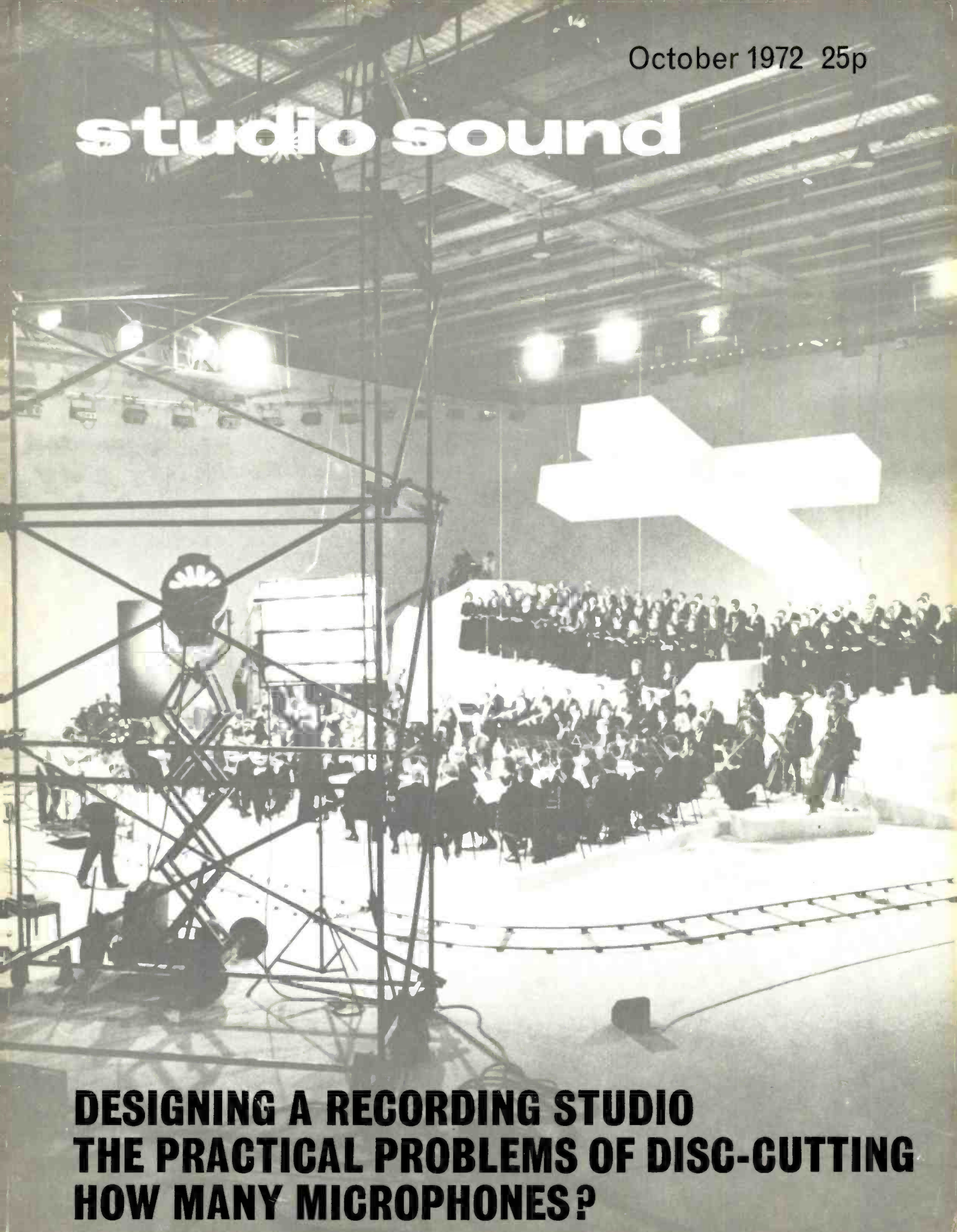
October 1972 25p

studio sound



DESIGNING A RECORDING STUDIO
THE PRACTICAL PROBLEMS OF DISC-CUTTING
HOW MANY MICROPHONES?

studio sound
INCORPORATING TAPE RECORDER

OCTOBER 1972 VOLUME 14 NUMBER 10

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IT APPEARS TO be taken for granted, in some circles, that recording companies, studios, national and commercial broadcasting stations are supported by limitless financial reserves. These most fortunate organisations consequently have little concern for the price of a service or item of equipment which catches their eye. The result, in STUDIO SOUND, has been a large-scale omission even of basic prices from advertisements and (we readily admit) editorial references. While the majority of companies co-operate in supplying technical specifications for equipment surveys, only a minority volunteer prices.

Meaningful comparison of the facilities and relative value-for-money offered by a particular group of manufacturers is thus pushed from the difficult into the virtually impossible. In this industry at least, manufacturers tend to hide behind the tradition that prices are negotiated for individual installations—a polite way of saying that the more you order, the cheaper you'll get it. Importers are generally more closely involved with their price lists though even they have been put into a state of minor confusion by the so-called 'floating' pound.

Those companies who take the trouble to quote prices are placed in an immediate dilemma since there are two dissimilar methods of marketing. Some publish a realistic price, only to be agitated by customers trying to wheedle, as they put it, 'their usual discount'. Other companies, particularly those with American connections, prefer to inflate their basic prices both to imply inherently superior equipment and to permit enticing discounts to favoured customers. It is this practice that makes us reluctant to argue prices out of secretive manufacturers: the figures, if obtained, are usually misleading. Nor can STUDIO SOUND help by researching discount structures, until such time as every reader possesses an electronic calculator. We can only suggest that the industry assist itself by being less furtive about what it has to offer.

FEATURES

COLUMNS

REVIEWS

COVER

Karl Richter conducting the Munich Bach Orchestra and Choir in a Bavaria Film Company studio. W. A. Wettler describes on page 20 some of the problems which arise in the simultaneous filming and sound recording of such productions; problems which, he insists, could not be solved by conventional coincident microphone techniques.

CORRESPONDENCE AND ARTICLES

All STUDIO SOUND *correspondence should be sent to the address printed on this page. Technical queries should be concise and must include a stamped addressed envelope. Matters relating to more than one department should occupy separate sheets of paper or delay will occur in replying.*

Articles or suggestions for features on all aspects of communications engineering and music will be received sympathetically. Manuscripts should be typed or clearly handwritten and submitted with rough drawings when appropriate. We are happy to advise potential authors on matters of style. Payment is negotiated on acceptance.

SUBSCRIPTION RATES

Annual UK subscription rate for STUDIO SOUND *is £3 (overseas £3.80, $8 or equivalent).*
Our associate publication Hi-Fi News *costs £3.24 (overseas £3.66, $8.64 or equivalent). Six monthly home subscriptions are £1.50* (STUDIO SOUND) *and £1.62* (Hi-Fi News).

STUDIO SOUND *is published on the 14th of the preceding month unless that date falls on a Sunday, when it appears on the Saturday.*

PAST ISSUES

A small number of certain past issues may still be purchased from Link House, price 31p each including postage.

Photostat copies of any STUDIO SOUND article are available at 25p including postage.

BINDERS

Loose-leaf binders for annual volumes of STUDIO SOUND *are available from Modern Bookbinders, Chadwick Street, Blackburn, Lancashire. Price is 85p. Please quote the volume number or date when ordering.*

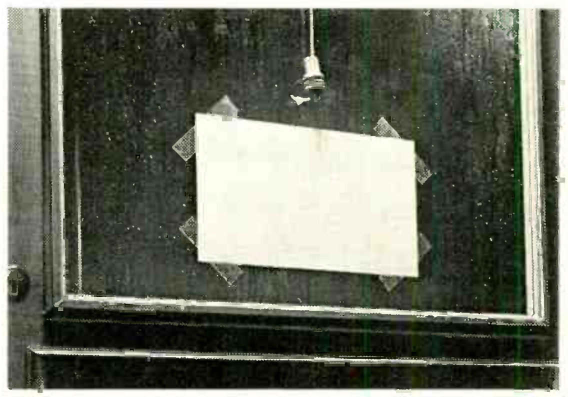
What you get when you use

Very Visible. Ordinary tape makes a very visible join and spoils the appearance of the item.

Not 'Waterproof'. If exposed to moisture ordinary tape tends to crinkle and peel off.

What you get when you use

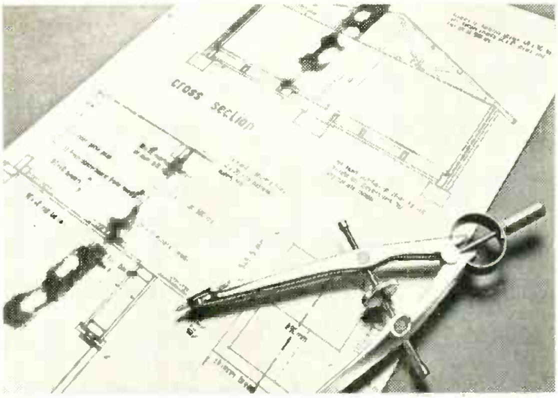
Almost Invisible. Magic Transparent Tape is much less obvious. It tones in with the background so the join becomes almost impossible to spot.



'Waterproof'. Magic Transparent Tape is unaffected by moisture. If it gets dirty you simply wipe it clean again with a damp cloth.

You notice the improvements.
Not the tape.

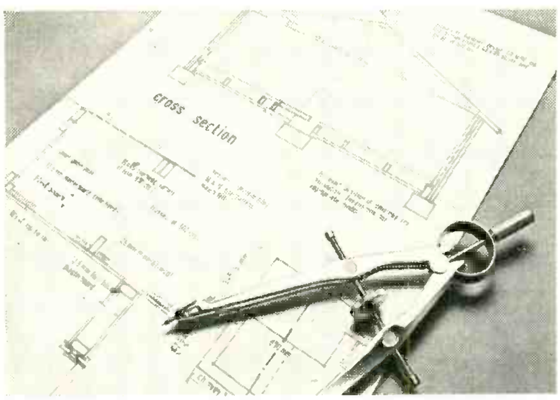
ordinary transparent adhesive tape

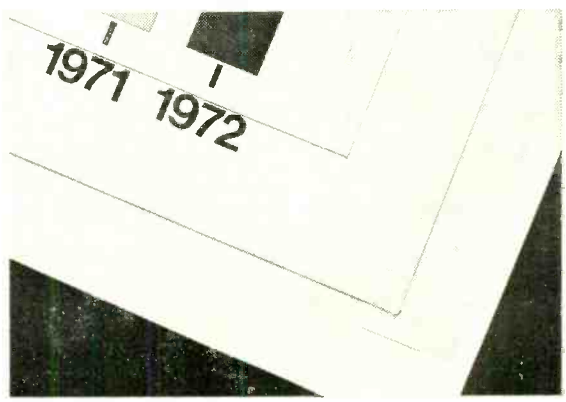


'Ghost' effect on copies. Ordinary tape gives a ghost effect on copies and photographs making the subject underneath difficult to read.

Shrinks and discolours. Ordinary tape shrinks and yellows with age leaving an unsightly black line round the edges.

Scotch Magic Transparent Tape





'Ghost free' copies. You get perfect copies with Magic Transparent Tape. The taped section is just as legible as the rest.

It's permanent—so it won't shrink or discolour. Magic Transparent Tape isn't affected by sunlight or temperature change. It stays just the way you stick it down no matter how long you leave it.

If you would like a free sample of Scotch Magic Transparent Tape and details of how you can get an attractive tape dispenser absolutely free, fill in this coupon and send it to: Commercial Trades Marketing, 3M United Kingdom Ltd., 3M House, Wigmore Street, London, W1A 1ET.

Name ....

Company ....

Position ....

Address ....

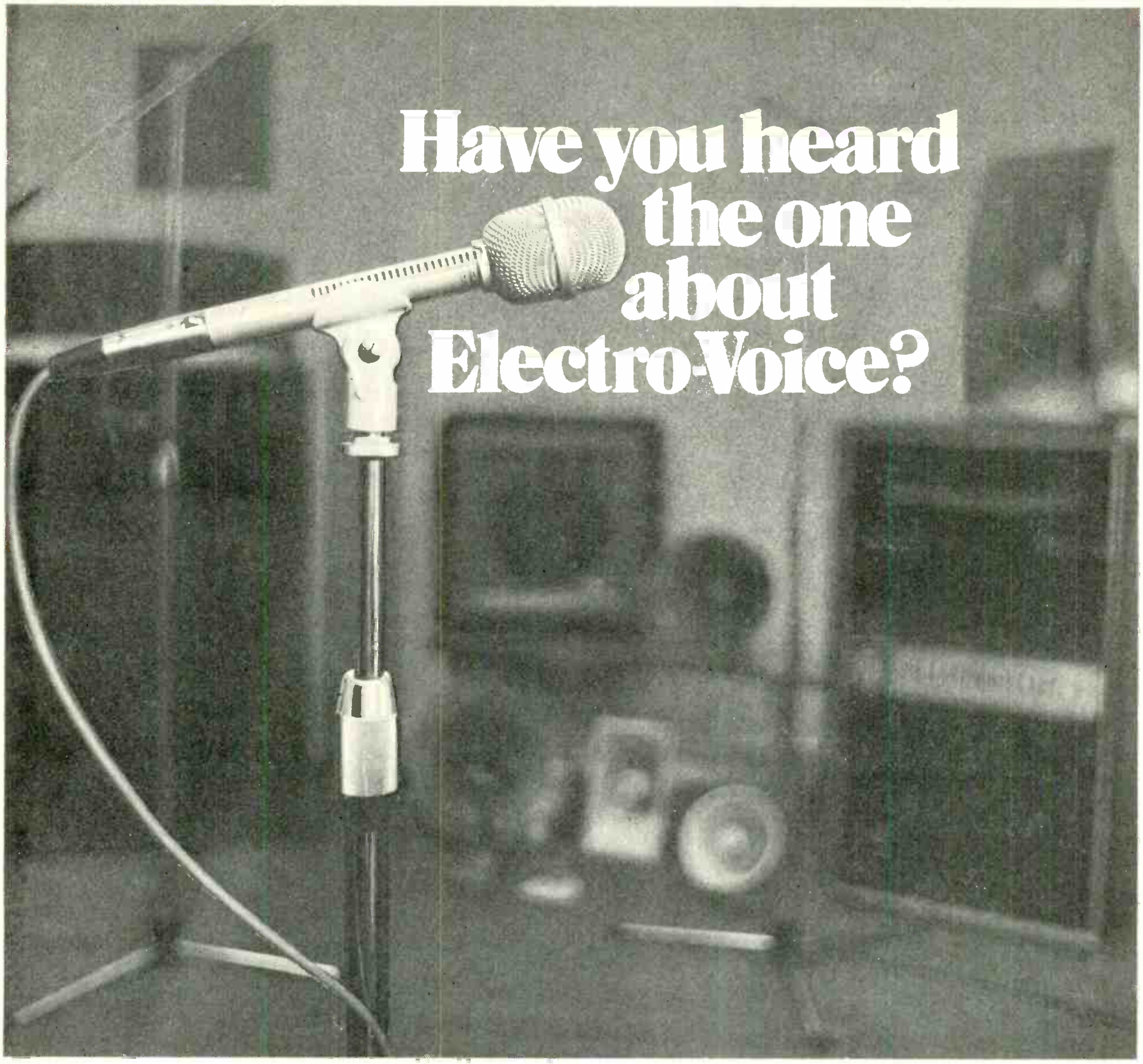
....

....

SS/10/72 3M2049

3M and Scotch are trademarks

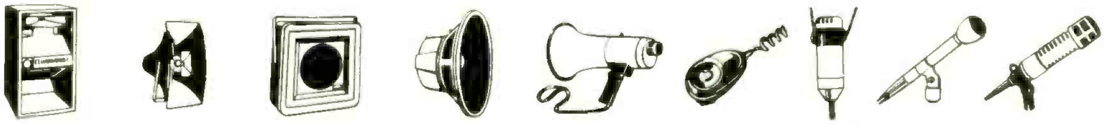
3M



It's more than likely that you have because most people in this business have heard of or experienced the world famous Electro-Voice range of products. So they don't need any introduction.

Gulton Europe are now manufacturing and marketing Electro-Voice broadcast, p.a. and recording microphones together with a range of sound reinforcement speakers.

This most important development means that Electro-Voice and their customers are already benefiting from Gulton's wide, technical and electronic experience.



As an expert you really do need a copy of our new product information brochure, so why not contact Gulton Europe for yours.
Your personal contact is Len Cole.

Gulton EUROPE

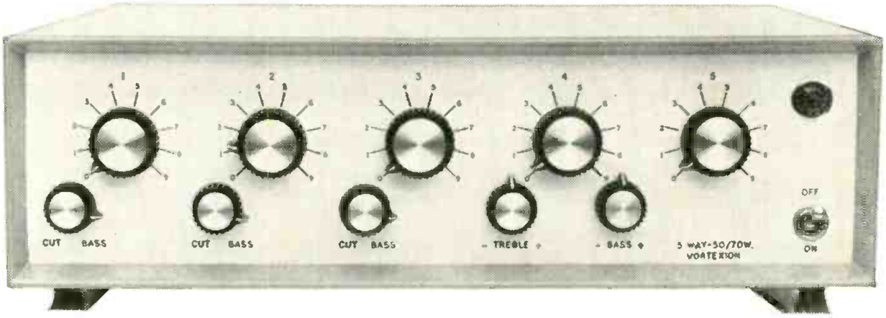
Gulton Europe Limited
ELECTRO-VOICE DIVISION
The Hyde, Brighton BN2 4JU
Telephone: 0273 66271 Telex 87172

Electro-Voice®
MICROPHONES AND PUBLIC ADDRESS EQUIPMENT

Vortexion

50/70 WATT ALL SILICON AMPLIFIER

WITH BUILT-IN 5-WAY MIXER USING F.E.T.s



THIS is a high fidelity amplifier with bass cut controls on each of the three low impedance balanced line microphone stages and a high impedance (1·5 meg.) gram stage with bass and treble controls, plus the usual line or tape input. All the input stages are protected against overload by back to back low self capacity diodes and all use F.E.T.s for low noise, low intermodulation distortion and freedom from radio breakthrough.
A voltage stabilised supply is used for the preamplifiers making it independent of mains supply fluctuations and another stabilised supply for the driver stages is arranged to cut off when the output is overloaded or over temperature. The output is 75% efficient and 100V balanced line or 8-16 ohms output are selected by means of a rear panel switch which has a locking plate indicating the output impedance selected. The mixer section has an additional emitter follower output for driving a slave amplifier, phones or tape recorder, output ·3V out on 600 ohms upwards.

50/70 WATT ALL SILICON AMPLIFIER WITH BUILT-IN 4-WAY MIXER

(0·3% intermodulation distortion) using the circuit of our 100% reliable 100 Watt Amplifier with its elaborate protection against short and overload, etc. To this is allied our latest development of F.E.T. Mixer Amplifier, again fully protected against overload and completely free from radio breakthrough. The mixer is arranged for 2-30/60Ω balanced line microphones, 1-HiZ gram input and 1-auxiliary input followed by bass and treble controls. 100 volt balanced line output or 5/15Ω and 100 volt line.

100 WATT ALL SILICON AMPLIFIER

A high quality amplifier with 8 ohms -15 ohms or 100 volt line output for A.C. Mains. Protection is given for short and open circuit output over driving and over temperature. Input 0·4V on 100K ohms.

THE 100 WATT MIXER AMPLIFIER

With specification as above is here combined with a 4 channel F.E.T. mixer, 2-30/60Ω balanced microphone inputs, 1-HiZ gram input and 1-auxiliary input with tone controls and mounted in a standard robust stove enamelled steel case. A stabilised voltage supply feeds the tone controls and pre-amps, compensating for a mains voltage drop of over 25% and the output transistor biasing compensates for a wide range of voltage and temperature.
Also available in rackpanel form.

20/30 WATT MIXER AMPLIFIER

High fidelity all silicon model with F.E.T. input stages to reduce intermodulation distortion to a fraction of normal transistor input circuits. The response is level 20 to 20,000 cps within 2dB and over 30 times damping factor. At 20 watts output there is less than 0·2% intermodulation even over the microphone stage at full gain with the treble and bass controls set level. Standard model 1-low mic. balanced input and HiZ gram. Outputs available 8/15 ohms OR 100 volt line.

CP50 AMPLIFIER

An all silicon transistor 50 watt amplifier for mains and 12 volt battery operation, charging its own battery and automatically going to battery if mains fail. Protected inputs, and overload and short circuit protected outputs for 8 ohms—15 ohms and 100 volt line. Bass and treble controls fitted. Models available with 1 gram. and 2 low mic. inputs, 1 gram. and 3 low mic. inputs or 4 low mic. inputs.

200 WATT AMPLIFIER

Can deliver its full audio power at any frequency in the range of 30 c/s—20 Kc/c±1 dB. Less than 0·2% distortion at 1 Kc/s. Can be used to drive mechanical devices for which power is over 120 watt on continuous sine wave. Input 1 mW 600 ohms. Output 100-120V or 200-240V. Additional matching transformers for other impedances are available.

F.E.T. MIXERS AND PPMs

Various types of mixers available. 3, 4, 6 and 8 channel with Peak Programme Meter. 4, 6, 8 and 10 Way Mixers. Twin 3, 4 and 5 channel Stereo, also twin 4 and 5 channel Stereo with 2 PPM's.

VORTEXION LIMITED **257-263 The Broadway, Wimbledon, SW19 1SF**

Telephone: 01-542 2814 and 01-542 6242/3/4 **Telegrams: "Vortexion, London S.W.19"**

News

Poly sound courses

THE POLYTECHNIC of North London are holding two courses on sound starting in October. One, called 'High quality sound reproduction', includes lectures by J. Linsley Hood, Ralph West, Roger Driscoll, Angus McKenzie and James Moir and the subjects covered include amplifier design, loudspeaker design, radio sound, equipment testing and stereo sound. There is a forum half way through the course, which begins on October 26, and the final lecture will be about Quadraphony, given by John Moseley. The fee is £6.30 for the course of 14 lectures.

The other course, which runs from October 26 until the examination in May, is called 'Sound studios and recording' and will include lectures on acoustics, recording techniques, control equipment, signal distribution, design and testing and power supplies. The fee for the course, which is of City and Guilds part three standard, is £10.50.

Stage sound

A COURSE in theatre sound starts on September 29 at the Cockpit Theatre. The series of classes will run every Friday evening until next July and will include doing the sound for a production at the Cockpit, which is in Gateforth Street, Marylebone, NW8. The course will be run by Mr Harry Jacobs. For further details ring Judy Gray at 01-262 7907.

Stock exchange visuals

A RECEPTION was held at Unit Five Design studios on August 1 to announce details of the design and production of the London Stock Exchange permanent exhibition. The new exchange will be opened in February of next year. The audio-visual part of the exhibition has been designed by Electrosonic. In the exhibition there will be six double sided display panels, a six section multiscreen, two television displays and, in the reception area, an electronic calculator display which will show the current FT ordinary share index.

The display panels have six earphones beneath them which can be fed independently with any of five languages. 1,500 slides on 24 projectors provide the multiscreen presentation. The projectors are synchronised with a stereophonic soundtrack with an English commentary; other languages can be heard on earphones. Of the two television sets, one is a colour receiver for all the broadcast channels, and the other displays news, from any or all of 22 channels of changes in stock market prices.

The contract for the design and content of the exhibition has been awarded to Unit Five Design, that for the production of the exhibition and reception areas to Pollards of London, that for supplying electronic control, projection and audio equipment to Electrosonic.

Specialised Mouldings will be supplying all the exhibition display units in glass reinforced plastic.

Millbank amplifiers

WE REGRET that Millbank were omitted from the power amplifier survey in our September issue owing to a wavering in the thin grey line that separates power amplifiers from integrated amplifiers. They make nine amplifiers: a 30, 50 and 100W model with a choice of one, three or five inputs for each rating. The single input version has two sensitivities, preset bass and treble lift and cut, master gain control and VU meter. The three and five input version has the above tone controls and a master gain control and can equalise any of the input to match the source.

Local radio postscript

JOHN CORDEAUX was surprised to read in our September issue that he had once worked for Radio Moscow. Mr Cordeaux, manager of Radio Humberside, informs us the nearest he has been to Radio Moscow was to be interviewed on it with his wife. We offer our apologies to Mr Cordeaux, and to Radio Moscow, for any inconvenience caused.

People

DAVID ANNETT has been appointed sales manager of Action Video, the closed circuit television company. Mr Annett formerly worked with Philips, where he was manager of the professional recording department in charge of UK sales of recording, instrumentation and communication equipment for five years. In his new job, he will be in charge of the development of equipment sales.

Alan Say, technical manager at Garrard, head of the team who designed the *Zero 100* turntable, retired at the end of July. Mr Say joined the Plessey company in 1936 and reached the position of technical director in 1963. He finally represented Garrard at the HiFi Show in Sydney, Australia.

Address change

CHADACRE ELECTRONICS have now moved to 63 Stratford Broadway, London E15, where their telephone number is 01-555 0411.

Film production and equipment hire company, Chippenham Films, have moved to Chippenham Studios, 192-198 Villiers Road, London NW2. Telephone: 01-794 3438.

Shure service changes

SHURE ELECTRONICS have moved their service department to 12 West Street, Bromley, Kent BR1 1RF. Their new telephone number is 01-464 7521. The department at Bromley is managed by Mr G. C. Ingle. Goods sent by post or rail should be addressed to Bromley but inspection, reception and collection will still be provided at 84 Blackfriars Road, London SE1 8HA. Telephone 01-928 3424.

New magnetic film laboratories

FILM FACILITIES (Magnetic) are opening new striping, treatment and environmental control laboratories for the treatment of 16 and 8 mm film. The laboratories, at 3 Springbridge Road Mews, Ealing Broadway, London W5 2AB, will be opened on October 2. Film Facilities were formed by Mr Elwyn Callaghan and Mr Tony Attwater and have just taken over the magnetic sound striping, ultrasonic cleaning and scratch removal and film treatment that were formerly carried out at Zonal House for Technicolor Film Services.

Crown king at Cambridge

CROWN POWER amplifiers were used for pa at the Cambridge folk festival at the end of July. A single *DC300* was used in the main marquee along with Crown 150W monitor cabinets. *DC300*s were also used elsewhere; Isaac Guillory and Moxy used one on the open air stage for their act on Saturday afternoon.

