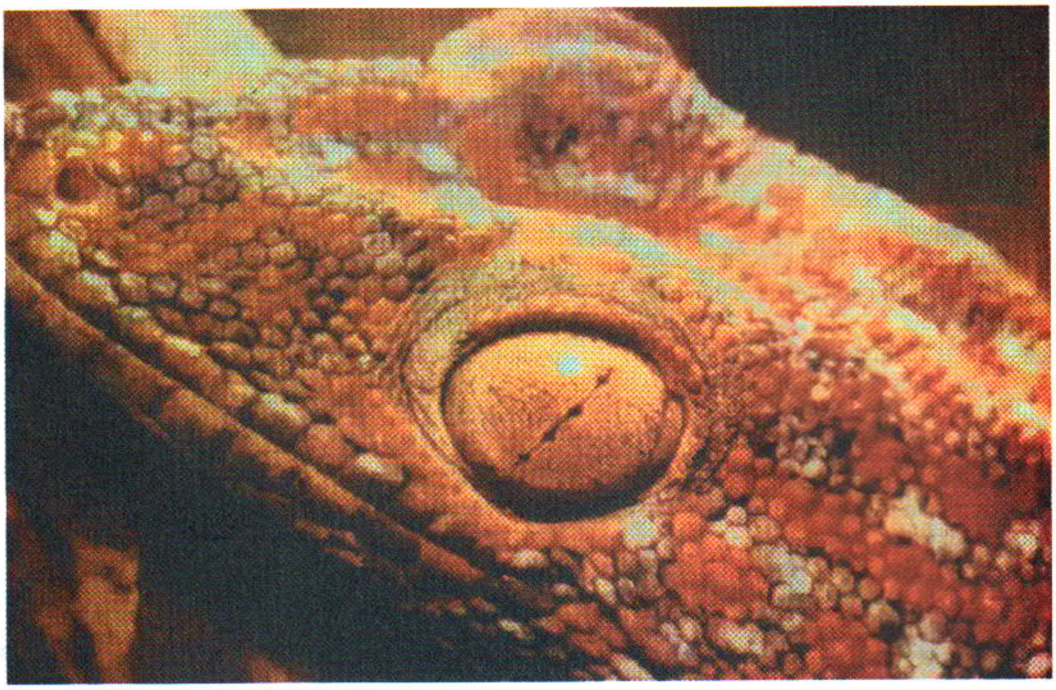
More Power to the PC

The GUI, however, is only the first of the massive computational requirements needed to mask the complexity of the underlying technology so that we are able to extract value from the information super-highway. As the data itself becomes more visual even greater processing power will be needed to analyze and recognize images intelligently. The 'agents' that will process data intelligently for you, so that you only receive information on the subjects you want to learn about, will not develop on non-PC platforms; other technologies - the multimedia games machines, for instance - whilst they are maturing towards compatibility with the PC, do not have the competitive pressures of the PC platform. They cannot double in power and halve in price every eighteen months.

Today's PCs offer a vast installed base and a broad spectrum of niche markets (therefore providing low barriers to entry for software developers). This in turn leads to a proliferation of titles and applications, and still greater demand for PCs and power.

Seeing is believing

The key principle of all these developments, and the factor which links them all, is of course their dependency on the use of images. Where advanced compression techniques such as the Fractal Transform fit into the scheme of things is obvious - even with the fastest modem speeds and/or ISDN lines, raw images/ visuals will always take up enormous space - with moving images the problem is simply exacerbated.



The fractal approach is a key enabling technology NOW and will continue to develop as the super-highway itself does. From still and moving image compression to intelligent image recognition the fractal architecture of images will, as we move into the late 1990s and the next millennium, be as important to the development of the digital super-highway as Euclidean geometry was to the development of CAD/CAM in the 70s and 80s. It may be unseen and (as yet) unknown to many but the developments in the fractal world will be key to the delivery, use, extension and development of the super-highway.

Biography

Alan McKeon, now International vice president of Iterated Systems, set up Iterated Systems Ltd. in 1991 as the UK subsidiary of Iterated Systems Inc. The UK was the first country to host an overseas office. Prior to joining Iterated Systems, Alan worked for Microsoft UK Ltd. where he was responsible for setting up a new sales channel via OEMs. He joined Microsoft from Dun & Bradstreet subsidiary Nielsen Marketing Research, where he established a pan-European specialist software division offering high value information analysis tools on the PC platform to leading FMCG retailers and manufacturers.

Alan has a BSc in Mathematics and Management Science from Manchester University.

Iterated Systems has grown since its inception to supply a range of products across a broad international base, offering ultra-high compression ratios to very small file sizes, fast decompression times and resolution independence.

The Fractal Transform's resolution independence enables pictures to be zoomed without blockiness. The first zoom (B) using traditional techniques, reveals the individual pixels that comprise the image. The second, fractal image (C), although zoomed to an even higher degree, shows only a slight bluriness and is much easier for the human eye to 'read'.

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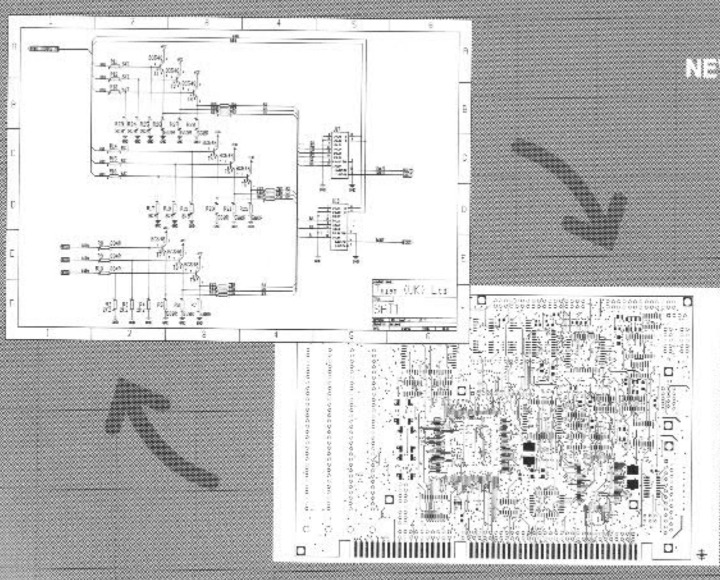
- Smooth scrolling
- Intelligent wires (automatic junctions)

- Dynamic connectivity information
- Automatic on-line annotation
- Integrated on-the-fly library editor
- Context sensitive editing
- Extensive component-based power control
- Back annotation from BoardMaker2



BoardMaker

BoardMaker1 - Entry level

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- Easy and intuitive to use
- Surface mount and metric support
- 90, 45 and curved track corners
- Ground plane fill
- Copper highlight and clearance checking

BoardMaker2 - Advanced level

- All the features of BoardMaker1 +
- Full netlist support - BoardCapture, OrCad, Schema, Tango, CadStar and others
- Full Design Rule Checking both mechanical and electrical
- Top down modification from the schematic
- Component renumber with back annotation
- Report generator - Database ASCII, BOM
- **NEW** Thermal power plane support with full DRC

BoardRouter

Gridless re-entrant autorouter

- Simultaneous multi-layer routing
- SMD and analogue support
- Full interrupt, resume, pan and zoom while routing

Output drivers - Included as standard

- Printers - 9 & 24 pin Dot matrix, HPLaserjet and PostScript
- Penplotters - HP, Graphtec & Houston
- Photoplotters - All Gerber 3X00 and 4X00
- Excellon NC Drill and Annotated drill drawings (BM2)

Call, write or fax for more information
or a full evaluation kit

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