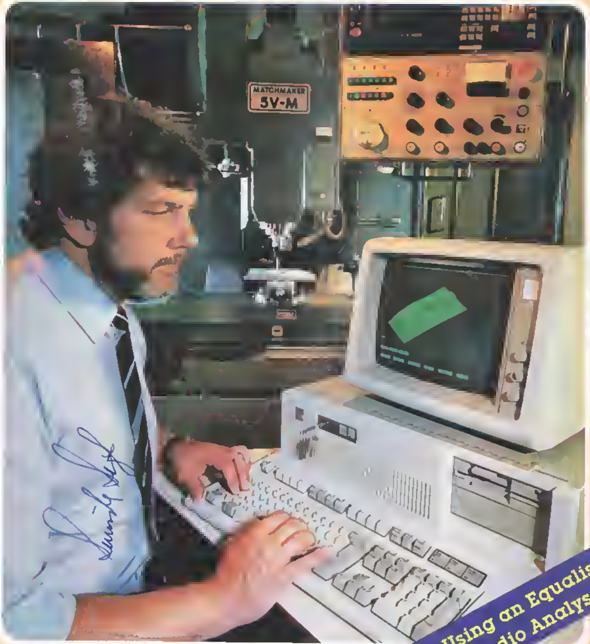
ISSN 0970-3993

63
July
1988

elektor INDIA electronics

Rs. 8.00

THE PRACTICAL MAGAZINE WITH THE PROFESSIONAL APPROACH



Using an Equaliser
Audio Analyser

Publisher: C.R. Chandarana
Editor: Surendra Iyer
Technical Adviser: Ashok Dongre
Circulation: J. Dhas
Advertising: B M. Mehta
Production: C.N. Mithagari

Address:
ELEKTOR ELECTRONICS PVT. LTD.
52, C Proctor Road, Bombay - 400 007 INDIA
Telex: (011) 76661 ELEK IN

Overseas editions:

Elektor Electronics
1, Harlequin Avenue,
Great West Road, Brentford TW8, 9EW U K.
Editor: Len Seymour

Publitron Publicacoes Tecnicas Ltda
Av Ipiranga 1100, 9° andar CEP 01040 Sao Paulo – Brazil
Editor: Juliano Barsali

Elektor sarl
Route Nationale; Le Seau, B P. 53
592270 Bailleul – France
Editors D R S Meyer,
G C P Raedersdorf

Elektor Verlag GmbH
Susterfeld Straße 25 100 Aachen – West Germany
Editor E J A Krempelsauer

Elektor EPE
Karaiskaki 14 16673 Voula – Athens – Greece
Editor: E Xanthoulis

Elektuur B.V.
Peter Treckpoelstraat 2.4
6191 VK Beek – the Netherlands
Editor: P E L Kersemakers

Ferreira & Bento Lda
R D Estefania, 32 1° 1000 Lisboa – Portugal
Editor Jorge Goncalves

Ingelek S A.
Plaza Republica Ecuador 2.28016 Madrid-Spain
Editor A M Ferrer

In Part:
Kedhorn Holdings PTY Ltd Cnr Fox Valley Road &
Kiogle Street Wahroonga NSW 2076 – Australia
Editor Roger Harrison

Electronic Press AB
Box 63
182 11 Danderyd – Sweden
Editor Bill Cedrum

Printed at: Trupti Offset
Bombay - 400 013
Ph. 4923261, 4921354

elektor electronics

Volume-6 Number-7
July-1988

Editorial

Audio & Hi-fi

Computers

General Interest

Test & Measurement

Information

Guide lines

Selex -36

Telecom Board, a Non Starter

The committee of secretaries set up by the Prime Minister to formulate a proposal for restructuring the Telecom Board and for setting up a Telecommunication Commission, has failed to arrive at a consensus and the proposal is being referrred back to the cabinet secretariat.

The proposal to restructure the Telecom Board and to create a separate advisory body for planning and development of telecommunication system, mooted two years ago is facing rough weather from various departments. The Department of Electronics and the finance ministry opposed the move of the Department of Telecommunication. Originally, the role of the board was to be restricted to the implementation of policies and development of telecommunication system. The commission was to enjoy wide powers and work out long-term strategies. The DoT proposal envisaged the ranks of secretaries to the six members of the board and that the board should have powers to present an independent telecom budget. The finance ministry opposed both these suggestions. The DoT agreed to give up the demand for presenting an independent budget but insisted on the rank of secretary to government of India to be given to the board members.

The Telecom commission met with resistance when the DoT said that the commission should have powers to control all the production units manufacturing telecom equipment. It sought even the licensing powers which encroached upon the power now enjoyed by the DoE and the Directorate-General of Technical Development.

Splitting up of functions hampered the telecom equipment manufacturing plan, the Dot argued. It took more than four months for the public sector Manikpur unit to get clearance for import of equipment and it delayed the production plan. The Dot's grievance is that the industry ministry and the DoE do not approve the proposals expeditiously.

The DoE, while commenting on the transfer of power to the proposed commission said that it was looking after the electronic industry and the telecommunication equipment units since their inception without any serious problems and that the department had the necessary infrastructure to monitor the growth of the industry while the Telecom Board was to be only a service organisation. DoE also pointed out that it was responsible for standardisation of the technology relating to electronics. Though DoT was the monopoly user of telecom equipment, DoE did not feel it necessary for the DoT to gain full control over the industry, including the unfettered right to make imports.

The DoT is hoping that the Union cabinet will sort out the issues raised by the Committee of secretaries. Incidentally, the DoT has a new secretary . Mr Sathya Paul has replaced Mr D.K. Sangal who retired after superannuation.

World Telecommunication day

Every year, May 17 is observed as the World Telecommunications Day and the theme for this year was "Technology Transfer".

Mr. T.H. Chowdry, managing director of Videsh Sanchar Nigam Ltd., on this occasion, made certain pertinent points in an article. Touching on this year's theme, Mr Chowdry said in India it could as well be the realisation of the technologies we have developed, at least in the switching area, but what would a switch do if it was not interconnected. What was needed now was another mission to develop radio and optical fibre systems and several other transmission equipment, he added.

Telecommunications are too serious and too valuable business to be left to technicians and engineers alone. The business provided annual service revenues of about Rs. 2000 crores and it called for an annual investment of more than Rs. 2000 crores. It required the attention of economists and also that of public policy makers.

The world telecommunication business is worth 400 billion dollars or about Rs. 520,000 crores. This is nearly 70 per cent of India's gross national product. In the second five-year-plan, 30 years ago, India invested a mere Rs. 66 crores in telecommunications. In the current year, the investment is 30 times more and yet not sufficient.

Having achieved universal availability of telephone, in the west, telecommunications are being perceived as a means to enhance competetiveness in national and international business, higher productivity of men, machines and systems. Every business is now able to afford its own in-house worldwide communication and this specific customer-driven market is causing the realisation of technologies quickly.

The development of new systems like a new generation of digital exchanges or communication satellites incorporating new facilities and increasingly customer oriented control is costing a lot. The development of a new generation of switch costs upto 1000 million dollars or Rs. 1300 crores in the west. To keep it in service and to incorporate service enhancement costs another 40 per cent every year.

Telcom network is global and so is the market. The global nature of the market requires worldwide consumption of productions and services. Unless the market share is about eight per cent, no company can stay in business. It means that only about a dozen companies can play safely in the global telecom market. Hence, the emerging trend in the world is to merge operations and seek bigger markets. This is a compulsion.

The AT & T of the US and Philips of Holland have a joint company; The GTE of US and Siemens of Germany have combined their operations; CIT-Alcatel of France and the multinational ITI are merged; The GEC and Plessy of the UK have combined.

The Canadian Northern Telecom and Germany's Siemens have made serious inroads into the US telecom market. Ericcson of Sweden has got a slice of the British market. The Japanese domestic market is under seige by the British and American companies. These developments are leading to the demand that telecommunications, which is a trade in information transport service, must be covered under the General Agreement on Trade and Tariff (GATT) to end all restrictions on free trade in telecom equipment and services.

Korea is a shining example of first chossing to produce entertainment electronics for mass consumption as a thrust area seeking collaborations and later gaining the capability to produce reliable and high quality products. Having got the components base established, it sought collaboration with foreign telecom equipment manufacturers and within a short time mastered the production technologies. At the same time, a number of companies joined together to mount research, design and develop Korean products as the C-DOT has been doing in India.

Malaysia has shown how to benefit from the world technology in communications. For example, to cost effectively commission telephones in rural areas, it chose Automatic Telephone Using Radio, which is also known as cellular mobile telephone. This was developed in the first instance to provide mobile telephones but this can also be readily utilised for providing telephones in far flung areas, remote and low density communities, very quickly.

Indonesia has given a franchise to a private company for a period of five years to implement ATUR. In return the company had to find all the capital and the foreign exchange to develop the market. The ATUR system will become the property of the country after the franchise period.

Things are becoming bright in India because the poor quality of communications hurts every section of the society. It can be hoped that the Malaysian or Indonesian example in providing rural telephones is emulated by Indian authroities too.

SPACE INDUSTRY

The major components of the Indian Space Programme comprise the launch vehicle development, the spacecraft development and the space applications programmes for communications and earth observations.

Indian industry had not only been contributing directly to the space projects and missions by developing and fabricating the various hardware that go into ISRO's launch vehicles and satellites but has also been building the ground infrastructure for all the components of the national space programme.

Currently, three major launch vehicle projects are underway namely the Augmented Satellite Launch Vehicle (ASLV), the Polar Satellite Launch Vehicle (PSLV) and the Geostationary Launch Vehicle (GSLV).

ASLV project, whose first developmental flight took place in 1987 was supported by over 70 industries and high-tech institutions. As the first flight failed, efforts are on to launch the second flight.

The motorcase hardware including the nozzles for its lower stages were fabricated by Larsen & Toubro, Powai, Walchandnagar Industries Ltd. and Anup Engineering, Ahmedabad. The fabrication of light alloy structures such as interstages, base shrouds and heat shield was undertaken by the space cell of HAL, Bangalore. The forgings of Alloy steel were supplied by BHEL, Hardwar, Republic Forge, Hyderabad and Echjay Industries, Rajkot.

For PSLV, contracts worth Rs. 150 crores were placed for various special materials, chemicals, machinery and equipment, electronic systems and special fabrications as well as general fabrication services. For its numerous electronics systems, PSLV is drawing major support from BEL, Bangalore and KELTRON, Trivandrum among others.

A crucial system presaging the development of geostationary satellite launch capability for India is the cryogenic propellant powered upper stage. A host of new technologies in advanced materials, precision fabrication, thermal insulation, liquid prpulsion and material handling will be introduced to the country in the process.

Long-term production and supply for the space programme has been established in various Indian industries for rocket propellants as well as the high-tech ingredients of solid propellants.

ISRO's two-decade old sounding rocket programme totally relies on Indian industry for supply of its hardware. The metallic hardware for stable sounding rockets have been under regular production at Ramakrishna Engineering Works, Madras, Anup Engineering Works, Ahmedabad and Alwar Engineering Works, Trivandrum.

Liberal policy suggested

A panel set up by the government of India under the chairmanship of Mr Abid Hussain, planning commission member, has suggested a liberalised approach for the private sector participation in "core industries".

The Abid Hussain panel has also recommended that a package of fiscal incentives be prepared for encouraging such a diversification. The panel is of the view that the core sector, where the public sector plays a prominent role, should be opened to the private sector. This would encourage investment from the private sector and provide competition for the public sector units. The areas suggested include heavy industries, high-tech areas such as electronics, telecommunication and biotechnology. Though private sector has been allowed entry in these areas on a lmited scale, the panel feels that the licensing policy should be further liberalised.

Dry cell battered

The dry cell battery industry is in doldrums and four major manufacturing companies in India have gone out of business in the last five years. Even after their exit, the existing manufacturers are saddled with excess capacity.

Five manufacturers now have an installed capacityof 1713 million batteries and their production is only 1208 million batteries. The excess capacity and negligible growth in demand may edge out even some of the existing companies.

In the 1970s, many business houses were lured into the dry cell battery business because of the phenomenal growth in the domestic market. Exports were also encouraging. In 1975-80, a 12 per cent growth was noticed in the market and additional capacities were installed on this basis. But, in 1980-85, the growth rate declined to 5.5 percent and it now stands at 3.5 percent.

The situation worsened further with the drying up of the export market. During 1980-85, the industry exported 54 million batteries in a year. The Soviet Union discontinued its imports since 1986 as they had enough domestic capacity.

In India, about 45 per cent of the dry cells are used in radios. The phenomenal growth in television coverage has hit the battery industry. With 72 per cent of the population covered by the Doordarshan, the radio listenership decreased. As more villages are electrified, the demand for torches is bound to be affected. About, 46 per cent of the batteries in India re used in torches. In January, 1988, the total villages electrified stood at 426,000. Rural India accounted for two-thirds of battery consumption and drought in the last three years hit the sales. If product price is lowered, battery usage will improve but the government does not accord priority to the product.

Chernobyl Today

The accident at the Chernobyl nuclear power station on April 26, 1988, gave the world its first real encounter with a severe nuclear power plant accident and implications that went beyond national boundaries.

The Chernobyl accident killed 31 and injured 300 and all of them were plant workers and firefighters within the plant area. The Soviet Union's hardwon expertise in decontamination is paving the way for a return to a more normal pattern of life in the area.

The three neighbouring reactors shut down after the accident are back in service at Chernybol, with units 1 and 2, back within nine months while unit 3, which shared certain systems with the destroyed unit 4, was recommissioned in December, 1987. Construction of units 5 and 6 has been halted now.

Unit 4 has been permanently entombed in a steel and concrete sarcophagus that confines the residual radioactivity in and around the reactor. More than 300 devices in the entombed unit monitor temperatures, radition levels and technical system performance.

For starting units 1, 2 and 3, the building complex had to be decontaminated. Liquid sprays, steam injection, dry metyhods based on polymer coatings, and clothing soaked in special solutions effectively reduced radiation to the present dose that allows compliance with the recommendations of the International Commission on Radiation Protection.

Working and living conditions in and around Chernobyl plant will soon become normal when the new town at Slavutich becomes home for thousands of plant workers and their families. Most of the workers are now housed in hostels built after the accident in the areas just outside the 30 – km zone around the plant.

In addition to decontamination, top soil removal, food and livestock monitoring and destruction and agricultural restrictions are reducing radiation dose levels to well below worst-case assessments made shortly after the accident. By summer 1987, 60,000 houses and other structures in 600 population centres had been decontaminated and residents of 16 villages in the outer part of the 30-km zone were able to return to their homes; clean up of towns closer to the plant continues.

The immediate visible health effects of radiation exposure are understood while the long-term implications are still unfolding. Immediately after the accident, the Soviet Union began a comprehensive medical surveillance of more than one million people. A long-term epidemiological study of the most exposed population group is underway.

The Chernobyl accident prompted Soviet specialists to take a fresh look at how safety systems could be further enhanced at the nuclear reactors.

Speaking Computer

Though electronic sensors are now widely used for laboratory measurements, many instruments, especially the more accurate ones, still have to be read by the human eye. This human element can and does lead to errors.

The National Physical Laboratory at Teddington, London, has developed a hand-held computer terminal to help the observers record readings from instruments faster and reliably.

Most metrologists believe that observations made by eye are, as a rule, recorded more easily by jotting them down on paper than by using a key board that calls for extra concentration to avoid mistakes. They generally prefer to write their observations and transfer them to a computer later, when data input can be carefully compared with their observation book.

The NPL device is a speaking computer terminal which dictates back the number entered into it through the key board. It also warns against improbable readings. The synthesiser has a vocabulary of 70 discrete words selected specifically for metrological purposes. It may be actuated by the terminal's keyboard to speak digits, or by the laboratory computer to speak any words from its vocabulary.

The essence of the device is its ability to speak the value of numeric input data. It speaks digits quickly and under very fast operation, truncates them to avoid a delay between pressing the key and starting to speak. For example, very rapid entry of say, "678", would cause "siseveight" to be spoken, but that is quite intelligible.

Another salient aspect of the system is that when a gross mistake is committed in entering the data, instead of the usual silent display on the screen, the computer orally gives a warning. The first NPL speaking computer was built three years ago and now eight are in service.

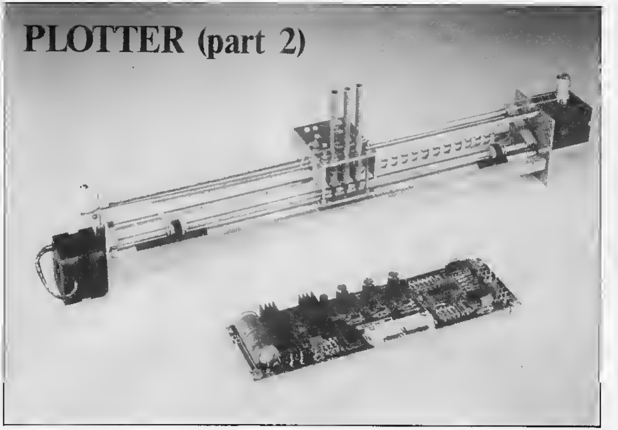
TV for elementary schools

About 100,000 television sets will be supplied to elementary schools in the country in a phased manner in 1988-89 and 1989-90. Also, 500,000 radio-cum-cassette players will be supplied to these schools under a scheme sponsored by the Union government.

The central government will bear 75 per cent of the cost of the project. All the suitable schools in a particular district or block in the states of Andhra Pradesh, Gujarat, Maharashtra, Orissa, Uttar Pradesh and the Union Territories where the INSAT was being already utilised to telecast educational programmes would be covered under the scheme.

The criteria for extending this scheme are that the schools should have at least two classrooms, two teachers, and electricity. They should be within the range of a TV transmitter. For supply of radio-cum-cassette player, electric supply is not essential. In the north-eastern region, 5000 TV sets would be installed in selected villages of Arunachal Pradesh, Assam, Manipur, Meghalaya, Mizoram, Nagaland and Triupura.

PLOTTER (part 2)



No hardware without software, and vice versa. In this month's final instalment we lend a hand to all constructors of the plotter who are eager to write control software, but need general algorithms, flow diagrams and elementary programming procedures as guidance for tailoring the communication patch between their computer, available graphics programs, and the plotter.

Before attempting to write a plotter interface program, it is necessary to acquire a basic understanding of computer control (*software/hardware*) in combination with graphics (or, more specifically, *drawing*). An algorithm needs to be devised for translating graphics information (on screen or in any form of memory) to actual pen positioning commands. The low-cost plotter described last month has no "on-board" intelligence, and must, therefore, be controlled at the bit level by the computer. In order to obtain reasonable drawing speed, it is necessary to write part of the control program in machine language, which accepts commands or command strings from a line editor, and translates these into pen movement commands by actuating the relevant control lines on the plotter interface board.

The bit assignment in the plotter control word is shown in Fig. 9. A stepper motor performs a full or half step, depending on the logic level of bit 2, on each positive transition of the clock signal (bit Ø). The clock pulse must remain logic high for at least 10 μs. Straight lines can be drawn by actuating the X or Y motor alone. Lines under an angle of 45° are drawn when the motors are actuated simultaneously. When both motors are actuated, but one is operated in the full step mode, and the other in the half-step mode, the slant angles become 26°34' or 63°27', corresponding to the tangent of 0.5 and 2, respectively.

Elementary routine

A number of routines and algorithms are given below to provide a basis for developing one's own software. It should be noted that the information given is intended as guidance for those who have little or no experience in handling computer graphics. It is beyond doubt that there are other, perhaps more efficient, ways of controlling the plotter, but the methods outlined here have the advantage of being illustrative and relatively simple to put in practice on a particular computer system.

The suggested elementary routine does what its name implies: it provides control of the most fundamental capabilities of the plotter. Depending on the structure of the 8-bit control word sent along as a parameter, a single full or half step is performed in the X or Y direction, and/or a particular pen is selected. Bits Ø and 3 determine which motor, or which motors, is or are actuated. A step is performed by the relevant motor when the associated clock bit goes logic low. Direction of travel and full/half-step operation are controlled by the remaining four bits.

9

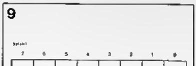


Fig. 9. Bit assignment in the plotter control word.

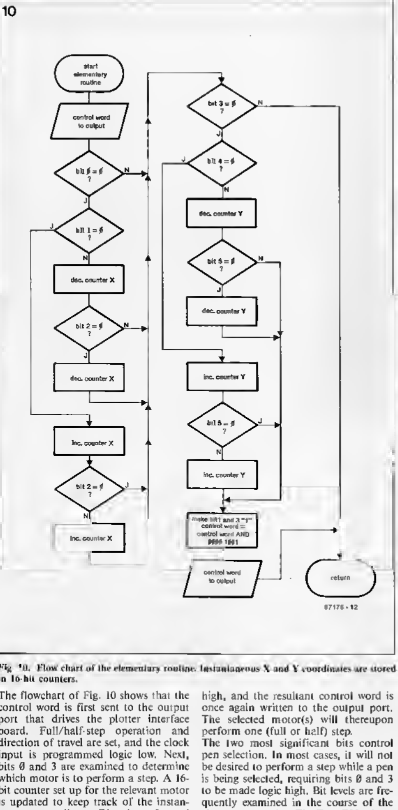


Fig. 10. Flow chart of the elementary routine. Instantaneous X and Y coordinates are stored in 16-bit counters.

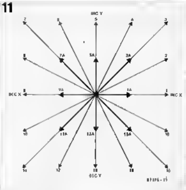


Fig. 11. Oblique lines of certain fixed angles are relatively simple to draw using the control words listed in Table I.