Fiarex 72

THE DUTCH electronics trade exhibition will have 105 exhibitors when it opens on September 25. The exhibition is the fifth 'Fiarex' and will be held at the RAI exhibition centre in Amsterdam. Among the exhibits will be components, semiconductors, ICs, test equipment, electro acoustic devices, intercoms, and aerial systems. The 1970 exhibition was attended by over 18,000.

English offices for AR

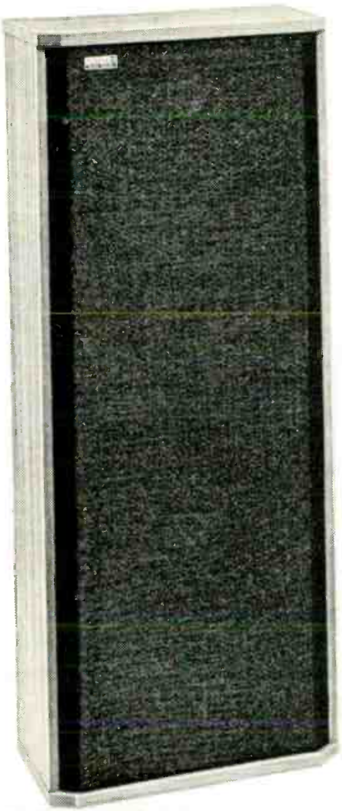
ACOUSTIC RESEARCH are opening offices, including assembly and test centres, in Britain. Buyers of AR speakers, in particular, will benefit from the resultant reduction in costs: the *AR3a*, for example, which sold at £178, will now be available for £100 before purchase tax. If purchase tax is taken into account, the savings will represent about 30 to 40 per cent of the old price. The new manager of the offices, which open on October 2, is Mike Glover, who can be reached at Acoustic Research International, High Street, Houghton Regis, Bedfordshire.

Celestion

Loudspeakers for the Perfectionist

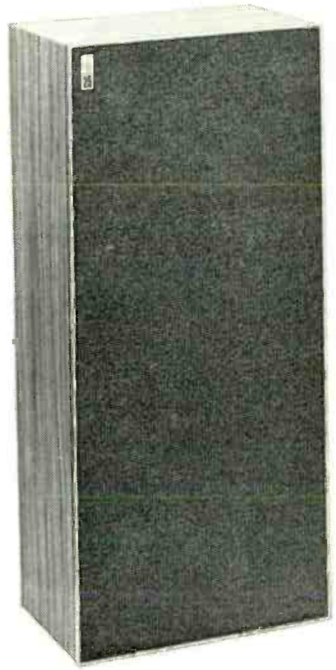
TELEFI

A remarkable innovation exclusive to Celestion for use in conjunction with Hi-Fi and Audio systems for providing high quality television sound reproduction from 625 line television receivers.
*No direct connection to the TV is required, the coupling being effected by an inductive pick-up.



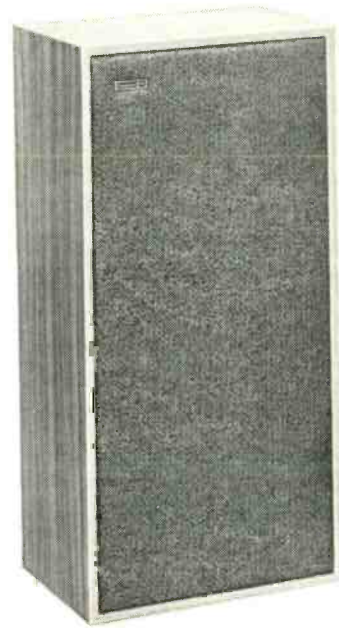
DITTON 66 STUDIO MONITOR

A new Loudspeaker of advanced design suitable for studio use and for home installations of the highest quality.
UNITS: HF 2000 (dome 'pressure' type) MF 500 (Mid-range Dome 'pressure' type) Ultra linear 12" bass driver and 12" A.B.R. The crossover has resulted from considerable research and crossover points are at 500 Hz and 5000 Hz 80 Watts Maximum, 4-8 ohm. This monitor loudspeaker system has an exceptionally wide and flat frequency response. Very low order harmonic and inter-modulation distortion. Precise response to transients. Beautifully maintained polar response ensures absence of unwanted directional effects and provides a highly satisfactory stereo image throughout the listening area. Matched pairs.
SIZE 40 × 15 × 11½ Natural Teak or Walnut Cabinet



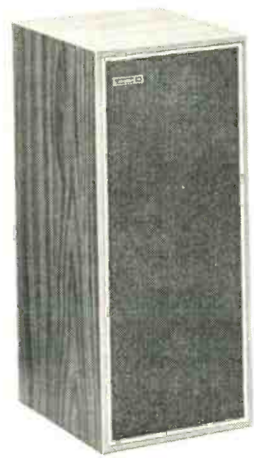
DITTON 25

Recommended for luxury Domestic Hi-Fi Installations. A system having extremely low harmonic distortion and high sensitivity. This well established loudspeaker is excellent value and will delight the most fastidious audiophile. Noted by reviewers for complete absence of 'listener fatigue'; smooth and effortless in performance.
UNITS: HF 2000, HF 1300 MK II (two), long throw 12" bass driver plus 12" A.B.R. (auxiliary bass radiator). Substantially level response from 25 Hz to 30 kHz 60 watts maximum 4-8 ohms. Matched pairs.
SIZE 32 × 14 × 11 Natural Teak or Walnut Cabinet



DITTON 44 MONITOR

Designed for the discriminating listener . . . wide, smooth frequency response. Exceptional transient performance, superb controlled bass, accurate mid-range—and smooth extended highs.
UNITS: HF 2000, MF 6 and Ultra linear 12" long throw bass speaker. Crossover system of superior design at 500 Hz and 5000 Hz. Substantially level response from 30 Hz to 30 kHz 44 watts maximum—4-8 ohms.
SIZE 30 × 14½ × 10 Matched pairs
Natural Teak or Walnut Cabinet



DITTON 15

The world famous high performance 'bookshelf' loudspeaker 30 watts maximum. 4-8 ohms.
UNITS: HF 1300 MK II, heavy duty 8" long throw bass speaker plus A.B.R. (auxiliary bass radiator). Substantially level response 35 Hz to 15 kHz.
SIZE 21 × 9½ × 9¼
Matched pairs
This truly remarkable loudspeaker never fails to impress.
Natural Teak of Walnut Cabinet

'COUNTY'

Designed for budget Hi-Fi systems but without sacrificing quality.
UNITS: HF 1300 MK II and special 8" long throw speaker. Substantially level response from 45 Hz to 15 kHz, 25 watts.
SIZE 19 × 10 × 9½
Natural Teak or Walnut Cabinet. Matched pairs

DITTON 10 Mk II

True Hi-Fi Sound from a tiny precision speaker 20 watts, 4-8 ohms.
UNITS: HF 1300 MK II and heavy duty 6" long throw bass speaker. Substantially level response from 45 Hz to 15 kHz.
SIZE 12¾ × 6¾ × 8½
Matched pairs
Natural Teak or Walnut Cabinet

Write for details of Celestion sound equipment. **ROLA CELESTION LTD.** DITTON WORKS, FOXHALL ROAD, IPSWICH, SUFFOLK IP3 8JP

Patents

THE FOLLOWING list of complete Specifications Accepted is quoted from the July issues of the Official Journal (Patents). Copies of specifications may be purchased at 25p each from The Patent Office, Orpington, Kent BR5 3RD.

1972, July 5

1285210
Westinghouse Electric Corporation
Power supply for camera system including image intensifier

1285221
Post Office TV
System with PCM encoding

1285222
Post Office PCM-TV
System with bandwidth compression

1285308
Ampex Corporation
Tape player having an automatic tracking control circuit

1285315
Licentia Patent-Verwaltungs-GmbH
Signal combiners and multiple receiver installations incorporating the same

1285321
Standard Telephones & Cables Ltd
Transmission line circuits

1285446
Plessey Co Ltd
Record-changer gramophones

1285469
Tektronix Inc
Graphic display system

1285475
Nippon Victor KK
Signal processing e.g. for magnetic recording and reproducing apparatus

1285542
Davies, G B
Stringed musical instrument

1285554
National Research Development Corporation
Display means

1285555
Plessey Co Ltd
Gramophone pickup cartridges

1285592
Marconi Co Ltd
Television cameras

1285611
Dual GEB Steidinger
Photoelectric stereophonic pickup

1285715
Thompson Ltd, John
Ultrasonic testing equipment

1285720
Post Office
Digital communication systems

1285745
International Business Machines Corporation
Magnetic recording system

1285759
Minnesota Mining & Mfg Co
Method of producing a coloured film from a black-and-white picture

1285837
Picker Corporation
Image reproduction systems

1285921
Telefonaktiebolaget L M Ericsson
Picture telephone arrangement

1285976
Western Electric Co Inc
Diversity switching for digital transmission

1285999
Johnson, P R Melchior, K W and Hanson P L
Recorder and reproducer system

1972, July 12

1286128
Matsushita Electric Industrial Co Ltd
Signal processing arrangement for an electronic musical instrument

1286157
Kokusai Denshin Denwa Co Ltd and Fujitsu Ltd
Time division multiplex communication

1286166
Commissariat A L'Energie Atomique
Apparatus for sampling random pulses

1286311
Marconi Co Ltd
Aerial systems

1286313
Wheeler, S E
Thermoplastic film or tape jointer

1286393
Cossor Ltd A C
Apparatus for displaying and editing information

1286454
Cambridge Scientific Instruments Ltd and Oatley, C W
Surface potential analysis by electron beams

1286470
Cartridge Television Inc
Tape cartridge and apparatus therefor

1286487
Westinghouse Electric Corporation
Speech unscrambler

1286546
Licentia Patent-Verwaltungs GmbH
Signal recording and reproducing apparatus

1286564
Borg-Warner Corporation
Apparatus for cleaning phonograph cartridge needle

1286635
Eastman Kodak Co
Anchoring member for reels

1286687
Manger, J W
Electro-dynamic transducers

1286743
Mitsubishi Denki KK
Product/quotient deriving system

1286757
Raytheon Co
Predetection signal processing system

1286762
Fernseh GmbH
Colour television apparatus

1286766
Grundig EMV
Apparatus for the transmission of information by television

1286787
Fernseh GmbH
Beam-splitting prism system

1286790
Sumlock Anita Electronics Ltd
Magazines for tape

1286806
EMI Ltd
Video mixers

1286827
Sony Corporation
Signal supplying circuit for a colour picture tube

1286828
RCA Corporation
Turn-off method and circuit for liquid crystal display element

1972, July 19

1286912
Compagnie Generale D'Electricite
Electro-optical cell

1286981
Colorado Video Inc
Television systems

1287030
Minnesota Mining and Mfg Co
System for controlling film motion

1287064
International Business Machines Corporation
Document scanning machine

1287169
Koishikawa, Y
Tambourine

1287195
RCA Corporation
Test signal generator

1287203
Solartron Electronic Group Ltd
Oscilloscope time base circuits

1287205
Nippon Electric Co Ltd
Solid state image scanners

1287375
Philips Electronic & Associated Industries Ltd
Device for transmission of information at varying frequency

1287407
EMI Ltd
Optical systems for colour television cameras

1287434
Philips Electronic & Associated Industries Ltd
Image transmission system

1287460
RCA Corporation
Optical system for a single tube colour television camera

1287480
Soc Italiana Telecomunicazioni Siemens SPA
Carrier frequency generator

1287535
Philips Electronic & Associated Industries Ltd
Diode mixer circuits

1287552
Sonoptics Corporation
Ultrasonic vibratory cleaning device

1972, July 26

1287767
Philips Electronic & Associated Industries Ltd
Optical relay device particularly for use in television

1287799
International Business Machines Corporation
Frequency detection circuit

1287801
Graphic Sciences Inc
Graphic data transmission system

1287804
Eastman Kodak Co
Cinematographic apparatus including mechanism for removing leading end of film roll from a spool

1287864
International Business Machines Corporation
Magneto-optic transducers

1287889
British Broadcasting Corporation
Telecine machines

1287914
RCA Corporation
Electronic colour tonal separation scanner

1287923
Telefunken Patent-Verwertungs GmbH
Mixing stages

1287935
Hiller, W
Sound reproducing devices

1288046
RCA Corporation
Tape transport mechanism

1288047
Minnesota Mining & Mfg Co
Arrhythmia recording and control system

1288050
Eastman Kodak Co
Cinematographic apparatus

1288092
Muirhead Ltd
Facsimile apparatus

1288144
Fuji Photo Film Co Ltd
Motion picture projector

Take a QUAD 50E Amplifier (a good start for any installation)

plug it into your monitor system and it bridges 600Ω lines to drive your speakers.
Take that same amplifier and, without changing it in any way, plug it into another installation to deliver 50 watts into 100 volt line * from a 0.5 volt unbalanced source. This versatility and its attendant easing of stocking and maintenance problems is one reason why large organisations use the Quad 50E.

**or indeed any other impedance from 5 to 250 ohms.*

See and hear QUAD on Stand No 39 International Audio Festival & Fair, Olympia, 23rd-28th October 1972

Other advantages appropriate to users of all sizes include:

Excellent power and frequency response (—1dB).
Low distortion (0.1% at 1kHz at all power levels).
Low background (better than 83 dB referred to full output).
Pre-set level control adjustable from front panel.
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How Many Microphones?

By W. A. Wettler (Sonographic Recording Company, Switzerland)

THERE is more to Switzerland than tourists, clocks and cheeses: we even have recording studios. Our own company, Sonographic Recording, has a worldwide reputation in the field of classical music recording. In one particular branch of recording, demands have been increasing for quite a while: the recording of soundtracks for music films for eventual distribution in form of video cassettes and similar media.

For film sound recording we use six channel Killi 35 mm machines and sprocketed magnetic tape. A Studer four channel 12.5 mm tape recorder operates in parallel to this. The Studer recordings are used for monitoring and experimental editing, the 35 mm machine being too cumbersome for these purposes.

The 35 mm recordings are made on six channels, to permit more elaborate sound manipulation later on. Some of these films are broadcast on television but unfortunately only with a mono soundtrack. To achieve a performance as authentic as possible, picture and sound are recorded at the same time.

Three to five cameras work together, keeping strictly to a preconceived plan. The cameras are guided through headphones from the directing room. They use normal negative colour film but are additionally equipped with a television camera for the director's monitors.

The cameras are started and stopped from the directing room. Specially trained personnel using marked scores give instructions to the camera operators about location and focusing. A camera is brought into operation a few bars before its picture is actually needed. This technique offers us the possibility of recording whole movements in one piece, much to the benefit of the quality of music and of sound recording.

We work with a great number of microphones which gives us the problem of concealing them. In overall pictures this cannot always be managed but in close-ups the microphones are always outside the visible field, or placed in such a way that they are hidden behind musicians.

The directors are usually understanding enough to see that we must find a compromise between placing cameras and microphones. In this respect it might be mentioned that we always use small and unobtrusive microphones —Schoeps and Neumann. For special cases we make use of tightly bunched unidirectional microphones. Of course, these so-called gun mics, like the Sennheiser *805*, must be used with care but very often they represent the best solution.

All these recordings are made in visually and acoustically suitable rooms. The architectural style, whenever possible, corresponds with the type of recorded music. Thus baroque music is recorded in a baroque church, classical music in a concert hall, opera in an opera house, and so on.

An exception to this that I would like to mention was the recording of the Bach *St. Matthew Passion*. The film director, Hugo Käch, was looking for a new rather more abstract approach to the making of music-films, an approach that, hopefully, opens up interesting new prospects for the creation of such films in the future. However, it would be digressing too far from my present purpose to give details of this aspect of the work fascinating though it may be. I can talk here only about my particular part, the sound recording.

To allow the director's ideas to be realised, the recordings had to be carried out in the largest film-studio of the Bavaria Film Company in Munich. The room measured some 120 by 60m with a height of about 20m. The walls were soundproofed and the acoustics were thus more akin to an open air theatre than a concert hall. The problems of recording were considerable. Up to 20 microphones, two EMT plates and a reverberation chamber were brought into action to give an adequate acoustic background to the music.

Difficult conditions

Under these difficult acoustic conditions, it was impossible to put the microphones far enough away from the sound sources. We hardly ever put microphones on stands or booms since these disturb the picture too much; they also restrict the freedom of movement of camera operators and cast unwanted shadows. Most microphones are suspended on nylon-strings. Many of these microphones, which are placed with some difficulties, are only in use for a few bars, just to give the necessary profile to a certain passage or to a particular instrument.

It was comparatively easy for us, with the means at our disposal, but far more difficult for the actual performers. However, under the guidance of Professor Karl Richter, who mastered the tricky and unusual acoustics in a masterly way, a performance was achieved which, I believe, will satisfy the most discriminating tastes. Certainly the performers taking part were first class. Only with artists of this calibre can such a delicate task be accomplished. The Munich Bach Choir, the Munich Bach Orchestra, Peter Schreier as the evangelist, Helen Donath (soprano), Julia Hamari (alto), Walter Berry (bass), and many other excellent soloists, mastered the situation superbly.

The whole *Passion*, 3½ hours, was recorded within ten days. For readers who are not familiar with this work, it might be added that the total 'cast' consists of about 200 singers and musicians. These were placed on a stage rising up to a height of 6m at the back, on a 30m square base. Two positive organs were placed left and right of the orchestra, each with its own microphone (Schoeps hypercardioid *CMT 341*)—suspended above the organs at a height of about 3m.

The soloists stood on platforms in the space between the orchestra and the choir, which in turn was placed on a higher level. Peter Schreier stood in front of one of the organs. He had particular problems in that he could not see the conductor while singing since his camera had to be placed outside the orchestra. He had to sing turned away from the orchestra for the camera to encompass him at all. Professor Richter accompanied the recitatives himself on the harpsichord. Of course, this kind of situation often occurs in a live performance but with one big difference in our case: the tenor stood about 10m away from the harpsichord which he could no longer hear when he started singing. It says much for Peter Schreier that he dealt with this difficulty superlatively well, never showing uncertainty of intonation. Professor Richter for his part could hear the soloist though he could not see him.

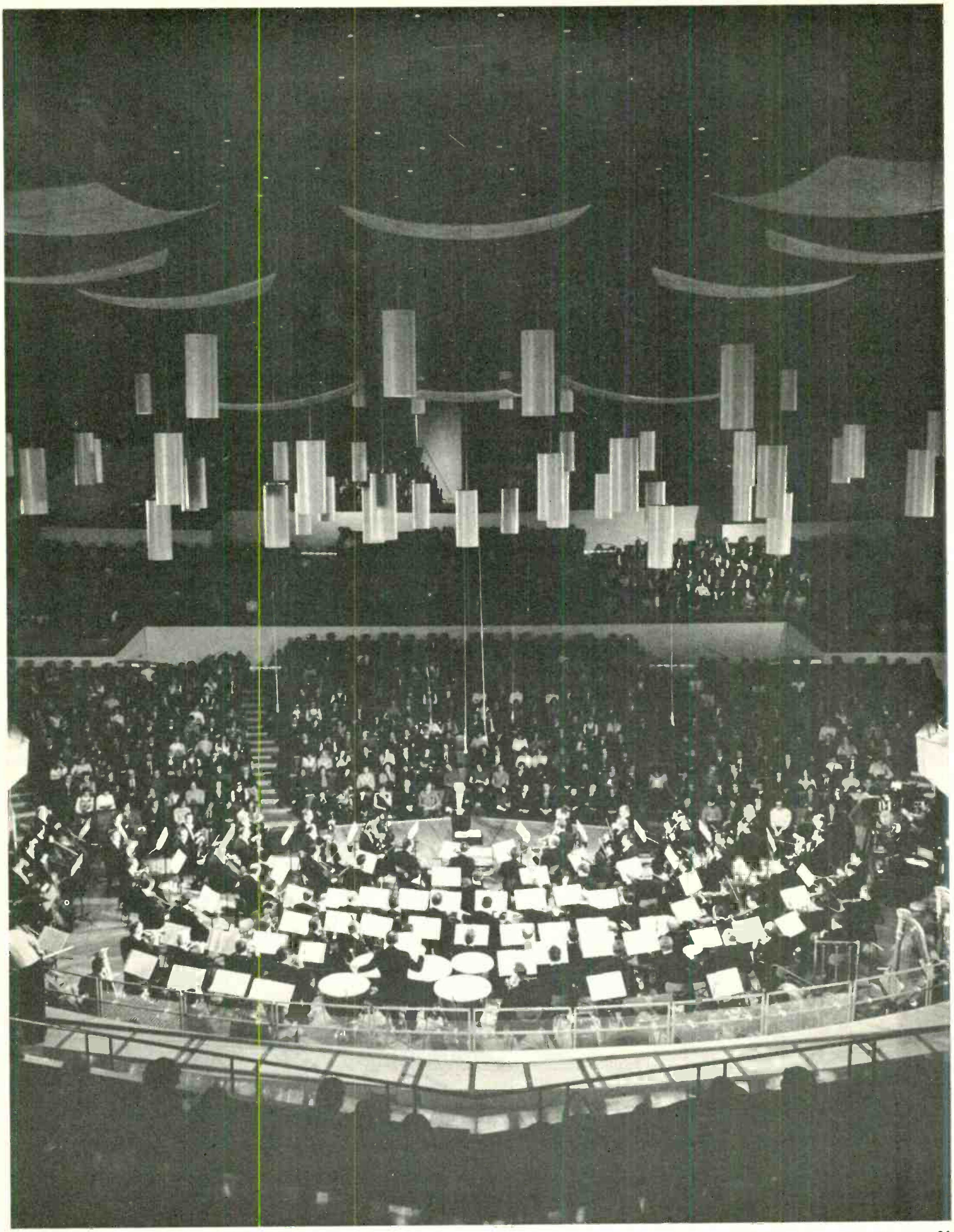
For the male soloists, we used Sennheiser *805* at a distance of 3 to 4m; for the women Schoeps *CMT 341*, again at a distance of 3 to 4 metres. In the case of the harpsichord we use (for film-recordings) a special Schoeps double microphone which can be placed underneath the instrument and which produces we find good quality sound. The cello also had a microphone of its own, a Neumann *KM53* embedded in a block of foam-rubber which was fixed to the soloist's music stand. These microphones were only brought into operation when necessary.

The number of artists actually performing at one time can vary from, say, three up to the entire 200, with a number of variations in the arias. Nevertheless we recorded large sections of the music at one go, six or seven pieces, so that the musical flow would not be interrupted. The chorals were sung and played at the original place but they were filmed in playback with varied lighting effects.

One must often compromise when recording sound for films. We are never able to achieve the quality of sound which has earned us our reputation for gramophone records. This is

continued 23

Leonard Bernstein conducting the Vienna Philharmonic Orchestra (Mahler *Ninth Symphony*) at the Berlin Phylharmonie.



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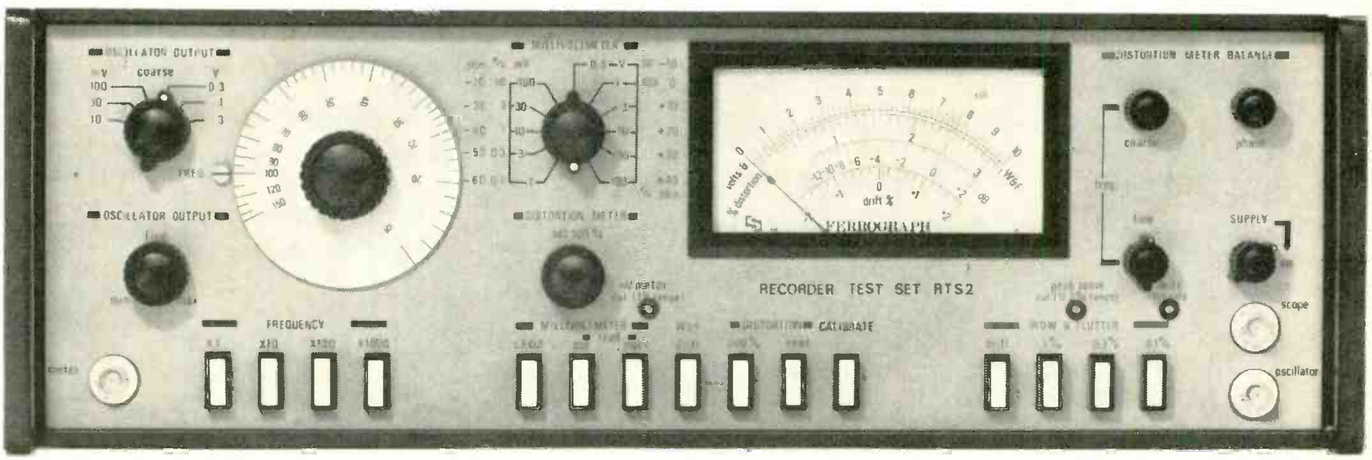

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unavoidable since the placing of soloists is done according to visual criteria and we simply make as light of this difficulty as we can. However, with goodwill and co-operation, acceptable qualities can always be achieved.

Why do we not work to playback throughout? This is not really helpful since, for obvious reasons, the sound recordings have then to be made under severe time restrictions and the results are in our experience largely unsatisfactory.

To tell the truth, I had arrived at the point where I intended to give this work up entirely and again devote myself exclusively to gramophone recordings when certain film-makers realised that the results obtained from the playback approach were not to their liking either. In many cases the soloists would make use of the playback situation to put on their best 'Sunday faces', be it said sympathetically, not always the most expressive aspects of themselves and certainly, all too often, far removed from the spirit of the music and the emotions one might reasonably expect them to record while performing it. So we decided to record pictures and soundtracks simultaneously. This secures both a good musical performance and suitable visual material.

For all concerned, this became an exciting venture which often stimulated the artists and engineers to very high levels of achievement. Of course this problem would be easy enough to solve, at least in so far as the sound is concerned, if one made use of coincident microphone technique. And with this I have come to my essential point—that problems of the kind which I have described, the recording of such a complex work in an acoustically impossible room, can in my opinion only be solved with multimic techniques.

Early experiments

For months now I have read with the deepest interest statements from both camps in the controversy about multimic and coincident recording techniques. Much of what I read often gives cause for a quiet smile. In all humility, I was probably among the first people in Europe to make experimental stereo recordings—as early as 1953. Since no stereo mixing desk was available, and I was still fully occupied with mono-recordings, stereo recordings were made at the same time in many variations of two microphone techniques. We found that pure coincident technique gave good (even excellent) results but only for headphones, especially when we had small formations which produced a good sound balance within themselves.

Even today I still make recordings with only two microphones when the room is acoustically suitable and when we record say a good string quartet or a small capella choir. This is done with an a/b setting since I believe that, with a good setting of this kind, better results for loudspeakers can be achieved. If we have a more complex musical event, or if the acoustic conditions are not good, then multimic technique is used, for in such condition this technique alone gives good results or, as we might say, a recording corresponding to the score.

Utopian theories

As can easily be seen from the various articles and contributions from readers, opinions on this question are diametrically opposed. In general I share the opinions of Robert Auger, whom regrettably I do not know personally. On the other hand, I have the pleasure to know Jerry Bruck and I must say that his ideas about microphone technique amused me somewhat. The theories he propounded were utopian to say the least and were so, I feel, because Jerry has not been through the hard school that the practical recording engineer must endure.

Finally, in respect of multimic technique I should like to introduce another point which has not yet been given the attention it deserves: the dynamics (i.e. the difference between a pianissimo and a fortissimo must not noticeably exceed 30 dB. Through multimic working a skilled sound engineer has the possibility of keeping the difference in dynamics within a 'frame of tonal reference' without making use of possibly idiosyncratic means like volume controls which ruin the whole sound picture. I had the honour of recording Mahler's *Ninth Symphony* with Leonard Bernstein and the Vienna Philharmonic Orchestra. Mr Bernstein is notably fond of extreme dynamic contrasts and I suppose the dynamics of the orchestral sound moved somewhere in the range of 60 dB.

Now in recording this work, let us assume that I had followed Michael Gerzon's advice and simply placed a pair of coincident microphones in the most suitable positions. I would have had no alternative than to have changed the volume control of the two microphones according to the dynamics of the sound at a rate of around 30 dB to reduce the original dynamics of around 60 dB to the technically possible rate of a little more than 30 dB. (I have made recordings for about 20 years now and know these problems very thoroughly. I maintain that this problem would be insoluble even under better acoustical conditions.)

Emil Gilels, the Russian pianist (right), in discussion with the author at a church in Ossiach, Austria.

Remixing for the mono television version, I would have had to reduce the dynamics by a further 10 dB since mono tv sets can hardly operate with higher dynamics. These reductions of volume are much more disturbing than a good multimic recording because the sound balance in relation to the acoustics of the room changes tremendously.

If we had used coincident microphones placed at quite some distance from the orchestra, which would have had to pick up the sound of a string soloist, the results would have been quite disastrous. The unavoidable noises from the orchestra would have ruined the whole atmosphere, particularly at the end of the fourth movement.

There is one thing about which we really must have no illusions: to record an orchestra 120 strong, coincident mics would have to be placed at a distance of at least 10m from the sound source.

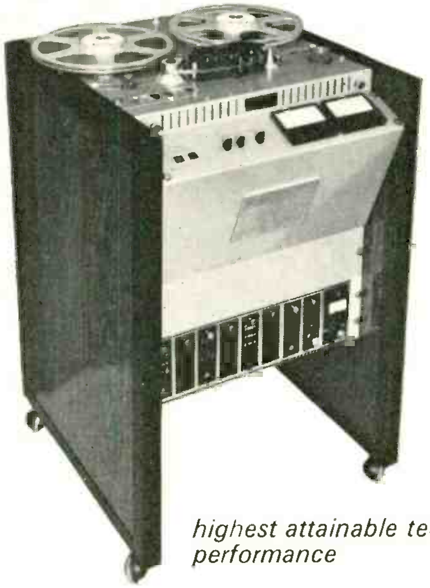
Know your score

I respect absolutely the arguments of the opponents of multimic technique for it is clear that a bad sound engineer can crudely distort, or even ruin completely, the intentions of musicians and conductors. Nevertheless it cannot be denied that the best gramophone recordings of large orchestras have all been made with the multimic technique. The important thing is that the sound engineer should know the full score of a given work as thoroughly as possible. Understanding its needs, he is thus in a position to know how best he can serve the intentions of the actual interpreters of the work. This is really the salient point.

Insufficient or even nonexistent musical education in engineers, very often with otherwise excellent technical education, is in my opinion the main reason for bad recordings. Again, many sound engineers do not have a sufficiently developed intuitive capacity to be of real service to the interpreters of the music. In my experience, really good recordings can only be achieved when there is considerable collaboration between the musicians and the technical realiser.

It follows from this that many famous artists have 'their' sound engineers who make all their important recordings, guaranteeing thereby a certain stability and consistency of quality. I am extremely lucky, and indeed honoured, to belong to this group of realisers—those whose skills are actually requested by a particular performer. This is due largely to a well applied multimic technique. It is true to say that not one of the conductors for whom I have recorded would be satisfied with a two-microphone recording. On the contrary, for a long time now the modern techniques of recording and the particular skills of sound engineers have been consciously used by great conductors and other interpretative artists to serve their interpretations. Be it said that performers of the kind to whom I refer usually have very exact ideas concerning such matters as sound balance and the profiling of single instruments.

In the light of the foregoing it would seem that a return to the one-point recording technique is not really a matter for further discussion.



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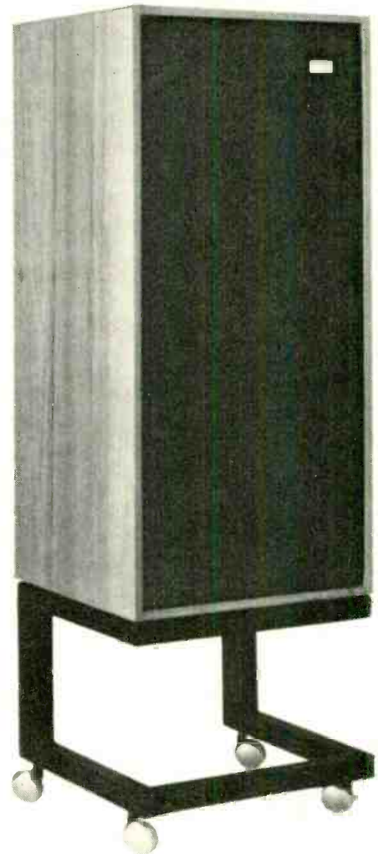

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Interference

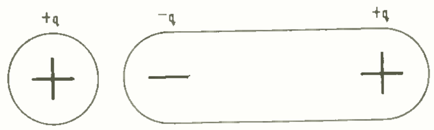
By Trevor Attewell

THE intrusion of interference, particularly at radio frequencies, is an ever-increasing problem which can become acute where low level signals have to be conveyed over long distances. Queries received from readers from time to time indicate that there is a good deal of uncertainty about the theory and practice of screening, and this article is offered as a simplified guide to a subject which can appear alarmingly complex—and often diffuse—in standard text-book form. Experts should be forewarned that some formal complications will be swept gently under the carpet, but only, it is hoped, where they will not cause anyone to trip up.

The first essential is to consider the nature of interference, other than that conducted through incoming cables, against which no amount of screening is likely to do any good. We start from a simple, fixed electrostatic charge, as obtained, for example, by stroking the cat with a plastic rod (no, she won't understand it, either). Associated with this charge, there is an *electrostatic field*. We need not worry about the physical nature, if any, of this field, but may merely consider it to be the region around the charge within which it is capable of exerting any detectable influence on other bodies. In particular, if a conductor is brought into the field, two charges of equal size but opposite sign appear on it. One of the charges will be found on that part of the conductor nearest the original charge, to which it is opposite in sign, while the other will be found at the farthest part of the conductor (**fig. 1**). The exact way in which this happens is slightly more involved than some would have us believe. Basically, it comes about because some of the electrons in the outer regions of the atoms of a conductor are readily removable under the influence of an external field. An atom which has lost one or more electrons is no longer electrically neutral but has a net positive charge. The atoms in a solid are unable to move appreciably from their mean positions but the electrons can move freely in a conductor. It is the movement of these electrons that gives rise to the observed charge distribution.

We note that an electrostatic field has a strength, measurable by its effects, which is proportional to the charge giving rise to the field and inversely proportional to the square of the distance from the charge to the point of measurement, assuming that this distance is great enough to allow the charge to be considered as a point source. In theory, some trace of the field should exist at all distances out to infinity. In most practical cases, however, the field is negligible at a metre or two away from the source. While such electrostatic fields may have their own nuisance value, for example in pulling dust on to gramophone records, they do not produce what we normally think of as electrical interference.

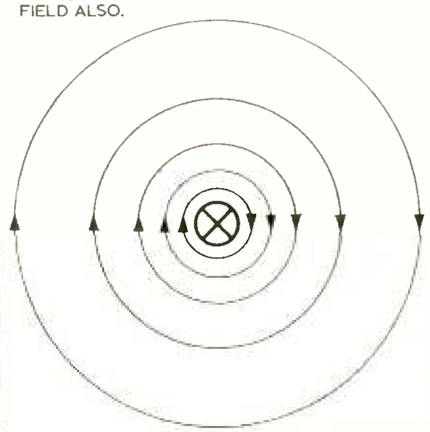
FIG. 1



If we now make the charge move or vary its strength, however, then the field associated with it will vary in sympathy. This, in turn, will cause corresponding changes in the sizes of the charges in neighbouring conductors. This is only another way of saying that there will be a varying potential difference between the charges in any such conductor. If the conductor happens to form a part of the signal path of, let us say, an amplifier, then this varying potential difference is an interfering signal which the amplifier has no means of distinguishing from a genuine one. This is a case of electrostatic interference (strictly we ought not to use the suffix 'static' which implies an unvarying field but the term is widely used) and we note that such cases arise when conductors are near any point of varying electrical potential.

By continuously producing electric charge and allowing it to flow along a conductor, we have what is better known as an electric current. An important effect associated with electric currents is the magnetic field which always accompanies them. A steady direct current gives a stationary magnetic field, shown for the case of a cross-section through a straight piece of wire, in **fig. 2.** The field is indicated by the concentric circles, which simply join those points at which the field strength is equal, just as contours on a map join points of equal height above sea-level. The field completely surrounds the wire at all points along its length and the diagram only indicates the form of the field at one such point. The arrowheads give the field direction which is arbitrarily taken to be the direction in which the N-pole of a magnet would move if it were placed in the field, the N-pole being the end which points north when the magnet is freely suspended in the earth's magnetic field. Again, we need not concern ourselves overmuch with the exact meaning of the term 'field' but may think of it as the region around the conductor in which the magnetic effect of the current can be detected.

FIG. 2 MAGNETIC FIELD AROUND A CROSS-SECTION OF A WIRE, CARRYING CURRENT AWAY FROM THE OBSERVER. REVERSING THE CURRENT DIRECTION REVERSES THAT OF THE FIELD ALSO.



The fixed field associated with a steady (direct) current, or with a stationary magnet, does not give rise to any electrical effect in nearby conductors, though it may present specific hazards such as ruining master tapes or saturating the cores of small transformers. However, if the field is changing, an electromotive force, of magnitude proportional to the rate of change of the field, is set up in adjacent conductors. A changing field results from a change in current and the everyday example of a constantly changing current is, of course, the ordinary alternating power supply.

It follows that a varying flow of charge can give rise to interfering emf in nearby conductors, both by electrostatic and electromagnetic means. If you wish to try both at once, arrange a recording session directly underneath a supergrid overhead power line.

The fields discussed so far are known as *induction fields*, because they *induce* effects in neighbouring bodies. Where electrostatic and magnetic fields coexist, there may be no easily definable relationship between their strengths. The ratio can, and more often than not does, vary widely from place to place in the vicinity.

In the case of currents varying at frequencies above the audio range, another type of field can appear. This is the *radiation field*, produced when electromagnetic energy 'escapes' from a conductor. Sometimes the escape is intentional, the conductor in question being specifically designed as a transmitting aerial. A radiation field consists of two components, both alternating at the same frequency as the current in the aerial, one being a magnetic and the other an electric field. These two fields occupy planes at right angles to one another and also to the direction of propagation in which the composite wave travels outwards from the source (**fig. 3**). The energy being radiated is shared equally between the two components, which

continued 27

are always in phase with one another—they are mutually sustaining, and one cannot exist independently of the other. The amplitude of both is proportional to the distance from the source, for distances large compared with the source dimensions.

Near an aerial there exist both induction fields (due to the current in the aerial) and radiation fields but the former become negligible a few wavelengths away, at most, because their amplitude follows the inverse-square law already mentioned. Hence only the radiation field is significant at greater distances.

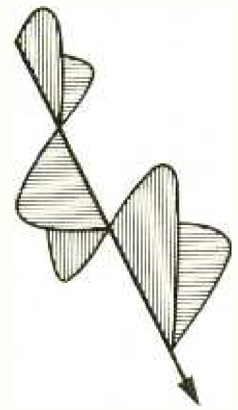
Like induction fields, radiation fields give rise to emf in any conductor placed in their path. The voltage distribution so produced can be calculated for a conductor oriented at a known angle to the direction of propagation, placed at a known height above ground of specified conductivity, conforming to an analytically amenable shape, and terminated in a defined impedance. On the other hand, the voltage which might appear at the business end of *n* metres of unbranded microphone cable, looped drunkenly round the walls of a steel-framed town hall and terminated with something out of the junk box, is anybody's guess. In practice, we must accept what we actually get and concentrate on trying to reduce the ill effects as much as possible.

Fields of both types are attenuated if the affected apparatus is screened (i.e. enclosed in a container). There are two basic types of screening material—magnetic materials of good permeability, and non-magnetic materials which are good conductors. Faraday showed that conductors make excellent screens against electrostatic fields. He proved that there is no electrostatic field inside a closed hollow conductor (as long as no charged body is actually put inside, but not touching it) regardless of the size of any external charges near, or even on, the conductor. An enclosure of this kind is known as a Faraday cage, and its applications in high-voltage engineering are well-known; when all around are being struck by lightning, stay in your car!

It follows that a conducting enclosure offers complete protection against any electrostatic field and there will be no purely electrostatic field inside, even if the external field is varying. But we must not lose sight of the fact that a changing electric field will cause current to flow in any conductor near it, including a metal screen, and the current will have a magnetic field associated with it which might be able to penetrate a non-magnetic screen. Clearly, we have to consider how currents flow in screens and how screens affect the passage of magnetic fields.

Surprising as it seems at first sight, a non-magnetic, conducting screen can also be effective against changing magnetic fields. What happens is that the magnetic field induces an emf in the screen and the emf causes circulating currents in the conducting material of which the screen is made. Now these currents also have associated magnetic fields and these are always in such a direction as to oppose the original field which gave rise to them—this opposition of effect and cause is embodied in

FIG. 3 ELECTROMAGNETIC FIELD, SHOWING DIRECTION OF PROPOGATION, WITH ELECTRIC AND MAGNETIC FIELDS IN PLANES AT RIGHT ANGLES.



a law called, after its proposer, Lenz's law. The result of all this, as far as we are concerned, is that an alternating magnetic field finds it difficult to penetrate a conductor. Moreover, since we know that the magnitude of an induced emf is proportional to the rate of change of magnetic field, it follows that the effectiveness of a conducting screen will improve with frequency. The converse is also true—the lower the frequency, the less effective the screening.

Another way of looking at this property of a conducting screen is to note that it represents, effectively, an increasing reactance as the field penetrates deeper into the conductor. This is, also, the basic cause of the well-known 'skin effect' in which an rf current, flowing along a conductor, becomes increasingly concentrated near the outside as the frequency is raised. In fact, the attenuation of a magnetic field by a given screen can be estimated—and indeed calculated—from the skin depth occupied by a current at the same frequency as the field flowing in that screen.

From this it follows that a complete solid conducting screen around any component, circuit or piece of connecting wire will give good protection against interfering fields of either type, provided the field frequency is high enough to make the appropriate skin depth less than the screen thickness. The skin depth is inversely proportional to the square-root of the frequency, and is approximately 70 μm at 1 MHz in the case of a copper screen. By way of an example, the outer braiding on a good quality screened cable is about 180 μm thick and this thickness is roughly equal to the skin depth at 100 kHz. Thus we should expect a useful degree of protection at all radio frequencies but very little at low audio frequencies.

Leaking screens

Any screen should be solid and should totally enclose the area to be guarded if it is to be fully effective. The half-dozen loosely woven wires which constitute the screening on cheap and cheerful cables can hardly be expected to be as efficacious as the near-solid braid on a decent specimen. Screens around components or apparatus should be continuous, and may need to be virtually watertight in exceptional cases where interfering signals are of large amplitude and high frequency. In most practical cases, holes of reasonable size, such as for ventilation, do not matter, though it is always better to cover them with metal gauze wherever practicable, using the smallest mesh possible.

Our next problem is what to do about alternating magnetic fields at low frequencies—these will be induction fields—since we know that non-magnetic conducting screens will not be very effective at these frequencies. One might expect that a screen made of magnetic material would be useful for this job and this proves to be the case. Such a screen works by providing a low reluctance path which guides the magnetic field round, rather than through, the space to be protected. Of course, this also works for static magnetic fields, and screens made of high-permeability materials such as Mumetal and Permalloy are commonly placed around cathode-ray tubes to render them less susceptible to stray fields, either static or, more generally, at supply frequencies from local transformers.

Why does this artifice not work for high frequency alternating fields as well? Simply because the permeability of all materials suitible for use as screens falls dramatically with rising frequency, the hysteresis losses rising equally fast, the screening effectiveness becoming increasingly a function of the electrical conductivity rather than the magnetic permeability. The conductivity is generally poor, and copper or aluminium screens are more effective at all radio frequencies, for a given thickness.

It is sometimes claimed that a tin (as in tin cans) screen is a good general compromise, since it combines a magnetic material with a conducting coating. Like most compromises, it isn't actually very good for anything in particular since rolled steel sheet has only a moderate permeability at low frequencies, and the conductivity of the tin (which is, in any case, miserably thin) is almost the same as that of the steel, both having a conductivity of about one sixth that of copper. Consequently it is far from ideal at any frequency, though it is certainly convenient and may be good enough in some practical cases.

There are a few components, notably microphone transformers, which are particularly sensitive to interfering fields, and which also operate at low signal levels. To protect them adequately, it is sometimes necessary to use several screens, one inside the other like Chinese boxes, putting magnetic screens in between the conducting ones to give good protection over a wide frequency range.

Some examples of the wide range of interfering signals now freely available to most of us are broadcast radio and tv programmes (normally radiation), local communications such as police and taxis (radiation, also induction at close range), switching transients from thyristor control circuits, machine commutators, thermostats and ignition systems (induction and/or radiation), and that ubiquitous favourite, hum pickup (induction). The complete frequency spectrum of this little lot alone extends from 50 Hz to at least 1,000 MHz.

Even if a screen gives complete isolation from a particular field, it may itself actually introduce an interfering voltage into a circuit which it is supposedly protecting. Perhaps the best

continued 29

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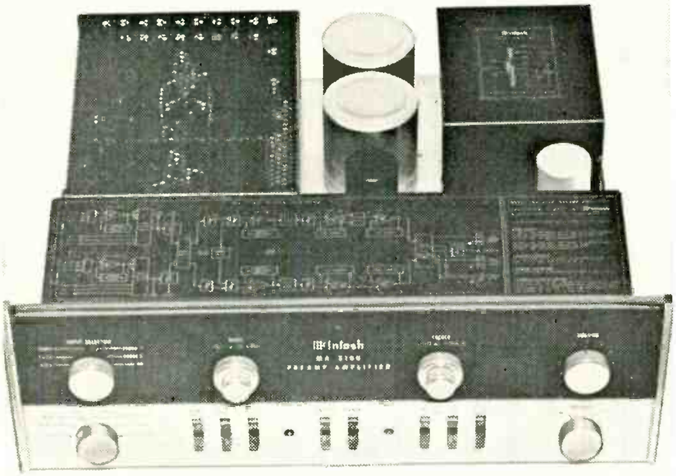

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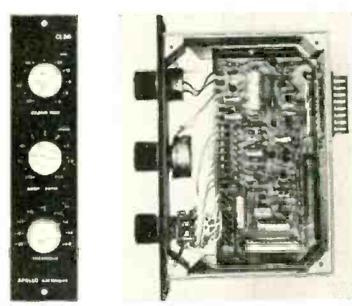

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example is a coaxial cable carrying a wanted signal and placed in an interfering field. The situation is shown in **fig. 4**, in which v_s is a generator representing the source of the desired signal, connected to one end of the cable. An amplifier is shown connected to the other end. We assume that no interfering voltage can appear on the inner conductor of the cable but there will be an induced voltage, due to the interfering field, along the length of the outer screen. For convenience, this induced voltage is represented by a single generator, v_i, at one end of the screen, and this generator is clearly in series with v_s, so that a proportion of v_i will be applied to the amplifier input. The actual proportion will depend on the impedance of the signal source r_s and the amplifier input impedance r_a. In most practical cases, the former will be low and the latter high, so that almost the whole of v_i will be applied to the amplifier input. The arrangement will be recognised as an example of unbalanced working from (for example) a microphone into a mixer, or an aerial into a receiver, and we see that it is far from ideal. However, it is only fair to add that one can expect to get away with it in some situations, especially when using capacitor microphones with their comparatively large output.

Fig. 5 shows a generally superior arrangement in which the cable braiding screens the signal return path as well as the input line. The input is still unbalanced but the generator v_i is open-circuited as far as the amplifier input is concerned (bear in mind that v_i is in series with the whole circumference of the screen, not just one edge as the figure might seem to suggest. Note also how the screening has been extended, as it would probably have to be in practice, to cover the input circuit, the amplifier, and the leads to the following circuit, and how it is connected to the actual circuit at only one point to prevent circulating currents from causing trouble. How could they? Imagine a second connection between the screen and the signal return lead (the bottom one) at the input end of the cable in **fig. 5**. The generator v_i would then cause current to flow out through the screen towards the left, returning via the left-hand connection and the signal return path to the generator. In flowing through the resistance of the signal return lead, this current would give rise to an emf, by Ohm's law, and the emf would again be in series with the amplifier input. With only one connection between screen and circuit, this particular current flow is prevented.

Unfortunately, **fig. 5** embodies another route for an interfering current. There is capacitance between both signal wires and the screen, indicated by the dotted lines in the figure, which shows single, 'lumped' capacitors for convenience, although the capacitances are really distributed evenly along the cable. One of these capacitors (C_L) allows some alternating current from v_i to flow through it, and hence into the amplifier input. Moreover, because C_E is actually distributed along the signal return wire, small circulating currents will be set up, giving rise to emf in the return wire, in series with the amplifier input. (There is also capacitance between the wires, omitted here to avoid needless complication.) Although the circuit of **fig. 5** is an improvement on that of **fig. 4**, we see that it will still be prone to interference to some extent, depending on the various impedances involved.

The arrangement of **fig. 5** is readily adapted to balanced working and **fig. 6** shows a commonly used scheme, a transformer being interposed between line and amplifier. The distributed capacitances indicated in the figure should be equal in a good cable, so that equal circulating currents from v_i will flow to both wires, returning to the generator by way of the transformer centre-tap. The winding arrangement ensures that the magnetic fluxes due to these currents will cancel and this circuit gives excellent protection against interference as long as it is properly balanced.

It is sometimes necessary to provide a preamplifier with a power supply by 'phantoming'—that is by using the same cable to carry both dc power and signals—an obvious example being the preamplifier and polarising supply to a capacitor microphone. Generally, the screen is used as the return connection for the power supply and the scheme of **fig. 7** is usually adopted to minimise the polarisation of the cores of the transformers by the dc flowing through them. The arrows show the current path and it can be seen that, in a properly balanced system, the total current divides equally between the two signal leads. The magnetic fluxes produced in the core of each transformer by the currents in the two halves of its winding are in opposite directions and are mutually cancelling. If we now consider the equivalent interference generator v_i, we see that it is in series with the power supply so that the resulting circulating interference current follows the same path as the supply current, though in alternate directions since it is ac. By the same reasoning, this interfering current also divides equally, and in opposite senses, between the halves of the transformer windings. Thus there is no output due to v_i, from either transformer.

In practice the screen is not completely effective, especially at low frequencies, and some interfering voltages will appear on the wires inside, as well as on the screen itself. To investigate the result, we consider two wires, running straight and parallel, in a plane at right angles to an alternating magnetic field travelling past them. **Fig. 8** shows, diagrammatically, the instantaneous magnetic field as a function of distance. It is a sort of snapshot of the field

continued 31

FIG. 4

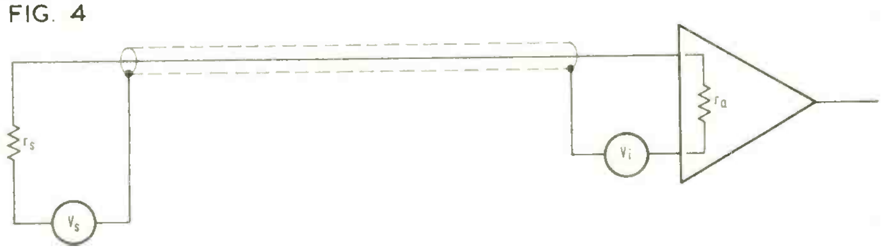


FIG. 5

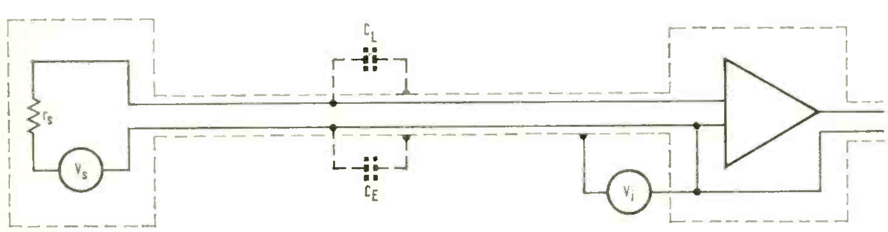
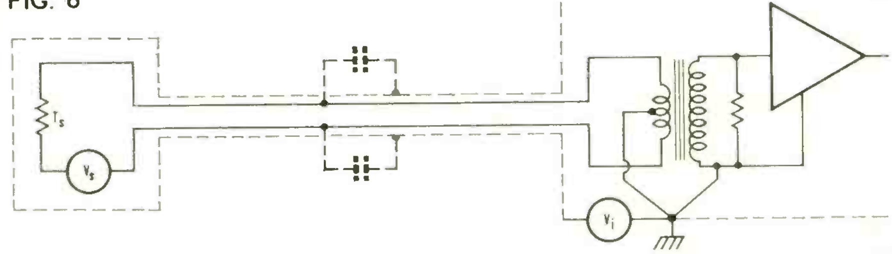


FIG. 6



in the region of the wires at one instant of time, showing the field strength at any point as the vertical distance from the horizontal axis, an upward displacement representing a field in one direction, and a downward displacement in the other. A fixed field, as we have seen, would induce no emf in either wire and the field is to be regarded as moving steadily across the diagram, as shown by the arrow. The wires are shown as dots, since they are assumed to pass vertically through the paper. In **fig. 8a** the wires are very close together and therefore subject to almost the same field strength, in the same direction. Much more important, they are subject to virtually the same rate of change of field and the voltage induced on the two wires will be practically the same, both in direction and instantaneous magnitude, i.e. they will be equal, in-phase alternating voltages.