Table 1.

vector (Fig. 11)	control word BIN. 7 6 5 4 3 2 1 0	HEX.	DEC.
1	x x x x 1 0 0 0	0 8	0 8
1A	x x x x 1 1 0 0	0 C	1 2
2	x x 1 0 0 0 0 0	2 0	3 2
3	x x 0 0 0 0 0 0	0 0	0 0
3A	x x 1 0 0 1 0 0	2 4	3 6
4	x x 0 0 0 1 0 0	0 4	0 4
5	x x 0 0 0 x x 1	0 1	0 1
5A	x x 1 0 0 x x 1	2 1	3 3
6	x x 0 0 0 1 1 0	0 6	0 6
7	x x 0 0 0 0 1 0	0 2	0 2
7	x x 1 0 0 1 1 0	2 6	3 8
8	x x 1 0 0 0 1 0	2 2	3 4
9	x x x x 1 0 1 0	0 A	1 0
9A	x x x x 1 1 1 0	0 E	1 4
10	x x 1 1 0 0 1 0	3 2	5 0
11	x x 0 1 0 0 1 0	1 2	1 8
11A	x x 1 1 0 1 1 0	3 6	5 4
12	x x 0 1 0 1 1 0	1 6	2 2
13	x x 0 1 0 x x 1	1 1	1 7
13A	x x 1 1 0 x x 1	3 1	4 9
14	x x 0 1 0 1 0 0	1 4	2 0
15	x x 0 1 0 0 0 0	1 0	1 6
15A	x x 1 1 0 1 0 0	3 4	5 2
16	x x 1 1 0 0 0 0	3 0	4 8
pen selection			
pen 1	0 0 x x 1 x x 1	0 9	0 9
pen 2	0 1 x x 1 x x 1	4 9	7 3
pen 3	1 0 x x 1 x x 1	8 9	1 3 7
lift all pens	1 1 x x 1 x x 1	C 9	2 0 1

x = don't care; assumed logic 0 here

The flowchart of Fig. 10 shows that the control word is first sent to the output port that drives the plotter interface board. Full/half-step operation and direction of travel are set, and the clock input is programmed logic low. Next, bits ∅ and 3 are examined to determine which motor is to perform a step. A 16-bit counter set up for the relevant motor is updated to keep track of the instantaneous coordinate. Direction of travel and full/half-step operation is taken into account as the counter is decremented or incremented. Bits ∅ and 3 are made logic high, and the resultant control word is once again written to the output port. The selected motor(s) will thereupon perform one (full or half) step.

The two most significant bits control pen selection. In most cases, it will not be desired to perform a step while a pen is being selected, requiring bits ∅ and 3 to be made logic high. Bit levels are frequently examined in the course of the elementary routine. The Type Z80 microprocessor offers special bit checking instructions. These can be simulated by other processors by, for example, logic ANDing of the control word with a mask byte in which the bit to be examined is logic high. The result of the check can then be read from the status of the zero-flag.

Straight lines and pen selection

A small addition to the elementary routine makes it possible to draw straight lines at certain fixed slant angles. To begin with, the command word is set up, taking the bits for pen selection into account. The desired line length can be related to a specific X or Y coordinate. Each step is followed by a check for arrival at the end position. When this has not yet been reached, the next step is performed after a short delay. The delay time can be generated with the aid of a simple software loop, or a timer as available in, for instance, the 6522-VIA or Z80-CTC. A hardware timer has the advantage of making the final step rate, within limits, independent of the program routine that is to be executed between two steps. It is the task of the programmer to ensure that the motors operate smoothly in both the full and the half-step mode. Motor control can be enhanced by programming equal acceleration and deceleration rates for both motors when these are being stopped and started. This reduces the risk of one motor lagging because it misses out on a few steps, and in addition keeps longitudinal vibration of the pen carriage to a minimum (this effect is caused by inertia in combination with elasticity of the string).

The wind-rose shown in Fig. 11 is drawn by actuating one or both motors, in combination with direction of travel and full/half-step operation. The number of full or half steps is always constant. Reverse the polarity of one stator when a motor revolves in the wrong direction. Table 1 may be examined to see how the various control words for drawing the wind-rose were built from individual command bits.

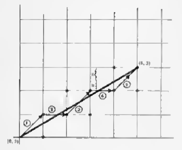
Bits 6 and 7 allow four logic combinations: three for putting the invidual pens on paper, and one (combination 11_2) for lifting all three pens simultaneously. Pen-down commands are preceded by a small, fixed, displacement in the X direction (offset, 58 or 116 steps) to compensate the distance between the pens in the carriage.

Random lines: Bresenham's algorithm

The drawing of oblique lines under slant angles other than the fixed ones discussed above is relatively complex. In most graphics applications, the working area is considered a system of coordinate axes. In this, a plotter should be able to draw a straight line between two random coordinates. In practice, however, the line drawn by the plotter will deviate from the desired, ideal, line owing to the limited number of discrete pen positions. Bresenham's line algorithm allows close approximation of the ideal line between random points in the coordinate system.

The drawing and Table in Fig. 12 illustrate the theory behind Bresenham's line algorithm. It is assumed that a line is to be drawn from starting point X1,Y1 — set at coordinates 0,0 for convenience's sake — and destination X2,Y2 at coordinate 5,3. Assuming the slant angle of the line to be smaller than 45° (Y2≦X2), the line can be drawn by actuating the X motor one step per increment, or the X and Y motor simultaneously. The choice between these options is determined by the difference between *a* and *b*. When *a* is greater than *b*, only the X motor is actuated, else the X and Y motor simultaneously. In essence, the procedure entails measuring the

12



(X1, Y1) = instantaneous coordinates
(X2, Y2) = end coordinates
dX = X2 - X1 dY = Y2 - Y1

step	steps	error	decision	movement
1	8	1	error > 0	2
2	6	-3	error <= 0	1
3	5	3	error > 0	2
4	3	-1	error <= 0	1
5	2	5	error > 0	2
6	0	—	return	—

Fig. 12. Bresenham's line algorithm. The ideal line is shown in bold print, the dots in the raster form the discrete positions that can be reached by the pen. The choice between stepping in the X or Y direction, or stepping obliquely (X and Y simultaneously) is made after calculating the difference between *a* and *b*.

13

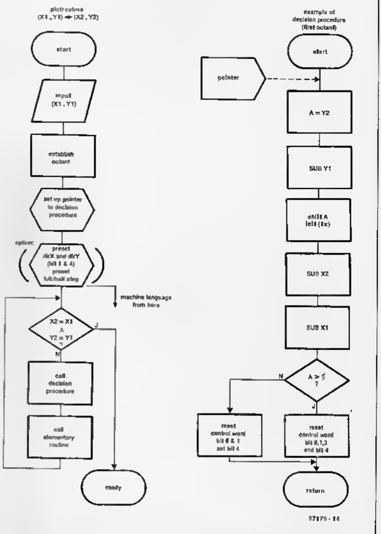


Fig. 13. Suggested flow chart for drawing lines to Bresenham's algorithm. The right-hand sequence is an example of one of the eight decision routines listed in Table 2.

angle of the line that can be drawn between the instantaneous and destination coordinates. When this angle is greater than 22°30' ($2dY-dX>0$), the next discrete position, X+1,Y+1, is stepped to at an angle of 45°. Otherwise, only the X motor performs a step.
The above algorithm is attractive because it allows simple calculations to be used for the decision procedure. Displacements dX and dY are deduced by subtraction, while multiplication by two is effected at machine code level by a single shift-left operation in the accumulator.
The same algorithm can be used for lines of angles between 45° and 90°, provided X and Y are exchanged. Lines in the remaining three quadrants are also fairly simple to draw to the above method. It is necessary, however, to determine beforehand in which octant (half quadrant) the destination coordinate will be with respect to the start-coordinate.
The flow diagram of Fig. 13 shows how lines between random coordinates can be drawn using Bresenham's algorithm. A routine is included to find out in which octant the destination coordinate is going to be with respect to the start-coordinate. Depending on the result, a pointer is preset to point to one of eight decision routines listed in Table 2a. In these, the control word is set up to define which motor (or motors) is to perform a step in a certain direction.
The actual stepping is done by calling the elementary routine. After each step, the instantaneous coordinates are compared to the destination coordinates (X2,Y2).

Table 2a

octant	Δa	Δb	movement 1		movement 2	
0... 45°	+dX	+dY	inc X	—	inc.X	inc Y
45 . 90°	+dY	+dX	—	inc.Y	inc.X	inc.Y
90. .135°	+dY	−dX	—	inc Y	dec X	inc.Y
135...180°	−dX	+dY	dec.X	—	dec X	inc.Y
180 . 225°	−dX	−dY	dec.X	—	dec.X	dec.Y
225. .270°	−dY	−dX	—	dec.Y	dec.X	dec.Y
270...315°	−dY	+dX	—	dec.Y	inc.X	dec.Y
315...360°	+dX	−dY	inc.X	—	inc.X	dec.Y

dX = X2 − X1 X1, Y1 = start coordinates
dY = Y2 − Y1 X2, Y2 = destination coordinates
ERROR = 2 · Δb − Δa initial error
dERROR1 = 2 · Δb error change after movement 1
dERROR2 = 2 · Δb − 2 Δa error change after movement 2

Table 2b.

	control word
inc. x	x x x x x x 0 0
dec. x	x x x x x x 1 0
inc. y	x x x 0 0 x x x
dec. y	x x x 1 0 x x x

Table 3.

```
dX = X2 - X1
dY = Y2 - Y1
error = 2dY - dX
derror1 = 2dY
derror2 = 2dY - 2dX
steps = dX + dY

WHILE steps > 0 DO
  IF error <= 0 THEN
                 step X
                 error = error + derror1
                 steps = steps - 1
               ELSE
                 step XY
                 error = error + derror2
                 steps = steps - 2
               ENDIF
ENDWHILE
```

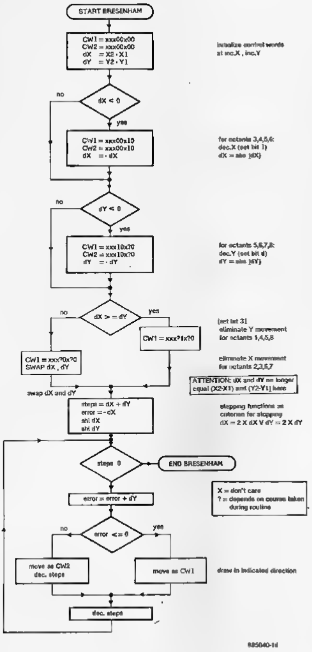
Algorithm for octant one

Bresenham's line algorithm derives step information from the distance to be covered in the X and Y direction (dX and dY respectively). The algorithm for the first octant (angle between 0 and 45°) is shown in Table 3. First, dX and dY are calculated to obtain the initial value of decision variable "error", which must be corrected (updated) after each step. Depending on the direction of travel, "error" is corrected with d-error1 (after movement 1) or d-error2 (after movement 2). Variable "steps" holds the number of steps to be performed in the X and Y direction, and is used for stopping the plot routine in time. The actual plotting is done in a WHILE-DO loop. Depending on the value of "error", steps are straight (X or Y) or oblique (X and Y). Variable "steps" is decreased by one or two in accordance with the movement performed (remember that one oblique step is one step in the X direction and one in the Y direction, i.e. two steps in all).
The Table in Fig. 12 and the flowchart in Fig. 13 illustrate the operation of the algorithm with the aid of some variables. As already stated, the routine is only valid for the first octant. It is, however, fairly simple to modify for drawing lines in other octants. Depending on the octant in which the line is drawn, it will be necessary to:

- use the absolute value of dX and/or dY;
- swap dX and dY;
- adapt the two elementary movements.

Table 2a provides an overview of the above functions for each of the eight octants. The drawing of a line between two points in an arbitrary octant requires an extended version of the line routine. The flow diagram of this is shown in Fig. 14. The first part of the program (up to ATTENTION) is, in fact, a programmed version of Table 2a. This part of the routine ensures that the actual plot routine (the loop at the end) draws a line in the correct direction. The calculation of "error" is scattered over several branches, but is still in accordance with Table 2a when the decision routine is called.
The listing in Table 4 is a Pascal procedure written after the flow-chart of Fig. 14. It should be noted that the program is intended to draw lines on a computer screen, so that instantaneous coordinates X and Y are read and updated for use as end criterions. Variables

14



STEP1 and STEP2 correspond to control word 1, and variables STEP3 and STEP4 to control word 2 in the flow diagram.

Circles and ellipsoids

The plotter will have to draw circles frequently. A set of coordinates of a circle can be computed with the aid of two tables: one holds data of one period of a sine function, the other data of one period of a cosine function. Table entries are rounded off to the nearest integer. The sine table then holds X coordinates, the cosine table Y coordinates. The amplitudes form the radius in the X and Y direction. Equal amplitudes result in a circle, unequal amplitudes in an ellipsoid whose major axis runs in parallel with the X or Y axis. Ellipsoids which are oblique with respect to the X or Y axis are obtained by mutual shifting of the tables. This effectively creates phase shift variation.

Calculation of coordinates lays a rather heavy claim on processor time, and is, therefore, done beforehand. The result, in the form of two tables, is stored in memory. The circle can then be drawn by having the plotter step from point to point using the Bresenham algorithm.

Extending the control program

The previously discussed elementary routine and general algorithms should enable programmers to develop a suitable control program for their computer. The bulk of the plotter control program may be written in a higher programming language, but there is no way to go round machine code for time critical routines. The final program should enable drawing

- lines between arbitrarily chosen coordinates (absolute function);
- lines between the current pen position and a coordinate defined with respect to that position (relative function);
- standard figures such as circles, squares, etc.;
- characters (letters, symbols and numbers).

Each character should have a corresponding set of relative coordinates, which can be multiplied by a fixed factor for enlarging or reducing character size.

TW

Fig. 14. Flow diagram of the extended line drawing routine.

```
PROCEDURE BRESENHAM (X1,Y1,X2,Y2): WORD; ATTRIBUTE: CHAR; PAGE: BOOLEAN);

VAR X, Y, dX, dY, ERROR, STEP1, STEP2, STEP3, STEP4: WORD;

BEGIN
  dX := X2 - X1;
  dY := Y2 - Y1;
  STEP1 := 1; {initialize all steps at +1}
  STEP2 := 1;
  STEP3 := 1;
  STEP4 := 1;
  IF dX < 0 THEN BEGIN {initialize for octants 3, 4, 5, 6}
                   STEP1 := -1; {step backwards in X direction}
                   STEP3 := -1;
                   dX := -dX     {dX := ABS (dX)}
                 END
  IF dY < 0 THEN BEGIN {initialize for octants 5, 6, 7, 8}
                   STEP2 := -1; {step backwards in Y direction}
                   STEP4 := -1;
                   dY := -dY     {dY := ABS (dY)}
                 END
  IF dX >= dY THEN {eliminate Y direction in movement1 for octants
                     1, 4, 5, 8, and initialize decision variables.}
                   STEP2 := 0;
             ELSE {eliminate X direction in movement1 for octants
                     2, 3, 6, 7, and initialize decision variables.}
                   BEGIN
                     STEP1 := 0;
                     {dX and dY must be swapped.
                      ERROR serves as an auxiliary variable}
                     ERROR := dX; dX := dY; dY := ERROR;
                   END;
  {start plotting algorithm}
  X := X1; {make instantaneous and start coordinates equal}
  Y := Y1;
  ERROR := -dX;
  dX := 2 * dX; {these two lines prevent}
  dY := 2 * dY; {multiplications in the loop}
  HPLOT (X,Y,ATTRIBUTE,PAGE); {plot first pixel on screen}
  WHILE (X<>X2) OR (Y<>Y2) DO
  BEGIN
    ERROR := ERROR + dY;
    IF ERROR <= 0 THEN BEGIN {movement 1}
                         X := X + STEP1;
                         Y := Y + STEP2;
                       END
                  ELSE BEGIN {movement 2}
                         X := X + STEP3;
                         Y := Y + STEP4;
                         ERROR := ERROR - dX;
                       END;
    HPLOT (X,Y,ATTRIBUTE,PAGE) {plot pixel X,Y on screen}
  END
END;
```

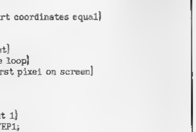
Computer user groups and individual programmers are invited to write universally applicable software drivers for the plotter described. Publication in this magazine can be arranged in cooperation with the editor and a technical assessment committee. We are particularly interested in programs to the Hewlett Packard Graphics Language (HPGL) standard for the following computers: Acorn Archimedes, Acorn Electron, Amstrad CPC464, Atari ST, BBC B, Commodore Amiga, *Elektor Electronics* BASIC computer, IBM PC XT/AT and compatibles (Amstrad 1512/1640), MSX-2, Sinclair Spectrum and Quantum Leap.

Writers who have submitted programmes found suitable for publication are offered a publication fee and a year's subscription to this magazine. The closing date for sending in suggested programs is 1 December 1988.

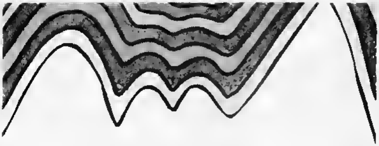
Editorial Office
1 Harlequin Avenue Harlequin
BRENTFORD TW8 9EW
England.

Rain Synthesiser

This simple circuit has proved very reliable and effective as a background sound effect generator for use by organists etc.
Other simple devices of this type often use several stages of amplification or make use of special noise diodes which are comparatively expensive. The advantage of this design is that it employs an ordinary OA 91 or similar diode. The internal noise produced by the diode is amplified by the single stage preamplifier, consisting of T1 and its associated components, which is designed for high gain and low cost. T1 can be almost any silicon NPN transistor, a BC107 being used in the prototype.



The output at X may be taken straight to an amplifier if only a white noise output is required. However, the addition of the passive filter, comprising C3 and P1 enables a variety of effects ranging from light rain to a heavy storm to be obtained.



using an equaliser

Although there are many different types of equaliser, they all perform the same basic task, namely the correction of deficiencies in the frequency response of one's speaker system and/or listening environment. As such they represent an extremely useful tool in the quest for 'perfect' hi-fi. Unfortunately, however, equalisers are all to often misused, and in extreme cases actually do more harm than good. The following article takes a look at the various types of application for which equalisers are most suited, and also explains how to get the best out of this versatile instrument.

The great advantage of an equaliser is that, unlike conventional bass and treble tone controls, which can provide only a fairly limited amount of boost or cut at the extremes of the audio spectrum, it is possible to iron out (equalise) peaks or dips in a response over the entire range of audio frequencies. Not only that, but with a *parametric* equaliser, the centre frequency, Q and gain of the equaliser filters can all be tailored to exactly compensate for non-linearities in the response of any given system.

Although the use of equalisers was originally limited to professional sound recording studios, their undoubted benefits have led to an increasing number of amateur applications: dedicated hi-fi enthusiasts, having lavished considerable attention and expense on cartridges, pick-up arms, turntables, amplifiers and loudspeakers, are now resorting to equalisers to 'upgrade' the last link in the audio chain, namely the listening room.

Unfortunately, however, many amateurs fail to make the most of the facilities offered by a sophisticated parametric equaliser, and simply end up using it as a sort of 'super-duper' tone control, twiddling the knobs to get a bit more bass here, less treble there and so on. This article is therefore intended to provide a few insights on how to achieve effective room equalisation, whether it be for domestic or PA-system applications.

Equalising your living room

In recent years the subject of room equalisation has become something of a fad. Various audio design consultants and well-known manufacturers of audio equipment have conducted extensive research into the response of domestic listening environments. Bruel and Kjaer, for example, offer a comprehensive measurement and equalisation system for listening rooms, whilst Philips loudspeakers are specially designed to compensate for the deficiencies of the 'average living room'. The subject of room equalisation, with particular reference to the effect of the placement of loudspeakers, has been discussed in a spate of recent articles, and numerous hobbyist magazines have produced designs for (graphic) equalisers. There is no doubt that people are now generally aware of the effect of the shape and contents of the listening room on the reproduction of the audio signal.

That the room has considerable effect is hardly surprising, especially when one considers how much care and attention is paid to the internal construction of loudspeakers (bracing ribs, damping

1

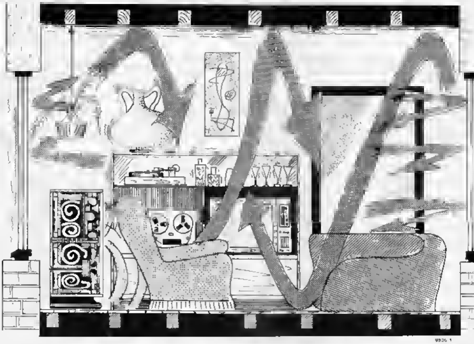


Figure 1. Considerably more attention is spent on the internal design of loudspeakers than on the interior of one's living room, despite the fact that the latter has a profound effect upon the sound of the music signal being reproduced.

materials, air-tight seals etc.): in a sense, the listening room is simply a giant loudspeaker cabinet, in which the listener sits. However, as a rule little or nothing is done to improve the response of the room. Of course it is possible to take such steps as to change the curtains, fit wall-to-wall carpeting, experiment with different loudspeaker placings, swop the furniture around etc. Although whether the living room will remain liveable-in is another question!