In **fig. 8b**, the wires have been widely separated so that they are in quite different parts of the field. The instantaneous values of the field strengths at the two places occupied by the wires are very different, of course, but we are again concerned mainly with rates of change. These, too, differ in magnitude and direction at the positions shown. Consequently, the emfs induced in the wires will also differ in instantaneous magnitude and direction. However, the same magnetic field sweeps past both wires, and at the same speed, so that the *peak* values of the two emfs must be equal, giving rise to two alternating emfs, equal in amplitude but differing in phase according to the separation between the wires expressed as a function of the wavelength of the interfering field.

If one wire were placed at each of two points exactly one wavelength apart in the field, they would again carry equal in-phase voltages. This would hardly be practicable for a number of fairly obvious reasons (one wavelength at 50 Hz is about 6,000 km), and it is also important to keep the wires as close together as possible to avoid any undue variation of phase with frequency.

When two such wires are part of a balanced input circuit (e.g. **fig. 6**), the amplifier will 'see' the vector difference between the emfs induced on them reduced, as discussed above, by a factor (usually small) depending on the relative source and amplifier input impedances. The difference will be zero if the wires are coincident, and finite in other cases. It might appear that the difference should be negligible in a practical cable, in which the wire spacing is perhaps a millimetre or two, especially in a hum field. Unfortunately, actual hum fields are not of the tidy variety indicated in **fig. 8** but are usually very localised, very strong, and disgustingly non-linear, so that the amplitudes of the induced emfs are unequal, as well as their phases.

Similarly an input lead separated from its return path (often the chassis) can act as a single turn of wire which, coupled to the stray field from a power transformer in the same unit, can make an unpleasantly large hum contribution to the input signal. It is always worth taking pains to minimise this effect by using a twisted pair, or running a single wire, kept as short as possible, close to the chassis.

Any leakage of an interfering field into a screened area does not affect the conclusions already reached about the screen itself. For example, we might be tempted to think that the arrangement of **fig. 4** would be good enough at low frequencies, where the field penetrates the screen, especially if thin cable were used to reduce the separation of inner from screen. A little further thought shows that this is not so —the screen will always give at least some shielding, thus positively guaranteeing a difference between the emfs induced on inner and screen. Worse still, the screen will be highly effective at rf so that almost the only induced emf will be that on the screen, and this will be applied, in full, to the following amplifier! Thus **fig. 5** is always preferable for unbalanced operation, while balanced working is better still.

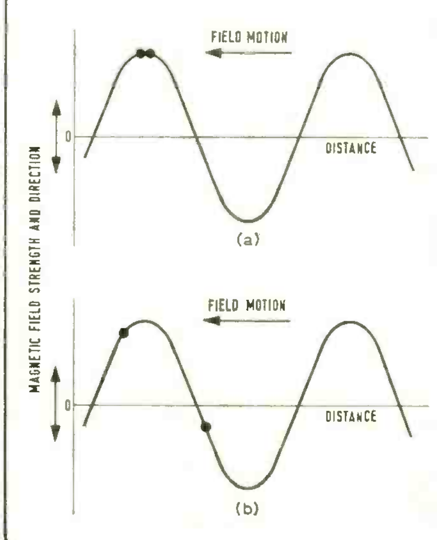
Effectiveness of a balanced system

The effectiveness of a balanced system does also depend on the accuracy with which the input circuit forms the difference of the two interfering emfs. It is most important to achieve a really good resistive and reactive balance, and the use of one per cent terminating resistors is no exaggeration. Every reasonable precaution should also be taken to maintain good capacitative balance by adopting a symmetrical layout of components and wiring. Manufacturers' advice should be sought, and followed, regarding the correct method of terminating multi-sectional windings on transformers.

Up to this point we have considered only means for keeping interference out of equipment. It may be useful to discuss, briefly, what happens when it does get in, since this can result in some rather obscure troubles. If the interference is of a frequency within the passband of the equipment, such as hum in an audio amplifier, the result is clearly predictable, but signals outside the passband can give rise to equally undesirable symptoms. For example, rf can be demodulated by a non-linear element in the amplifier, making the carrier modulation audible. The most usual rectifiers are diodes, emitter-base transistor junctions, and even dry joints which may be good conductors at audio frequencies. Another common problem is the overloading of one or more stages, causing bias shifts, so that an unmodulated interfering carrier can make its presence felt by the introduction of distortion and intermodulation. The result is not always particularly apparent. The author still shudders at the memory of a valve amplifier, investigated to trace a barely audible distortion, in which each half of the output stage was found to be giving 300V squarewaves at 200 kHz, amplitude-modulated by the *wanted* signal.

Another potential source of trouble is interaction between the interfering signal and a frequency used in the equipment itself. In radio receivers, this can lead to some exceedingly complex relationships which take a good deal of sorting out, but a simpler and more common

FIG. 8

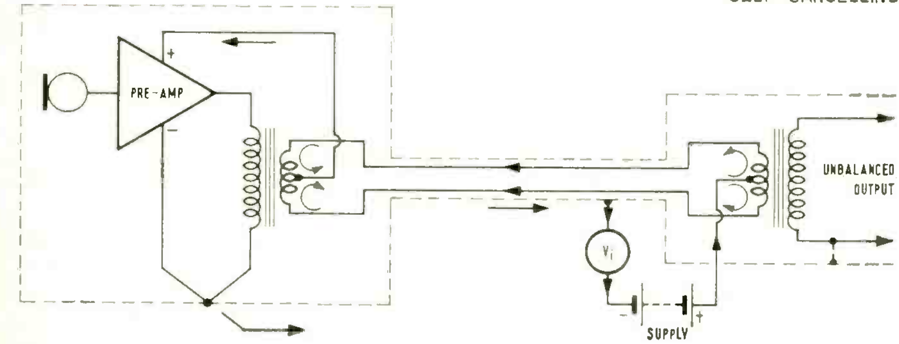


example is the beat between a bias oscillator and a spurious signal in a tape recorder. Since harmonics of either may be involved (though the bias oscillator shouldn't have any, as we all know), the total number of possibilities is large.

It should be pointed out that many a crafty bit of screening is rendered impotent by interference sneaking in through the mains cable. It is surprising how much rubbish can sometimes be borne along on the crest—and just about every other part—of the CEGB's waves,

continued 36

FIG. 7 IN THIS PHANTOM-POWERED PRE-AMPLIFIER CIRCUIT THE TRANSFORMERS ENSURE THAT ANY INTERFERENCE CURRENTS CIRCULATING THROUGH THE SCREEN ARE SELF-CANCELLING



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Analysis: Magnetic Tape

By Angus McKenzie

IN the November 1970 issue of STUDIO SOUND I compared the various magnetic properties of a number of standard play recording tapes used by the recording industry and explained the significance of the different properties. I felt that it would be better to leave the reader to judge from the published results which tape would be the best for his particular application, although I did give some basic recommendations. Several conclusions were reached and it might be helpful to summarise these.

The general magnetic properties of tapes seemed to fall into two distinct categories. The first had very low tape background noise but always a lower than average middle frequency distortion point, with a very good high frequency performance relative to middle frequencies. The second category in the main had a rather higher background noise with a general tendency to very much higher overload levels at medium frequencies, although the high frequency performance was usually slightly inferior to the first section's. One or two tapes fell in between the two categories with the benefits of both and relatively few pitfalls, an example being EMI *815*.

In the first category were tapes such as Ampex *434* and Scotch *202*, which had a three per cent distortion level of 1 kHz at an average of 4.75 dB above 320 nWb/m. The 10 kHz 1 dB squash point was usually only 0.75 dB below this point. The second category of tape had three per cent distortion points of 1 kHz between 2 and 5.5 dB higher than tapes belonging to the first, but their high frequency squash points were no better than the first and often inferior. It was therefore suggested that the first category of tapes, mostly made or designed in the US, were more suitable for use with the European DIN curves at 38 cm/s since these apply more high frequency boost in the recording process. The European tapes of the second category would give a better performance when used with the NAB recording curve for 38 cm/s. This curve requires 3 dB less high frequency energy to be applied, allowing a higher general modulation level to be used, in view of their lower middle frequency distortion. This finding seemed ironical, but was subsequently confirmed by many readers.

A further situation has become obvious. This is the tendency for VU meters to be more usually used with American type equipment and American tape, whereas meters of a peak program type seem to he more common where European tapes are used. VU meters tend to give a consistently false impression of the real peak levels being recorded on to tape and it has been found in general that rather more distortion is recorded, particularly when tapes with a lower medium frequency three per cent distortion point are used.

The neatness of spooling was compared between shiny and matt backed tapes. A general conclusion was reached at the time of the last report that, although matt backed tapes spooled very well and shiny tapes tended to spool rather badly on many machines, matt backed tapes frequently caused more head wear. This could be a most important economic factor when the cost of replacing eight or 16 track heads is considered.

If a noise reduction system is used, some factors which would otherwise be of major importance become relatively unimportant, an example being a particular tape's signal-to-noise ratio and overload point. The 12 dB weighted noise advantage with the Dolby *A* system, for example, means that engineers using it can reduce their recording levels by at least 4 dB and still have up to 8 dB less background noise on the tape. In practice, when using the Dolby system, any noise on the tape will usually be found to originate from the microphones and mixing equipment employed. With this in mind, therefore, the main considerations for a tape used with noise reduction systems are the cost of the tape itself, the modulation noise, the accuracy of the coating and the tape distortion produced when recording at reasonable levels. The signal-to-print ratio is also significant since, although the print-through is effectively reduced by a noise reduction system, the background noise of the tape itself is also reduced. Subjectively, print-through could therefore be just as audible under some conditions if the recording has a very wide dynamic range.

Since the last report, some important new tapes have become available and, in addition, a number of lp tapes have begun to be used more often, particularly when long playing times are required for recording long events. For this reason three examples of lp tape have been evaluated in addition to some new standard play tapes.

Scotch *206* is now also available as a matt backed lp tape, type *207*. Type *206* is still very expensive compared with other tapes but, nevertheless, in a few respects it is marginally better than EMI *816*, although the sample of *206* which we tested came from the States.

BASF decided to produce a new tape which they considered would be particularly suitable for use with noise reduction systems and also for a 25 mm format in loop bins required for cassette and cartridge duplication. BASF *SP50M* has a shiny back, required by loop bins, and also has one of the highest high frequency squash points of any of the tapes measured. Its tape noise is not particularly good but this does not matter when noise reduction is used and, in any case, was not a design criterion. It has good modulation noise and print-through ratios and winds quite well. This tape can be particularly recommended for use with noise reduction systems and will perform quite well even without.

To get the best performance out of the tape at high frequencies, an IEC/DIN curve is best used. Its excellent 19 cm/s high frequency performance will be found particularly useful, especially with the NAB recording curve which requires quite a considerable high frequency pre-emphasis.

EMI have now produced a matt backed version of their *815* tape with virtually identical magnetic characteristics. These were found to be very close. A Philips *Pro 36* was biased and equalised carefully on *815* to give a sensibly flat 9.5 cm/s response across the audio range. A reel of *816* tape was subsequently measured and the overall response was within 1 dB of the previously measured response, even at 15 kHz. This proves that the two types of tape are no more different from one another than two of the same type taken at random. When the noise level of *816* tape was measured, I was surprised to find that it measured 1 dB lower than my original samples of *815*, despite its matt backing. Later batches of *815* have since been measured and, despite their originally good figures, the tape is now measuring 1 dB better, bringing it into very close alignment with *816*. Other than this, no significant difference in performance has been noted in the last two years. EMI have been studying the difference between shiny and matt backed tapes for some years and have, in the past, been reluctant to make a matt backed tape because of the head wear, inferior modulation noise and higher low frequency tape rumble it produces on many makes of recorder. Their type *816*, however, would appear to be a compromise between shiny and fully matt backed tapes. It has a sufficiently matt backing to give very good spooling, though not quite as good as BASF *LR56*, *LGR30* and Agfa tapes. On the other hand it has much better head wear characteristics, with modulation noise and tape rumble almost identical to *815*. They therefore appear to have produced a winner, the closest competitors probably being Scotch *206* and Zonal matt backed low noise, both which are unfortunately rather more expensive.

Zonal now come into the picture with both a low print tape and a type they term as low noise. It was not possible to carry out quite such extensive tests of their tapes since only relatively small samples were forthcoming. All

continued 35

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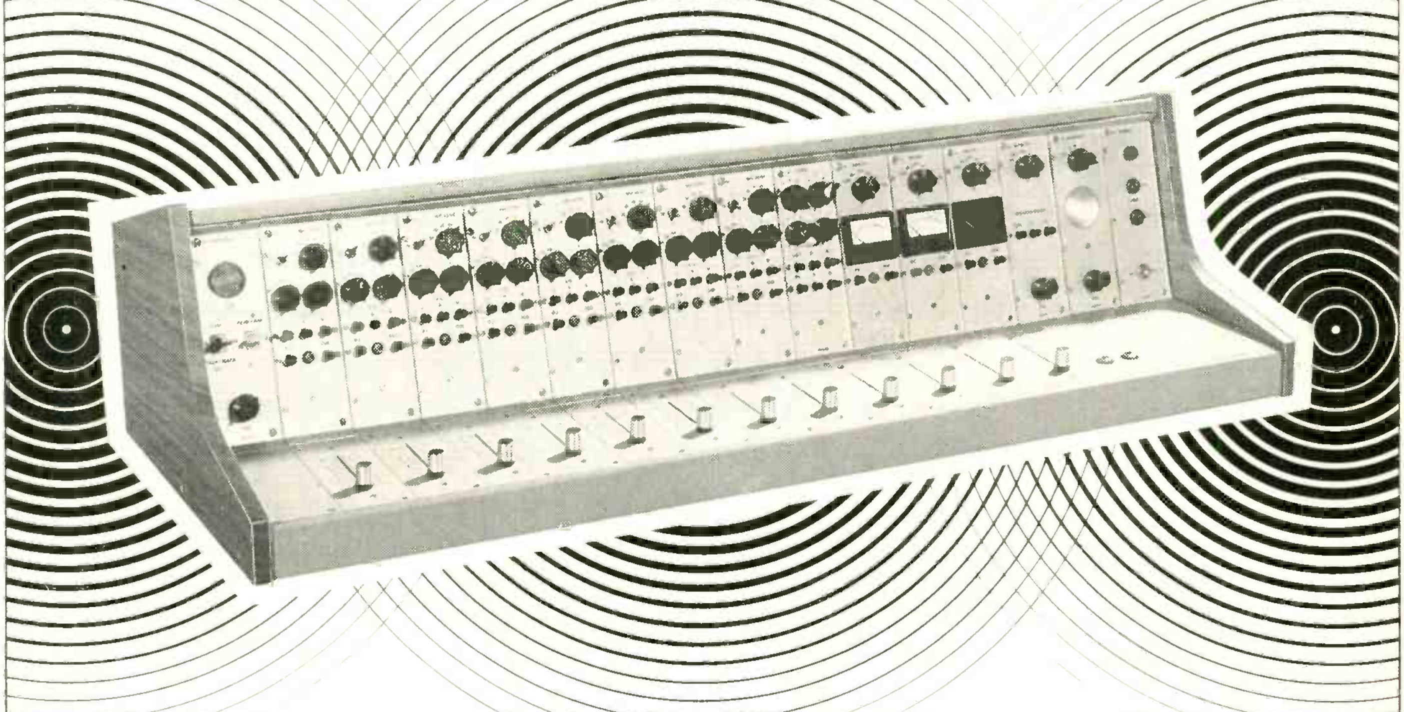

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Distortion:–	Typically 0.02% at normal levels. Less than 0.1% 20Hz–20KHz at +18dBm.
Noise:–	Microphone channels:– noise factor less than 2dB ref. 600 ohms. All other inputs:– –75dBm.
Gain:–	Maximum system gain, microphone 100dB. Line 20dB.

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continued

the magnetic properties previously mentioned were checked but it was not possible to do any user tests. I therefore had to take a concensus of opinion from their customers. The low noise tape appears to have very similar magnetic properties to EMI *815* and it is well known that one of the largest users of tape in the country regard this as an important requisite. Unfortunately, however, the signal-to-noise ratio of the low print tape appears to be approximately 1.25 dB inferior to that of EMI *815* and in general use some problems have been experienced with dropout although I only noticed dropout tendencies on the low noise type.

I also checked a sample of tape that Zonal are making for a very large consumer and found once again that its properties were very similar to *815* but in this case the noise level was some 2 dB inferior to *815*. It was noted that this special tape had a pvc backing rather than the more usual polyester. The Zonal low noise tape could become a close competitor to EMI *816*, but I suggest that potential users check the tape both for oxide coating consistency and head clogging, which can of course cause long term dropouts at high frequencies.

Lp tapes are now frequently used by professionals where a playing time of 50 minutes or more is desired at 38 cm/s. BASF have for some time been making a low noise high output tape known as ***LP35LH*** with very fine general magnetic characteristics. I have however noticed inconsistencies between samples of this tape supplied on 19 cm spools and those on 27 cm plastic spools containing 1,280m, the latter having slightly inferior distortion and squash point characteristics. Only last year did I find out at the BASF factory, that, to accommodate 1,280m of tape on to a 27 cm/s spool, it was necessary to use a slightly thinner oxide coating. This latter tape should therefore perhaps be designated ***LP30LH*** rather than ***LP35***. It is more difficult to edit lp tape and, furthermore, spooling is normally significantly inferior to that of standard play tape. To overcome this BASF have released a matt backed version of their ***LH*** tape known as ***LPR35LH*** with almost identical characteristics to the thicker ***LPLH*** tape. They are winding some 1,200m of this tape on to 27 cm plastic spools. The ***LPR*** tape has marginally better distortion figures at middle frequencies and will not accept quite such high levels at high frequencies, although the tape is superior to all other single play tapes at high frequencies with the sole exception of BASF's ***SP50M***. ***LPR35LH*** as would be expected, spools extremely well—better than any other long play tape known to me—although unfortunately one must pay a very high price for it.

One new long play tape has recently come to hand with quite extraordinary magnetic properties, and this is TDK ***LPSD***, made in Japan. At 38 cm/s it has the lowest third harmonic distortion at 1 kHz and 320 nWb/m of any tape measured. The distortion figure, 0.2 per cent, is achieved by using a fairly thick oxide coating combined with a special

continued over

	EMI *815*	EMI *816*	BASF *SP50M*	BASF *LR56*	SCOTCH *206* (US)	ZONAL *SPECTRUM* low print	ZONAL *SPECTRUM* low noise	ZONAL *PVC*	BASF *LP35*	BASF *LP35LH* 18 cm	BASF *LP35LH* 27 cm	BASF *LPR35LH*	TDK *1800SD*	SCOTCH *207*	
Backing/Type	Shiny std	Matt std	Shiny std	Matt std	Matt std	Shiny std	Matt std	Shiny std	Shiny lp	Shiny lp	Shiny lp	Matt lp	Shiny lp	Matt lp	
Spooling	Fair	Excellent	Reasonable	Excellent	Good	Fair	***	Fair	Reasonable	Fair	Fair	Excellent	Fair	Fair	***Insufficent sample
Bias 1 dB overdrop at 1 kHz (dB ref EMI *815*)	—	0	—2·75	+0·75	—0·25	—0·5	—0·75	—0·5	—3·5	—2·25	—2·25	—2	—0·5	—0·25	
1 kHz sensitivity (dB ref EMI *815*)	—	0	+2·0	+1	+0·5	—0·5	+1	0	—1	—0·5	—0·5	0	+1·5	+1·5	
10 kHz sensitivity* (dB ref 1 kHz)	0	0	+2	—2	0	+0·75	—1	+0·75	+2·5	+2·75	+3	+1·5	+0·75	+0·5	*after compensating at 1 kHz
1 kHz 3rd distortion at ref level	0·7%	0·7%	0·7%	0·3%	0·6%	0·9%	0·5%	0·8%	1·7%	0·8%	0·9%	0·7%	0·2%	0·45%	
3% distortion of 1 kHz (dB above ref level)	+6·25	+6	+6	+9·5	+7·5	+5	+7·5	+6	+1·5	+5·5	+5·25	+6·25	+9·75	8·25	
10% distortion of 1 kHz ref level (dB)	+11·5	+11·5	+11	+12·5	+12	+10·5	+12·5	+11	+7	+10·25	+10	+11	+14	13·25	
10 kHz 1 dB squash above ref level**	+4	+4	+7	+3·5	+4	+4·5	+4	+3	+3	+6·5	+5·5	+5·5	+5·5	4·5	**As read on IEC 35 µs PB curve
dBA weighted noise ref level (dB)	—65	—65	—62	—62·5	—65	—63·5	—64	—62·5	—63	—65	—65	—64·15	—64·15	—65	
dBA weighted noise 10 kHz squash point (negative)	69 dB	69 dB	69 dB	66 dB	69 dB	68 dB	68 dB	65·5 dB	66 dB	71·5 dB	70·5 dB	70 dB	70 dB	69·5 dB	Ref level = 1 kHz at 320 nWb/m using 35 µs time constant
dBA weighted noise/ref peak NAB recording level ratio (dB) (negative)	70·25	70	68·5	69 dB	70·75	68·25	69·75	67	65·25	71	70·5	70·5	72	70·75	

crystalline structure and a technique of accurately orienting the magnetic particles longitudinally along the tape. The noise is only slightly inferior to that of their low noise lp tapes but, if an NAB recording curve is used, the tape can be driven particularly hard to achieve an exceptionally high signal to noise ratio at relatively low distortion. Despite its being an lp tape these magnetic properties are closer to those of a standard play tape. Its rather high cost, however, must be borne in mind and, in addition, the recorder's record gap should be sufficiently wide to enable the biased record signal to penetrate fully into the oxide. For this reason little advantage would be gained by the use of this or other very high output tapes on many amateur or semi-professional machines which have narrow record gaps.

I also tested as a comparison a reel of BASF's normal *LP35* tape, made just before the introduction of their *LH* tape, and the results of the tests shown will indicate the dramatic improvements that lp tapes have undergone in the last few years.

The noise level of tapes is largely governed by the variation of particle size in the oxide used and also by the crystalline structure of the particles. It has been found that if the particles are all of a fairly similar size the background noise of the tape is reduced and the modulation noise is also improved. An improvement of orientation of the particles results in lower harmonic distortion and greater output while the presence of a number of smaller than average particles can result in a poorer signal to print ratio, since the small particles tend to become magnetised paramagnetically by strongly magnetised particles on an adjacent turn. Fully matt backed tapes usually seem to have a thicker oxide coating and, since the matt backing is rougher than a shiny backing, the actual surface of the tape can cause head wear.

In order to obtain a good, steady high frequency response from such tapes, by keeping them in better contact with the record and replay gaps, a slightly higher back tension than usual is often desirable. But, again, this will cause still further head wear. So although matt backed tapes spool very well they are not advised for use with most domestic and semi-professional machines.

I have also noticed that either a relatively small degree of dc magnetisation of the record head or an impure bias waveform will tend to give more cockling on matt backed tapes, in some cases so bad as to cause severe dc modulation noise. At speeds slower than 38 cm/s lp tapes, because of their better wrap round characteristics, will give better high frequency and dropout performances. In the case of recorders having a combined record/replay head the record gap will be fairly narrow in order to reproduce short wavelengths. Such machines will therefore perform best with other than standard play tapes since it is virtually impossible for them to record deep into a thick oxide coating.

I have been somewhat puzzled as to why the Philips *Pro 36* should be giving rather lower distortion figures at 320 nWb/m than the Telefunken *M10* used for the previous tests. It appears possible that up to their saturation point ferrite heads can have a lower inherent distortion in the gap material than normal type heads and this is confirmed by listening carefully to tapes made on both types of head. A small improvement in the distortion measurements has also been realised by using an oscillator itself having an extremely low harmonic content of 0.002 per cent.

For the interest of readers who prefer to record at rather high levels I have also quoted the point at which distortion reaches ten per cent on some of the tapes. It will be seen that the rate of increase of distortion of a tape is not necessarily equal for all tapes; some behave rather like a pentode valve whereas others have a more gradual onset, as with a triode valve without feedback. The significance of the ratio of the dBA weighted noise to the 10 kHz squash point for tape recorders using an IEC/DIN curve as opposed to the same weighted noise with respect to the average between the 10 kHz squash point and the three per cent distortion point of 1 kHz for NAB recording characteristics was explained in the 1970 article, and I feel that they are still highly significant. The new measurements are approximately 10 dB lower than the old figures since a precise correction was made, allowing 1 kHz peak level to reach 0 dBm, whereas previously the loss of the dBA filter was incorrectly not allowed for.

Some readers commented that I had made no reference to the measurements of dropouts on the different tapes. Again I must point out that almost no trouble was experienced during all the tests, although I know that in practice quite a considerable amount of trouble can be experienced on some machines. Bad tape weave and dropouts can become noticeable if tape guides are not changed or rotated regularly, let alone the problems introduced by too infrequent replacement of heads. Tape weave can also be introduced by rubber idlers exerting unequal pressure on the tape and capstan. So often consumers, both professional and amateur, claim a tape is faulty when it is their own equipment that is at fault.

Finally, I must deprecate the use of pressure pads in any form: they not only dramatically increase head wear but also introduce scrape, flutter and squeal; they can encourage oxide shedding, and hence long term dropouts, on an otherwise satisfactory tape. Too small a radius on a tape guide can also produce similar problems, particularly when the tape wraps a long way round a guide.

Some while ago I could not see that gamma ferric oxide tapes could be much improved; it seems that I have been proved wrong. Furthermore I understand that in the next few months both Scotch and EMI are likely to be releasing some still newer and improved tapes—these I hope to evaluate in due course.

including transients that have been known to trip trigger circuits set as high as 650V. Mains filtration is a subject on its own—suffice it to add here that suppression is best fitted at source, or, failing that, close to the affected equipment, and the very low impedance of the mains makes it highly desirable to fit series inductors, shunt capacitors alone being sometimes insufficient.

A common method of dealing with hf interference inside a unit is by the use of 'rf spoilers'—capacitors strategically placed to reduce the circuit gain at high frequencies. In many cases, this gain reduction is also useful or even necessary for other purposes and the technique is a valuable one, provided it is used in a carefully designed manner taking possible side effects into account. As an example, a capacitor introduced into a feedback amplifier, without due thought for the consequences, might reduce the stability margins, possibly giving rise to oscillation with tolerance-limit components or with particular sources or loads. Moreover, a capacitor in the collector circuit of a transistor won't prevent the base/emitter diode from being overloaded! The aim must always be to keep interference out in the first place, rather than rely on palliatives. A capacitor hung hopefully in a circuit, its value and position determined largely by guesswork, may prove about as useful as a bucket placed on the upstairs landing to protect your house against rising floods.

To summarise, we have seen how interfering signals are produced, and how they can enter equipment and cables by induction or radiation (or by conduction, which is not our present concern). Induction fields may be either electrostatic or magnetic and either or both may be present, while radiation fields are always electromagnetic. A conducting screen keeps out all purely electrostatic fields and is also effective against magnetic fields at high frequencies, though less so at low frequencies and we know that the effectiveness, and hence the thickness required, can be found from the skin depth for ac flowing in the screen. At low frequencies, where the skin depth becomes too large, screens of magnetic material may be neeeed.

The screen must be solid and must completely surround the volume to be protected. It should not be used as part of any signal path and should be connected to the signal earth, or return path, at only one point. If connected to a physical earth, this should be at the same point on the screen as the signal earth. If concentric screens are used (critical applications only), they should similarly be connected at only one point, any other necessary mechanical connection (e.g. supports) being insulated.

Balanced working gives better protection against stray fields, especially at lf, and should be adopted wherever possible, taking care to obtain really accurate balancing of the differencing circuit (normally a transformer). Whether the circuit is balanced or not, the spacing between 'send' and 'return' conductors should always be minimised, particularly where low level inputs are involved.

WHAT IS GOOD RECORDED SOUND?

Stereo lp produced by Discourses Ltd (34 High Street, Tunbridge Wells, Kent), in conjunction with *Hi-Fi News & Record Review*. Price £1.60. (Offered at £1.20 including p & p until October 31.)

THERE is something just a little terrifying about being sent for review a record with the title *What is good recorded sound?* with some remarks about being unbiased and impartial. It doesn't help. Any more than when the wife takes one look at it and stalks off to bed with the remark 'See you next week'.

I've always had a suspicion the late Terence Long was right when he said that, transferring a tape to disc, you lost nothing but added a lot. So long as you've a fairly adaptable brain and start to accept one or two shortcomings in reproduced sound, it is surprising just how far you may go on kidding yourself you are reproducing the original sound.

To make matters worse, this record is a joint venture between *Hi Fi News & Record Review* and Discourses Ltd in the 'All About Music' series of lps and booklets aimed at the person exploring music through recorded material. Those taking part on the production side included Bob Auger (Granada Recordings), John Crabbe (editor of *Hi Fi News & Record Review*), Donald Aldous and Peter Gammond, technical and music editors respectively of the same publication.

The disc booklet is intended to introduce the listener to the basics of stereo, to an appreciation of separation, ambience, reverberation, close and distant microphone techniques, tape edits, coincident and spaced microphones, and similar technicalities which get bandied about in the better reviews and which may be individually unfamiliar to the listener, while nevertheless contributing something characteristically nice or nasty to the sound of records he buys. It is a mammoth task and neither a record nor a booklet alone could cope. Even this pack turns out to be the barest skim through the subject but it's a start and anyone seriously interested could go on from there, enlightened.

The record starts with some dry, panpotted mono commentary which is used to establish the correct reproducing level for the record (very important) and correct setting up for good stereo reproduction; the booklet gives remedies if all is not well. The commentary, spoken by Bob Hardcastle, continues in the body of the Conway Hall, to demonstrate the possibilities of movement being reproduced in stereo, the effect of a reverberant acoustic, and the collapse of this ambience if the channels are paralleled for mono. This band continues with a recording of solo piano as the producers think it should sound. I suspect, though it is not made clear, that this was made with the coincident stereo (Neumann *SM2c*) microphone used throughout as the reference for tonal balance and realism. Despite being a practitioner of multi-mic techniques on practical grounds, Bob Auger does use a stereo microphone where appropriate, as I know myself from the odd occasion on the other end of his mic. Certainly, whatever the microphone arrangement used on this particular piece, the sound is to my ears very realistic and the recording produces acceptable mono when the channels are paralleled, usually a good sign. This, then, establishes a fairly natural piano sound and pleasing balance between direct sound and reverberation.

The next piece is a demonstration of how not to record a piano, with the microphone (or spaced pair?) very close in so that the piano occupies the full width of the sound stage, complete with action noise in some cases, and with individual notes coming from all over the place. A self-duet as it is termed here. Interestingly, and for a moment puzzlingly, the high notes come from the *left* and the lows from centre or *right;* this of course is as it should be because, close in, the short treble strings would be on the left if one faces the soundboard of a grand piano. The sound is initially very dry and clangy, reverberation from a distant spaced pair being added gradually to lessen the effects of the over-close mic: it hardly improves the sound. Very nasty, and reminiscent of many commercial piano recordings. Also, very difficult to track without distortion. After that, another snatch of 'realistic' piano as in the first snatch. The music, by the way, is the Mendelssohn *Andante and Rondo Capriccioso in E, Opus 14*, played by Thomas McIntosh, on a Steinway *B*.

Having re-established a sense of proportion, the next piece examines the problem of balance between two performers, clarinet and piano, in an arrangement of the Weber *Concertino in Eb* for clarinet and piano. Again, it is not entirely clear from the notes whether the recording is made using a stereo pair plus spot mic on the clarinet or using several microphones but the former sounds more likely and the clarinet feels far too close for comfort moving as the instrument moves; one is sitting on the edge of the stage. Next the microphone or microphones move back—too far back—and the music becomes very bathroomy. Next the 'optimum balance', perhaps not everybody's but pleasant and feasibly realistic. I'm not sure whether there is a spot on the clarinet or not but the instrument is placed a little more prominently than the piano. Finally on this band, keeping the same balance, an intermezzo by Hurlstone for clarinet and piano, which is quite delightful (Thea King on clarinet).

Having explored the recording of two instruments, the next comparisons are on the recording of singer plus piano, first with a Schubert song performed by Beth Boyd and Thomas McIntosh, recorded with an intimate but natural sound; would that all song recordings were as nicely balanced. There is a more lively acoustic for the next, rather lighter song sung by Bryan Corrie. Little separation between singer and piano in either of these but the performers are set in a clear acoustic. Once more, it *sounds* as though a single stereo mic is being used as everything is so clear and the change into mono so subtle but the pamphlet is again not particularly helpful about the exact setup.

For the last band on the first side, the natural progression is to a string quartet (Fidelio). The examples start with a glaringly unnatural sound, one microphone on each performer, positioned by panpotting. The result is a harsh close sound, probably worsened by the microphones being above the instruments instead of in front as the audience usually hears them, with very clear definition of the instruments' positions and a rather dead and horny space between. Reminds me of the sound of one cheap label whose musical content often exceeded its recording standards. Having learned to recognise that particular technique, one is taken back to the rather distant single stereo pair, the reference microphone. The sound is much more natural and indeed one has heard concerts sound like this live, but it is obviously too reverberant from the hall. Curiously, it doesn't sound to me as though one has simply moved closer in from the previous balance. I can only suppose this has something to do with multiple mic stereo; perhaps two stereo pairs would have been better or the stereo mic on its own a little further away? I also suspect that the stereo mic may have been a bit high, it has a hint of the harshness of the spaced mic sound.

Finally on this side, one is treated to a touch of Pop technique with a lot of artificial reverberation added to the sound of the string quartet to produce a quite unreal swimmy effect—just to show, in an exaggerated way, the limitations of adding artificial reverberation to compensate for over-dry sound in close-mic working.

The second side starts, curiously, with the sound of a good amateur recording and the preferred professional balance of the same material being compared by splicing together

continued 39

at the same level. Both are good in their way and I frankly preferred the amateur recording on balance, being mildly dissatisfied with both. The recording is of Thea King and the Fidelio Quartet playing part of Mozart's *Clarinet Quintet in A, K 581*. I don't know what the amateur microphones were beyond the detail 'Revox'. The sound was pleasant, if very slightly boomy, the strings perhaps slightly distant but full and natural. The changeover is obvious even without the point being illustrated after. The subtle change in hiss, the perhaps slightly better internal balance between instruments (though we can't know how they sounded, only what the producers wanted), a slightly thinner and more wiry sound from the strings (due to closer and higher microphones?) losing a bit of body in mono, perhaps slightly wider spacing of the strings and a more airy quality. The point is then made that it would be difficult for the amateur to produce as good results with a large body such as a symphony orchestra. If it were simply a question of accurately reproducing the sound heard at the location, I do not think this is necessarily so. On the other hand, given that most recordings have to take place in buildings that are in some degree not ideal, it is probably easier to achieve a pleasant and lastingly acceptable sound in the majority of halls using elaborate microphone arrangements, though perhaps less elaborate than those used in some commercial recordings. The BBC often use far fewer microphones for a concert than would be used by a commercial recording company and nevertheless produce a clear and natural sound that many people prefer. Anyway, to illustrate what a good orchestral recording should sound like, an excerpt from a Unicorn recording of the Raff *Symphony No 5* is included, played by the LPO under Bernard Herrman. This has a very wide dynamic range, is beautifully clear and unmuddled, with a natural ambience, a lot of body seems to disappear in mono, possibly a function of multimiking? Nevertheless, an excellent example of orchestra on record, as well as a delightful work.

After the full orchestra, the next band goes back to solo voices and piano to examine the question of balance and layout with voices. Beth Boyd sings Offenbach's letter song from *La Perichole*, starting with the voice in one speaker and piano in the other. The use of the panpot is then demonstrated, passing the performers to opposite sides of the stage before being brought back to their proper places near the centre. The sound is then more natural, with a pleasing ambience around the voice and piano, presumably done with spot mics plus atmosphere mics. This excerpt does demonstrate how objectionable the technique of ping-pong stereo is, with performers unnaturally divided to the two speakers. A duet then follows between Beth Boyd and Bryan Corrie with a 'natural' balance, perhaps a shade live but the Conway Hall, where all the recordings were made, is a live place.

The final band moves on to massed voices (members of the Royal Tunbridge Wells Choral Society). One sees from photographs in the booklet that there are three spaced microphones in use, and one hears the (Bach) chorale three times—first with the microphones in close, polar pattern set to cardioid, giving a sound that is too close and dry, breathy and full of individual voices. The microphones are then moved back and switched to figure-of-eight pattern to pick up more reverberation from the rear and a more integrated sound from the choir (which sounded decidedly rough in the first example). Third time round, the microphones are switched to omni to include more indirect sound from all sides and this certainly sounds better.

We are told that only on this chorale did the theoretically correct single stereo microphone produce a sound that was in all respects as good as the suggested optimum (actually, the stereo microphone was left in the same position throughout, while the other mics were moved, so this is an unfair comparison). Moving a single stereo pair intelligently might have produced comparable or even better results than the 'optima'.

Finally, soloists, piano and chorus join in Stephen Foster's '*Some Folks Do*', which has a very wide dynamic range. This illustrates the use of multimic techniques to produce an integrated sound picture where practical limitations prevent the actual layout of performers from being that one imagines from the sound—the chorus are in fact strung out along the front of a stage, with soloists and piano further back on the floor. The piece also demonstrates a rather contrived (and, when you have heard it a couple of times, very tedious) tape edit to remove a fluff in the piano part.

Obviously, as the booklet admits, it is impossible to cram into one disc and one booklet an example of every technique that could be used in every particular case. It is impossible to illuminate every pitfall, just as it is impossible to provide a perfect microphone arrangement for all occasions.

So does the record work? An acquaintance who kindly lent me his equipment and ears for one evening was decidedly unflattering: 'Who on earth would want to buy that lot?' I don't think a lot of people would—they are less concerned with sound quality in general than with the quality of the particular records they buy and their ears are the arbiters of that.

What *is* good recorded sound? I'm not sure that the record, or any disc, could hope to answer that one. The purist says it should be the most accurate possible reproduction of the original sound. But again, what is that? Without a totally periphonic system of unlimited bandwidth, introducing zero noise and distortion, one is making compromises all the way. Anyone who has heard coincident quadraphonic recordings knows that the difference between them and two-channel stereo is akin to that between stereo and mono. These are all steps towards the real thing, never actually getting there. Part of the fun is trying.

Or again, a producer may want to get away entirely from the bounds of the concert hall. Why not go for the 'ideal' sound—the sound the conductor probably hears. Is this any the less 'good recorded sound' because it is not what one would hear in the concert hall?

I can say that I found nothing to complain about in terms of frequency response on the record; the sound, reproduced through chains I consider to be respectable, was consistently natural without the use of tone controls or filter. Hiss was negligible except possibly during the amateur tape, and then only minute. The dynamic range was very wide, deliberately so in an attempt to reproduce the range likely to be encountered in live listening; unfortunately this leaves the disc surface in a rather critical position, compared with commercial recordings which make use of compression to give a higher overall level and consequently lower surface noise. Apart from the need to clean the record scrupulously to avoid particle noise, there were minor surface blemishes in the form of occasional surface roughness and what is sometimes known as stitching distortion, a burst of distortion caused during the actual pressing of the disc.

Distortion in the 'elongated piano' excerpt, on the brass in the Raff, on the Bach chorale (particularly the first take), and the Stephen Foster, bothered me. The Shure *Supertrack* cartridge with its elliptical stylus had a subtly cleaner sound than the conical stylus but, despite adjustment of tracking weights, bias etc, we kept hearing traces of distortion at these points.

The end of the *Gloria* in the Argo recording of the Haydn *Harmoniemesse* (ZRG515) seemed a fair comparison with the choral pieces at the end of the review record. The peak levels were very much the same but there appeared to be compression so that much of the material was being held around ppm 5 in loud passages. St John's chapel acoustic may also have something to do with this effect. Despite the boys' voices and the soprano solo, there was very little trace of unease from the pickup.

My one regret on the production of this record is that a good fifth of the sleeve is devoted to sales blurb for the series in a style mildly reminiscent of those untempting inertia-sales offer-of-a-lifetime record promotions that flop through the letterbox from time to time. But it *is* worth listening to, if only to overcome the right and proper prejudice that all recordings should be done with a single coincident pair and the method adopted, with the coincident pair placed to best advantage. Yet in a way I am grateful for the area of uncertainty; if it sounds good, it *is* good, however achieved.

John Fisher

The Practical Problems of Disc ~ Cutting

By Sean Davis

SO much has been written about studio practice and techniques that there can hardly be anybody in recording who wouldn't recognise a 16 track 38 cm/s NAB Dolby A master at 100 paces. But most sessions result, eventually, in a 17 cm or 30 cm disc via a process kept largely secret by that strange group of backroom introverts, the disc cutting engineers.

Perhaps because of this introversion, the cutting engineer is often blamed for all and any shortcomings in the final produce, sometimes unjustly. I hope to show some of the possibilities and limitations of the disc recording process and how to get the best out of your chosen cutting service.

Firstly, a brief resumé of the principles. Both tape and disc systems have an amplifier driving a transducer. But, whereas the tape recording head produces a magnetic field, the disc cutterhead produces a physical movement in a cutting stylus; obviously the cutterhead will require more power than the tape head. Most cutterheads have a coil suspended in the field of a permanent magnet so that a current in the coil will produce a varying field which interacts with that of the magnet, resulting in movement of the coil. The cutting stylus is mounted to the coil.

For reasons of groove spacing and signal-to-noise ratio, the amplifier feeding the cutterhead is not fed with a 'flat' signal from the tape machine. A curve is superimposed giving a bass cut (−17 dB at 50 Hz) and a top lift (+17.7 dB at 16 kHz, both relative to 0 dB at 1 kHz). Now the output power of the cutting amplifier doubles for 3 dB increase (Power dB = 10 log V_2/V_2) so that, at high frequencies, a cutterhead coil can overheat and burn out. For good transient response, and to restrict the number of resonances, the mass of the coil and stylus assembly should be low (which conflicts with high power handling capacity), and the design and manufacture of cutting coils is intricate and costly.

In addition to the drive coil, most modern cutterheads have a second winding which provides a feedback voltage, ideally derived from the movement of the system, not from flux linkage with the drive coil field. The amount of feedback is typically 30 dB. Those acquainted with feedback systems will know that, with this amount, one has to be careful with phase shift or the whole system will oscillate. Furthermore, the phase shift must be controlled over a bandwidth of about 40 kHz. Very interesting, but where does this get us? Well, it explains why cutting amplifiers and heads are so expensive.

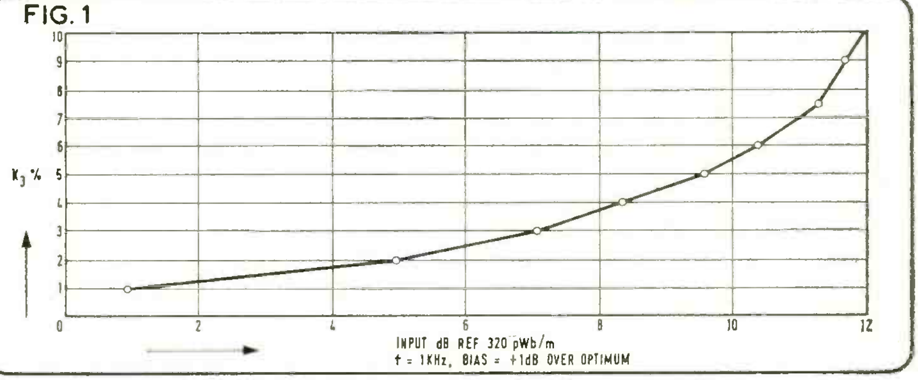
One well-known make of stereo head with its associated amplifiers costs around £5,000, that is without the cutting lathe. So, if you go to the cheapest cutting service, you can't expect a perfect match between tape and disc. With skilled operation, amazing results are possible from some pretty ancient gear. If you are doing a demo on a tight budget, however, don't pile more top on the tape to compensate for the disc loss. It overloads the cutting amplifier and creates severe distortion.

Another difficulty arises from woolly bass, balanced at a high level relative to the rest of the instruments. This produces large amplitudes in the groove which can cause pickups to jump. This is no use to anybody so the cutting engineer has to either keep the overall level down or roll off a lot of bass in order to affect the 'wool' frequencies around 100 to 150 Hz. A tight bass makes a world of difference.

If your disc is an acetate, remember that the material (cellulose nitrate lacquer) is softer than the vinyl of which pressings are made, so the cutting engineer may have kept the level down a bit to avoid jumping and to reduce playback wear. The softer lacquer also yields slightly less top on playback than a comparable pressing.

Perhaps the biggest arguments occur over distortion—the disc is undeniably distorted, the cutting engineer says it's the tape, the recording engineer says the tape is clean. To understand what is really happening we should look at **fig. 1**, the distortion curve of a typical tape. Note how, beyond a certain point, a comparatively small increase in recording level causes a large increase in distortion: the tape is approaching saturation. Additionally, for a given record level, the distortion of most tapes rises with frequency. It is at high frequencies that pickup tracing distortion becomes evident, all this aggravated by the top lift applied to the disc; a nice case of all parameters summing on the debit side. If the recording engineer has pushed the tape to the verge of breakup (which on most programme materials becomes evident at mid frequencies), the top will already be on the steepening part of the distortion curve. Add to this the slight but inevitable distortion of even the best disc cutting system and the sound will have become distinctly unpleasant.

FIG. 1



If the tape is an original, there is little the cutting engineer can do except keep the last few minutes well away from the label, either by reducing level or bass or a bit of both. However, if the 6.25 mm tape is a reduction, a big improvement is often made by re-reducing at a lower level. When using Dolby, there is no need to fry the tape on the record head; it is far better to sacrifice a few of the 10 to 15 dB noise reduction to get a really good cut.

Mention of Dolby brings me to the sins of omission. Cutting engineers are not clairvoyant, they need to be told whether a tape is NAB or CCIR (DIN, BSI, IEC), perhaps even a special such as AME, ARD, whether it is stereo or mono, Dolby or non. If Dolby, *always* put a Dolby tone at the head (not on a different batch of tape or a machine not used on the master). I know all Dolby tones are supposed to be the same but the plain fact is that people don't always check their machines and tape batches often enough and some amazing variations occur.

Some cutting rooms have tape playback machines fitted with ferrite heads, the gaps of which are both more accurate and, in two track versions, better aligned than is possible with the laminated metal type. This normally helps playback accuracy but difficulties can occur when the master tape is an assembly of items recorded at different times or on different machines (e.g. *Cringe's Greatest Hits*). Where the recording machine(s) have metal heads, a mono playback of the stereo tape will probably sound all right but the ferrite head will detect the slight difference in azimuth existing between items. When the disc is played back in mono (perhaps on radio) many tracks may sound horribly off azimuth. The way out of this one

continued 45

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Recording Studio Techniques

BY ANGUS MCKENZIE

Recording in the Albert Hall

OVER the last year or two the management of the Royal Albert Hall have been doing a lot of work to improve facilities for those that use the Hall. Instead of using the Green Room behind the stage, as in the past, arrangements can now be made to use a recording room; this has come to be known as the transcription room, because it is frequently used by the BBC's transcription recording unit. The Green Room is, in fact, the main artists room and, having recently been redecorated, cannot be used by engineers under any circumstances.

The transcription room is on the East side of the hall, on the same level as the bottom balcony, and it has excellent facilities. Permanent screened pairs have been installed from the stage, organ loft and gallery. The leads terminate in a post office type tip, ring and sleeve jack field in the recording room and in XLR female sockets in the hall which are wired with the normal convention of pin one screening, pin two positive and pin three negative; the positive appears on the tips of jacks inserted into the jack field. Some 20 sockets are available on the stage, ten in the organ loft and five at gallery level. In addition to these, the BBC have, for their own use only, some special multiway connectors and some XLR sockets which terminate in the BBC's own permanent control room next door to the transcription room.

Naturally a charge has to be made for the use of the transcription room in addition to any charge for the use of the hall itself, so that readers who wish to record a live event will have to contact the Hall in plenty of time, otherwise it may not be possible to use the facilities. Should mains be required in the hall itself for powering capacitor mics this can be arranged provided adequate notice is given to the Hall authorities. However, I recommend that phantom powered mics are used, since my own experience is that the earthing in the hall is such as to produce clicks under some circumstances. However, no clicks were noticed with phantom powered lines, many of which I used recently. Only one mains socket is available at present in the transcription room. The room has a floor area of just over 30 m^2 and the floor covering is lino. There are a few acoustic tiles on one wall and we used two acoustic screens which had been left in the room to dampen the reverberation time slightly. The Hall cannot provide any jumper leads for the jack field and users are recommended to bring leads at least 3m long to extend from the jack frame to a convenient place for a control desk.

Almost no noise was audible from outside the building but unfortunately the ventilator fan in the room had to be turned off because of its noise. Furthermore, a large motor, presumably driving the Hall's ventilation system, was within earshot and caused a slight rumble-type hum to be noticeable in the room, though this was not audible in the hall itself.

The Hall's acoustics have been changed dramatically by the addition of the flying saucers, which noticeably reduce the echo from the dome. Since the Hall is so very large, and despite the absorption given by the seating, the reverberation is quite long—approximately 2.6s—but I noticed that a capacity audience cuts down the reverberation to such a degree that under many circumstances it becomes necessary to add additional reverberation to a microphone balance.

The Hall is probably the best one to show the advantages of a fairly simple microphone technique, and a crossed figure-of-eight stereo microphone can give a superb sound when placed correctly. Although surprisingly little coloration, other than reverberation, is added to the sound picked up from a well placed microphone, the sheer size of the hall means that, for many applications, quite a high degree of microphone amplification may be required. The Hall's size also encourages musicians to perform with wider than usual dynamic range. For these reasons, therefore, capacitor microphones with very low noise pre-amps should be used, especially for the pickup of reverberation. A very marked difference in hiss level can be detected, even when recording an orchestra, between, for example, an AKG *C24* and a Neumann *SM69* fet, or the newer low noise AKG type *412*. Because of this the BBC now often use quieter microphones than the *C24* for their relays.

Hot air draughts

Another surprising problem is a continual upward draught of hot air from the audience in the centre of the hall. This draught can produce quite severe bass rumble on some types of capacitor microphone. The AKG *451E*, for example, should never be used without a wind shield if an audience is present since both the BBC and I have had trouble with microphones of this type when using them without wind shields. Plastic foam shields cause less loss of high frequencies than the metal type, and furthermore I understand that the foam type has less effect on the polar diagram at higher frequencies.

The stage is large enough to allow plenty of room for floor stands but it is essential to use stands with anti-vibration mounts, since the floor transmits vibrations all too easily. Microphones can be suspended above and well behind the conductor from the top balcony rails. For safety reasons the Hall authorities insist on the use of very strong slings for any microphones hung over the audience as well as independent tie to the sling and to the microphone cable itself. If the sound of the organ is to be picked up adequately it is necessary either to have the canopy considerably raised or, for solo organ, fully lowered; if other performers are present a microphone situated above the canopy may be necessary, although a reasonable pickup of the organ can be obtained by slinging a microphone fairly tightly between the fronts of the two top balconies. It is unfortunate that, because of the canopy, the best position for orchestral pickup on one stereo mic is about the worst for the organ.