A simpler solution to the problem of 'upgrading' your living room is to employ an equaliser, which will compensate for the inherent deficiencies in the room's frequency response

Assuming, for example, that the room in question has the response shown in figure 2a. Using an equaliser the response of the audio system can be tailored to look like that shown in figure 2b, i.e. the inverse of the room's response, with peaks at 1600 Hz and 4 kHz, dips at 50

2

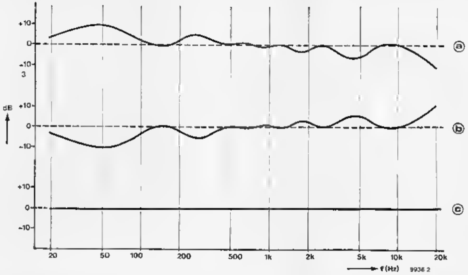
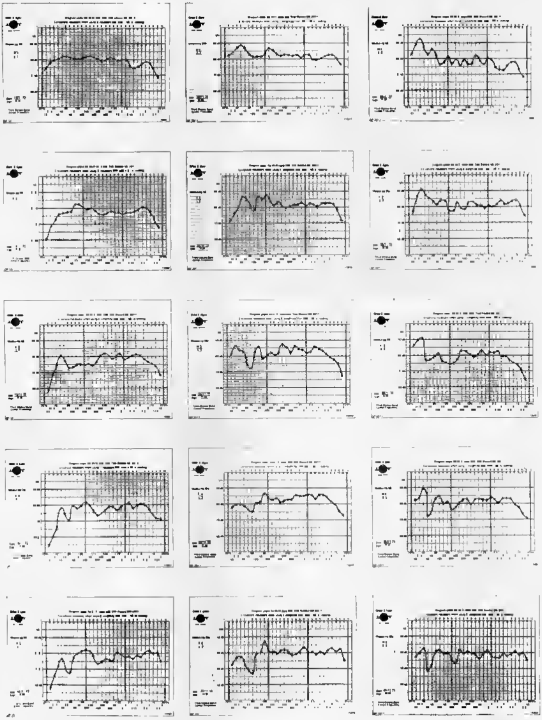


Figure 2. An example of how, in principle, it is possible to obtain a uniform frequency response with the aid of an equaliser. The irregular response of figure (a) is smoothed out by setting up the inverse response (shown in figure (b)) on the equaliser filters. The result (figure (c)), in theory at least, is the desired perfect reproduction.



A page from Bruel and Kjaer application note 13-101, which throws an interesting light on the topic of room acoustics. The frequency responses shown here were measured using 5 different loudspeakers, set up in 3 different living rooms.

3

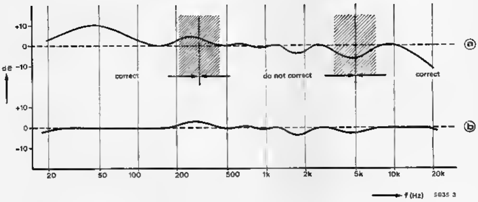


Figure 3. In hi-fi applications it is neither necessary nor indeed advisable to attempt to iron out every single peak and dip in the response. In particular, the band of mid-range frequencies between approximately 300 Hz and 5 kHz is best left untouched, so that the resultant corrected response will look something like that shown in figure 3b.

and 250 Hz and treble boost above 10 kHz. Thus, in theory, the resulting combined frequency response (i.e. that which, so to speak, reaches the ears of the listener) should be the perfectly flat line shown in figure 2c.

Unfortunately, however, as one might expect, things are not quite so simple in practice. The situation is complicated by the fact that the signal which reaches the listener is a mixture of direct and indirect sound. The direct sound is that which travels straight from the loudspeakers to the listener's ears, whilst the indirect sound is that which has first been reflected off the walls, ceiling, floor and furniture. It is the indirect sound, therefore, that is 'coloured' by the acoustics of the rooms. This fact has two consequences:

The relative proportions of direct and reflected sound will vary at different points in the room. Due to path length differences between the direct and indirect signals, either phase cancellation or phase reinforcement may occur, creating nodes and anti-nodes at *different* locations in the room. For this reason it is only possible to equalise the frequency response of a particular listening position. If that position is altered the frequency response will have altered also.

4

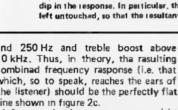


Figure 4. In most cases it is a relatively simple affair to incorporate a switch selectable 6 dB attenuator into a P.A. system. A resistance R_V of approximately the same value as the volume control (P_V) is connected in series with the latter, and a pushbutton switch S_V is then connected in parallel with the resistance.

Secondly, the human ear responds differently to direct and reflected sound, particularly at frequencies within the vocal spectrum between roughly 300 Hz and 5 kHz. The direct sound is recognised as the primary factor determining the 'quality' of the sound source, whilst the reflected sound provides information relating to the listening environment. Excessive equalisation can therefore lead to highly undesirable results, namely strong colouration of the direct sound in an attempt to compensate for a reflected signal heavily influenced by the room acoustics. As already mentioned, careless or over-enthusiastic use of an equaliser can do more harm than good. However the prospective user should not be put off by this fact, since an equaliser can offer tangible benefits to the hi-fi enthusiast who, for practical reasons, is constrained to listen to his system in a small and acoustically-poor room, with his speakers positioned in non-ideal locations.

The advantages of an equaliser can be illustrated by taking a closer look at the frequency response of a typical living room, as shown in figure 2a. The same curve is shown again in figure 3, with several 'critical' areas emphasised. For the band of frequencies from roughly 300 Hz to 5 kHz, the golden rule is 'leave well alone' (assuming that it is the acoustics of the room and not deficiencies in the response of the loudspeakers which are responsible for irregularities in the response). However peaks and dips in the response which occur at frequencies outside this band can be flattened out with the aid of an equaliser; at frequencies which are at the junction of these regions (i.e. around 300 Hz and 5 kHz), limited equalisation may be useful in certain cases. What this means for the response curve of figure 3a is this:

- the prominent resonance at around 50 Hz can be completely eliminated (that this also results in an improvement of approximately 10 dB in the signal-to-noise ratio is an added bonus).
- The smaller peak at around 250 Hz lies in a transitional area, thus partial equalisation is possible, if desired. The most sensible procedure is to audibly compare the results obtained with and without equalisation.
- The barely perceptible 'bump' at 150 Hz is really too small to be worth considering; furthermore it lies right in the middle of the critical mid-range of frequencies and should therefore be left untouched.
- The dip at around 1600 Hz is likewise inside the critical vocal spectrum which should be avoided.
- The somewhat larger dip at approximately 5 kHz straddles the second crossover area, thus once again a partial or limited equalisation may prove worthwhile.
- Finally, the roll-off in the response above 10 kHz can legitimately be corrected with the equalizer; care should be taken not to apply excessive amounts of boost, however, since there is the danger of damaging one's tweeters (!)

After the above corrections have been carried out (and assuming that the dip at around 1600 Hz is the result of the room acoustics and not one's loudspeakers), the overall response which is obtained, should look something like

5

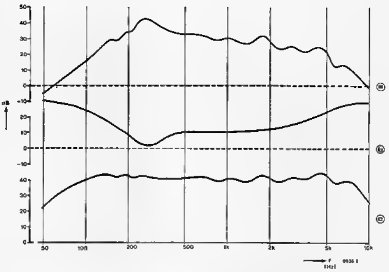


Figure 5. The frequency response of P.A. systems is frequently fairly poor. That shown in figure 5a is a typical example. With relatively simple equalisation, however, (figure 5b) one can obtain a response like that shown in figure 5c, which in practice improves the quality of reproduction to a quite amazing degree.

that shown in figure 3b — and hopefully there should be a correspondingly discernible improvement in the resulting sound!

As the above example illustrates, it is not necessary to make a large number of corrections in order to obtain an 'acoustically' flat response. All that is required in this example is a circuit to provide treble boost, and three variable resonance filters — in fact those facilities offered by the type of parametric equaliser.

The following paragraphs describe how to go about actually setting up an equaliser for optimum results in a variety of practical situations.

P.A. systems

P.A. systems used in conference halls and auditoria are usually installed by professionals. However there are many situations such as local community meetings, school prizegivings etc. where smaller halls have to be set up acoustically by comparative 'amateurs'.

The most common problems encountered in this type of case are 'lack of intelligibility', 'not loud enough', and persistent acoustic feedback. Before explaining the main causes of these problems a few preliminary remarks on the nature of P.A. systems would not go amiss. The primary aim of a P.A. system is not to achieve 'high-fidelity' reproduction, but rather optimum intelligibility. Unfortunately, in practice this is often confused with maximum volume. Of course, in some cases intelligibility can be improved by bumping up the volume, but it is often true, particularly in badly designed or wrongly set-up systems, that increasing the output from your speakers simply produces the dreaded acoustic feedback or 'howlround'. One must therefore attempt to (a) make the system less susceptible to feedback, and (b) search for other ways of improving intelligibility than simply winding up the volume control.

To take the problem of acoustic feedback first: most people know that this irritating phenomenon is caused by sounds from the loudspeakers being picked up — either directly or via reflections off the walls, ceiling, etc. — by the microphone(s). These are then amplified, fed back to the loudspeakers, only to be picked up once more by the microphones, and so on until a nasty high-pitched howl is produced (hence the name 'howlround'). In order to increase the volume without provoking this unpleasant effect, the only answer is to ensure that less of the loudspeaker signal is picked up by the microphone(s). This can be done in several ways:

- by using directional (cardioid) microphones, which are less sensitive to sound from the rear.
- by using loudspeakers which also have a directionally dependent response. It is probably not so well known that cardioid loudspeakers exist. By positioning these with their backs to the microphones, acoustic feedback can be considerably reduced.
- by not positioning the loudspeakers right next to the microphones. This may appear rather an obvious point, but it is surprising how many people fail to observe this elementary precaution.
- by setting the output level of those speakers which are nearest the microphones lower than that of speakers situated further down the hall. Many loudspeakers already have a facility for reducing the output level; in those that do not it is a simple matter to incorporate a small value series resistor to provide the desired level of attenuation. This step may at first appear a little self-contradictory, however it allows the amplifier volume to be turned up without significantly increasing the feedback signal to the microphones.
- at any given time, do not have more microphones switched on than is necessary. If there is only one person speaking, then one microphone is all that is required. Switching additional mikes on will simply increase the chance of feedback.
- ensure the volume control is adjusted correctly! This may also appear to be rather an obvious point, but in practice it is often more difficult to observe than it may seem. The following couple of tips should help:

— acoustic feedback is more liable to occur in an empty hall than in a full one. For this reason it is often sufficient to adjust the volume control so that the system is just on the point of 'howlround', with an empty hall. Once the hall has filled up the volume setting should prove spot-on.

– The difference between a correct volume setting and one which is just on the verge of howlround is about 3 to 6 dB. It is often possible to tell when a system is on the verge of howlround by the fact that it sounds decidedly 'echoey' – the effect is slightly similar to that obtained with artificial reverberation units. One can capitalise on the above fact by incorporating a switched 3 to 6 dB attenuator in series with the volume control (see figure 4). With the attenuator switched out of circuit, one first adjusts the volume control unit the P.A. system just starts to howl-round (bear in mind that acoustic feedback builds up gradually), then one simply switches in the attenuator, and the system should be ready for use.

Once acoustic feedback has been reduced to a minimum, the next step is to attempt to increase the intelligibility of the P.A. system without recourse to the volume control. There are basically two main ways of doing this: reduce the amount of reverberation generated in the hall, and improve the quality of the sound itself. The former point basically boils down to improving the acoustics of the hall by installing heavy curtains, thick carpeting, etc., and unfortunately is normally fairly expensive. The second measure, i.e. improving the reproduction of the speech signal is where electronics, in the shape of an equaliser, come in. It is not generally appreciated that the *quality* of the reproduced sound signal plays an important part in determining its intelligibility. It has been proven time and again in practice that a flat frequency response over a reasonably wide spectrum – roughly 100 Hz to 10 kHz – will lead to a considerable improvement in the intelligibility of the average P.A. system. Unfortunately, however, there are a number of prevalent misconceptions regarding the ideal frequency response and how to obtain it. These have led to the appearance of such monstrosities as bass cut 'speech switches' which roll off the response below 200, 300 or even 400 Hz, special 'speech' (loudspeaker) cabinets, which often have a truly horrific response, and speech microphones (whose response is sometimes little better that that of the loudspeakers). All that is needed is for the bass tone control on the amplifier to be set to minimum and the 'presence filter', which, more likely than not, has also found its way into the P.A. system, to be switched in, and one has all the ingredients for a full-scale acoustic disaster!

Figure 5a shows the measured response obtained from such a set-up, with the tone controls set to their mid-positions(!).

Using a simple parametric equaliser, the attempt was then made to iron out the grosser irregularities by employing the filter response shown in figure 5b. The resultant overall response is shown in figure 5c. What cannot be shown however is the amazing improvement in the intelligibility of the sound signal as a consequence of this measure. Whereas previously the speaker could barely be understood in an extremely quiet environment, after the equaliser had been used every word was clearly intelligible even with the noisiest of audiences.

Practice has proven that an equaliser is an extremely useful and effective tool for obtaining clear and readily comprehensible reproduction when working in halls with difficult acoustics. However, the way in which an equaliser is used in P.A. applications differs from that when employed with domestic hi-fi systems. It has already been stated that, when equalising the response of an audio chain and/or listening environment, the band of frequencies between roughly 300 Hz and 5 kHz should be left well alone. In the case of a P.A. installation, however, almost exactly the opposite is true: precisely this range of frequencies between 300 Hz and 5 kHz – or to be more accurate, the slightly broader band of frequencies between 100 Hz and 10 kHz – should be corrected with the equaliser. The extremes of the audio spectrum are of little significance for the intelligibility of the resultant speech signal.

Furthermore, whether the response of the reproduced signal is *completely* flat or not is also of secondary importance. For example dips in the response of up to 4 or 5 dB will often have little audible effect. The crucial factor as far as P.A. systems are concerned, is the presence of large resonant *peaks* in the response, since the highest peak effectively determines the maximum setting of the volume control which can be used without causing howlround. Consequently, the equaliser should be employed to ensure that all the peaks in the system's response are on the same level. This process is illustrated in figure 6. Although, at first sight, the response curve of figure 6a may appear to be slightly better, in practice superior results will be obtained with the curve in figure 6b. Of course, as it stands the latter response is far from perfect, and with judicious filtering it is possible to achieve the optimum response shown in figure 6c.

For those readers who are still less than convinced as to the advantages of an equaliser in this type of application, it may be worth pointing out that the cost of a (home-built) equaliser is nothing compared to the price of new microphones or speakers.

Electronic music

A less common but nonetheless important area of application for equalisers is in electronic music, where their flexibility and tone-shaping capabilities make them a useful addition to electronic synthesisers and organs. In direct contrast to the procedure

6

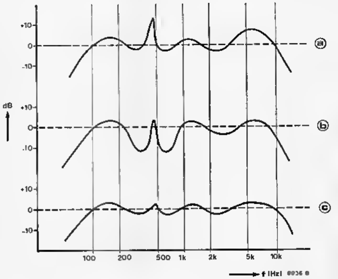


Figure 6. In the case of P.A. systems the equaliser should be set up such that all the peaks in the response have approximately the same amplitude. Although the curve in figure 6a may at first sight appear the better of the two, the fact is that the response of figure 6b will give superior results in practice. That is of course not to say that the latter represents an ideal case; using the same filters it is also possible to obtain the response shown in figure 6c.

7

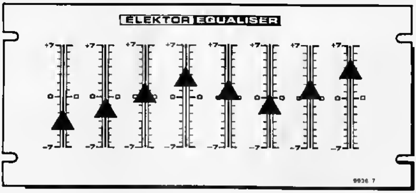


Figure 7. The 'graphic' equaliser owes its name to the fact that by arranging the (slide) potentiometers for the filter controls in a row across the front panel they provide an immediate graphic representation of the frequency response of the equaliser.

adopted in domestic hi-fi and P.A. applications, the filter parameters are not preset and thereafter left untouched; rather the filter settings are varied constantly as demanded by the (live) performance of the passage of music being played. For this reason the filter controls on the equaliser must be well-calibrated and ergonomically designed — a precondition which has led to the popularity of graphic equalisers, where the pattern of the slider potentiometers on the front panel provides immediate visual feedback regarding the overall filter response (see figure 7). However that is not to say that parametric equalisers are unsuited for this type of application — quite the reverse. Their greater scope (control of all the filter parameters) renders them much more flexible and affords the skilled user the possibility of achieving a wide range of different effects.

Setting up an equaliser

Before discussing the specific problems encountered when attempting to equalise the frequency response of domestic hi-fi and P.A. systems, there are several general points which can be made.

Firstly, and most importantly, it is essential that the frequency response which is to be corrected is already known. At the risk of sounding repetitive, fiddling around with the equaliser controls and 'playing it by ear' will almost certainly produce little in the way of tangible benefit and more likely than not will do more harm than good. However, measuring the frequency response in question is not such a fearsome undertaking as one might imagine and worried readers should banish any ideas about expensive Bruel and Kjaer measuring equipment that might be needed. In fact all that one requires is the audio spectrum analyser described elsewhere in this issue, a little patience, and a certain understanding of what one is trying to achieve. The point here is that exceptionally precise filter settings (within ± 0.5 dB) are not necessary, nor does one have to have an *absolutely* accurate picture of the frequency response. It does not matter whether a particular peak or trough happens to occur at *exactly* 225 Hz — what is more important is that irregularities in the frequency response can be detected (without necessarily knowing their precise location) and then corrected. Frequency response curves such as those shown in figures 2, 3, 5 and 6 may well be be interesting for the audio consultant or engineer, but as far as the hi-fi owner is concerned the only thing that counts is the sound reaching his ears!

The measurement and correction procedure for a domestic listening room can be carried out in a number of ways, although in each case the general principles involved are the same. The choice is basically one of ancillary equipment, whether one uses a measurement microphone, headphones, test records etc.

Setting up an equaliser for a P.A. system is somewhat simpler in that it only makes sense to utilise the existing microphone(s) to obtain the results of the spectral analysis. Since this step in fact forms the basis of the various procedures which can be adopted with domestic hi-fi systems we shall examine it first, before going on to discuss how to obtain the best results from an equaliser in domestic audio applications.

8

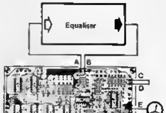


Figure 8. Before the equaliser is incorporated into the P.A. system it must first be adjusted for a flat response. This can be done with the set-up shown here.

P.A. systems

It goes without saying that, as far as possible, the performance of the P.A. system should be optimised before the equaliser is introduced. That is to say that the positioning of the microphone(s) and loudspeakers should be carefully chosen; ideally, cardioid microphones should be used, and, if necessary, the output level of the frontally situated speakers lowered. Only when no further improvements of this nature can be achieved should the equaliser be brought in. The setting-up procedure discussed here assumes that one possesses a parametric equaliser and the audio spectrum analyser.

The procedure
followed with an octave or third-octave graphic equaliser is broadly similar; any differences will be mentioned as they arise.

1. The first step is to adjust the equaliser controls to obtain a linear frequency response. This is done by connecting the noise generator direct to the equaliser input and the analyser filter and display to the output of the equaliser (figure 8). The analyser filter should be adjusted for

maximum Q (1/12 octave bandwidth). With this arrangement it is a simple matter to trace and correct any peaks or dips in the response which are caused by the equaliser itself (the filter sections of a graphic equaliser should be adjusted one at a time).

2. One next has to find a suitable point in the amplifier at which to connect the equaliser. If the amplifier has a monitor input, then in most cases one need look no further (see figure 9a). Figures 9b and 9c however, illustrate how it is possible to incorporate a monitor switch oneself

3. The output of the equaliser should then be connected to point B in figure 9, the noise generator connected to the equaliser input, and the analyser filter and display to point A in figure 9. This arrangement is depicted in figure 10.

4. The frequency response of the system can now be measured; first of all however, it is important that the potentiometer control which sweeps the centre frequency of the analyser filter up and down the audio spectrum has been provided with a (calibrated) scale (from, say, 1 to 10). If several microphones are used in the P.A system under test, only the main mike, i.e. the one used most often, should be switched on. The results obtained can be plotted to form a graph such as that shown in figure 11a The points most worth plotting are the highest values of a peak and the lowest of a dip. If an octave or third-octave equaliser is used then the analyser filter should be varied stepwise in octave or third-octave increments. The readings obtained for each frequency band are then plotted as shown in figure 12a.

5. Using a ruler one then draws a line approximately mid-way between the highest peak and lowest dip (see figures 11 b and 12b); this represents the theoretically ideal response to which one is approximating.

6. The Q of all the bandpass filters in the parametric equaliser are set to maximum (if a graphic equaliser is being used points 6 to 13 are omitted) and using the analyser filter the first peak or dip in the measured response is located; in figure 11b for example, this is the peak between measurement points 2 and 3. Since it is a peak, the first equaliser filter is set for maximum cut and the centre frequency of the filter slowly adjusted until there is a (fairly sudden) drop in the analyser reading. The centre frequency of the equaliser filter is then fine-tuned until the reading on the analyser display is at a minimum. Finally, the attenuation of the filter is reduced to the point where the meter reading coincides with the theoretically uniform response.

7. The analyser filter is then tuned up the audio spectrum until the next irregularity in the response is encountered. If, as in figure 11b, this is a dip, the second equaliser filter is set for maximum boost, tuned in to the appropriate frequency, and the gain of the filter varied until the desired reading on the analyser meter is obtained. If further deficiencies in the frequency response exist, this procedure is then repeated with the remaining equaliser filters.

9

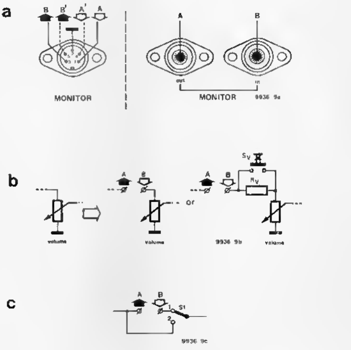


Figure 9. It is necessary to find a suitable point in the amplifier at which one can connect the equaliser. In general this will be in the region of the volume control. If the amplifier already possesses a monitor input then this can be used.

10

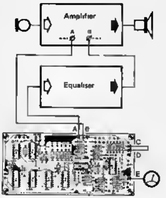


Figure 10. Once the equaliser has been adjusted for a flat response and a suitable connection point in the amplifier has been found, the analyser and equaliser are connected as shown.

8. The next step is to tune the analyser filter to the frequency at which the bass response of the system begins to roll off sharply. This point is indicated with an arrow in figure 11b. The Baxandall bass control on the equaliser should then be set for maximum cut, and its 3 dB point adjusted until the meter reading falls to 0.7 of its original value.

9. The turnover frequency of the treble filter in the tone control network is adjusted in exactly the same way. Were one to measure the resultant overall response (not that this is necessary), it would look roughly like that shown in figure 11c.

10. The centre frequency of the analyser filter is now tuned down to the point just below that at which the turnover frequency of the bass control was adjusted. The gain of this filter should then be increased until it coincides with the theoretical 'flat' value. The same procedure is performed for the treble control.

11. The analyser filter is tuned to a frequency on the 'flank' of the first

11

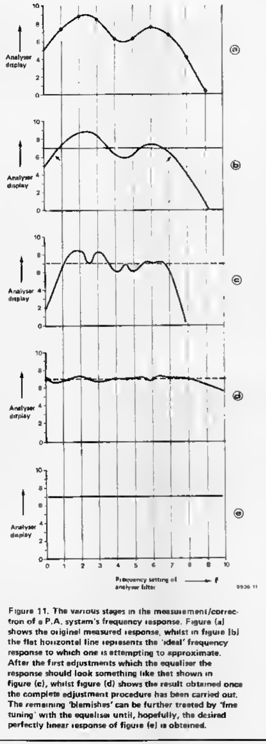


Figure 11. The various stages in the measurement/correction of a P.A. system's frequency response. Figure (a) shows the original measured response, whilst in figure (b) the flat horizontal line represents the 'ideal' frequency response to which one is attempting to approximate. After the first adjustments which the equaliser the response should look something like that shown in figure (c), whilst figure (d) shows the result obtained once the complete adjustment procedure has been carried out. The remaining 'blemishes' can be further treated by 'fine tuning' with the equaliser until, hopefully, the desired perfectly linear response of figure (e) is obtained.

peak or dip in the response and the Q of the first equaliser filter is reduced until the reading of the meter at this point reaches the nominal 'ideal' value. This procedure is repeated for the rest of the equaliser filters.

12. Theoretically, the equaliser should now be set up correctly and the response curve of the system should resemble that shown in figure 11e, i.e. flat over the range of the spectrum analyser. Unfortunately, however, this will rarely be the case in practice, and it will be necessary to repeat the above procedure from point 4 onwards in a slightly modified form. The reason for this can be explained if one looks at the curve shown in figure 11d, which represents the probable frequency response obtained so far. The curve exhibits the following faults:

– The turnover frequency of the bass tone control is too low, with the result that the response slopes too sharply at this point. The remedy – increase the turnover frequency and reduce the gain slightly

– The centre frequency of the first (equaliser) bandpass filter is too high, the consequence being that the filter introduces too much attenuation and has too large a bandwidth. Each of these filter parameters should therefore be adjusted.

– The second bandpass filter is correctly adjusted, however the centre frequency of the third is slightly low, causing over-attenuation and resulting in too small a bandwidth.

– The turnover frequency of the treble control is too low, causing the response to roll off at high frequencies; once again this should be corrected.

13. With an octave or third-octave (graphic) equaliser the adjustment procedure is considerably simpler; this is in fact one of the main advantages of this type of equaliser. A filter with switchable centre frequency (in steps of an octave or 1/3 octave) is employed as analyser filter. The adjustment procedure consists simply of setting up each frequency band in turn and varying the gain of the corresponding equaliser filter until the analyser reading coincides with the nominally flat value. As expected, the resultant response curve (see figure 12c) has a certain waviness, which is unavoidable when using a graphic equaliser. However this is of only minor importance in this type of application.

14. Irrespective of the type of equaliser which is employed, the adjustment procedure, once completed, should be checked with the aid of the following test. The system should be set up as for normal use, i.e. the equaliser is connected to point A in figure 9 and the pink noise generator removed. The analyser filter and display, however, are left connected to point A (see figure 13) for the time being. The volume control of the amplifier is then turned up to the point where acoustic feedback just starts to occur. Using the analyser filter it is a simple matter to detect the frequency at which the signal is oscillating, whereupon the gain of the corresponding equaliser filter should be reduced a fraction.

If the equaliser has been optimally set up, the system should no longer oscillate at the same frequency. If, however, it should continue to do so, then it means that the equaliser has not been correctly set up and the adjustment procedure should be repeated point for point.

16. If more than one microphone is used in the P.A. system, the above procedure is only carried out with the

main mike. The response obtained with each of the other microphones is measured separately as described in point 4. Should these all prove to be reasonably flat, the system is ready for use as it stands. If this is not the case, however, then one of the following steps may prove necessary. If one mike has an irregular response and it is of a different type to the main mike, then one should consider replacing it. If the discrepancies are only minor, then basic equalisation (one equaliser filter per mike) for each microphone may be adequate. Bear in mind that a dip in the response of the other microphones is less important than the presence of a peak. Finally, a compromise solution is also possible: i.e. one switches on all the mikes and adjusts the equaliser for the optimal response.

In conclusion it is worth pointing out that all the above measurements were carried out using a pink noise test signal. This type of signal source was in fact chosen for a very good reason. Were the response of the system measured using e.g. a sinewave generator, the response shown in figure 5a would look something like that in figure 14. The response is characterised by countless dips and peaks separated by little more than a couple of Hertz and varying in amplitude by between 20 to 30 dB. These very sharp dips and peaks are intrinsic to the response and cannot be corrected. If attempting to equalise a response measured using a sinewave generator the important thing is to align the tops of the peaks; the average and minimum amplitude levels are of minor importance, since, as already mentioned, it is the signal peaks which determine at what point the system succumbs to acoustic feedback.

Although the measurements obtained with a sinewave generator are more accurate, they are also considerably more time-consuming. In addition, when plotting the response of a system, there is the added difficulty of ensuring that one is recording only the peak signal levels.

12

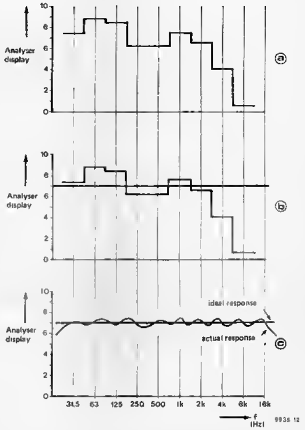


Figure 12. With octave and third-octave graphic equalisers the response can only be varied in octave or third-octave steps, hence there is little point in measuring the response of the system more accurately than this. Figure (a) shows the measured response with an octave/third-octave analyser filter; in figure (b) the nominal 'flat' value is drawn in, whilst figure (c) shows the response obtained with the equaliser optimally adjusted. The 'waviness' of the response is an inherent result of employing a graphic equaliser and cannot be rectified. However in practice it has little effect upon the final sound quality.

13

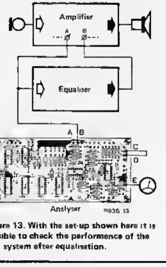


Figure 13. With the set-up shown here it is possible to check the performance of the P.A. system after equalisation.

The living room

As in the case of P.A. systems, the most suitable point in the reproduction chain to incorporate the equaliser is the monitor input of the amplifier. If such an input does not already exist, then, as already mentioned, it is a relatively simple matter to incorporate such a facility oneself.

For stereo hi-fi systems a 'stereo' equaliser in the shape of two independently variable mono equalisers is required. Quad fans need not worry, since generally speaking there is little to be gained from using an equaliser for the rear channels.

Once installed there are several methods which can be adopted to set up the equaliser. The simplest is to use the complete audio analyser described elsewhere in this issue in conjunction with a measurement microphone. However other approaches in which only part of the audio analyser is used together with a pair of high impedance headphones are also possible (it is even possible to dispense with the audio analyser entirely!). Each of these methods will be described in detail.

a. Analyser and measurement microphone

The adjustment procedure with analyser and measurement microphone is essentially the same as that adopted with P.A. systems. By 'measurement' microphone is meant a mike whose frequency response is sufficiently flat to ensure that it does not introduce a significant degree of error into the

14

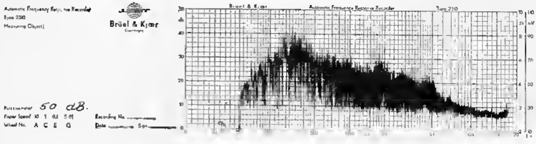


Figure 14. Until now the frequency responses shown have all been 'idealised'. However if the response is measured extremely slowly (15 to 20 minutes for one complete response curve) using a swept sinewave generator then the resultant graph looks rather different from that shown in figure 5a! One can clearly see that there are a large number of quite sharp peaks and dips which are only a few Hertz apart. These rapid variations in amplitude cannot be corrected however, and consequently there is a little point in measuring them. When using a noise generator as a test signal source, one obtains an 'averaged' response curve, which is much more useful when it comes to practical adjustments with the equaliser.

measurements. A good quality microphone of the type intended for use with reel-to-reel tape recorders should fit the bill.
The connections for the analyser and microphone are illustrated in figure 15. The microphone should be situated in the 'ideal' listening position within the room and care should be taken to exclude extraneous noise sources (wives, children etc.!) One then works through the same procedure as described for P.A. systems, but with one notable exception. As already mentioned, any dips or peaks in the response occurring between roughly 300 Hz and 5 kHz should generally be left alone. Until now, however, there has been no need for the frequency scale on the analyser filter control to be calibrated, which means that there is no way of telling where these frequencies occur! Fortunately, however, there are alternative methods of determining this frequency band with sufficient accuracy: e.g. the use of test records which have a number of specified frequencies recorded on them; alternatively one can utilise the knowledge that on a piano (or the 8' register of an electronic organ) 300 Hz coincides roughly with d^1 — the d above middle c, and 5 kHz with e^5 (i.e. four octaves above middle e).
In figure 3a the frequency response exhibited a dip at around 1600 Hz, and it was stated that if this was a result of the room acoustics, it should not be equalised; if however it was caused by the response of the loudspeaker, then it was legitimate to remove the dip using the equaliser. The simplest method of ascertaining which of these two situations is in fact the case is to measure the loudspeaker response in two different rooms. The most suitable room for this purpose (assuming it is large enough!) is the bathroom! However one must of course be extremely careful when using electrical equipment in the vicinity of water taps etc. At any rate, if the same dip in the response occurs when the loudspeaker has been set up in a different room, then one can safely assume that it is the fault of the loudspeaker itself.
Since a stereo equaliser actually consists of two separate mono equalisers, in theory the adjustment procedure should be carried out twice, once for each channel, and in each case with the other channel completely disconnected. In practice, however, it is sufficient to feed the noise signal to the desired channel and simply to turn the balance control on the amplifier to the appropriate end stop. Any crosstalk between channels should be too small to affect the resultant measurement.

15

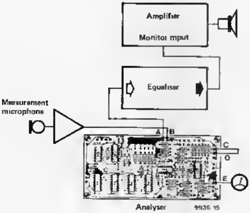


Figure 15. If a reliable measurement microphone is available the arrangement shown here can be used to measure the response of a hi-fi system and living room.

Test records

Certain hi-fi stores stock various test records which often include pink noise test signals. In principle, these can be used in place of the pink noise generator of the audio analyser. The adjustment procedure then becomes slightly more inconvenient, since one must constantly search for the right spot on the record for each measurement; however this in no way interferes with the accuracy of the adjustment procedure.

Sinewave test signal

It is also theoretically possible to use a pure sinewave (whether from a sinewave generator or a test record) as a test signal, however this approach is not recommended. As has already been

16

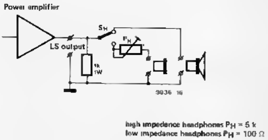


Figure 16. When using headphones to measure the room's response a volume control is necessary to be able to adjust the signal from the 'phones' until it sounds the same as that from the loudspeaker. In addition one must be able to switch between the two, so as to make a direct comparison.

explained, the actual frequency response of the system consists of a large number of very rapid variations in signal level. Were a sinewave generator employed as a test signal source these peaks and dips would be reflected in the measurement. One would then have to determine the 'average' frequency response of the system before one could set about equalising it. A small drift in the oscillator frequency, a fractionally incorrect setting of the controls, could lead to differences in signal level of from 5 to 10 dB. Such is the risk or error using a sinewave test signal that it is best to avoid this approach altogether.

Headphones

There may be those who do not wish to purchase a measurement microphone (and suitable pre-amp) solely for the purpose of setting up an equaliser. If that is the case an alternative solution is to use a pair of high-quality headphones. The adjustment procedure is simplest if one has a pair of 'open' headphones, i.e. which do not acoustically isolate the ears from external sounds. Figure 16 shows how the headphones are connected to the amplifier. This set-up allows one to switch from loudspeaker to headphones and to vary the volume of the headphone signal until it sounds the same as that from the loudspeaker (It is important that the headphones do not muffle or distort the loudspeaker signal in any way).

Since the switch and volume potentiometer must be operated from the desired listening position, a sufficient length of suitable cable is required.

The connections between the amplifier equaliser and analyser are shown in figure 17.

Once again it is possible to use a test record as a pink noise source in place of the noise generator on the analyser, although it is less convenient. The display or meter section of the analyser is not used with this set-up (no measurement mike), instead one trusts to one's ears to distinguish between signal levels. This does require a certain amount of concentrated listening, however in practice this has proven to work quite well. The adjustment procedure is as follows:

1. The analyser filter control is set to roughly its mid-position, and with the S_H switch (see figure 17) in the 'loudspeaker' position, the noise signal is adjusted to a reasonable room level. If the volume of the noise signal is too high it is not only extremely disagreeable, but there is also a risk of damage to the speaker!
2. Potentiometer P_H is set for maximum resistance, switch S_H is moved to the 'headphones' position, and P_H is then adjusted until the signal from the headphones sounds to be at the same level as that from the speaker was.
3. The frequency of the analyser filter is gradually moved up and down the entire spectrum and the differences between the signal levels of the loudspeaker and of the headphones are noted — loudspeaker slightly louder, much louder, the same, etc. At the same time one should observe at what points the highest peaks (i.e. greatest signal levels) and lowest dips (smallest signal levels) occur. A useful method of recording one's observations is illustrated in figure 18a; figure 18b shows the corresponding frequency response. With this information one can now proceed to set up the equaliser in the manner described above, using the signal level established in point 1 as the nominal 'flat' value. As already mentioned, the band of mid-range frequencies should normally be left unaltered.

Summarised briefly, the remainder of the adjustment procedure is as follows:

4. All the equaliser (bandpass) filters are set for maximum Q. With the aid of the analyser filter the first peak (in figure 18 this lies between test points 1 and 2) is detected, the first equaliser filter is set for maximum cut and its centre frequency adjusted until it coincides with the top of the peak. The amount of attenuation introduced by the filter is then adjusted until the signal level of the loudspeaker and headphones is the same. This procedure is repeated with the remaining equaliser filters for any other irregularities which require correction (in figure 18 the other prominent peaks and dips fall within the

17

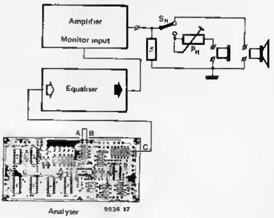


Figure 17. Connections between amplifier, equaliser and analyser when using headphones to measure the room's response.

18

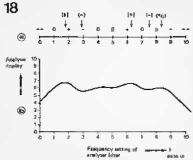


Figure 18. An example of how one can audibly chart the response of the room when using headphones. ++ signifies: loudspeaker much louder than headphones, 0 means both are equally loud, etc. The tops of peaks and bottoms of dips are marked with an arrow. The actual curve which corresponds to this notation might look something like that shown in figure (b).

critical mid-range of frequencies to be left alone).

5. Using the analyser filter, find the frequency at the lower end of the spectrum at which the loudspeaker begins to sound perceptibly quieter than the headphones (just below point 1 in figure 18); set the bass control filter of the equaliser to its lowest frequency and adjust it for maximum cut. Then gradually increase the turnover frequency until the loudspeaker sounds even quieter still. Repeat the above procedure for the equaliser treble control (in figure 18 the reference frequency will probably lie just above test point 9).

6. Set the analyser filter frequency to minimum and increase the gain of the bass control until the 'flat' level is obtained; adjust the treble control in the same way.

7. On the sides of the original first peak in the response there should now be two new peaks. Adjust the analyser filter until it coincides with one of these new peaks and reduce the Q of the first equaliser filter until it has disappeared. If necessary repeat this procedure with the remaining equaliser filters.

8. Finally, sweep the analyser filter up and down the entire audio spectrum and check to ensure that all the adjustments that have been made are correct. It will generally prove necessary in practice to make a few additional corrections or alterations. Once done, the system is now ready for use and can be subject to the crucial test of introducing a suitable music signal and listening to hear (hopefully) the improvement in the resultant sound.

Bibliography:

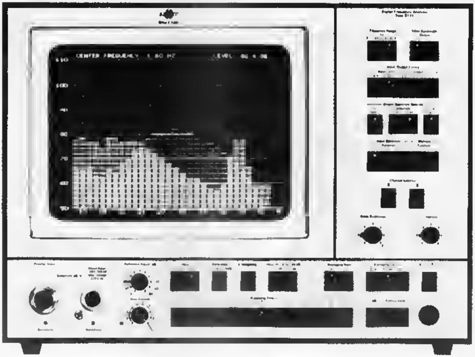
J. Moir: Interactions of loudspeakers and rooms. Wireless world, June 1977.

Bruel and Kjaer: Relevant Hi-Fi tests at home. Paper at the 47th Audio Engineering Society Convention. Also available as a Bruel and Kjaer application note.

Bruel and Kjaer: Hi-Fi tests with 1/3 octave weighted, random noise. Bruel and Kjaer application note no. 13-101.

Philips: Sound equalisation using Philips K and Q-filters. ELA application note 17.8100.35.331011.

2



An example of an extremely sophisticated (and expensive) spectrum analyser used for professional applications. The model shown here is the 2131 Digital Frequency Analyser from Brüel and Kjaer, which splits the audio spectrum up into octave or third-octave frequency bands and displays the corresponding signal levels on a CRT.

HF OPERATION OF FLUORESCENT TUBES

A circuit is described that enables HF control of fluorescent tubes. This not only increases the already high luminous efficacy of these lamps, but also enables them to be dimmed gradually.

Although fluorescent tubes have a much higher luminous efficacy* (80-90 lm/W) than ordinary, vacuum light bulbs (about 15 lm/W), and have a much longer life expectancy, they are nowhere near as popular for use in the home. This unpopularity is caused by the 'cold' character of the light, the difficulty of controlling (dimming) the light, and the objectionable behaviour (flickering) immediately after switch-on. Although the present circuit cannot alter the character of the light (manufacturers are already producing much 'warmer' fluorescent tubes), it does obviate the other two undesirable aspects.

Economy of HF control

High-frequency control units for fluorescent lamps have been available for some time, but so far these are mainly used in factories, office blocks, and other large buildings. The principal reason for their use there is that they provide a higher luminous efficacy. This comes about because the transformation of electrical into luminous power is more efficient at higher frequencies, and also because the losses in the control units are smaller at such frequencies (the choke of a domestic 40 W fluorescent lamp dissipates about 9 W). These advantages are, of course, not of such great importance for domestic lighting, because the resulting savings on the electricity bill are small. The main reason for adopting the present circuit in the home is seen primarily in the dimming facility.

Conventional set-up

A fluorescent tube usually consists of a long glass tube T (see Fig. 1), which is internally coated with a fluorescent powder, although other shapes are now also on the market. The tube contains a small amount of argon together with a little mercury. At each end of the tube there is an electrode E that invariably consists of a coiled tungsten filament coated with a mixture of barium and strontium oxides. Each electrode has attached to it two small metal plates, one at each end of the filament. These plates act as anodes for withstanding bombardment by electrons during the half-cycles when the electrode is positive. During the other half-cycles, the adjacent hot filament acts as the cathode, emitting electrons.

Before the gas in a fluorescent tube can be ionized, certain conditions must be met by the control circuit, consisting of choke L and starter switch G. Before the gas is ionized, the resistance measured between the two electrodes is high.

Switch G, called a glow switch, is, strictly speaking, a small glow discharge lamp filled with a mixture of argon, helium, and hydrogen at low pressure. The contacts of the glow switch are normally open, but when the supply voltage is switched on, a glow discharge is started between the electrodes of the switch. The resulting heat is sufficient to bend the bimetallic strips until they make contact and close the circuit between electrodes EE of the tube. A fairly large current then flows through these electrodes, the value of which is determined by choke L. The current heats the electrodes, which, by thermal emission, results in a number of free electrons in the tube. These electrons are necessary for the onset of ionization (avalanche effect).