For years now I have heard engineers claiming that the Albert Hall is very difficult for recording. I have also heard some rather poor commercial recordings made in the Hall; either they have been very badly balanced or have had a quite intolerable hiss level, particularly on organ recordings. A multimic balance could probably give quite good results but no matter how large the forces I feel that most engineers will find a simple stereo microphone 6 to 8m above floor level and 5 to 10m behind the conductor will give an excellent sound on which to base a stereo balance.

A high position of the main mic may well improve the pickup of a choir and the presence of woodwind and other instruments towards the back of the orchestra, but such a mic position may well produce lack of presence on a solo instrument such as a piano or violin in the front of the orchestra. It will frequently be unnecessary, if a fairly low microphone position is used, to close mic the soloist, but such a position would almost invariably mean that a small or medium sized choir would have to be separately miked and pan-potted into the balance.

I suggest that separate, backward-facing mics should be installed, their main axes pointing up towards the top of the back of the hall; such a set up is particularly advised if a capacity audience is expected when recording a live event. It is recommended that these backward-facing mics should be at a similar distance from the performers as the main stereo pair. This is to reduce any cancellation of bass frequencies caused by the long wavelength cancellation effects which result when the outputs of two microphones at dissimilar distances from a low frequency source are mixed together.

The direct sound picked up from the orchestra in the Albert Hall cannot only give very

continued 45

RECORDING STUDIO TECHNIQUES

continued

wide stereo but, because of the excellence of the acoustics in general, the positions of all the performers should be very accurately determinable. I have noticed on many occasions that brass and percussion instruments, although usually at the back of the orchestra, can sound surprisingly forward in the balance, and little or no spot miking is therefore necessary on the orchestra. On occasions some types of percussion instruments can bring out a resonance of the flooring at the back of the stage giving these instruments an undue prominence which can only be rectified by asking the players to perform more quietly.

The acoustics of the hall make it an obvious choice for quadraphonic recording of classical music and recently I had the opportunity of recording both in stereo and quadraphonics a concert in which the forces employed were full orchestra and soloists and a choir of 700. During the rehearsal the quadraphonic sound was very exciting, but the position of the four coincident cardioids gave a little too much reverberation, since they were quite high up and a long way behind the conductor. At the actual performance, however, their position turned out to be just about ideal; although the reverberation time was only marginally less, the damping of the reverberation by the capacity audience reduced the amount of sound audible from the back of the hall.

The coincident placing of the AKG *452E* mics with *CK1* capsules was achieved by having their amplifiers vertical and using four right angle bends, allowing the capsules to point from left, front right, back left and back right. The capsules are arranged in the shape of a square and back to back, the distance between opposite capsules being approximately 7 cm. This type of mounting avoided the back of one microphone distorting the sound field entering the front of the opposite mic, and it also reproduced each quadrant accurately with respect to the others. The result was a surround sound very similar to that heard by an audience seated in the centre of the hall. A separate *SM69* fet mic was suspended slightly closer to the orchestra and a few feet lower.

One extra microphone was placed near the piano for Tchaikovsky's *First Piano Concerto*, and a slight amount of this mic was added to the stereo balance only, so that a later comparison could be made between the stereo and quadraphonic recordings. Some output from the back pair of *452E* was added to the stereo balance to give extra reverberation. In Dvorak's *Te Deum* it was not necessary to use any close mics at all for the soprano and bass soloists, a very natural balance being obtained from the main mics.

Having listened to much stereo from the Royal Albert Hall I feel that the BBC's balance is extremely good, although the microphones that they have used have often had an inferior noise level to that which I obtained. Many commercial recordings have failed because the main mics have been too close to the orchestra and, in particular, too low, so that in one instance a large chorus was almost completely drowned by the orchestral strings. The use of separate mics for the choir, although occasionally necessary, is, unfortunately, fraught with snags, the main one being a tendency for some voices to stick out more than others if the choir mics are too close. The use of an insufficient number of choir mics, if they must be used, also fails to give a complete spread from left to right.

Recently I recorded the Albert Hall organ for six complete days and, for the occasion, the management lowered the canopy on to the stage, exposing the entire pipework of the organ to the microphones. I understand this is the first time this has been done for a recording. The organ, originally built by Father Willis, was extensively rebuilt before World War Two by Harrisons, who also added many stops. I must admit that, unfortunately, rather a lot of air escapes from various pressure chambers, although I found that only in the very quietest passages did this become troublesome.

In addition to using a feed from the back quad mics to add reverberation in the stereo balance I used a small amount of feed from two microphones placed in the highest gallery. Considerable trouble was experienced during the session from noises made by workmen and by the lift motor, which was clearly audible in the gallery. I therefore advise any potential user to stipulate in the contract that during recording no workmen shall be working on the ground floor or above, particularly not on the roof, which can be reached by ladder from a high balcony.

I found it necessary to install reverse talkback from the artist to the control room to hear what was being said in the hall. Since it takes quite a time to reach the hall floor from the transcription room I also installed headphones for an engineer in the hall, in addition to normal talkback. Any special connecting leads running from the recording room to the hall will need to be at least 60m in length. Ordinary lap screened cable should not be used, interwoven screening being essential because there is so much electrical machinery being used in the building that is poorly suppressed.

I found it necessary to place my main mics a considerable distance away from the organ to achieve a reasonable balance between the choir and solo departments against the swell and great sections. The choir organ is right at the back of the instrument and can become almost completely obscured by the great organ whose pipes are right in the front of the instrument. On the days that we were recording I did not notice any violent temperature changes in the hall yet I found it vital to have an organ tuner available throughout the sessions as some of the pipes were rather temperamental and needed constant attention.

One outstanding feature of the Royal Albert Hall is the remarkable quality of bass frequencies; these can only be faithfully reproduced with simpler microphone techniques because of the cancellation difficulties with multi-mic configurations that I have already mentioned. With pop and light music however, I fully appreciate that a multimic technique is essential. I have no doubt that this hall can produce some of the most exciting sounds, and I hope that this article may encourage some classical record companies to record there, particularly in quadraphonics.

Any readers who are interested in the possibility of recording should contact the hall's office staff who are very helpful and enthusiastic.

The deputy manager, Mr Charlton, tells me that he will always be pleased to give information about facilities and can be reached on 01-589 3203.

DISC-CUTTING

continued

is to make a copy master on one machine at one time, checking the azimuth of each item on the original tapes by listening in mono and adjusting the playback head of the playoff machine accordingly.

Finally, the vexed question: 'How much playing time can I get on a side?'. There is no straightforward answer as so many factors must be considered.

First, is it an acetate or a master for pressings? Master lacquers are always larger than the finished pressing (e.g. a 36 cm lacquer is used for a 30 cm master cut), so that the recorded area avoids the surface bumps of the lacquer at the edge of the disc (an inevitable result of the coating process). A 30 cm acetate is on a 30 cm blank so the first minute or two of the recording is on the bumpy portion, necessitating a deeper average cut to avoid losing the groove on the shallow portion of the bump. Of course, a deeper groove means wider spacing and/or lower level.

Second, how many bands on the side? Each normal-sized scroll uses the equivalent of up to 30 seconds playing time.

Third, what is the dynamic range of the programme material? On machines equipped with automatic variable pitch control, quiet passages occupy much less space than loud ones. This is why a piece of classical music of say 24 minutes (which usually has a fairly wide dynamic) can peak at a higher level than a 20-minute side of heavy group music which keeps the pitch open all the time.

It should by now be obvious that every disc presents the cutting engineer with a number of factors to be weighed, each against the rest, and this is where personalities come in. Some balance engineers want the best possible sound from the disc and, if it involves a bit of cooking or limiting, that's okay by them. Others want the disc to be exactly like the tape, and scream blue murder if it isn't (or just keep silent and go elsewhere for their future work). As long as the cutting engineer knows his client's views and requirements, he can act accordingly. So, if you are trying a cutting service for the first time, decide what you want beforehand and let the engineer know. If the disc is not quite as you expected, and it came from a reputable cutting service, check the points listed above before picking up the phone. And if you're going to query the frequency range of the disc, make sure you have a standard frequency test disc of your own. Most cutting rooms have.

Diary

BY JOHN DWYER

BILL Foster of **Roger Squire's** studio says that the studio is building up its commitment to steady work as opposed to hourly bookings. They have just done a pilot promotion tape for one of the big record companies, who will send the tape to dealers to advertise forthcoming records. This means that a new tape will have to be made every two or three weeks as new releases appear. Roger Squire already do such work for Polydor and United Artists.

Bill Foster, who only recently joined Roger Squire from CBS, also mentioned that he was still grateful for all he had learned during his days at **R. G. Jones'** studio, now in Wimbledon. Most engineers remember their beginnings but not always fondly; especially, as in this case, after eight years. I decided, therefore, to go and see for myself.

When I arrived, Gerry Kitchingham and Nick Sykes were preparing for a session for Valley music. Gerry, the studio manager, has been at the studio for seven years now, and Nick has been training as an engineer for six months; before that, he worked with the other side of the firm which manufactures and hires sound equipment.

The studio floor is 90 m^2 in area and will accommodate 40 musicians comfortably. The studio has a Neve desk and has just gone eight track, having acquired a Studer *A80*. Previous customers include Status Quo, the Rolling Stones, Georgie Fame, Jo-Ann Kelly, Second Generation, Fairweather and McGuiness Flint.

The present session was being produced by Ronnie Scott of Valley Music, who should not be confused with Ronnie Scott the saxophone player. Ronnie has been using the studio regularly and has produced in it the demos of the hits *I'm a Tiger*, *Ice in the Sun*, *It's so Easy*, and the Eurovision song *Beg, Steal or Borrow*. R. G. Jones also provided the pa in the Usher Hall for the Song Contest itself.

This month the R. G. Jones studio has been used by John Hiseman with Dick Heckstall Smith, Leapy Lee, Gallagher and Lyle, Larry Grayson, Christie, John Lewis of Terry Dactyl and the Dinosaurs, and an American group called the Stooges.

The hourly charges are £8.50 for four and £12.50 for eight track recording. Robin Jones, who has now taken over most of the firm's running from his father, told me that although the studio is heavily booked they can always use more bookings. 'People ring up when they find out what we charge and say, "What do we get for that?". I tell them and they clearly think there's a catch. There's no catch; here in Wimbledon we don't have West End overheads, that's all.'

During the month I dropped in to **Marquee** where Phil Dunne let me hear part of a single by Margaret Powell. Mrs Powell is the author of *Below Stairs* and other works, and is what is known in the trade as a radio and tv personality. The record is based on the *Laughing Policeman*, which Charles Penrose recorded for Columbia in 1926; diligent research informs me that the recording date of the original was April 22. Phil, who engineered the Margaret Powell session, told me it was great fun. It was produced by Don Gould for James Fisher's Very Good music company. Marquee have also recorded the Sleaz band, a Scottish group, who laid down a couple of tracks under the production of Stuart Taylor, Chris Barber mixed an album and Rory Gallagher and Ashton, Gardner and Dyke also did sessions.

Penny Farthing, a production company who frequently use Marquee, did a promotional jingle which was produced by Larry Page. Larry also produced Daniel Boone's latest single, *Annabel*, which was recorded and mixed at Marquee. Tony Visconti produced a live album for his own Good Earth Productions of a group called Gasworks, a folk duo.

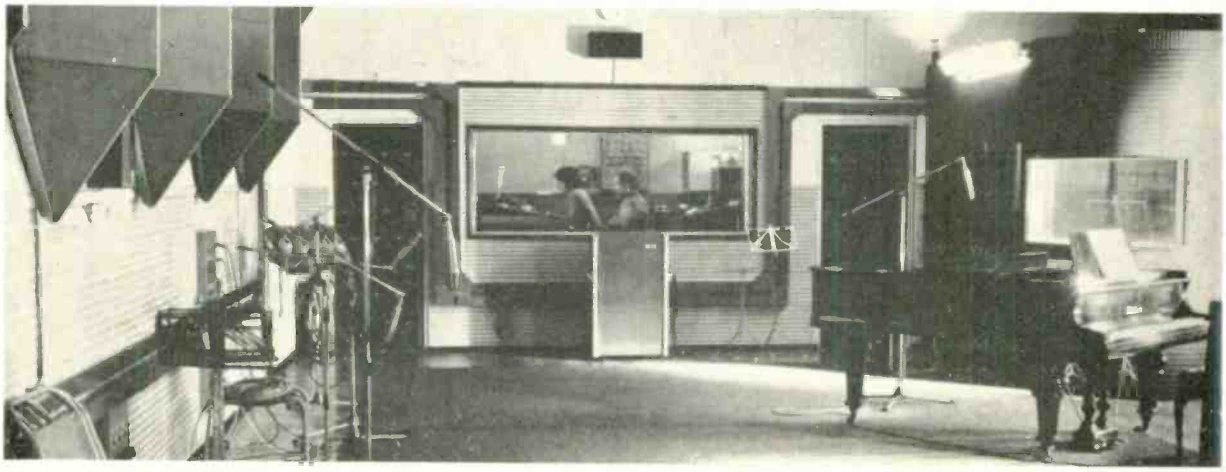
Geoff Calver joined the studio from Pye at the beginning of August and one of his first jobs was to do overdubs of a group called Justin and Wilde for Pye Records. Marquee tell me that, after a couple of slack weeks—which one can expect anyway at this time of year—they are back to being fully booked, which is always good to hear.

Sound Developments of Chalk Farm tell me they have just installed a new Ampex *AG440-8* eight track machine. The rates for using the facilities are £15 per hour and a reel of 25 mm tape also costs £15. Sound Developments have also built a ten input stereo mixer, designed by Harry Day, for use as commercial broadcasting console. As part of the setup they have fitted two Gates turntables with Gray arms, two NAB broadcast standard cartridge machines and a stereo EMT plate.

The studio has recorded Bacardi and Brillo ads for the J. Walter Thompson advertising agency as well as a Mike D'Abo promotional session for his new American album; which is called *Down At Rachel's Place*. The session, recorded in mono, was also filmed. Ivor Raymonde has used the studio for a number of sessions.

Great changes, I am told, have been wrought at **Gooseberry**, the Gerrard Street studio owned by Peter Houghton. In the last six weeks Mike Day, the studio manager, has almost rebuilt the place, he tells me. Mike, himself a drummer, has made the acoustics of the drumming booth 'really dead' by putting 2.5 cm wads of foam in the ceiling and 7 mm wads on the walls, and

continued 53



R. G. Jones, Wimbledon

Akai VTS-110 and VT700

MANUFACTURER'S SPECIFICATION (VTS-110)

Battery portable helical-scan video recorder employing 6.25 mm tape.

System: Twin rotating heads, frequency modulation
Recording time: 24 minutes.
Tape speed: 23.85 cm/s.
Spool capacity: 13 cm.
Resolution: 200 lines.
Signal-to-noise ratio: 40 dB video; 38 dB audio.
Video input: 1.4V p-p at 75 ohms.
Audio input: —65 dBm at 600 ohms.
Audio bandwidth: 100 Hz to 10 kHz.
Fastwind time: Six minutes.
Power source: 12V dc (two rechargeable 6V batteries).
Power consumption: 14W (without camera or monitor).
Dimensions: 255 x 112 x 263 mm (whd).

Manufacturer: Akai Electric Co Ltd, 12-14 2-chome, Higashi-Kojiya, Ohta-ky, Tokyo, Japan.

UK Agent: Rank Film Equipment, PO Box 70, Great West Road, Brentford, Middlesex.

THE Akai system consists of a very compact battery portable video recorder with instant playback facilities, a compatible mains recorder with a large reel capacity, and a range of accessories. The vtrs are unique in their use of 6.25 mm tape, and this makes them interesting variations from the 12.5 mm systems with which they compete in price and performance.

The video synchronising and sound tracks together occupy about 1 mm of tape width, leaving a mere 5.25 mm for the slanting video track. The next important measurement is the video head drum diameter, which at 76 mm is smaller than any other type of twin head vtr and enables the battery portable to be more compact than its competitors. The video head writing speed is also proportional to the drum diameter and on European 50 Hz standards is simply 76 x 25 π = 600 cm/s.

The writing speed sets an upper limit to the resolution or video frequency range; with normal heads this would be about 1.7 MHz or 140 lines whereas 2.5 MHz or 200 lines is specified and this difference must be attributed to the Akai fine gap ferrite heads.

For a given tape width and drum diameter, the linear tape speed is fixed by the video track width which, for the Akai, is 0.193 mm (i.e. 0.15 mm plus 0.043 mm spacing) and the linear tape speed of 23.85 cm/s can be derived from the above figure by some tedious trigonometry. This fairly wide track is probably chosen to give greater reliability and compatibility between machines, albeit at the expense of playing time and tape economy. Although the Akai is very economical when compared with say the Sony *CV2100*, it is consequently not better than either the Shibaden *SV700* series or the newer EIAT 12.5 mm recorders (which, apart from their wider bandwidth, record colour). Based on retail prices for tapes, the figures are:

Sony *CV2100* 18 cm reel (40 minutes): £10.75.
Shibaden *SV700* 18 cm reel (70 minutes): £10.80.
Akai *VT700* 27 cm reel (90 minutes): £16.

Both retail prices and discounts may vary but the comparison will still be valid.

The *VT110* portable recorder together with the *VM110* video and sound monitor fit neatly into a leather case a mere 355 x 265 x 44 mm and can be carried on a shoulder strap as easily as a Nagra *4* sound recorder. Loading the 12.5 mm reel of tape was quite easy, despite the dual capstan and over-lapping reels. Sound and vision levels are maintained automatically. There are no meters and, as the lens focusing and zoom are the only operational controls, the system really is simple to use. In common with other portables, the vtr can be left on record standby and started by the 'trigger' on the camera grip (or from any other remote switch).

A small amount of use exposed two annoying features: firstly the three-digit tape counter was hidden by both lid and recorder case. Secondly, as the monitor screen with the vtr gave a far better idea of the recorded picture than did the camera optical viewfinder, one was constantly moving ones eyes backwards and forwards between the recorder and the camera.

On bringing these problems up with the distributors, it was found that a new camera with built-in electronic viewfinder (Model *VC115*) had just been released. (Both the Sony and Shibaden portable cameras have these miniature crt view-finders built-in and they really are essential for non-studio use.) It is also worth noting that the *VC115* camera will

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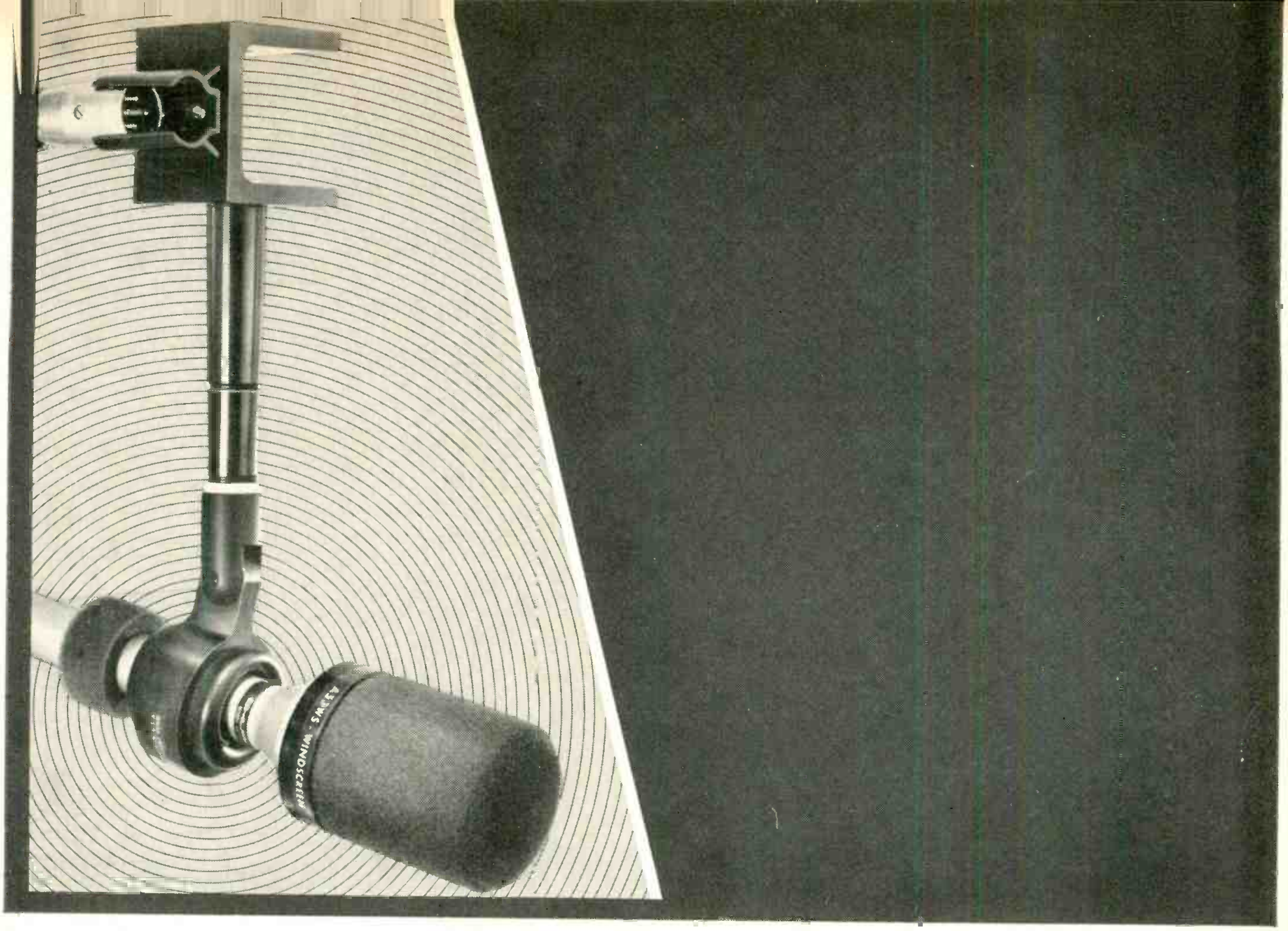

The ubiquitous SM53.

The *Shure SM53* professional unidirectional microphone is seen with increasing frequency in the best of company because it affords eight distinct performance advantages: (1) a wider front working angle with uniform tonal quality; (2) effective noise rejection through a true cardioid pickup characteristic; (3) a built-in shock mount for effective mechanical noise isolation; (4) extraordinary ruggedness for performance consistency after severe shocks; (5) a superior hum rejection system; (6) an integral breath "pop" filter; (7) a minimized proximity effect for constant tonal quality; and (8) full field serviceability. Interested?

Shure Electronics Limited
84 Blackfriars Road
London SE1 8HA, England

SHURE

continued

monitor the playback so that, with an earphone for sound, one can dispense with the *VM110* monitor and have a portable recorder only 255 mm square.

As questions about the picture quality attainable from 6.25 mm tape are most likely to be the first ones asked about the Akai system, this was checked quite carefully. As explained earlier, the attainable resolution is limited by the writing speed and the 200 line figure specified is *just* met; that is to say 200 lines is the limiting resolution and finer detail is not reproduced at all. **Fig. 1** shows an 'off-air' recording of BBC testcard *F* and the 2.5 MHz frequency grating (right of circle, 2nd block from top) is visible. **Fig. 2** is a Marconi camera test card which (shifted vertically for another test covered later) just shows the 200 line section of the resolution wedge. A decision as to the suitability of this should be made with care and be based on actual tests. For example, in drama schools, used as a visual notebook in filming, or as an addition to an estate agent's still camera, the picture is detailed enough to satisfy most users. For recording a more complex picture with writing or technical detail, the Akai system may not be sharp enough.

Aside from resolution, recordings were found to be more stable than most 12.5 mm vtrs and this is probably due to having the head switching in the bottom ten lines of picture rather than (normally the case) during field blanking. The switching disturbance is not visible on most overscanned domestic or industrial monitors, but can be seen by shifting the vertical sync.

Video signal-to-noise ratio was an adequate 38 dB but a small section of the tape, recorded before the machine was handed over, was about 40 dB so adjustment of bias or playback equalisation might improve on the 38 dB figure.

Unless special techniques are used (as in the Sanyo *100 SL*) there is always some vision breakup when still-framing a vtr, and the Akai is about average here. This is a function of format rather than machine; a typical still frame is shown in **fig. 3**.

Joins between shots caused the expected field disturbance on playback (akin to a splice on films) and the drum servo took between one and two seconds to regain a steady picture.

The agc on the *VT110* made some audio measurements difficult so a Sony *ECM21* microphone was used to record samples of speech which were then replayed through a good sound system. Although clear and distortion-free, some field frequency buzz was detectable; it measured 36 dB below 320 nWb/m. Wow and flutter measured only 0.12 per cent peak, DIN weighted.

The tv tuner unit connects between the *VM110* monitor and the vtr, producing a combined tv receiver/recorder unit of comparable sensitivity to a domestic television (150 μV for 28 dB signal-to-noise ratio). As this unit appears to consist of an rf and an if module in a box, the cost did seem rather high. (There are several complete uhf television receivers using the same components which cost less.)

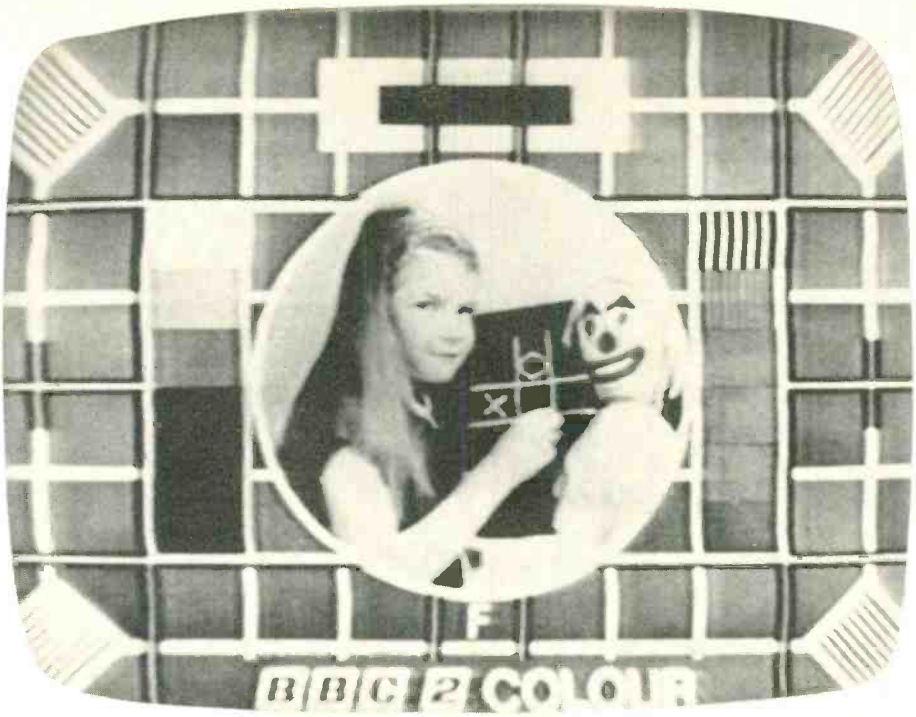

Fig. 1

The rf adaptor converts sound and vision signals to a modulated uhf output that can be plugged into the aerial socket of a domestic television; the model tested produced correct sound but negative pictures. I doubt if this fault would escape the normal pre-sales tests.