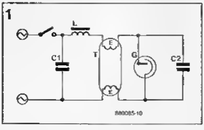


Fig. 1. Circuit of conventional low-pressure fluorescent tube.

Because the contacts of G are closed, the dissipation in this switch diminishes rapidly. This causes the bimetallic strips in G to cool and after a second or two the contact between these strips is broken. The consequent sudden reduction in current induces an e.m.f. of about 1000 V in L. The sum of this e.m.f. and the mains voltage is sufficient to ionize the argon in T. This reduces the resistance of the tube and the choke limits the current to a value specified by the manufacturer. The voltage drop across T is then of the order of 100 V, which is lower than the voltage required to ignite the glow switch.

The reason that fluorescent tubes flicker before they ignite properly is that the reduction in current caused by the bimetallic strips opening happens randomly with respect to the period of the mains voltage. If they open at the instant when the current through the choke is small, the induced e.m.f. may not be large enough to ionize the argon in T. In that case, the starting process repeats itself until ionization does take place.

The power factor of the circuit is raised from about 0.5 to 0.9 (lagging) by capacitor C_1.

Capacitor C_2 is an RF suppressor.

Most energy of this type of fluorescent lamp is radiated at a wavelength of 253.7 nm, which is in the ultra-violet region. The fluorescent coating of the tube absorbs this energy and converts it into visible radiation. Different coatings reradiate the absorbed energy at different wavelengths: zinc-beryllim silicate gives yellow to orange; cadmium borate and yttrium red; magnesium tungstate pale blue; and zinc silicate green. The use of appropriate mixtures of these powders make it possible to attain any desired colour.

Dimming

Dimming of fluorescent tubes operating at the mains frequency is troublesome.

* The term 'luminous efficiency' would be incorrect, since that is the ratio of output power to input power when both are expressed in the same unit.

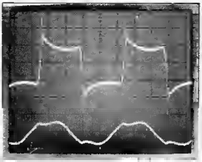


Fig. 2. Waveforms of voltage and current in a conventional fluorescent tube.

The reason for this may be seen in Fig. 2, which shows the voltage and current as functions of time. It is seen that after each and every zero crossing the voltage must rise substantially before the tube relights. Although the light output of all fluorescent tubes therefore fluctuates at twice the mains frequency, the visible effect of this is fortunately considerably reduced by the persistence of glow of the fluorescent coating.
If the tube is dimmed with the aid of a conventional triac circuit, the length of time that the current through the tube is zero becomes longer, and the risk of the tube being extinguished becomes greater. There are a number of ways of preventing this situation. The first is to maintain the high temperature of the electrodes with the aid of an external holding current. The second is to use a resistance strip along the tube as an aid to ignition. This strip is connected at one end to the electrode via a high-value resistor. At the other end it causes a kind of pre-ignition (the effective distance between the electrodes is reduced, which causes the fieldstrength to be locally much more intense). The third is to increase the frequency of the mains to a value where the period is small with respect to the recovery time of the ionized gas in the tube. The circuit described here uses this last method.

Block schematic

The circuit is, in fact, an a.c.–a.c. converter. The mains voltage is first rectified (full wave) and smoothed. The resulting direct voltage of 300 V is then converted to a square-wave voltage with a frequency of 80 kHz (at start-up) or 30 kHz (normal operation). The fluorescent tube is part of a series LC circuit that is shunted by a capacitor. As long as the tube is not lit, it has a high resistance and does not load the circuit. At the relatively high start-up frequency, the reactance of the capacitor is relatively low. When a voltage is applied across the circuit, a current will flow that causes the electrodes of the tube to be heated. Just after switch-on, the frequency will decrease gradually. As soon as it approaches the resonant frequency of the circuit, the impedance of the circuit will drop rapidly, which will result in a much larger current through the electrodes. At the same time, the voltage across both L and C is increased greatly. Since the tube is in parallel with C, it will light readily. As soon as this happens, the tube resistance drops considerably and this will damp the LC circuit. The current through the electrodes will then become much smaller. The control circuit further reduces the frequency until it reaches a value of 30 kHz. The currents through the tube and capacitor will be small, because the ignition voltage across the lamp (and thus the p.d. across the capacitor) is relatively low and also because the reactance of the capacitor at 30 kHz is relatively large.
Dimming of the tube is effected by controlling the current through it. In contrast to conventional triacs, the present circuit is a real control loop. The current is measured with a current transformer and fed back to the control circuit. The latter circuit varies the duty cycle until the measured current has the same value as the set current. This arrangement enables dimming of the tube to near-extinction. Quenching it completely is not possible, because that would necessitate a new start cycle (with the consequent frequency swing). The current regulation also ensures that at start-up, when the lamp current is zero, the duty cycle of the output signal is automatically optimized. In this manner, the tube will always start smoothly, independent of the position of the dimmer control.

Circuit description

In Fig. 6, fuse F_1 and chokes L_1 and L_3 are shunted by varistor R_{25}, which suppresses spikes on the mains supply. The mains voltage is rectified in bridge $D_3-D_4-D_5-D_6$ and smoothed in C_5. The peak current through C_5 is limited by R_{26}. It should be borne in mind that switch-on may occur at any moment during the mains cycle: the peak charging currents that may occur should not be understimated. To keep the dissipation in R_{26} low, an NTC type is used here. Immediately after switch-on, this heats up, which causes its resistance to drop from 50 ohms to about 2 ohms, effectively limiting the dissipation.
Capacitors C_4 and C_6 and diodes D_1, D_2, and D_7 form a pre-control for the supply voltage to the drive circuit. This voltage is stabilized at 12 V by IC_4. The maximum current that can be drawn from this supply is 30 mA (determined by C_4). The drive circuit draws about 20 mA.
The power stage consists of T_1 and T_2, which are connected as a half-bridge. The voltage at the junction of T_1 source and T_2 drain swings between 0 V and 300 V (= the rectified mains voltage). The d.c. component of this voltage is blocked by capacitors C_2 and C_3. One capacitor would have been sufficient, but two in series give some extra decoup-

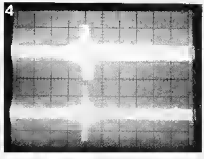


Fig. 4. Electronic starting: the frequency swings from 80 kHz to 30 kHz. When it is about 50 kHz, the tube lights.

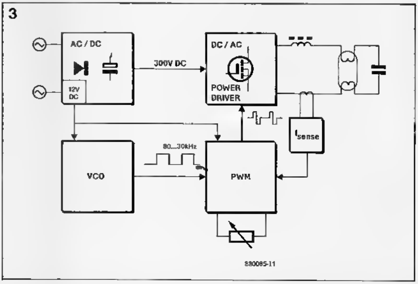


Fig. 3. Block schematic of HF controller.

ling of the high-voltage supply. As far as the a.c. through the lamp is concerned, the two capacitors are in parallel.

The power FETs contain parasitic free-wheeling diodes that are active during the dimming of the lamp. During dimming, both FETs are switched off for part of the period of the applied voltage. The voltage at the junction of T_1 source and T_2 drain, because of series circuit L_1-C_1, will swing several times between 0 V and 300 V during that time, which causes the free-wheeling diodes to conduct alternately (see Fig. 5b). A new period starts with T_1 being switched on. Now assume that D_{17} is shunted, D_{16} is not there, and that the free-wheeling diode in T_2 conducts just at the instant T_1 is switched on. During the recovery time of the free-wheeling diode in T_2 a short peak current will flow through both T_1 and T_2, which will affect the dissipation adversely. Since this problem is caused by the relatively long reverse-recovery-time of the internal free-wheeling diode in T_2 (typically of the order of 1.8 μs), it is obviated by connecting diode D_{17} in series with T_2, because this prevents the parasitic diode from conducting. The series-connected diodes can then be shunted by a much faster free-wheeling diode, D_{16} (T_{rr}=25 ns typically).

The series LC circuit is formed by L_1 and C_1. The circuit is damped by R_{23} and R_{24}. Without these resistors, the damping of the circuit would be dependent solely on the resistance of the tube electrodes. Because this is very low, very large values of current and voltage might ensue before the tube lights. Resistor R_{23} guarantees a given minimum series resistance in the circuit. The resistance of varistor R_{24} will drop as soon as the voltage across C_1 exceeds a maximum value of about 1 kV. The clamping of the potential across C_1 will prevent too high an upswing of voltage and current in the circuit. As soon as the tube lights, its resistance will further damp the circuit. Since the final potential drop across the lamp is relatively low, additional dissipation in R_{24} is prevented because the varistor has a high resistance at that voltage.

Since the operating frequency of 30 kHz is much higher than the conventional 50 Hz, the self-inductance and dimensions of choke L_1 can be accordingly smaller. Although it would be possible to limit the lamp current to a given value with the aid of the current regulating circuit, it is better done by the choke. The self-inductance is chosen so that at maximum duty cycle the lamp current does not exceed the value specified by the manufacturer of the tube.

Control circuit

The control circuit has two tasks:

- the generation of a frequency that within about 2 seconds from switch-on swings from 70-80 kHz via the resonance frequency of 50 kHz to the normal operating frequency of 30 kHz.
- the controlling of the lamp current in accordance with a variable desired value to enable dimming of the lamp. The current is controlled by varying the pulse width of the drive signal.

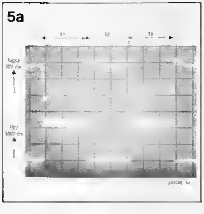


Fig. 5a. Gate signal (upper trace) and the voltage at the junction of T_1 source and T_2 drain at maximum duty cycle. The 'broad band' in the lower trace is caused by the 50 Hz ripple.

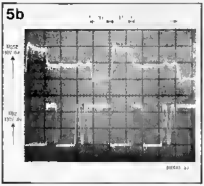


Fig. 5b. The same signals as in 5a, but with the tube dimmed. During the freewheeling period neither of the MOSFETs conducts, and the drain–source junction swings several times between 0 V and 300 V.

Frequency synthesis is provided by the VCO in IC_1, a Type 4046 CMOS PLL. The supply voltage is kept steady by zener D_{15}. Should the supply drop below 11 V, both T_7 and T_8 are switched off. The 4046 is then inhibited. When the input voltage is not lower than 11 V, C_7 is connected to the positive line via T_7. Since the capacitor at first has no charge, the VCO input will also tend to rise to 12 V, but is prevented by D_{12} from exceeding 4.5 V. From this voltage, a signal at a frequency of about 70 to 80 kHz is generated. Capacitor C_7 is then charged via R_{16}, which causes a drop in the potential at the junction of C_7 and R_{16}. When this voltage drops below 4 V (the earlier mentioned 4.5 V less the drop across D_{13}), the VCO input is pulled down and the frequency of the output signal drops. The operating VCO input, and thus the operating frequency, is determined by potential divider $R_{17}-R_{16}$.

Multivibrators MMV_1 and MMV_2 provide the pulse width modulation. The VCO signal has a duty factor of 50% (square wave). MMV_1 is triggered at the leading edge of this signal. Immediately on termination of the mono period of MMV_1, the other multivibrator, which has an identical mono period, is triggered. The mono period of the multivibrators is variable because C_{15} and C_{16} are not charged via a fixed resistance, as is usual, but by a variable current source (strictly, current mirror): T_4 and R_6 and T_3 and R_7 respectively.

The magnitude of the current, and thus the mono period and duty cycle, is constantly adjusted, as required, by the current regulating circuit. The mono periods can not become longer than the half-periods of the VCO signal. Were one of the multivibrators likely to generate a longer period, this would be terminated prematurely by the reset input. In this manner, it is ensured that the maximum duty cycle of the circuit is exactly 50% as determined by the 50% duty factor of the VCO signal. This is, of course, essential to guarantee symmetrical control of the output stage.

The output stage is driven by a pulse transformer, Tr_1, which is contained in bridge $T_5-T_6-T_8-T_{10}$. Any d.c. components caused by small deviations of the mono periods are blocked by C_{12}. Such d.c. components would cause an unnecessarily large current in the low-ohmic primary of the pulse transformer, which might lead to saturation of the core of the transformer.

The MOSFETs are driven direct by the secondaries of Tr_1. It is, of course, imperative that these windings are connected in anti-phase to make sure that the MOSFETs cannot be switched on simultaneously. Resistors R_2 and R_3 serve to damp any oscillations caused by parasitic self-inductances. The zener diodes in the gate circuits limit the amplitude of the gate voltage.

To make current regulation possible, the lamp current is measured by a current transformer, Tr_2. A complication here is C_1, which is in parallel with the tube. This means that not only the current through the lamp, but also that through the capacitor, is measured. When the lamp is dimmed, and the current through it is, therefore, small, the current through the capacitor is relatively large and would put paid to any current regulation. Direct measurement of the lamp current alone is not possible, and it is, therefore, measured indirectly. This is done by first measuring the total current (winding 1) and deducting from this the current through the capacitor (winding 2 — wound in anti-phase to winding 1).

6

Readers should note the difference between 'duty cycle' and 'duty factor', a source of confusion for many! Duty cycle refers to variations of the load with time. Duty factor is used in pulse techniques and is the ratio of the average pulse width to the average pulse spacing of pulses in a pulse train.

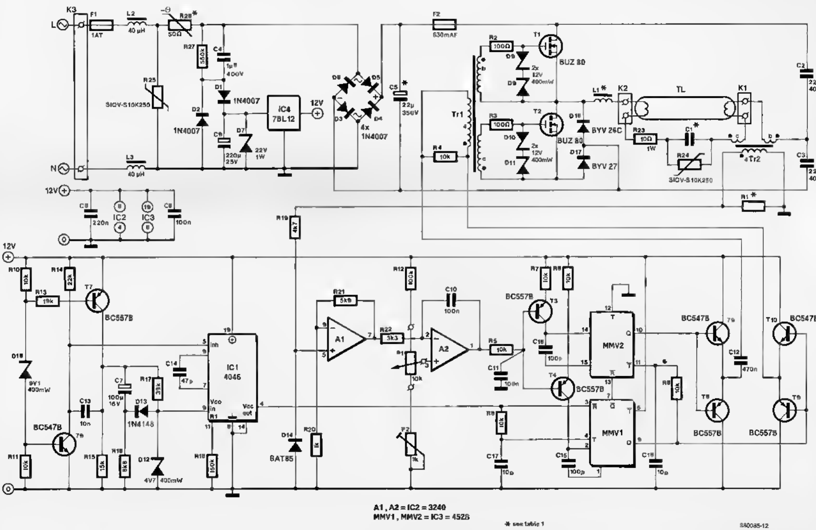


Fig. 6. Circuit diagram of the HF controller.

The secondary current of Tr_2 is converted into a voltage by R_1. Of this voltage, the positive half is amplified by A_1 and its average value is then compared with a voltage whose level is preset with P_1. If any differences are measured, A_2 increases the drive to the bases of T_3 and T_4, which varies the duty cycle until the two voltages are equal. The minimum lamp current (when the lamp just does not get extinguished) is preset by P_2.

Construction

Since the circuit is connected direct to the mains, it cannot be stressed too much to BE CAREFUL.

The circuit is best constructed and tested in stages. It is strongly recommended to use an isolating transformer during tests on the circuit.

Start with the control section at the centre of the PCB. That is, mount all ICs, except IC_4, and all associated components, including the transistors. Resistors R_1 and R_4 may also be fitted, but the two transformers must wait a little. Potentiometer P_1 may also be connected with the aid of three (temporary) short wires.

Apply a stabilized voltage of 15 V in place of the wire links near C_4 (earth closer to the edge of the board). Check the output signal of the VCO (IC_1 pin 4) with an oscilloscope or frequency meter. This square-wave signal must remain stable at 70–80 kHz for about a second and then drop to 30 kHz ±5 kHz within a few seconds. Any deviations from the stated values of frequency are caused by tolerances in IC_1 and must be compensated by small changes in the values of R_{18} and C_{14}.

The same square-wave signal should be present across R_4, but here it is not a pulse train, but an alternating signal with a peak-to-peak value of about 12 V. Since at this stage there can be no lamp current, the current regulator will automatically optimize the duty cycle.

When the supply input is decreased to less than 11 V, the oscillator should stop functioning. When the voltage is then raised again to 12 V, a new start cycle should commence.

Check the current drawn by the control circuit: this should be 10–15 mA.

Choke and transformers

Choke L_1 and two transformers, Fig. 8 and Fig. 9, are not available commercially.

The choke, L_1, is wound on a readily available pot core with an air gap, measuring 30×19 mm, with A_l=1,000. The number of turns depends on the tube with which it is intended to be used — see Table 1. Since high voltages occur across the choke, particularly during start-up, it is essential to separate each

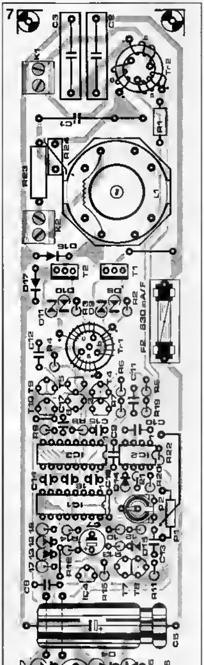


Fig.7. The printed circuit of the HF controller.

Parts list

Resistors (±5%).

R_1 = see text
R_2;R_3 = 100R
R_4 . . . R_{11} incl. = 10K
R_{12} = 100K
R_{13} = 18K
R_{14} = 22K
R_{15} = 15K
R_{16} = 6K8
R_{17} = 39K
R_{18} = 150K
R_{19} = 4K7
R_{20} = 1K0
R_{21} = 5K6
R_{22} = 3K3
R_{23} = 10R; 1 W
R_{24};R_{25} = varistor S10K250 (ElectroValue*).
R_{26} = NTC 50 Ω, 1 W e.g. Mullard no. 2322 610 11509.
R_{27} = 560K
P_1 = 1K0 preset
P_2 = 10K linear potentiometer with plastic shaft.

Capacitors:

C_1 = see text.
C_2;C_3 = 220n; 400 V
C_4 = 1µ5; 400 V
C_5 = 22µ; 350 V
C_6 = 220µ; 25 V; radial
C_7 = 100µ; 16 V; radial
C_8 = 220n
C_9;C_{10};C_{11} = 100n
C_{12} = 470n
C_{13} = 10n
C_{14} = 47p
C_{15};C_{16} = 100p
C_{17};C_{18} = 10p

Semiconductors.

D_1 . . . D_6 incl. = 1N4007
D_7 = zener diode 22 V; 1 W
D_8 . . . D_{11} incl. = zener diode 12 V; 400 mW
D_{12} = zener diode 4V7; 400 mW
D_{13} = 1N4148
D_{14} = BAT85 (Cricklewood)
D_{15} = zenerdiode 9V1; 400 mW
D_{16} = BYV28C (Mullard)
D_{17} = BYV27 (Universal Semiconductor Devices)
T_1;T_2 = BUZ80 (ElectroValue*)
T_3 . . . T_7 incl. = BC557B
T_8;T_9;T_{10} = BC547B
IC_1 = 4046
IC_2 = 3240
IC_3 = 4528
IC_4 = 78L12

Miscellaneous:

F_1 = fuse 1 A; delayed action
F_2 = fuse 630 mA; fast
2 off PCB-mount fuseholders.
K_1;K_2 = 2-way terminal block for PCB mounting.
K_3 = 3-way terminal block for PCB mounting.
L_1 = the following parts from Siemens are required for making this inductor:
1 off pot core B65701-L1000-A48;
1 off coil former B65702 B T2;
2 off washers B65705-A5000;
1 off mounting assembly B65705-B3;
1 off white screw core B65679-E1-X22;
1 off threaded flange B65679-L3;

These parts are listed in the Siemens Preferred Products Catalogue, and are available from ElectroValue*.
L2;L3 = suppessor choke 40 μH; 2 A.
TR1 and TR2 are wound on 2 ferrite cores Type RK60 (Mullard no. 4322 020 97060)
PCB Type 880085

* ElectroValue Limited • 28 St Judes Road • Englefield Green • Egham • Surrey TW20 0HB. Telephone: (0784) 33603. Telex: 264475. Northern branch: 680 Burnage Lane • Manchester M19 1NA. Telephone. (061 432) 4945

layer from the next with good-quality insulating tape. Use enamelled copper wire 24–26 SWG (0.5 mm dia.).
Both transformers are wound on the same type of ferrite toroid. The primary winding of the pulse transformer, Tr_1, consists of 40 turns enamelled copper wire, SWG 35 (0.2 mm dia.). Both secondary windings consist of 30 turns enamelled copper wire, SWG 14. It is important that the secondaries are wound in opposite directions from one another to ensure anti-phase drive of the power MOSFETs. Furthermore, the potential difference between the primary and the secondary windings is some 300 V: it is therefore important to keep the secondaries well away from the primary.
The current transformer is fairly easy to make. Both primary windings consist of 2 turns enamelled copper wire, SWG 25 (0.5 mm dia.), wound in opposite directions from one another. The secondary consists of 4 turns of the same wire as the primaries.

Final construction

Fit Tr_1 and Tr_2 in position on the PCB, followed by R_2, R_3, D_8, D_9, D_{10}, D_{11}, T_1, and T_2. Apply a voltage of 12 V from an external source and ascertain the current drawn: this should be 20–25 mA after about 5 seconds (i.e., at the normal operating frequency).
Next, check that the secondary windings are in anti-phase by temporarily interconnecting the source connections on the PCB and verifying that there is NO signal between the two gate connections. Then, mount K_3, F_1, L_2, L_3, R_{25}, R_{26}, R_{27}, C_4, C_6, D_2 and D_7. With a suitable mains cable, connect K_1 to the mains and switch on. Measure the voltage across D_7, which should be 18 V. REMEMBER YOU ARE NOW WORKING WITH MAINS VOLTAGES!
Disconnect the mains from K_1, discharge C_6 through a resistor, and mount IC_4. Then, fit the two wire links near C_5 (but not yet this capacitor). Again, connect the mains to K_1 and check the output of IC_4 as 12 V. Afterwards, measure

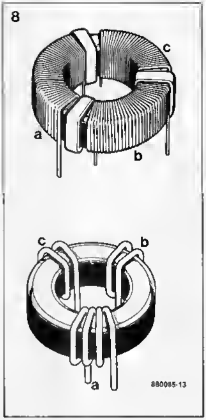


Fig. 8. Showing how the pulse transformer (a) and the current transformer (b) should be wound.

the gate signal with an oscilloscope (compare with Fig. 5a upper trace.).
Finally, mount all other components, and do not forget the wire link near T_1. The values of C_1, L_1, and R_1 are given in Table I. Take care not to confuse D_{16} with D_{17}: these components look very much alike!
When tubes with a power rating >30 W are used, it is advisable to mount T_1 and T_2 on a simple heat sink: an L-shaped piece of aluminium as shown in Fig. 10 is sufficient. Note, however, that the MOSFETs must be insulated from the heat sink. In view of the relatively high potentials involved, use ceramic, not mica, insulating washers.

Assembly and connecting-up

Connect the tube to the circuit, turn P_2 completely anti-clockwise, set P_1 to the centre of its travel, take a deep breath, and connect the mains. The tube should light after 1–2 seconds and it should be possible to dim it with P_1. It is possible that you experience odd running-light effects in the tube: these may be eliminated by turning the adjustment screw in the core of L_1.
Set P_2 to a position where the tube just remains lit. It will be noticed that a

Fig. 9. Showing how choke L_1 should be wound. The number of turns for a variety of tubes is given in Table 1.

Table 1

Tube rating	L_1	C_1	R_1
20 W	2.0 mH 45.5 turns	4n7 1500 V	2R2
30 W	1.8 mH 43.5 turns	5n6 1500 V	1R8
40 W	1.8 mH 42.5 turns	8n8 1500 V	1R8
60 W	1.1 mH 32.5 turns	10 n 1500 V	1R0

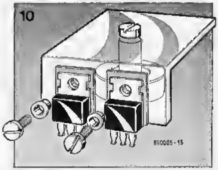


Fig. 10. When fluorescent tubes of rating >30 W are used, the MOSFETs should be cooled, for example, with the aid of a simple L-shaped piece of aluminium as shown here.

warm tube can be dimmed to a larger degree than a cold one. It is, therefore, best to set P_2 when the tube is cold.
In view of the operating frequency and the waveform of the output signal of the circuit, the connections between the cir-

Continued on p. 48

PAINTBOX: THE HIGH TECH APPROACH TO ARTISTIC CREATIVITY

by John Spurling*

David Hockney's PAINTBOX painting.

The great technical innovations of art have seldom been observed or reliably recorded in their early stages. Jan Van Eyck, at the beginning of the 15th century, was probably the first artist to make masterly use of oil painting, though he was not, as is sometimes supposed, its inventor.

Watercolour in the most general sense is a very ancient technique, but its full development did not take place until the 18th and 19th centuries in England. Graphite sticks seem to have been invented in the 16th century, but not until 1790 did the French chemist, Nicolas-Jacques Conté, manage to control their hardness and softness and transform them into "lead" pencils that have been used by artists ever since.

The use of brushes, on the other hand, goes back to ancient China and Egypt, and the Stone Age in Europe.

PAINTBOX[1] is a different matter. Originally unveiled in 1981 by Quantel[2] and continually refined since, it is basically a tool for graphic design on television, an electronic system for producing and editing images with great speed and sophistication.

One might loosely describe it as the visual version of a word processor, but instead of tapping out letters on a keyboard, the user sits in front of a smooth surface or table and simply draws or paints on it with an electronic stylus on the end of a wire. The result appears immediately on a television

John Spurling is art critic of the *New Statesman*.

monitor and there you can also mix a virtually limitless range of electronic colours. You use a cursor to pick them out and apply them, much as a word processor operator sends in his cursor among text to shape and colour his sentences.

Thinking aloud

To watch a novelist or a poet composing on a word processor would probably send a television audience to sleep. How would it be, though, if an artist — as opposed to a designer or editor — were let loose on PAINTBOX?

In a series of BBC TV programmes shown in Britain last year under the title 'Painting With Light', several artists with international reputations made the experiment. If the results were not yet quite as convincing as Van Eyck's exploitation of oil paint, the process made intermittently interesting viewing.

The artists — struggling manfully to understand and master the possibilities of the new medium — were assisted by a technician and chatted and thought aloud as they worked. What they said was often more telling than the images they produced on the screen, but then these were all well established people whose ideas, techniques and dodges have been developed over many years in quite different media. It was a little like asking Turner to paint a Persian miniature or Van Gogh to fool around with collage — curious, but slightly unfair.

David Hockney, the first of the artists, and a great one for technical experiments, was typically enthusiastic, saying: 'A barrier has gone. Now there is nothing between the viewer and the artist. What you are seeing developing on your television is the inside of the artist's head.'

The United States artist Larry Rivers, trying to adapt his habitual technique of painting over photographs to this new machine, sounded more frustrated: 'I feel as if I am working with one hand tied behind my back. It is a situation in which colour is light, but when I start to mix, I don't like what I get.'

Strangely frightened

He kept his sense of humour, however, especially when attempting a portrait of a well-known pop star, saying: 'I am going to spend five minutes on your nose.' Bending down over his electronic stylus, Rivers did look a bit like a cosmetic surgeon — or perhaps a dentist with his drill.

The British artist Richard Hamilton specializes in collages with pop associations — images lifted from advertisements and press photographs. One might have expected PAINTBOX, with its formidable cut-and-paste facilities and editorial wizardries, to be just his thing. But although he did admit that he was beginning to feel he should have one in his studio, Hamilton seemed strangely frightened of the machine.

He seldom trusted himself actually to handle the stylus, but mainly worked by issuing instructions to the technical assistant, and he was painfully cautious and undemanding in what he committed to the screen. He scarcely called on PAINTBOX's mighty range of colour and, having started with a rather powerful photograph of Protestants marching in Northern Ireland, he contrived by the end only to weaken and confuse it.

Howard Hodgkin, a winner of the Turner prize and noted for his small, densely painted and brilliantly coloured abstracts and figurative subjects, was much more adventurous than Hamilton, but still distinctly put out by the experience. The main problems for him were the lack of texture and the speed at which he was required to work.

Since he sometimes takes two or three years to finish a painting, this was hardly surprising. Even so, he managed to make the screen look just like a Hodgkin painting, or rather some ten Hodgkin paintings in succession, since he kept covering up one with another.

Medium of the future?

But whereas in an oil painting these successive layers would leave some trace of themselves in the finished work, adding to its depth and richness and often actually altering one's perception of the surface without appearing to do so, in a PAINTBOX creation they simply vanish as if they have never been. It proved, perhaps, what the politicians in last year's general election campaign had strongly suggested: that what you see on your television screen is all veneer.

It may be that PAINTBOX and its no doubt still more sophisticated successors will become the artistic medium of the future and make brushes, pencils, watercolours, oil paints and the rest obsolete, but I fancy not. Time, as Hodgkin demonstrated, is as important to art as technique — the time put into it and the time required to assimilate it.

The very qualities that make PAINTBOX such an ideal tool for television news items, commercials, animated sequences, and so on, make it little more than a toy for artists, since nothing is left of the process once the result is reached and so the result can only hold the attention for a few seconds.

What made good watching was the artists at work, not their works of art. But, of course, it is perfectly possible that there is an artist not yet known or born who will coax depth from the superficial and time from the instantaneous in some quite unimaginable way.

References.

1. Paintbox, c/o BBC Enterprises Ltd, 80 Wood Lane, London W12 0TT.

2. Quantel, Kiln Road, Shaw, Newbury, Berkshire RG13 2HA.

From page 46

cuit and the tube must be kept short. In practice, that means that the circuit will have to be built into the armature. This has been taken into account during the design of the PCB. Make sure that there will be at least 6 mm (¼ in) space between live parts of the board and metal parts of the armature. The existing starter and choke may, of course, be removed.

Potentiometer P_1 is connected to the PCB by a 3-core cable: remember that it is connected to the mains (neutral) via P_2 and L_3! It is, therefore, advisable to use a potentiometer with a man-made fibre spindle.

11

Fig. 11. The current transformer and choke L_1.

Fig. 12. Completed HF controller ready for fitting into the tube armature.

ELECTROSTATIC PAPERHOLDER

Phatagraphers, draughtsmen, campasitars, lithagraphers, artistic as well as technical designers, and, at caurse, architects use drafting tables which shauld allaw quick and safe exchanging, positioning and fixing at large sheets at paper. Far this purpase, an electrastatic paperhalder has significant advantages aver clip-on systems ar bits at drafting tape.

A wide range of equipment is currently available for putting graphics information on paper. Such equipment includes printers, plotters, X-Y and X-t recorders. In all of these, it is essential that a pen device or printer head can move with respect to the paper surface. In most cases, paper is held on a roll, which is rotated to achieve movement in the Y-direction, while a carriage is used to achieve movement of the roll, or the pen, in the X-direction. There are, however, also systems in which the paper is held flat and secured on the working table, while the pen is moved across it in both directions. This arrangement is essentially identical to that of the well-known drafting table, for which the electrostatic paperholder was developed about 20 years ago. The current trend in plotter design, however, is clearly towards the rotating paper roll.

To prevent the electrostatic paperholder falling into oblivion, this article aims at providing essential information on the operation, designing and building of this drafting aid.

Theory of operation

The general structure of the electrostatic paperholder is shown diagrammatically in Fig. 1. In principle, the construction is relatively simple, but some theoretical knowledge is required for explaining and understanding the basic operation and the effect of all parameters involved.

1

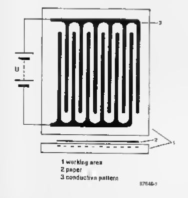


Fig. 1. Basic structure of the electrostatic paperholder.

The system can be analyzed in two ways. One is based on the theory of electrical fields. This includes the possible, but important, role of a large number of side-effects that other models fail to take into account.

The second way of analyzing the electrostatic paperholder is an essentially qualitative approach which has the advantage of being more illustrative and better comprehensible than the theory of electrical fields.

Figure 2 shows a schematic representation of a part of the electrostatic paperholder. The diagram shows voltage U present between two tracks of the conductive pattern. This voltage causes an electric field, E, between the tracks. The field strength is directly proportional to the voltage applied. Lines of force will cross the working area, but also extend beyond this, traversing the paper sheet. This will result in a certain degree of polarization of the paper due to dielectrical shift, which, in turn, is explained by the relative permittivity of paper, which is about 3 (ε_r = dielectric constant).

The force between paper and working table is then best understood in terms of a force between two charges: one is the apparent charge caused by polarization of the paper (proportional to field strength E and, therefore, voltage U), the other the charge on the electrodes of the working table (also proportional to U, and, in addition, to the capacitance). Since voltage U determines both the degree of paper polarization and the amount of charge on the electrodes, it can be safely assumed that the force is proportional to the square of U. In addition, the force between two charges is inversely proportional to the square of the distance, which means that the thickness of the insulating layer above the electrodes is an important factor. Also note that the number of lines of force traversing the paper decreases with an increase in the distance between paper and electrodes.

The above model allows simple deducing of a number of additional parameters that determine the adhesive force between paper and working table.

Relative humidity of the paper is an important parameter. Relative permittivity of water is as high as 70, caused by the dipole moments of individual water molecules. As a result, dielectrical shift in paper with high relative humidity will be considerable, causing increased adhesive force. It should be noted, however, that humid paper has conductive properties, which are augmented by impurities in water. Since electric field strength is effectively cancelled in a conductor, there will be no force at all on the paper when this is humid. In practice, it has evolved that a relative humidity of 40-50% is optimum for most applications.

2

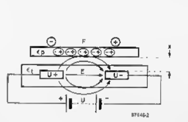


Fig. 2. The electric field causes dielectric shift in the paper, resulting in a force between paper and electrodes.

3

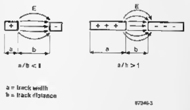


Fig. 3. The pattern of the lines of force is determined by the geometry of the track pattern.

A further important parameter to consider is the geometery of the electrode pattern, since this determines the pattern of the lines of force. Tracks whose width is small relative to the track-to-track distance cause the field to become so narrow that it does not act on the paper. A higher width/distance ratio gives a more favourable pattern of the lines of force (see Fig. 3). A ratio of slightly more than 2 was found to give best results in practice.
The final parameter to consider is the permittivity of the working table material. High relative permittivity results in high inter-electrode capacitance and, therefore, a high amount of electrode charge ($Q = UC$). Hence, adhesive force is also greater.

The curves in the graph of Fig. 4 were obtained from experiments. The y-axis shows force per unit of area at a certain voltage and electrode distance. Increasing this distance results in strong vertical shrinking of the curves. Increasing the voltage by a certain factor compresses the vertical scale with the square of the factor.

An experiment

Observing the above criteria, the following conditions should be met for obtaining reasonable adhesive force on the paper:

- Voltage should be as high as possible without causing arcing between tracks.
- Paper-electrode distance should be as small as possible.
- Relative permittivity of the working table should be high.
- Ratio of track width to track distance should be greater than 2.

A further important consideration not mentioned so far is safety. Clearly, the first two of the above conditions conflict in respect of safety. For an efficiently operating paperholder, paper-electrode distance should be of the order of hundredths of a millimeter, or one tenth at the most. A voltage of 1 kV already requires special properties of the upper layer of the working table in respect of insulation. Standard epoxy PCB material is unsuitable here because it is too thick. Considerable adhesive force is obtained when the paper is laid direct onto the copper tracks, but audible corona effects via the paper will be observed ($U = 2.5$ kV; track distance: 2.5 mm). Polycarbonate foil as used for *Elektor Electronics* adhesive front panels ensures sufficient electrical insulation, but has the disadvantage of reducing the electrostatic effect by increasing the electrode-paper distance. Better results should be obtainable with much thinner foil as used for covering model

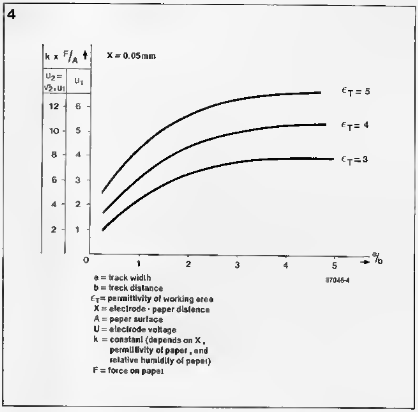


Fig. 4. Force per unit of area as a function of the ratio of track thickness to track distance, with relative permittivity of the working area as a parameter. Force is a square function of the voltage.

airplanes. This material is simple to secure on surfaces with the aid of a flatiron, but the insulating properties would have to be checked in practice.

Practical suggestions

The drawing of Fig. 5 shows a suggested structure of an electrostatic paperholder. Ordinary PC board material can be used as the base material. The track pattern is readily made with the aid of rub-off artwork transfers. A complete raster pattern on one sheet (track width: 3 mm; track distance: 1.5 mm) is rubbed off in one go. Alternate tracks are then shortened, and protruding tracks are connected at both sides. After etching, the panel can be smoothed with a thin layer of potting compound (car body repair material is suitable here). After this has stiffened, the layer is cleaned, polished, and covered in model aircraft foil (Fig. 5).

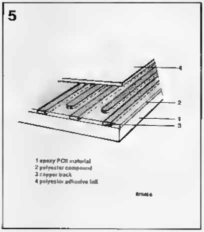


Fig. 5. Structure of a home-made paperholder to traditional design.

The high voltage source for the paperholder need not supply current because leakage current in the etched panel will be negligible. Figure 6 shows a suggested circuit for the high voltage cascade. The use of a mains transformer is obligatory. If a 1:1 safety transformer is not available, a step-down type (240 V/117 V) may be used with the corresponding number of cascade sections added. The actual voltage required depends largely on the foil thickness, so that the high voltage source is best constructed in a step by step manner by adding as many cascade sections as required. Commercially available electrostatic paperholders usually operate at 1 kV.
A prototype of the paperholder required 2...3 kV (track width 3 mm; track distance 1.5 mm; foil thickness approx. 0.05 mm). The circuit diagram of the voltage source used is shown in Fig. 6. Four cascade sections in each arm were

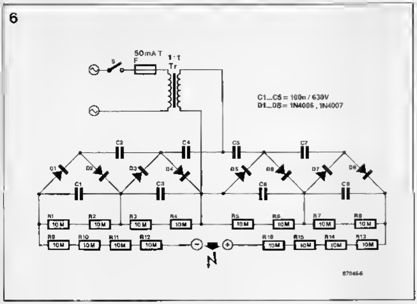


Fig. 6. Symmetrical high voltage supply for the paperholder.

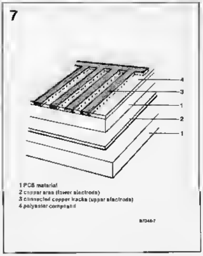


Fig. 7. Alternative construction of a paperholder, which is esentially a PCB sandwich. The HV electrode is formed by the unetched copper surface on the lower circuit board. The upper electrode is earthed.

used to give an output of 8×330 V=2640 V. The resistors fitted in parallel with the high-voltage capacitors ensure that the paper is released within 2 to 3 seconds after switching off. The high-value series resistors function as current limiters to safeguard users from lethal currents when the electrodes are accidentally touched. Every precaution should be taken to prevent this happening, bearing in mind that even small currents can be lethal when carried in or near the heart area.

An alternative

The electrostatic effect of the previously suggested paperholder is still relatively small, notably when using certain types of photographically sensitive or other PVC-based paper. An alternative paperholder was, therefore, designed and studied to overcome this defiency. The new structure is shown in Fig. 7: the working surface is essentially composed of double-sided PCB material. It is, however, recommended to use two separate sheets of single-sided material, since this automatically ensures insulation of the lower side. The lower electrode is simply a large conductive surface. The top side carries a fine pattern of interconnected tracks (a checkered pattern is also suitable) which forms the complementary electrode. Paper laid on the top surface will be at the potential of the upper electrode. The function of the etched pattern is to ensure that force is evenly distributed over the entire sheet. Adhesion is not obtained by dielectric shift in the paper, but as a result of the force between the charge transferred onto the paper by the upper electrode, and the charge on the lower electrode. There is no dielectric shift in the paper because this lies in an area of one potential only. This set-up has advantages in respect of safety and construction, because the upper electrode can be connected to earth, while the high voltage is only present well-insulated at the lower side.

The circuit diagram of Fig. 8 shows that the cascade used for the alternative paperholder is asymmetrical to prevent high voltages between the primary and secondary winding of the transformer. A 5-stage HV cascade was used to obtain an output of about 1700 V. Figure 8 also shows the use of two small low-voltage transformers whose secondary windings are connected to act as a 1:1 safety transformer.

A disadvantage of the alternative paperholder described is the need for the paper to be in galvanic contact with the upper electrode. This means a higher risk of oxidation of the copper tracks, unless these are tinned. The upper side of the work area can be smoothed with a thin layer of potting compound as discussed earlier.

It is hoped that this article provides a basis for further experiments in building an electrostatic paperholder of the required size. Your practical notes and comments are appreciated!

TW

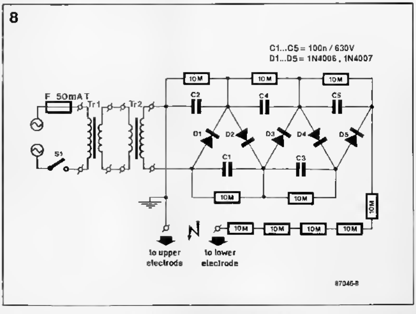
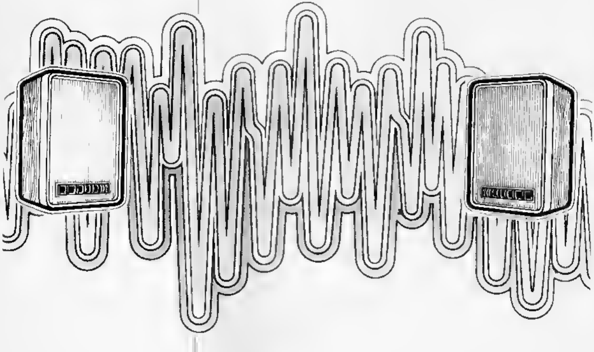


Fig. 8. Suggested HV source for the alternative paperholder. U_{out} is approximately 1700 VDC at an input of 240 VAC.

audio-analyser



Without an accurate picture of the frequency response of the sound reproduction system, the use of an equaliser can do more harm than good. For this reason an audio spectrum analyser, which can pinpoint the deficiencies in a particular audio chain and/or listening environment, is a virtually indispensable piece of equipment for the equaliser-user.

Attempting to set up a room acoustically by twiddling the controls on an equaliser and 'playing it by ear' is an almost certain recipe for heated tempers and high blood pressure, such is the difficulty of the task. To obtain any real benefit from an equaliser it is essential that the user knows exactly what changes he wants to implement in the frequency response of the audio system in question. It therefore follows that a reliable audio spectrum analyser is required to provide the acoustic information which is a necessary preliminary to effective equalisation.