In contrast to the *VT110*, the Akai *VT700* mains vtr was a heavy 20 kg but the 27 cm reel capacity, outer rotor spool motors and dc servo motor drives for both the tape transport and the head drum justify this weight. The video performance was identical to the *VT110* and tapes were interchangeable with the latter without need of tracking adjustments, but with a slight increase in video noise. Sound performance was better than the *VT110* and the switchable agc allowed measurements. Record/play frequency response was within ±3 dB from 70 Hz to 12 kHz. 0 VU on the meter (which reads playback as well as record levels) corresponded to 320 nWb/m and the video tape

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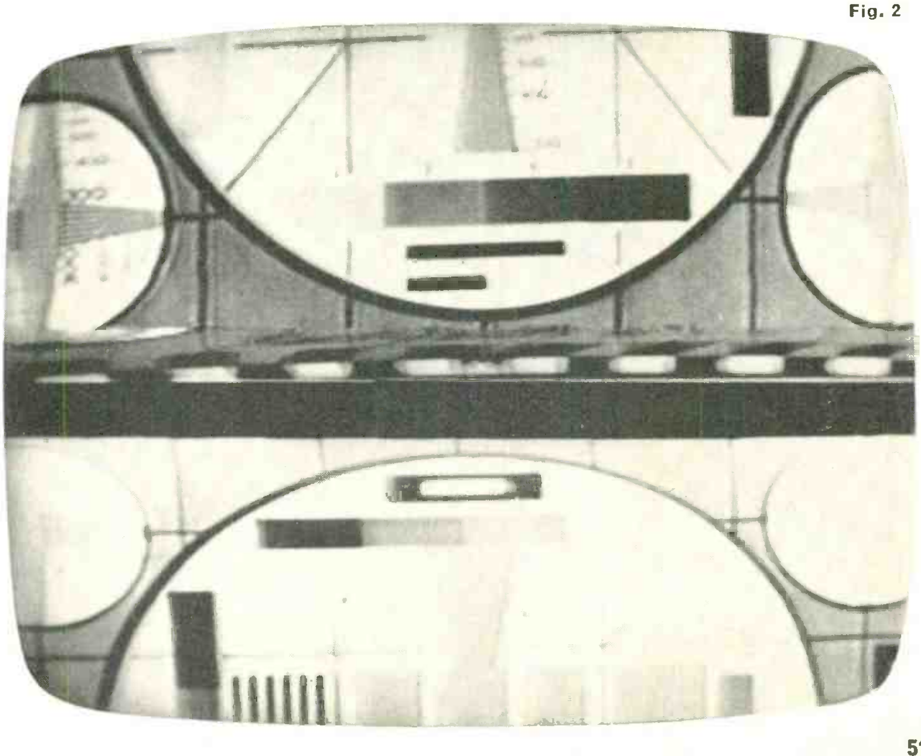
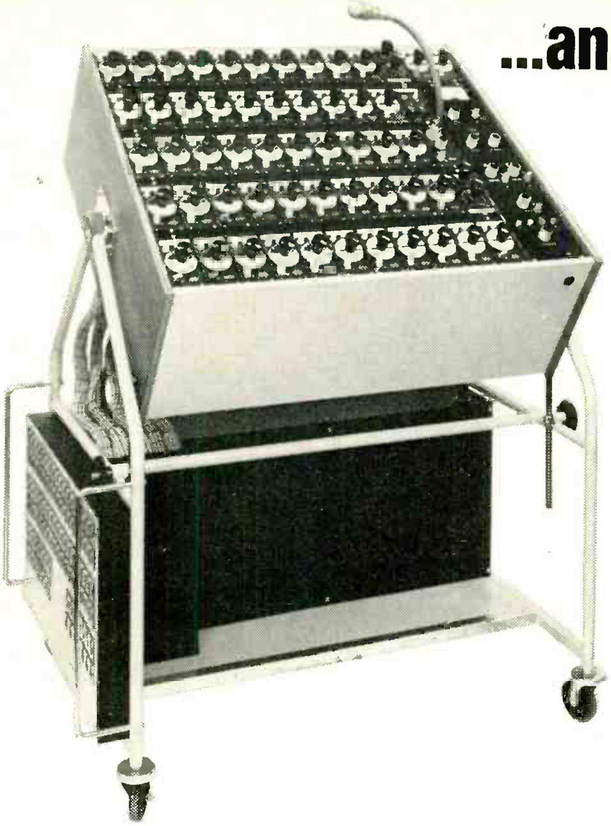
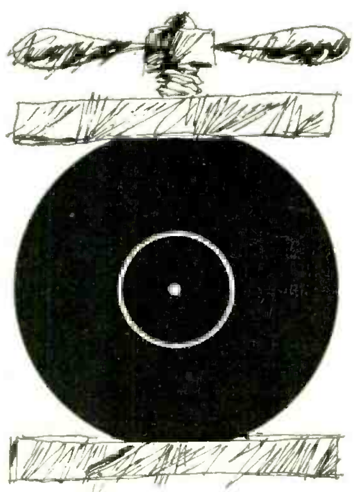
Fig. 2







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characteristics prevented recording at greater than about 6 dB above this. Unweighted noise measured 49 dB below 0 VU, which is very good considering the tape used and the 0.35 mm sound track width. Wow and flutter measured typically 0.05 per cent and never exceeded 0.08 per cent peak DIN weighted. As the above figures suggest, the *VT700* was able to do justice to music recordings and good broadcast tv sound.

The piano key controls were as light and positive as microswitched solenoids always are; the tape transport was extremely smooth and efficient and rewound a 45 minute tape in two minutes. Separate metering for sound and vision levels, two-position switched tape tension, and facilities for remote control put this machine a cut above the average 12.5 mm vtr with its single knob control and single drive motor coupled to the spools and capstan through belts and rubber wheels. The solenoid operation and remote control, together with the 90 minute playing time, suggest two particular uses for the *VT700*. Firstly, time switched off-air recording. Secondly, making a visual record of work in a sound studio, for which it could be run in parallel with the master sound recorder.

6.25 mm tape must be considered a success for both types of machine although it would need quite an improvement in tapes and video heads to squeeze an acceptable colour recording on to this format.

New types of battery portable recorders will soon also be offering instant playback facilities, together with greater resolution, but, if a really small and lightweight recorder is necessary, the Akai *VT110* is unlikely to be superseded for a long time. The characteristics of the *VT700* mains recorder also put it ahead of much of the 12.5 mm competition. Where extreme ruggedness, low tape costs, and remote control are needed, there are no alternatives at the moment.

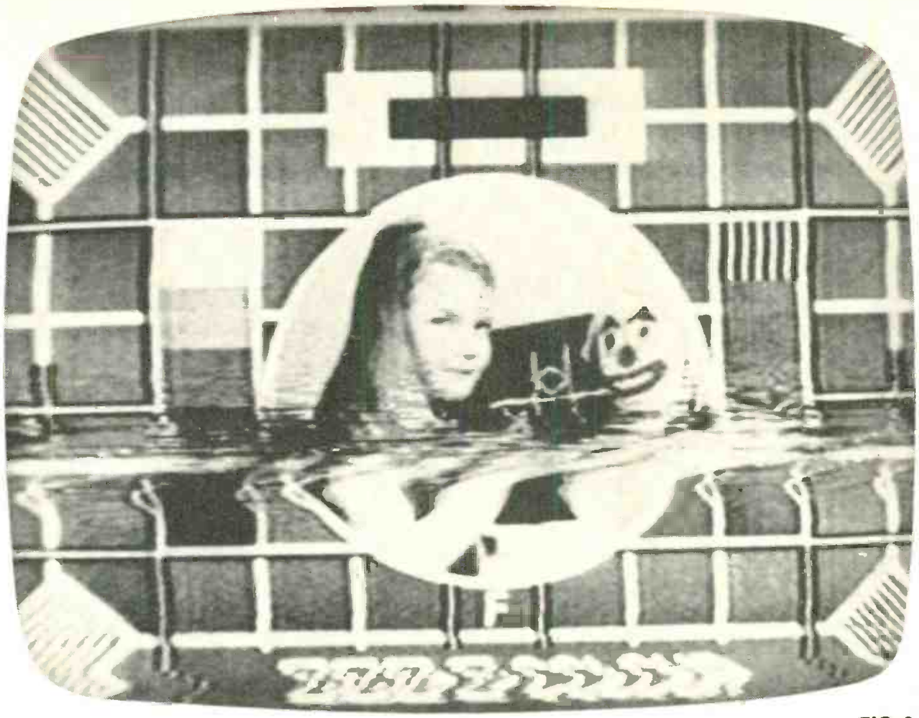

FIG. 3

DIARY

continued

he has covered the lot with hessian. The studio recently bought an AKG *BX20* reverberation unit, and Mike says he is very pleased with it.

Mike tells me that Gooseberry's reputation is improving all the time and they are now getting quite a bit of work from being recommended by people who have used the studio. Among others, Malcolm Koss of World Wide Artists came into produce a group called Sacrifice. The session was to make a record for a choreographer to use at rehearsals. Gooseberry have also recorded the dance sequences for a concert of classical music by a Chinese composer, which will shortly be held at the Festival Hall.

The sound installation for *Jesus Christ Superstar*, now running at the Palace Theatre, London, was provided by **Theatre Projects** for a total cost in excess of £20,000. David Collison, who was engaged for the work by the Robert Stigwood Organisation, said that his aim was to recreate the sound of the original lp recording. Each of the 25 musicians in the orchestra pit has a microphone and there are another eight for the rock group on stage. 'For the cast there are motorised rising microphones, suspended microphones, long range microphones and many hand held mics,' he said. Altogether 80 microphones will be used and a mixing desk has been built for the two engineers who will be needed for the show.

Turning now to the film world, I learn that sound mixing and dubbing engineer Hugh Strain is to join **World Wide Pictures** at the end of September. His past works includes dubbing on *Lawman*, *Get Carter*, *One Day in the Life of Ivan Denisovitch* and *The Nightcomers*. He will be working in the new World Wide centre in St Annes Court, off Wardour Street in London.

From Anvil's Denham studios I hear that they have been scoring for six pictures as well as making a *Top of the Pops* album for Pickwick and finishing records of eight Gilbert and Sullivan operas for BASF. The film scores were *Death Line*, music by Jeremy Rose; *The Great Waltz*, by Roland Shaw; *Carry On Abroad*, by Eric Rogers; *Lady Caroline Lamb*, by Richard Rodney Bennet; *The Sisters*, by Bernard Herrmann, and *Kronos*, by Laurie Johnson.

Because of the newspaper strike you may not have heard about the fire at London Weekend's Wembley studios. It started late on July 24 and stopped the clocks at six minutes past midnight. Most of the damage was done in the **Intersound** recording studios part of the building, which was gutted. The Neve desk in the control room was perhaps the least damaged of the equipment there but was nevertheless unusable.

Whatever its cause the fire has lent an air of melodramatic finality to the closing of the Wembley studios that would be worthy of the plot of one of Weekend's own soap operas. In January the owners of the site—Rediffusion Television, who are part of British Electric Traction—sold it for £2,250,000. Its book value at the time was put at £1,600,000. The new owners are Siege Estates, a company which is half owned by Peter Davies, a developer, and half owned by Metropolitan Property Holdings, part of the Stern Family Holdings group. Siege Estates said that no decision had been made as to how the studios would be used.

The new LWT building, built on a site owned jointly by the GLC and Coal Industry Nominees, has cost £7,000,000 to build. LWT has taken out a 125 year lease on it but will sublet some of the office floors of the 25 storey tower block and probably the five flats in the building as well. It has three main studios, the largest of which has permanent seating for 250 and room for 250 more. There are also dressing rooms, restaurants, smaller studios and technical areas.

But when LWT moved to the new building on the south bank of the Thames, early in September, the commercial recording venture, Intersound, was left behind in the ashes. The last recording session was held at the beginning of July and there are 'no immediate plans' to revive the project at the new headquarters.

Alan Evans told me that, before the fire and the closure, they were having to turn business away. Although I understand there have been no redundancies because of the move, there is less space at the new headquarters than there was in the three old buildings. Intersound staff will be wholly occupied in television work from now on.

Intersound had a large commitment to sound recording for Weekend itself, of course. In

continued 60

Designing a Recording Studio

By Kenneth Shearer

THE technological advances of recent years have brought about many improvements in the standard of equipment used in recording studios today but nevertheless many well equipped studios suffer from basic faults which severely complicate the work of both artist and recording engineer. This article outlines some of the more important considerations for either the design of new studios or the improvement of existing facilities.

Ambient noise level

With the wide dynamic range of present-day recording equipment it is essential to limit the ambient or background noise level in a studio to below the iso-noise rating curve NR.15. However, this criterion only applies to steady continuous noise devoid of discrete tones or transients due to impacts. The ambient noise spectrum is perhaps analagous to electrical circuit noise which is steady and characterless. This means that any discrete tones and transients (such as those caused by ducting, expansion and contraction of pipes and noise from light fittings) must be limited to a level at least 5 dB, preferably 10 dB, below the NR.15 octave-band criterion.

This can be achieved by designing to a '⅓ octave-band equivalent' of NR.15; i.e. by measuring the ambient sound pressure level spectrum (in dB ref 2 x 10^{-5}N/m²) in ⅓ octave bands and comparing this with a criterion reduced by 5 dB (i.e. 10 $\log_{10}3$). For rooms used with monitoring facilities (such as control rooms and copying rooms) the ambient noise level can be raised to NR.25.

Fig. 1 shows the two design criteria: NR.15 for studios and overdub booths, etc; NR.25 for monitoring rooms. These are plotted as octave-band sound level criteria and should be lowered by 5 dB if measurements are to be made in ⅓ octave bands.

Airborne sound insulations: Walls

Road, rail and air traffic are the principal sources of extraneous airborne noise and each of these is capable of producing a high noise level over a very wide frequency range. This usually renders single-shell studio construction inadequate. The sound insulation afforded by an airtight single-leaf wall or roof is generally a function of its mass and tends to increase by about 5 dB for each doubling of the weight per unit area. Thus the sound insulation afforded by a 114 mm brick wall is around 5 dB worse than that provided by a 228 mm brick wall. The sound insulation tends to improve by 5 dB/octave with increasing frequency.

Fig. 2 shows the theoretical sound insulation value or sound reduction index (R) for 114 and 228 mm single-leaf brick walls, ignoring the variations which inevitably occur in the gradient due to the effect of restraint by cross-walls, resonances, etc.

Since traffic is capable of producing octave-band sound pressure levels in excess of 90 dB at low frequencies (i.e. in the 63 Hz and 125 Hz bands) it is apparent that it would not be practicable to construct a single-leaf wall or roof with sufficient mass to give an adequate degree of sound insulation. It would, of course, be even less practicable to construct a single-leaf wall between a control room and a music studio where the sound of an amplified bass guitar can attain a level approaching 120 dB in the 125 Hz band.

It is therefore necessary to resort to double-leaf construction. Unfortunately the predicted sound insulation by the mass of each leaf cannot simply be added together since the leaves are coupled by the air in the cavity which tends to act as a spring. A double leaf wall (or roof) can thus be considered as two masses coupled by a spring. This mass-spring-mass system is analagous to a pendulum suspended on an infinitely long string, coupled to an adjoining identical pendulum by means of a spring as shown in **fig. 3**.

It is obvious that this system, if mechanically excited, will exhibit a natural frequency of resonance. If one pendulum is driven at a frequency below that of natural resonance, the second pendulum will tend to follow it with a greater degree of swing. As the driving frequency is increased, the swing amplitude of the second pendulum will progressively increase until, at the natural frequency of resonance, it will tend to approach infinity (i.e. the system will act as an amplifier).

If the driving frequency is further increased, the amplification will become less, reaching unity (i.e. no insulation) at the square root of twice the resonant frequency. From this frequency upwards, the transmitted amplitude will decrease progressively.

The theoretical sound reduction afforded by two structurally-unconnected 114 mm brick walls separated by a 153 mm cavity is plotted in **fig. 2** for comparison with that afforded by

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FIG. 1 RECOMMENDED OCTAVE-BAND AMBIENT SOUND PRESSURE LEVEL CRITERIA FOR RECORDING STUDIOS AND MONITORING ROOMS

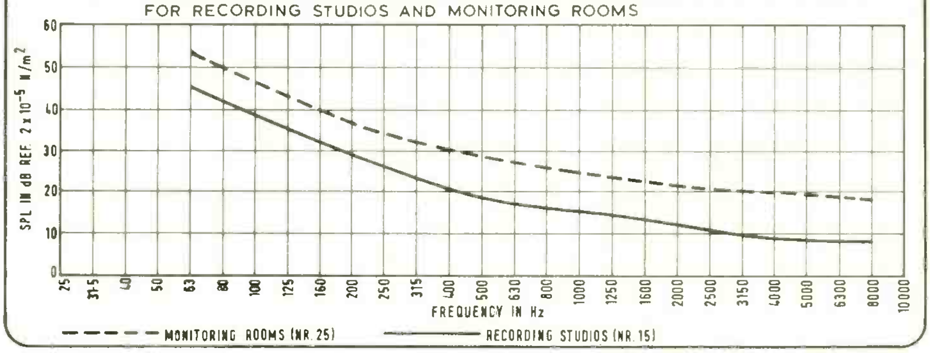
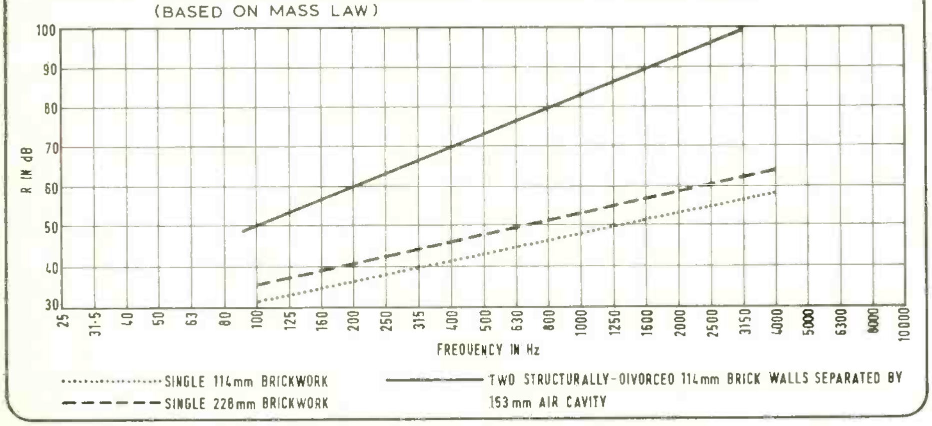


FIG. 2 THEORETICAL SOUND REDUCTION INDICES (R) OF BRICK WALLS (BASED ON MASS LAW)



singleleaf walls. It can be seen that the sound reduction index now increases by about 10 dB per octave above the resonant frequency, giving greatly improved sound insulation at high frequencies but only about 15 dB improvement at 125 Hz. The resonant frequency is proportional to the inverse of the square root of mass multiplied by the cavity width ($f_r \propto 1/\sqrt{M \times W_c}$). Thus it is necessary to increase, by a factor of four, either the thickness of each wall or the cavity width to lower the resonant frequency by one octave, thereby gaining 5 dB sound insulation at 125 Hz. It is usually feasible in practice to provide sufficient traffic noise attenuation and acoustic separation between studio and associated rooms by ensuring the following:

(a) structural room separation.
(b) the elimination of crosstalk via doors, windows, ventilation ducts and all apertures.

Doors

It is possible to construct a door with similar sound insulation to that of a brick wall. However, its weight and the mechanism required to attain an airtight seal would be such that considerable time and effort would be required to open and close it. In addition to this, a serious sound leak would occur each time the door was opened. Therefore it is preferable to employ two or more comparatively light doors (say 4 kg/30 cm^2) separated by lobbies sufficiently large to attain the required degree of sound insulation using the mass-spring-mass principle. In order to avoid losses in efficiency due to the build-up of reverberant sound in the lobby, the walls and ceiling need to be treated with a sound-absorptive material of sufficient thickness to damp out the lowest modes of standing-wave resonance in the space. The doors should be fitted with either a magnetic or very compliant seal to ensure a good air-seal. Magnetic seals are to be preferred for the lack of physical effort needed to cause sealing around the entire periphery. However, even the remote possibility of degaussing part of a costly recording tape by walking through the doors close to the seal should not be ignored when the choice of sealing device is made. Overall, perhaps the compliant-type door seal with a door-closing device is nearest to the ideal and will have the added advantage of guarding against noisy impacts when the door is closed. In any set-up the door frame should, of course, make an air-tight seal with the door-stop but compliant door seals do assist this.

Windows

The observation windows present acoustic separation problems, particularly when they are provided between two rooms which are occasionally used for different projects at the same time (e.g. studios, overdub and reduction rooms).

The sound insulation value of double windows depends upon the mass-airspring-mass principle in a similar way to double-leaf walls. Glass of practicable thickness is relatively light-weight and this necessitates the use of large air cavities. The two panes should be mounted non-parallel to avoid losses in efficiency due to standing-wave resonances between them, it not being very sensible to use the usual technique of sound absorptive material on the inner faces! However, the effective cavity between the panes can be increased by the use of coupled spaces and the top, bottom and sides of the cavity can be lined with an appropriate sound-absorptive material.

Inevitably the window offers less sound insulation than the surrounding walls. This effect can be minimised by the use of small windows but the present tendency is towards larger windows offering a greater field of vision. It is therefore necessary to design the window frames in such a way that the glass is held very stiffly so that the glass can 'borrow' mass from the walls.

Since it is impossible to guarantee that dust will not accumulate on the inner faces of the panes, it is advisable to arrange for access to the window void; either by opening one of the windows or by providing an entry from the void between the rooms.

Sheet glass is extremely elastic and has little inherent damping. Thus bending waves tend to occur at certain frequencies, causing the glass to 'ripple'. This phenomenon, known as 'coincidence', can seriously reduce the sound insulation of the window at a series of discrete frequencies determined by the thickness of the glass. For this reason it is usual to employ a different glass thickness for each of the two panes, partially 'cancelling' these out. This phenomenon does not occur in the case of laminated glass such as 'Triplex', due to the damping effect of the centre lamination, but this type of glass is at present only available in comparatively small sheets and would require the introduction of mullions. While comparatively narrow mullions do not significantly interfere with the field of vision, they can cause strain due to the tendency of the eyes to focus upon the window plane instead of the more distant objects on the other side.

Structural noise transmission

The most common cause of unwanted noise in many existing recording studios is the transmission of mechanical and impact sounds from other parts of the building. Unless the shell of the studio is structurally divorced from the remainder of the building, sound can be transmitted virtually without loss via the building structure eventually vibrating the wall, ceiling and floor surfaces of the studio to produce airborne sound. Among the most common sources of this type of noise are the following: lifts, ventilation and refrigeration plants, office machines, heating pumps, the expansion and contraction of heating pipes, surface and underground trains, workshop machines, door closure impacts, and footsteps.

Many of these structureborne noises can, of course, be dealt with at source but often the noises originate in adjoining premises where no noise control measures can be employed.

The structural isolation of a studio is not a simple matter. Firstly, it is usually necessary to construct the floor, walls and ceiling as an integral structure and to support this clear of the main building with damped steel springs or rubber-in-shear anti-vibration mounts. Here again is the mass-spring-mass principle (see **fig. 3**). If we make the springs very 'soft' (i.e. aim for a large static deflection under the dead load of the studio) we will attain good insulation at low frequencies. However, the studio construction will need to be very stiff and heavy with normal foot traffic causing a sensible movement—and possibly audible vibration of light objects.

If comparatively 'stiff' springs are used (i.e. we aim for a small static deflection) the natural frequency of the system may be sufficiently high for amplification of some lower frequency noises.

A compromise usually has to be achieved between the degree of insulation and the tolerable amplitude of deflection. For this reason it is preferable to employ a stiff sub-frame supported upon a comparatively small number of anti-vibration mounts and to provide access to these for the purpose of checking the static deflection at completion and replacing them if necessary.

Similarly the voids between the studio walls, floor and ceiling should be sufficiently large to facilitate access to check that the 'floating' structure has not become structurally connected to the main building or other 'floating' rooms, by builder's debris, pipework, conduits, ductwork or temporary struts. An access void is also useful for running additional audio and power lines to any point in the studio.