An audio analyser system basically consists of three sections: a test signal source (pink noise generator), a microphone to monitor the output of the audio system under test, and a suitable means of analysing and displaying the energy level of the incoming signal. Broadly speaking, audio analysers fall into one of two types, depending upon whether the analysis is real-time or not.

Real-time analyser

A *real-time analyser* is the most sophisticated, but also the most expensive way of obtaining a detailed picture of the spectrum of an audio signal. The operation of real-time analysers can be explained with reference to the block diagram of figure 1. A broadband test signal is fed to the audio system under test. Normally the test signal consists of pink noise, which has a uniform energy level over the entire spectrum. The output of the audio system is picked up by a measurement microphone and fed to a bank of octave or third-octave filters, which split the input signal into a corresponding number of adjacent frequency bands. The output voltage of each filter is then rectified and displayed. Various types of display are possible — a moving-coil meter, an oscilloscope, or, as in the commercially available spectrum analyser shown in figure 2, a matrix of LEDs. The advantage of a real-time analyser is that it enables the average energy level of the entire spectrum to be determined at a glance. However, in view of the large number of displays and filter sections which are required, real-time analysers are not cheap. The above-mentioned pocket analyser of figure 2, together with a suitable noise generator, costs in the region of £ 600 — and that is only a fraction of what some of its 'larger brothers' can cost!

Since however, the primary application of the analyser is to monitor the response of an audio system to a *constant* test signal (the output of the pink noise generator, which has a uniform spectral intensity) real-time analysis is something of a superfluous luxury. A much cheaper, but none the less satisfactory arrangement is to have a single tuneable filter, which can be swept up and down the frequency spectrum as desired. This is in fact the solution adopted in the Elektor audio analyser.

1

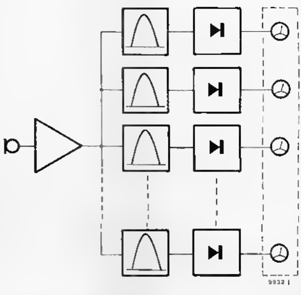


Figure 1. Block diagram of a real-time spectrum analyser.

The Elektor audio analyser

The block diagram of the Elektor, non real-time analyser is shown in figure 3. As can be seen, the basic principle of spectrum analysis remains the same, the only difference being that a single filter and display are employed, resulting in a considerable saving in cost. As far as the placing of the filter is concerned, three possible configurations come into consideration. In figure 3a the variable filter is situated between the pink noise generator and the input to the audio system, whilst in 3b it is fed from the output of the microphone. In figure 3c two filters are employed in an effort to obtain the best of both worlds. Although in theory there should be no difference between these three arrangements, things are not so simple in practice. With the configuration shown in figure 3a, all manner of interference and stray noise can reach the microphone and adversely effect the measurement. With the arrangement of figure 3b, this problem is effectively obviated, since only interference which lies within the passband of the filter can reach the microphone. A disadvantage of this set-up, however, is that only a very small portion of the pink noise spectrum is used, whilst the audio system in question is of course required to reproduce signals over the entire range of audio frequencies. The arrangement of figure 3c thus represents the ideal solution, however in view of the increased cost and complexity of two tracking variable filters, it was decided that, for this type of application, one of the simpler circuits (figures 3a and b) would prove sufficient.

The basic requirements for an analyser of the above type are therefore:

— a pink-noise generator
— a bandpass filter with stepwise or continuously variable centre frequency
— a suitable microphone with preamplifier
— a rectifier circuit
— a display circuit

2

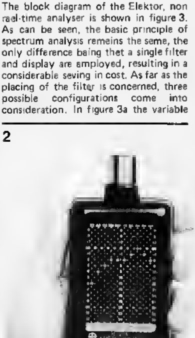


Figure 2. Photograph of a commercially available hand-held real-time analyser, incorporating a LED-matrix display.

As far as the choice of microphone is concerned, it is clear that, unless it itself has a fairly flat response, one cannot hope to obtain an accurate picture of the response of the audio system/listening room under test. For this reason it is important to invest in a reasonably good quality microphone capsule and preamp.

As a display circuit, a multimeter is as good as any, and has the advantage of being cheap and commonly available.

The remaining circuits, which form the heart of the analyser — and the substance of the rest of this article — are shown in figures 4a, 4b and 4c.

Noise generator

As can be seen from the circuit diagram of the noise generator shown in figure 4a, it in fact consists of a pseudo-random binary sequence generator, which has a longer than normal cycle time. This ensures that the noise has a high spectral density and that it is not characterised by the annoying 'breathing' effect obtained with short cycle times. The length of the shift register (IC1 . . . IC4) is 31 bits, and since the frequency of the clock generator (N5 . . N7, C1, C2, R3, R4) is roughly 500 kHz, the full cycle time is approximately an hour and a quarter!

EXOR-feedback is provided by N1 . . . N4. The circuit however has no anti-latch up gating. Instead there are two pushbutton switches, the START button ensures a logic 1 at the data input Q_0 of the shift register (pin 7 of IC1), thereby starting the clock cycle.

3

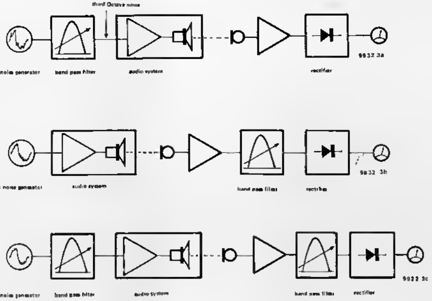


Figure 3. Three possible designs for a non real-time analyser.

4a

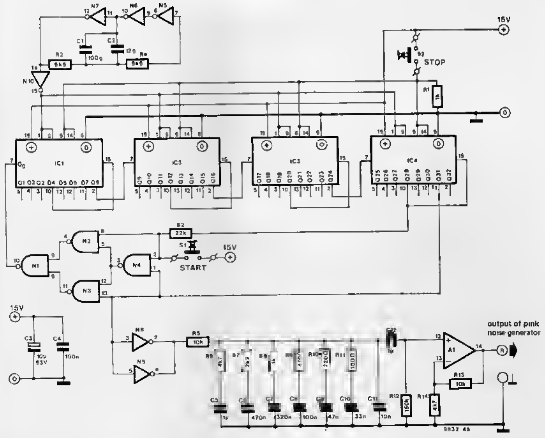


Figure 4a. The pink noise generator.

4b

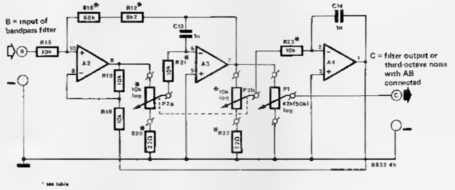


Figure 4b. The bandpass filter.

The cycle is inhibited by pressing the STOP button, S2. In this way it is possible to (temporarily) disconnect the noise source without switching off the supply voltage – a useful if not downright indispensable feature. The (pseudo-) white noise output of the shift register is fed to the pink-noise filter formed by R5 . . . R11, C5 . . . C11, before being amplified in the circuit round A1.

Bandpass filter

This section of the circuit (shown in figure 4b) is virtually identical to the third octave filter described in the article on the CMOS noise generator The output level of the filter can be varied by means of potentiometer P1, whilst the centre frequency can be varied between approximately 40 Hz and 16 kHz by means of the stereo potentiometer P2a/P2b. If stepwise control of the centre frequency of the filter is desired, P2a/P2b can be replaced by a pair of attenuator networks and a twin-ganged switch. The necessary modifications are detailed in figure 5. Resistors R20 and R22 are replaced by a wire link, the values of R21 and R23 are altered, and R40 and R41 are added. Table 1 lists the various resistance values required to give the ISO standard centre frequencies. When calibrating a parametric equaliser, a filter bandwidth of less than 1/3 of an octave is required. By altering the value of R16 to 220 Ω and replacing R17 by a wire link a bandwidth of approximately 1/12 of an octave can be obtained.

Rectifier circuit

It is of utmost importance that the amplitude of the test signal be measured accurately. If a pink noise test signal is used in conjunction with filters which

4c

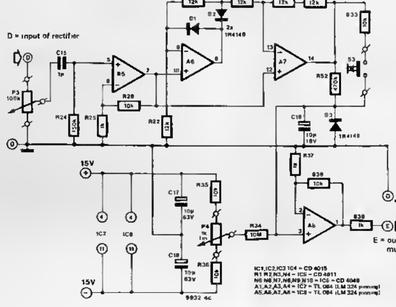


Figure 4c. The rectifier circuit.

have a constant octave or 1/3 octava bandwidth (i.e. filters with a constant Q) one should really maasura the RMS value of the noise — not an easy mattar. Fortunately, however, a reasonably simple alternative exists — namaly to measure the avarage of the modulus value, i.e. the average of the full-wave rectified noisa signal. This is obtained by feeding the output of the paak rectifier to a lowpass filter.

The rectifier circuit is built round IC8. The input level control is followed by an amplifier, A5. The actual (full-wave) rectification is performed by A6, A7, R27 . . . 31, D1 and D2. Tha output of A7, which always presents e low impedance, is connected via R32 to C16. Because this capacitor hes the same charge and discharga time, the voltage on the capacitor will aqual tha avaraga value of tha full-wave ractified noise voltage. The time that this voltage remains stored on the capacitor is detarmined by the RC tima constant, R32·C16, or, if S3 is depressed, R32/R33·C16. Depressing S3 causes C16 to charge and discharga much more rapidly, so that the cepecitor voltege will follow rapid variations in the noise voltage. Thus S3 is intended to provide a rapid overall view of the variations in noise level for different centre frequencies of the filter. For accurate measuremants, the longer time constant of R32·C16 should be used. After being amplifiad in A8, the voltage on C16 is displayed on the multimeter An offset control is provided (P4, R34 . . . R36) to enable the mater to be calibrated accurately (zero deflection under quiascent conditions).

Construction

A printed circuit boerd, which is shown in figure 6, hes been dasigned to accommodate the circuit of figures 4a, b end c.

Table

1	2	3	4	5
31,5	1/t	2Ω2 + 2Ω2	18 k	w
31,5	t/3	2Ω2 + 2Ω2	68 k	8k2
40	1/3	5Ω6	68 k	8k2
50	1/3	4Ω7 + 2Ω2	88 k	8k2
83	1/1	4Ω7 + 3Ω9	18 k	w
63	1/3	4Ω7 + 3Ω9	68 k	8k2
80	1/3	10 Ω + 1Ω2	68 k	8k2
100	1/3	10 Ω + 3Ω9	68 k	8k2
125	1/1	12 Ω + 5Ω6	18 k	w
125	1/3	12 Ω + 5Ω6	68 k	8k2
160	1/3	22 Ω	68 k	8k2
200	1/3	27 Ω + tΩ8	68 k	8k2
250	t/1	33 Ω + 2Ω2	18 k	w
250	1/3	33 Ω + 2Ω2	68 k	8k2
315	1/3	22 Ω + 22 Ω	68 k	8k2
400	1/3	56 Ω	68 k	8k2
500	1/1	68 Ω + 3Ω3	18 k	w
500	t/3	68 Ω + 3Ω3	68 k	8k2
630	1/3	82 Ω + 8Ω2	68 k	8k2
800	1/3	100 Ω + 18 Ω	68 k	8k2
1000	1/1	100 Ω + 47 Ω	18 k	w
t000	1/3	100 Ω + 47 Ω	88 k	8k2
1250	1/3	120 Ω + 68 Ω	68 k	8k2
1600	1/3	220 Ω + 27 Ω	68 k	8k2
2000	1/1	270 Ω + 47 Ω	18 k	w
2000	1/3	270 Ω + 47 Ω	68 k	8k2
2500	t/3	390 Ω + t8 Ω	68 k	8k2
3150	1/3	470 Ω + 68 Ω	88 k	8k2
4000	1/1	680 Ω + 47 Ω	18 k	w
4000	1/3	680 Ω + 47 Ω	68 k	8k2
5000	1/3	820 Ω + t50 Ω	68 k	8k2
6300	1/3	1 k + 390 Ω	68 k	8k2
8000	t/1	1k8 + 330 Ω	18 k	w
8000	1/3	1k8 + 330 Ω	68 k	8k2
10.000	1/3	3k3 + 390 Ω	68 k	8k2
12 500	1/3	5k6 + t k	68 k	8k2
16.000	1/1	39 k + 1k2	18 k	w
16.000	1/3	39 k + 1k2	68 k	8k2

Ramarks:

column t: cantra frequency in Hz
column 2. bandwidth in octaves
column 3. value of resistor to be connacted batwaan tha junction of resistors R40 and R21 and ground and between tha junction of R41 and R23 and ground, rounded up to values from tha E12 serias
column 4: value of R16
column 5. valua of R17 (w = wira link)

5

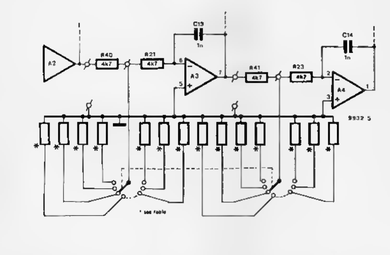


Figure 5. Modifications to the bandpass filter to obtain switched centre frequencies.

The design of the board is such that either of the configurations shown in figures 3e and 3b can be edopted. The construction of the standard version circuit should present no spacial problems. The wiring for the potentiometars and switches should be kept as short as possible. The connections for these components are arranged at one end of the board. Problems of a practical nature do arise, howevar, if one desires a number of switched filtar frequencies, since one then requires a switch with a corresponding numbar of ways. Since switchas with a larga numbar of ways are both expensiva and difficult to obtain, an alternativa solution is simply to use the desired number of double-pole singla-throw switchas. This of course involves operating two switches each tima one wants to altar the centre fraquency of tha filtar.

In addition to tha switch(es), the choice of fixed filtar frequancias involves the following alterations on the board (see

6

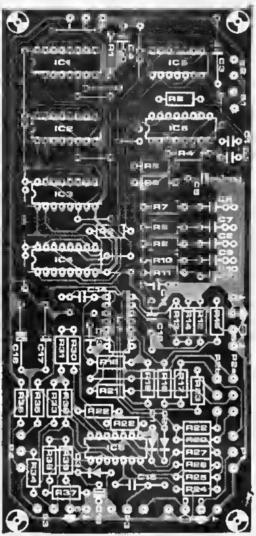


Figure 6. Printed circuit board for the circuit of figure 4.

parts list

Resistors:

R1,R8,R25,R37,R39 = 1 k
R2 = 22 k
R3,R4 = 6k8
R5,R13,R15,R18,R19,R21,R23,
R26,R33,R35,R36,R38 = 10 k
R6,R14 = 4k7
R7 = 2k2
R9 = 470 Ω
R10 = 220 Ω
R11 = 100 Ω
R12,R24 = 150 k
R16 = 68 k
R17 = 8k2
R20,R22 = 22 Ω
R27 . R31 = 12 k
R32 = 470 k
R34 = 10 M
P1 = 47 k (50 k) log potentiometer
P2a/P2b = 10 k log stereo potentiometer
P3 = 100 k log potentiometer
P4 = 1 k linear potentiometer

Capacitors:

C1 = 100 p
C2 = 12 p
C3,C17,C18 = 10 μ/63 V
C4,C8 = 100 n
C5,C12,C15 = 1 μ MKM
C6 = 470 n
C7 = 220 n
C9 = 47 n
C10 = 22 n
C11 = 10 n
C13,C14 = 1 n
C16 = 1 Uμ/35 V tantelum

Semiconductors:

IC1,IC2,IC3,IC4 = CD 4015
IC5 = CD 4011
IC6 = CD 4049
IC7,IC8 = TL 084 (Texas Instruments) DIL
D1,D2,D3 = 1N4148

Miscellaneous:

S1,S2,S3 = pushbutton switch, single-pole push to make

7

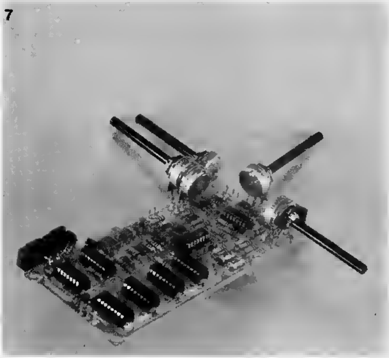


Figure 7. A prototype of the audio analyser.

figure 5): R21 and R23 become 4k7
R20 and R22 are replaced by a wire link
a 4k7 resistor (R40) is soldered between the 'top' two tags of P2a
a 4k7 resistor (R41) is soldered between the 'bottom' two tags of P2b

The resistor pairs forming the switched attenuator network are mounted externally on the switch(es). Suitable values are given in the table.

With [1]a continuously variable filter frequency it is useful to equip P2a/b with a pointer and scale. The scale can of course be calibrated in frequencies, but it is not strictly necessary. What matters is that one has a series of reference points – peak or dip at such and such a filter setting, etc. If, however an absolute frequency scale is desired, this can be obtained by using a tone generator and noting the frequency when the output voltage at point C is at a maximum, when feeding a pure sinewave into point B.

Using the analyser

The multimeter (10 to 12 V full-scale deflection) which is used to display the amplitude of the noise signal is connected to the output (point E) of the rectifier circuit. In the absence of an AC drive voltage (i.e. point D disconnected or else P3 turned right down) the DC voltage at this point should be set by means of P4 to *exactly* 0 (m)V. The correct setting for P4 is obtained by repeatedly switching down the voltage range of the multimeter and checking the reading by reversing the polarity of the probes. It should be borne in mind that, because of the long time constant of R34 and C16, it will take some time for adjustments to P4 to have any effect. The long discharge time of the storage capacitor in the rectifier circuit together with the natural inertia of the meter ballistics ensure that the needle responds only very slowly to changes in the level of the filter output. Thus when sweeping the filter up and down the audio spectrum, care should be taken to vary the filter frequency *gradually*, lest peaks or dips in the response are camouflaged by the slow response of the circuit.

If the analyser is used to measure a system with a completely flat response, the *mean* meter deflection (i.e. the mean between the maximum positive and negative deflections) should be independent of variations in the filter frequency. An audio system with a completely flat response would be pretty hard to find, however, something which does have a more or less flat response is a wire link! – by joining points A and B and C and D in this way (i.e. connecting the output of the noise generator to the bandpass filter and the output of the filter to the rectifier circuit) it is possible to test the operation of the audio analyser, and in particular, of the pink noise and bandpass filters. Variations of up to ±2 dB (0.8 . . . 1.25) in the mean meter reading are acceptable. To prevent the rectifier circuit from being overloaded, the mean meter reading can be adjusted to occur at around 3 . . . 4 V.

Finally a word of warning: care should be taken to ensure that the noise signal does not overload one's audio equipment. The risk of this happening is somewhat greater than in the case of a sine or squarewave input signal, since the distortion caused by overloading will be that much less noticeable (but none the less disastrous!). Tweeters in particular are susceptible to damage by being overloaded with high level noise signals.

Constructing the audio analyser is one thing, using it is another. The reader is therefore referred to the article on 'Using an equaliser', which deals with the subject of using the equaliser/analyser combination to measure and then correct a room's response.

HOLOGRAPHY AND LASERS PRODUCE SUPER PRECISE MEASUREMENT

by Anthony St E. Cardew*

Advances in the application of opto-electronics such as laser techniques and holography have brought industrial measurement to new realms of precision and speed. The ability to have workshop systems with inherent accuracies of a half wavelength of light has opened up a whole new vista. Also, developments in holography have extended its application to, for instance, use in the quality control of critical parts and the location of defects.

In the United Kingdom, a significant advance in laser interferometer design has been the pioneering of a novel system by Michael Downs and Kenneth Rains of the National Physical Laboratory[1], which was exploited by Linear Instruments[2] and forms the basis of its LIL 3000 series of laser interferometers.

Lower cost

Most laser interferometers use a sophisticated laser and electronic system to produce measurements and they are consequently very expensive. However, the laser transducers manufactured by Linear Instruments, which are of the remote interferometer type, achieve a performance equal to, or better than, conventional laser interferometers by using a simpler and less expensive method.

Instead of producing two identical signals with 90 degree phase difference, they produce three signals at 0, 90 and 180 degrees. By subtracting 0 from 90 and 90 from 180, the results are two signals with a phase difference of 90 degrees which switch about zero volts.

Apart from its simplicity and low cost, the great advantage of the system is that it is not dependent on any one laser but will operate with an unstabilized laser over short distances.

Although laser interferometers have a working range of 30 metres or more, this is dependent on the environment as the beam will move in turbulent air. Maintaining a good overlap between the reference beam and the measuring beam is essential and, therefore, over the longer ranges, it is better to use a larger beam diameter.

On the LIL series of laser heads it is possible to change the standard 3× beam expander to 5×, increasing the beam from 3 mm to 5 mm diameter.

Errors eliminated

In practice, the interferometer is supplied with a two-mode stabilized laser. The LIL models, based on the National Physical Laboratory design, are becoming established round the world in the fields of measurement and positional control.

The range of calibration equipment has been complemented by a tripod-based laser and interferometer system known as Uni-Cal which is specifically designed as a portable, length-measuring system for machine calibration. It consists of a laser mounted on a heavy-duty tripod. As the retro-reflector moves backwards and forwards, the interferometer measures its position.

With a mirror, it is possible to reflect the interferometer beam on to another axis so that three axes of the machine can be calibrated from one position. While this technique can produce errors in conventional systems, the special 'dead path' facility in the Uni-Cal system software eliminates this problem.

The latest addition is a software package written specifically for the calibration of transducers and ballscrews which allows up to five runs of 1000 points to be processed to provide statistics and graphs. A special feature with this package is provision for a chart-driven X-Y plotter which will enable the scale of the X axis of the graph (the measured axis) to be varied to suit the requirements from a single A4 sheet to a 25 metre long chart.

Shadow graph

The Beta Instrument Company[3] specializes in applying laser technology to the measurement of optical fibres, wires and filaments and also glass thickness.

Its most recent development is the enhanced Beta fast response fibre laser diameter gauge which provides a precise method of measuring optical fibres, fine wire drawing, wire/cable extrusion, and filament manufacture.

The unit operates on the principle of a fine laser beam sweeping past the product to be measured, which is located in the optical gate. The resultant signal is collected on a photocell, which produces a square wave that is related to the precise diameter of the profile being measured.

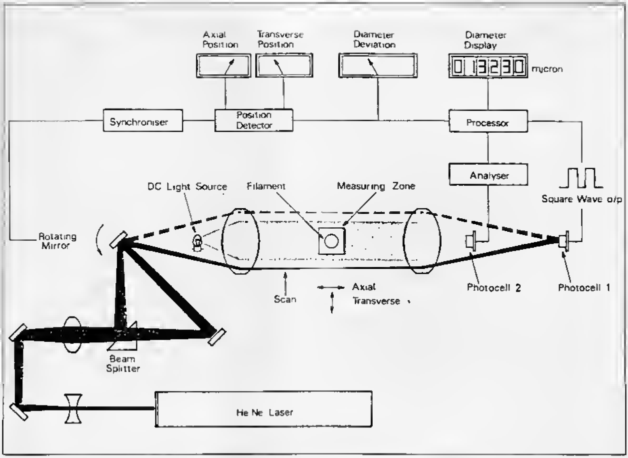
A second beam of light from a d.c.-operated continuous illumination source is projected on to the same product to produce a shadow graph image for examination of the surface imperfections and instant variations of the product under test.

This second beam of light is collected on a second photocell which produces a signal proportional to the instantaneous variations of the profile shadow of the product. The signals from the photocells are conditioned in the microprocessor which provides an actual and diameter deviation display.

Measuring reflections

Another laser-based system developed by Beta is the glass thickness gauge used for non-contact measurement of wall thickness of glass tubes, bottles, phials, capillary tubes and so on, as well as glass

* Anthony St E. Cardew is chairman of the Metrology and Inspection Technical Group of the Institute of Quality Assurance.



Principle of operation of the Beta SF fibre laser diameter gauge.

plate. Measurement is made as the glass is produced from the furnace, hot or cold.

The general method relies on a laser beam being made to fall on the surface of the glass tube or plate being measured, and produce reflections from the two surfaces. Both reflections pass through a lens system and on to a diode array, which is being scanned at a rate of 100 times a second.

Each scan produces a measurement that is checked for credibility and, after processing, the actual thickness of the material is shown on a six-digit display. In the first instance, the reflective index of the material has to be dialled into the unit by the user.

Alternatively, if the exact thickness of the glass plate is known, the instrument can be used to determine the refractive index.

Three dimensional data

For lower orders of accuracy, the Beamguide red-cross optical alignment system by Coteglade Photonics[4] provides a low-powered He-Ne laser system. An inexpensive, adaptable, general utility tool, it projects a visible laser line or cross-hair on to the target object, with finely adjustable accuracy to give precise planar positioning information.

Full orthogonal positioning accuracy can be obtained by mounting three Beamguides in suitable positions to give three-dimensional information. Typical applications include levelling in which a fixed, remote Beamguide gives precise positional orientation and levelling information for manned or automatic machinery.

It can also be used for centring where two or more beams can be focused on the central point of a piece of equipment whether static or moving.

Holography is finding increasing application in industry, particularly in the field of non-destructive testing where holographic interferometry enables small flaws or defects to be seen. In its simplest form, the process involves superimposing two holographic exposures on the same film and subjecting the object to mechanical or environmental changes between exposures.

Aid to design

The reconstructed hologram shows the object overlaid with a fringe pattern (caused by the interference of the two exposures) which is effectively a topographical map with contours defining every change in the surface of the object.

The distance between each fringe represents a movement of half a wavelength of the laser illumination. If a He-Ne laser at 632.8 mm is used, for example, a change of 0.3 μm will be detected. Holographic inspection is useful not just for checking production items but also as an invaluable aid in their design and development.

An example of a commercial holographic system is the Ealing Electro-Optics[5] Vidispec electronic speckle interferometer, which is finding increasing application in non-destructive testing. It measures the surface displacement of an object subjected to a mechanical load or an environmental change.

This is achieved with an accuracy of about a wavelength of light and enables any flaws or defects in the material or design of a component to be identified quickly.

Dual-size holograms

Unlike holographic non-destructive testing, Vidispec works well in daylight and artificial light and needs no film or special processing techniques.

The optical head houses a 10 mW He-Ne laser, a precision video camera, and all the necessary optical and mechanical

components to create and record a speckle pattern interferogram.
The electronic unit contains the video camera controls, digital frame store and electronic processing functions.
The Holocam model 70 camera from Ultrafine Technology[6] enables the user to make both 127 × 101 mm and 254 × 203 mm holograms from the one instrument. It incorporates an improved method for holding the film captive, using a single glass plate and a vacuum system. Recent applications include the testing of bonded structures.

Optical system

A significant application of the laser-to-surface metrology is the National Physical Laboratory's development of an optical system for the measurement of surface profile. The system incorporates a laser and takes advantage of the very high measuring sensitivity of polarization interference microscopy.
The practical result is a surface profilometer with sub-nanometre sensitivity for the measurement of smooth surfaces.
The system uses a birefringent lens in conjunction with a microscope objective to provide a double-focus objective in which the two foci correspond to the light of orthogonal planes of polarization.
When the surface under examination is placed on one of the focal planes, the light of one polarization is reflected from an area equal to the resolution limit of the objective. The light of the other polarization, on the other hand, is out of focus and is reflected from a larger area.
Each beam integrates the level of the surface over the area from which it is reflected. The larger area provides a mean reference level which should remain fixed as the area is scanned.

Double-focus objective

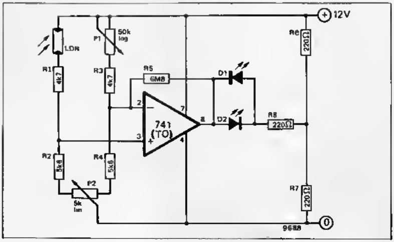
The two beams are combined in a polarizing interferometer and, as the surface is scanned, the variation in path difference between the focused and unfocused beams provides a measure of surface profile. An electro-optic system is employed with an electrical output directly proportional to this path difference.
The use of a common-path interferometer, in which both measurement level and the reference level are generated from the test surface, provides a measurement that is insensitive to movement in a direction perpendicular to the surface. Therefore, the use of this form of double-focus objective renders the system insensitive to tilt of the test table.
Patents for the system are held by the National Physical Laboratory and versions of the nano-profilometer are being manufactured by British Aerospace and the Cranfield Unit for Precision Engineering (CUPE), which is part of the Cranfield .itute of Technology.

References.

1. National Physical Laboratory, Teddington TW10 0LW.
2. Linear Instruments Ltd, 9 Raynham Road, Bishop's Stortford CM23 5PB.
3. Beta Instrument Company Ltd, Halifax House, Halifax Road, Cressex Industrial Estate, High Wycombe NP12 3SW.
4. Coleglade Photonics Ltd, Brunel House, 1275 Neath Road, Hafod, Swansea SA1 2LB.
5. Ealing Electro-Optics Ltd, Greycaine Road, Watford WD2 4PW.
6. Ultrafine Technology Ltd, 16 Foster Road, Chiswick, London W4 4NY.

Dark Room Aid

This circuit is intended as an aid in the dark room to ensure correct exposure time without effort. Before the paper is placed under the enlarger, the amount of light is measured by means of an LDR. This is the light-sensitive element.
The LDR makes up one branch of a bridge circuit, which is formed by the LDR, R1, R2 and part of P2. The other branch consists of P1, R3, R4 and a part of P2. With P2 in centre position, and P1 adjusted to a resistance value lower than that of the LDR, the voltage at the + input of the op-amp is lower than the voltage at the − input. With this condition, the output voltage of the op-amp is negative, and D1 lights up. On the other hand, when the resistance of P1 is higher than that of the LDR, the output is positive, and D2 lights. If the voltages on both inputs are equal, both LEDs light at half brilliance. This is due to the fact that hum is picked-up by the high sensitivity op-amp. Thus the circuit indicates when the resistance values of the LDR and P1 are equal. When the LDR is placed under the enlarger, its resistance value will correspond to the light intensity.
P1 is now adjusted until the bridge is balanced, after which the exposure time can be read from a calibrated scale attached to P1. By adjusting P2 slight changes to the bridge balance are possible. In this way it is possible to introduce corrections for different sensitivities of the photographic paper.
This dark room aid has only one drawback: the scale calibration of P1 can only be obtained by spending some evenings in the dark room. But, of course, for the real enthusiast, this is no problem at all!
The LDR must be mounted in a flat holder and be partly covered by a mask. This method allows spot measurement and also helps adapt the LDR to the circuit. To calibrate the unit, first of all ensure that the bridge can be balanced with P1 over the entire range, from extreme light to extreme dark. If not larger or smaller apertures in the LDR mask can be tried. After this, by using an ohm-meter, P1 can be provided with a scale.
For example: P1 = 500 Ω gives one second, 1 kΩ gives 2 seconds, and so on until P1 = 32 kΩ which corresponds to 64 seconds. By means of test strips and an 'average grey' negative, the exposure time can now be brought into accordance with the sensitivity of the photographic paper by means of P2. To this end, P2 is also provided with a scale with the type numbers or gradations for different papers.
The control range of P2 is equal to 4 stops. If this scale is shifted too far towards one of the extreme positions, the LDR mask must be changed.
Since only a small area of the picture is measured by the LDR, it is possible to determine the contrast, and choose the type of paper accordingly.



Humidity Indicator For Potted Plants

Compared to our barking, mewing and twittering friends, the potted plants are the most contented lot. They need the minimum of care, mostly just the regular watering is the main part of it. Too less or too much humidity will soon see the end of your green pride.

It is often not too easy to recognise the humidity contents of the garden soil from its looks. On the surface, it may be dry, but just a few centimetres below the surface it may hold sufficient humidity. Thanks to our electronics, it is not essential to dig in the pot in order to see, whether everything is in order.

Functional description:

If you see figure 1 closely, you will certainly wonder, why we are using the term "Electronics" here for such a simple circuit. There are none of those usual electronic components like resistors, capacitors, nor the semi-conductors! There is just one moving coil meter connected to two electrodes.

For a moment, one would doubt whether it will work at all? In fact it does work! A little background knowledge of physics is enough to find out why it works.

Both the electrodes, together with the humid soil, make a battery, the so called "Galvanic Cell". The conditions under which this battery works are that the two electrodes must be of two different metals, the soil must be moist and must contain some salts.

The higher the humidity in the soil, the higher is the current produced by this circuit. However, if you have already started thinking of using this type of batteries to reduce your electricity bills, you must immediately curb your thoughts. The current thus produced is just a few microamps and not even enough to light up the smallest bulb, even to a faint glow.

This requires a very sensitive meter to measure such a low current. An instrument with about 50 to 100 micro amperes full scale range should be adequate. A volume control indicator from an old tape recorder may serve the purpose.

The Electrodes:

The two electrodes can be formed by using a copper clad board etched in the form shown in figure 2 and a screw driver. Soldering the lead wires to the PCB is no problem, however, soldering a wire to the screw driver will turn out to be a futile excercise. The lead wire must be tied around the screw driver shaft and twisted tightly. Figure 2 also shows a suggestion for the assembly of our humidity meter. This type of design makes it easy for inserting the electrodes into the soil. To avoid corrosion of electrodes they must be properly cleaned after every humidity check. The length of the electrodes must at least reach upto half the depth into the soil.

Calibration:

Due to the large expected variations in all the parameters, data for absolute calibration of our instrument is almost impossible. Here, the only method applicable is "trying out!" This can be done as follows:

Take a pot with soil which is just watered to a sufficient degree, corresponding to an expected ideal condition. Now insert the electrodes into the soil and observe the meter deflection. If the needle deflects in the negative direction, than the instrument polarity must be reversed. Which of the electrode acts as the

1

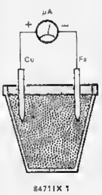


Figure 1:

The somewhat unusual "Circuit" of the humidity meter. It consists only of two electrodes and one meter, and of course, a pot full of soil. A current flows in the circuit if the soil is humid enough.

2

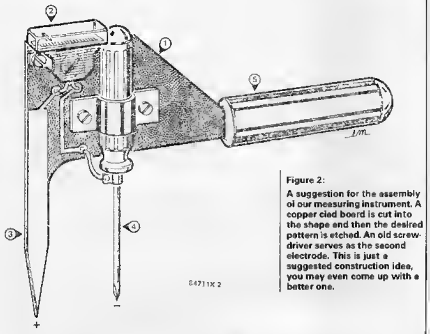


Figure 2:
A suggestion for the assembly oi our measuring instrument. A copper cied board is cut into the shape end then the desired pottern is etched. An old screw-driver serves as the second electrode. This is just e suggested construction idee, you mey even come up with e better one.

1 **Copper cied Boerd, cut to desired shepe**

2 **Meter (50 or 100 micro emp fuii scale)**

3 **Copper trecks**

4 **Screw Driver**

5 **Hendle.**

positive electrode depends upon the valancy of both the materials of electrodes. A short note appears at the end, on this subject.

If the needle of the meter deflects upto at least half the scale, it is a good indication. Howaver, if there is vary less or no movement at all, try changing the electrodas, or try using a more sensitive meter. When you are succassful in getting a sufficiant deflection, mark that as the maximum deflection. Your "humidity meter" is now raady to use. At this point, we also would like to warn you against putting this "meter" to any serious use, as the maesurement has no ebsolute calibration.

Note:

If a metal is immarsed in an acqueous salt solution, then positiva ions ara released by the metal. The metal thus becomes negatively charged. If two diffarant matals ara thus immersed in the same solution, depending on how easily the metals can ralease the ions, the two different metals get charged to diffarent levels. (For example, copper and zinc.) This creates a potential difference between the two metals. If they are now externally connected with a conducting wire, a currant will flow from one metal to the other, through that external wire. Idaally this current will continue to flow till one or both the metals are completely dissolved in the solution. The metals are classified according to their valancy, in the "Electro-Chemical-Series". The sequence is Platinum – Copper – Iron – Zinc – and so on. Every metal in the series is able to give out more ions than the previous one, and thus the metal in comparison with the next one always forms the positive pole of the battery.

A.C. Motors

'VISHAL' introduces SYN-341 motors which are 3 leads type, continuous duty rated, Low speed (60 RPM at 50 HZ), High Torque (3.5 Kg. CMS & 7 Kg. CMS). A C. Synchronous Motors having instant starting, stopping and reversing characteristics.
For servo application, Motors with 12 teeth delrin gears and both and extended shaft to be provided on request. Higher Torque upto 60 Kg. CMS Motors are under development. Geared Motors are also provided

Vishal Servotech • D-110 Bonanza Industrial Estate • 1st floor • Ashok Chakravarti Road • Kandivli (East) • Bombay-400 101.

Profile Controller

Micronix offers a new version of their Temperature profile controller. The instrument is designed to cater the needs in complex heat treatment applications. It faithfully follows the profile program stored in the memory and does not require operator monitoring even in case of emergencies like power failure.

The salient features include:
Easy flexible programming algorithms owing to advanced microprocessor technology. 8-digit display alongwith status indicators allow the user to monitor full system status at any given time.

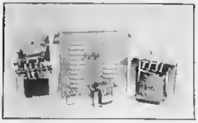
Full linearisation algorithms offered for all the standard sensors.

Fully modular design makes servicing easy and reduces system downtime. Optional support includes real time clock reference for programming the profile and printer interface for hardcopy of the program.

Micronix • D-74, Angol Industrial Estate • Udyambag • Belgaum-590 008. • Karnataka State.

Transformers

SHEPHERD TRANSFORMERS manufacture a wide range covering Auto, Control, Pulse L.T. Power, Motor Starting, Ignition, Current and Step-up/Step-down transformers.



M/s. Shepherd Transformers • Nityanand Nagar • Off Link Bridge • Ghatkopar (West) • Bombay-400 086.

Thickness Gauge

AA INDUSTRIES have developed "COATMETER" that measures thickness of non-magnetic coating on magnetic bases eg plating, paint, galvanising, powder coating, glass/rubber/FRP/lead linings etc. Scales offered are 0-80 microns, 0-250 microns, 0-600 microns, 0-1 mm, 0-3 mm, 0-6 mm, 0-9 mm, % FERRITE, Gm Zn/m^2 etc. This insturment can be used in all planes – vertically, horizontally or in an incline plane. When the scale is not visible, the reading can be locked and read afterwards.

AA INDUSTRIES • Samadhan • Kesarbaug • Mulund (East) • Bombay 400 081.

LCD DPM

ELINCO, in technical collaboration with LASCAR ELECTRONICS LTD. U K., offers a Low power LCD DPM module with True Digital Hold of displayed reading. It features Auto-Zero, Auto-Polarity, 200mV FSD, Low-Battery indication, 12.5 mm digit height and programmable decimal points. It has an accuracy of 0.05% ± 1 count (0.1% max.) It operates on 9 V battery (7.5 to 15 VDC) and consumes less than 1 mA. The inputs are protected upto ± 20 VDC.
A 0.1" header provides connection by a variety of methods and the module may be scaled to indicate different voltages, current or other engineering units
This meter is particularly suited to high volume applications. Supplied complete with DIN compatible bezel, incorporating protective window and mounting kit.

Electronics India Co • 3749 Hill Road Ambala Cantt- 133 001 India

NEW PRODUCTS • NEW PRODUCTS • NEW

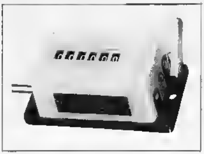
Revolution Counters

M/s. Controls & Equipments introduce series R400 Revolution Counters ideally suitable for coil winding machines to measure number of Turns.

The design incorporates lubricating bearing and gears.

Quick lever reset facilities bringing all the figures to zero instantly.

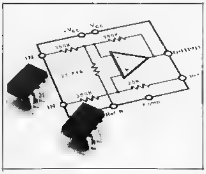
This model with a minor change can be used for linear measurement in Textile, Paper and other Industries as well.



M/s. Sai Electronics • Thakor Estate • Kurla Kirol Road • Vidyavihar (W) • Bombay-400 086. Phone 5136601.

Precision Difference Amp.

The INA117P, just released by Burr-Brown, is a differential amplifier consisting of a premium-grade operational amplifier and a precision resistor network all in a single 8-pin DIP. Its typical key features include 74dB min CMR, 0/+70°C temp range, 0.001% nonlinearity, 0.05% max gain error, 6.5 us settling to 0.01% Specified minimum common-mode input voltage range is ± 200V (DC or AC peak) and a differential input range is ± 10V. It finds applications in the monitoring of power supply or leakage currents, test equipment, and industrial/process control and data acquisition applications where total galvanic isolation is not required.



Oriole Service & Consultants Pvt. Ltd. • P.B. No. 9275 • 4 Kurla Industrial Estate • Ghatkopar (West) • Bombay 400 086.

Mini Switches

"IEC" have recently introduced Mini Push Button Switches specially for use in Electronic equipment & instruments. The mini Push Buttons are available in 'Push to On-Push to Off' type or with momentary contacts, with electrical rating of 2 Amps/250V AC and 28 Volts DC. The body is of electrical grade bakelite with Brass terminals, Silver plated.

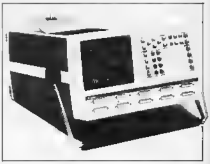
Indian Engineering Company • Post Box 16551 • Worli Naka • Bombay-400 018.

Logic Analyser

A New high-performance logic analyser which can monitor and capture data across 80 identical data channels at 100 MHz with 10ns resolution has been introduced by Gould Electronics Ltd., U.K., represented in India by Larsen & Toubro Limited. Alternatively, it can be configured from the front-panel keyboard to capture 40 channels of data at 200 MHz with 5ns resolution.

Designed K450B, the logic analyser has been designed to remove the restrictions on the examination of wide data paths imposed by earlier logic analysers. The use of identical inputs, 5ns glitch capture across all 80 channels and a sample memory up to 4 kbyte deep offer a number of benefits.

With the K450B, a user can analyse state and timing signals on 32-bit microprocessors without worrying about which signals will have the benefits of high-speed resolution and which must be relegated to low-speed measurement channels. All signals can be captured synchronously or asynchronously at the sample speed appropriate to the measurement.



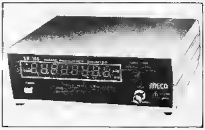
Instruments Division • Larsen & Toubro Limited • Venkataramana Centre • 563 Anna Salai • Madras 600 018.

150 MHz Frequency Counter

Meco has introduced an easy to operate Digital Frequency Counter incorporating LSI circuit.

The Frequency Counter gives upto 8 digits of resolution with a wide frequency range of 10 Hz to 150 MHz. Besides a memory system 'holds' the last input digits on the panel for future reference.

The counter can be used for adjustments, test and repairs of Electronic Clocks, Watches, AM/FM radio etc.



Meco Instruments Pvt. Ltd. • Bharat Industrial Estate • T.J. Road • Sewree • Bombay 400 015.

R N No 39881/83
MH BY WEST 228
LIC No 91