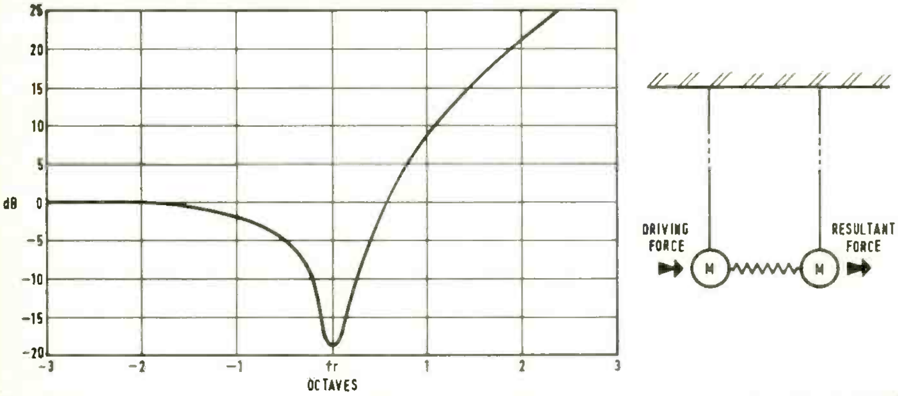
Ventilation

Since the sound insulation requirements of recording studios preclude natural ventilation via open windows, mechanical ventilation is essential to provide sufficient fresh moving air for bodily comfort and to dissipate the heat produced by the occupants, lighting and equipment. Adherence to an ambient noise level not exceeding NR.15 often results in a ventilation contractor installing large, slow-speed centrifugal fans coupled to large ducting designed to maintain very low air velocities in the ductwork. Although this arrangement produces comparatively little noise at middle and high frequencies, the combination of a centrifugal fan and ducts with large cross-sectional areas tends to create large-scale turbulence in the airflow and consequently low-frequency drumming of the ducts.

It is comparatively easy to attenuate high frequency duct-borne noise by means of sound-absorptive duct lining, splitter units, etc, but low frequency duct rumble is very difficult and costly to reduce. Of course, this type of installation would be considered very quiet in an office or even a cinema but the response of the human ear is such that comparatively low levels of low frequency noise become much more noticeable when a recording of a quiet passage is replayed at a higher level.

For this reason it is preferable to employ small high-speed axial fans for studio ventilation. This type of fan produces a high level o mid and high frequency noise but due to its axial disposition produces comparatively little low frequency noise. Smaller ducts carrying air at higher velocities tend to produce more airflow noise but the small dimensions preclude low frequency drumming and conventional duct lining or proprietary noise-attenuator units are usually adequate to deal with this noise which, in any case, is less than that generated by the fan. Terminal attenuators are invariably

FIG. 3 INSERTION LOSS OF A MASS–AIRSPRING–MASS SYSTEM SHOWING dB AGAINST OCTAVES RE RESONANT FREQUENCY – fr



required at the point of entry and exit via the studio shell to prevent the ingress of extraneous noise via the aperture and duct wall and these can serve equally well to attenuate airflow noise.

The size of the fan and ducting can be further reduced if air conditioning is employed since about 80 per cent of the airflow in an ordinary ventilation system is employed to dissipate heat. The use of a lesser quantity of cooled humidified air tends to reduce the weight and space occupied by the fans and ducting, also the size and cost of the noise-attenuation measures. In many cases, most of the noise attenuation can be achieved by installing the thermal insulation inside, rather than outside the ducts.

By careful design it is possible to install a relatively silent air-conditioning system for a cost only slightly in excess of a conventional ventilation system while maintaining a much closer control over temperature and humidity in the studio, and other noise-critical areas.

Conventional horizontal louvre-type diffusers with integral dampers should be avoided as these tend to 'buzz' at certain musical tones, particularly those produced by amplified bass guitars. The drum-louvre type of felt-mounted diffuser does not suffer from this disadvantage. It is, of course, advisable to acoustically-damp any plenum chambers or large ducts immediately behind diffusers and extract apertures to avoid feedback.

In order to avoid the possibility of crosstalk between rooms via the ductwork it is advisable to run separate ducts from the plant to each noise-critical room. This arrangement has the added advantage of facilitating individual temperature and humidity control in each room and allows the economy of switching off the air-conditioning in unoccupied rooms and studios while maintaining optimum conditions in, for example, the tape store.

Mechanical and Electrical Equipment

It is conventional to mount ventilation fans, pumps and other machinery on a concrete 'inertia block' and to then insert anti-vibration mounts between this and the building structure. The noise transmission is, of course, considerably reduced if the plant is mounted on the lightest framework consistent with stability and the concrete added to the mass of the floor or roof supporting the plant. In other words there should be the greatest possible impedance mismatch between the plant and the building structure. This is another reason for using axial flow ventilation fans since, for the same duty, they are lighter, smaller and, by virtue of plastics impellers, are generally balanced to closer tolerance than centrifugal types. The higher speed also results in higher frequency vibration which can be more effectively isolated by spring or rubber-in-shear anti-vibration mounts.

Fluorescent lighting is preferable to tungsten because the higher efficiency results in less heat generation. However, a large amount of 100 Hz noise (with the attendant harmonics) can be produced by loose choke laminates and the other electrical equipment normally housed in fluorescent light fittings. It is therefore important to locate this equipment in a ventilated space outside the studio, leaving the fluorescent tubes mounted, for example, by means of 'terry' type clips. Clips should be spaced at centres not exceeding about 600 mm and should preferably be lined with thin asbestos tape to damp tube resonance. Earthing busbars should be fixed to lighting recesses by means of screws *and* impact-type adhesive to avoid the possibility of music-induced resonance.

If dimmer transformers are employed, these usually need to be located within about 5m of their respective tubes but should nevertheless be outside the studio if possible. If this is not practicable they should be fixed by means of anti-vibration mounts to a massive element (for example the ceiling) of the structure.

Room proportion

If we consider a completely bare rectangular room in which sound is produced either by voice, musical instruments or loudspeakers, the level of the sounds will be increased at certain frequencies by the excitation of the natural modes (or rigentones) of the room. These occur when half the wavelength of the sound equals one of the room dimensions, i.e. when $\frac{\lambda}{2}$ = W or H. The interaction of these wave patterns produce modes at lower frequencies (equivalent to 'beat' frequencies) and the series continues for L, W or H $= \frac{2\lambda}{2}, \frac{3\lambda}{2} \ldots \frac{n\lambda}{2}$

It is apparent, in rooms where the highest common factor of L, W and H is large, that many of the modes will occur at the same frequencies resulting in pronounced resonances at a few frequencies. It is therefore preferable to select room dimensions with the smallest possible HCF in order to spread the resonant frequencies as evenly as possible. It is then only necessary to render the wall and ceiling surfaces non-parallel in widths not less than twice the HCF to avoid 'flutter' effects. Flutter echoes usually result in 'ringing' tones following transient sounds.

Reverberation time

The optimum reverberation time of a music recording studio depends upon the type of music to be recorded. Many small studios are too 'dead', i.e. the Rt is too short. Without some reverberant feedback from the studio, a musician or singer cannot hear himself sufficiently well to control his tonal quality. For the same reason it is advisable to incorporate a series of highly reflective randomly-spaced and angled surfaces to provide a degree of 'kickback'. These features affect the quality of the final recording partly by affording the optimum environment for the microphones and partly by the more subtle effect upon the performance of the artist, group or orchestra.

The reverberation time of rooms employed for control, reduction, mixing, etc, where the engineer operates his equipment in response to his subjective assessment of the sound from monitor speakers, should preferably be in the region of 400 ms from the lowest possible frequency up to about 10 kHz to simulate the conditions in an average domestic living room since the ultimate customer assesses the

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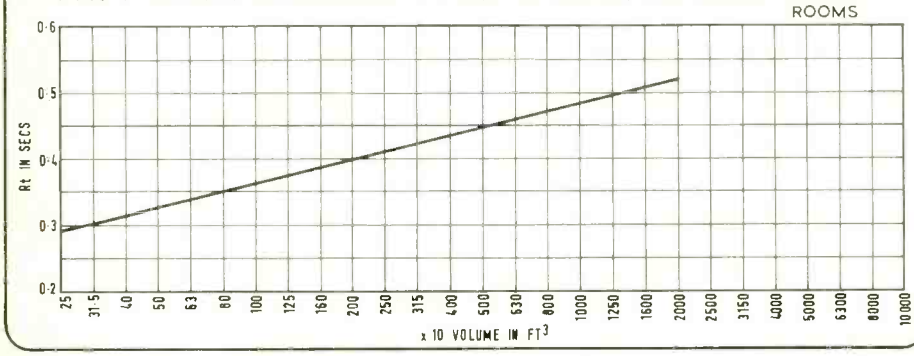
FIG. 4 OPTIMUM REVERBERATION TIME (Rt) FOR CONTROL AND SIMILAR MONITOR ROOMS



Internavex 1972

ORGANISED by the National Committee for Audio-Visual Aids in Education and the Education Foundation for Visual Aids, Internavex 1972 was held from July 25 to 28 in the National Hall, Olympia.

Animatic showed audio-visual presentation equipment suitable for the unattended control of slide projectors, dissolvers and dimmers. The basic equipment comprises *AN.P* projector controls units, ranging from a single-rate dissolver at £80 up to a two-rate presentation unit with cassette recorder, 10W amplifier and loudspeaker at £295. Two tape synchronisation encoders are offered: the £15 single rate *AN.S1* and £25 two rate *AN.S2*. These are supported by a range of mixers (matching high and low impedance microphones, magnetic and ceramic gram pickups and high impedance 'diode'), plus a series of 25W to 100W power amplifiers.

A complete 35W film and tv sound studio weighing some 28 kg was introduced by Evershed Power Optics, agents for Perfectone magnetic recorders. The *Bauer P6* (made by Robert Bosch, Stuttgart) provides 16 mm projection and synchronised twin track magnetic sound facilities in a single console with 20 different combinations of optical and magnetic recording, dubbing and mixing. Speeds are 24 or 25 f/s.

Clarke & Smith will be a familiar name to any readers who have indulged in servicing the RNIB's *Talking Book* machines. These, like the new Package Learning System shown on Stand 41, are based on the *1030* tape magazine containing 6.25 mm tape on small cine spools. The *1036* high speed copier produces six magazine dupes from a master tape in a total of 15 minutes.

A new general-purpose television camera for the cctv market was introduced by EMI. The 625/525 line *Surveyor* is suitable for remote operation and low light working. Other features include an intrinsically weatherproof casing and a layout permitting easy operation, installation and maintenance. Basic price of this monochrome unit is £300. Another monochrome camera on display was a new version of the *2004*, capable of using 26 mm Plumbicon or vidicon tubes. The *2005* three-tube colour broadcast camera was demonstrated, reflecting the introduction of colour broadcast systems into educational television (Aberdeen University were recently supplied by EMI with a colour vtr vehicle including three *2001* cameras). Continuing their role as distributors of Sony cctv equipment, EMI displayed 25 and 12.5 mm colour and monochrome vtrs and the colour video cassette recorder. On the audio side the recently introduced EMI *816* low-noise low-print tape with matt-coated backing. This is available in 6.25, 12.5, 25 and 50 mm widths and is supplied on 27 cm spools.

Video control desk in the Link Electronics advanced television mobile, shown for the first time at Internavex. The vehicle was built in conjunction with Dell Coachbuilders.

The EVR Partnership (now minus CBS who developed the system) soldiered on despite reports that Rank-Bush-Murphy had halted production of EVR players. An elegant but 'once-only' optical system (colour video stored in parallel frames of luminance and chrominance on monochrome photographic film) is combined with a less elegant audio medium, low speed magnetic stripe. Their demonstration included some particularly grubby material though it was impossible to determine at which stage dirt had entered the system. Curious because earlier demonstrations have been particularly impressive in this respect.

Main exhibit on the Philips stand was the *N1500* video cassette recorder. This incorporates a colour tuner which allows users to watch a broadcast on one channel while simultaneously recording another. An aerial pre-amplifier in the recorder ensures sufficient signal for both the receiver and vcr. Format is helical scan on 12.5 mm tape stored in cassettes of one hour capacity. Tape life is claimed to be sufficient for over 300 runs. The *N1500* is no prototype; it is actually being marketed. Due to restricted supplies, however, Philips will for the next few months concentrate on educational, industrial and training markets. In the audio cassette line, it is worth pointing out that Philips offer a functional battery portable (the *EL3302*) at £20.35 retail.

Full colour capability, exceptional still-frame performance, long tape life and minimal head wear were claimed by Racal-Zonal for their *HV25* range of video tapes. These are produced for 6.25, 12.5 and 25 mm helical scan vtrs in a variety of tape lengths.

Teldec offered the mixture as before, repeating their 1971 demonstration of colour video discs. Picture quality may have improved slightly in the intervening year though it was still reminiscent of a poor-to-average domestic receiver. Even at this standard, however, the medium has an appealing simplicity which should more than compensate for the lack of recording facilities. Prediction: twin-turntable prototypes (overcoming the five minutes per disc limitation) for Internavices 73 and 74 with something to sell by 1975.

On the Uher stand, the *124* stereo cassette recorder (field test next month) shared the limelight with the *4000L* open-reel portable and *Universal* teaching machine. The latter was offered as a home language laboratory in conjunction with tapes produced by PILL (Programmed Instruction Language Learning).

continued over

INTERNAVEX 1972

continued

Courses in French, Italian, Africaans, German and Spanish are offered at £25.50. Russian is also available at £29.50.

Unicol make platforms. More precisely, platforms, columns and bases for optical and audio equipment. Typical prices for standard units are £14.20 for a tape recorder trolley (flush-fitting a Ferrograph *Seven*) and £15.10 for a lectern. Bases, tubular metal columns and platforms are interchangeable, available column lengths being 400 mm to 1.4m (six sizes).

Two items of interest from Video Electronics (VEL) were the Tapecode and Cuecode systems. Tapecode applies to video techniques now being used in industrial audio recorders—writing a continuous chain of index pulses for subsequent time identification or position finding. Elapsed time, 24 hour clock time, denary digits up to 99,999,999, or a binary code output may be recorded on videotape without modification to the vtr and without affecting the picture content. Price of the simplest unit (four digits) is £267 while the eight digit version (binary coded decimal) is £330. Inlaid visible digits may be televised anywhere within the raster by using the *NG1* videonumeric generator (£180). The companion Cuecode system extends the Tapecode to operate mechanical or electronic switches a preselected points in a tape. Applications include assemble editing, dummy editing and process control. Single and multichannel versions are available, the basic Cuecode *CC1* costing £79.

DESIGNING A RECORDING STUDIO

continued

quality at home. However, the average living room has some sections of parallel reflective wall, ceiling and floor which produce flutter resonances; has poor diffusion and often some dimensions which cause 'coloration' of the sound from his replay system. The ideal monitor room should thus ideally represent a 'dimensionless' living room. This can be achieved in control or other monitoring rooms by careful selection of the room dimensions and the use of frequency-selective and diffusive acoustic treatments to damp the modes which would otherwise allow the listener to assess the size of the room subconsciously, and would tend to 'colour' the output from the monitors (see fig. 4).

Monitor speakers

Since a recording is passed from the control room via the reduction room, etc, finally reaching the copying or master cutting room, it is preferable that each room should offer the same dimensionless characteristics and that the monitor speaker systems should be identical in order that the engineers should not have to adjust their assessment of the playback in each room.

Quadraphonic monitoring

In theory, the four loudspeakers of a quadraphonic system should be equidistant from the listener and oriented at 45° to the room axis. Since loudspeakers propagate low frequencies more efficiently when placed in the corners of the room, this implies that the room needs to be square with the listener located at the intersection of the diagonals. However, the distribution of normal modes is very poor in a square room and it would be extremely difficult to damp the coinciding or 'stacking' modes without overtreating the room to the point of excessive 'deadness'.

It would thus seem preferable to employ a room with a length/width proportion of 1:0.8, maintaining the speakers in the four corners. Although, in theory, this would result in an incorrect phase relationship between the front and rear pairs. In practice the rear pair would normally be handling a greater proportion of reverberant sound than of direct sound and this would be random in phase. Listening tests seem to support this argument.

DIARY

continued

fact, the original idea of allowing the studio to be used for commercial recordings was merely to make better use of the large amount of capital which was tied up in equipment. For Weekend has had its ups and downs. There was one point, in November 1970, when the National Board for Prices and Incomes expressed doubts about LWT's viability and suggested a merger with another company.

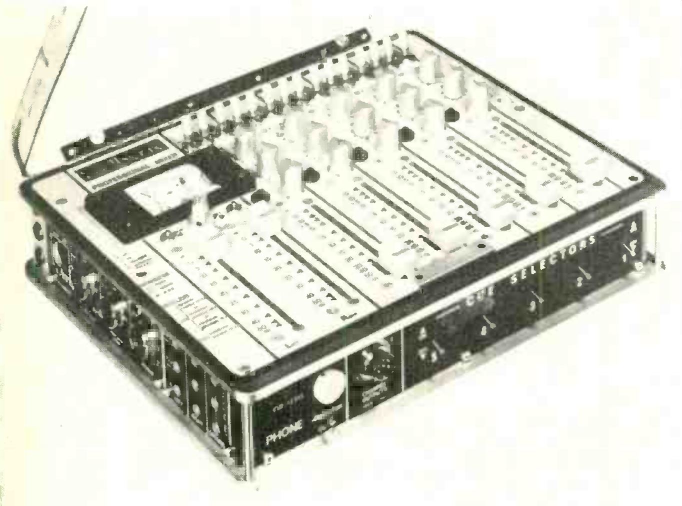
During the last months of 1970 and the first of 1971 a number of executives resigned and the company's image was not brightened by the Rupert Murdoch affair; some of the LWT board wanted Rupert Murdoch, head of a chain of newspapers ranging in quality from the *News of the World* upwards, to be managing director. The ITA said he was unsuitable and the resulting display of personalities was a little messy.

It was at this unfortunate stage in Weekend's fortunes that Aiden Crawley, then chairman of LWT, described the elaborate plans for the new centre on the south bank of the Thames. At the time it was said that the centre was Weekend's fun way of cutting its own throat.

With all this going on it was hardly surprising, therefore, that LWT should want to save money and even, if they could, make a little extra on the side. An operation like the Intersound recording studio was one of the obvious ways of doing this.

Perhaps because of these tactics, LWT is now doing much better than could have been foreseen eighteen months ago. Like most of the other commercial tv companies, Weekend are enjoying an advertising boom. In March of this year, for example, they announced a ten per cent improvement in advertising over the intake for 1971. In spite of the enthusiasm of all concerned, therefore, it is unlikely that Intersound will reappear as a record-making centre. Judging by what Keith Wicks said in the August 1971 Studio Sound, it seems a pity.

Stellavox AMI portable mixer



MANUFACTURERS' SPECIFICATION

Five identical inputs each having:
Balanced line input: up to 10V at 200 ohms.
Switch for using: dynamic microphones, 12V phantom powered capacitors, or AB parallel fed capacitors.
Bass cut switch: —3 dB at 120 Hz; —20 dB at 30 Hz.
Presence filter switch (broad): +8 dB at 5 kHz, +3 dB at 38 kHz and 65 kHz.
Bass control: approx ±18 dB at 40 Hz.
Treble control: approx ±18 dB at 10 kHz.
Main slide pot: +15 to —60 dB, length 5.70 mm.
Panpot: feeds progressively to both pots.
Frequency response: (overall): 20 Hz to 20 kHz ±1 dB.
Harmonic distortion (total): less than 0.1% at 1 kHz and 0 dB.
Intermodulation (50/3000/4:1): approx 0.1% at full level.
Temperature range: —20 to +70°C
Noise level: —124 dBm (mic inputs, 200 ohms).
Dimensions (overall): 80 x 215 x 270 mm.
Weight: 3.2 kg without batteries.
Basic price: £562.
Manufacturer: Stellavox G. Quellet Eng, —EPZ 2068, Hauterive/NE Switzerland.
UK Agents: AV Distributors London Ltd, 26 Park Road, Baker Street, London NW1 4SU.

THE initial impressions of the Stellavox mixer are of a very neat five channel unit in similar style and size to the Stellavox *Sp7* recorder. The mechanical layout and soundness of construction are to a high standard with all main functional controls protected in transit by a transparent plastic cover which may either be hinged back in use or easily removed by means of two small knurled nuts.

The right hand end of the mixer supports the five Preh *6973* microphone inputs, five pairs of 3.5 mm line input sockets on standard 25 mm spacing, and five mechanically-protected miniature three-way toggle switches which either select line input, microphone input, or microphone input attenuated by 20 dB.

On the left hand end of the mixer are a DIN socket for feeding external power supplies or for charging the internal batteries, three Preh *6973* type connectors for the mixer output, and the input from an additional mixer and for monitoring headphones. In addition there are sets of 3.5 mm sockets located under a removable plate. When the optional balanced output unit is fitted, the plate is removed and these sockets provide two balanced outputs each rated at either 4.4V into 600 ohms or 1.5V into 200 ohms.

Twelve pen-torch size batteries (*UM3*, *HP7* or whatever you like to call them) are located under a removable plate at the rear of the mixer and should be very easy to change when required. More anon.

Reverting to the main functional controls, the left hand end of the front panel is occupied by a twin 'modulometer', two miniature push-buttons (one of which gives a battery check on one of the modulometers and another which produces a 1 kHz 'beep' at zero level on the mixer outputs), two master faders of the normal slide type (for the left and right outputs) and a four position rotary switch. The first position of the rotary switch is the mixer 'off' position; the next two positions are mixer 'on', with or without illumination of the modulometers. The fourth position is mixer 'on' but with an automatic gain control operating on channels four and five. For some reason it has been assumed that modulometer illumination is not required when the automatic gain control is operational on two of the five channels available.

The remainder of the front panel is occupied by the five channels of input controls which are located 'vertically'. At the top of each channel are three miniature toggle switches which provide: (1) A high pass filter for speech with a 3 dB point about 60 Hz and an attenuation of 10 dB per octave. (2) A fixed presence lift of 8 dB at 5 kHz with 3 dB lifts at 4 kHz and 6.5 kHz. (3) A choice of microphone feeds for either parallel fed capacitor microphones or for moving coil and phantom powered capacitors. Under the switch sets are three rotary pots which control treble, bass and pan, and a standard slide fader. All these controls are clearly identified and well laid out but, for some peculiar reason, the five channel faders and two master faders are calibrated from +15 dB to minus infinity with a zero dB reference which appears to be meaningless.

The majority of electronic components are contained within encapsulated modules which plug into printed circuit connectors within the mixer. This means that most component failures will lead to the replacement of a complete module but, as the five channels use the same selection of modules, this should make fault location and rectification a very simple matter. However, in addition to the 14 plug-in modules, there are some eight printed circuit boards which are wired into the mixer. This seems to defeat the modular concept of the unit as it is rather surprising to find so many components located on wired-in boards, when they could well have been fitted with printed circuit connectors.

In spite of these criticisms it must be emphasised that the mechanical standard of the mixer is excellent and that good quality printed boards and components are used throughout.

Switch-on

As a first step, the base cover of the mixer was removed by releasing six allen screws and the optional type *S00* balanced line output unit plugged into the already fitted socket. While this unit is a fairly tight fit in its socket, it was provided with two holes for retaining screws. The mixer was provided with two tapped screwholes but no screws.

Next it was decided to fit the optional set of 12 Stellavox 'specially selected' Ni-Cad accumulators. The battery compartment was opened with ease, the batteries fitted with somewhat less ease, and . . . the battery compartment cover would not fit. This was found to be due to one excessively long battery (possibly the result of overcharging at some time) which obstructed the ends of the battery compartment cover. The review proceeded, mixed with some four letter words whenever the 12 batteries avalanched about the testbench.

Next, switch on—and nothing happened. The battery test showed that the batteries were okay, the modulometer lighting wouldn't work

continued 63

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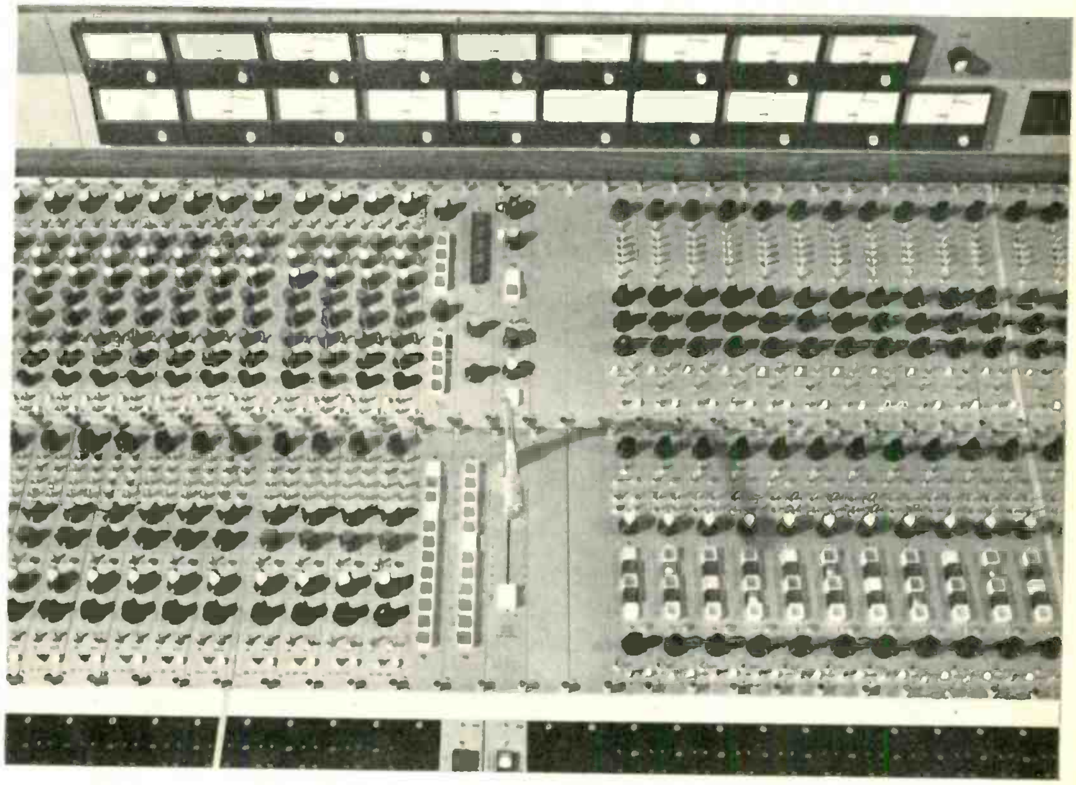

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MANUFACTURERS OF AUDIO CONTROL EQUIPMENT FOR THE SOUND RECORDING INDUSTRY and nor would anything else. It was then found that no internal fuse was fitted. What fusing current? No sign in the instruction book or anywhere.

Next take a guess and fit a fuse . . . sparks. Yes, you can short the 'live' end of the fuse-holder to chassis when replacing the fuse.

While on the subject of sparks, it may be well to deal with the optional mains power-supply/battery-charger. When this unit (type *APS*) is plugged into the DIN socket on the left hand side of the mixer, it charges the internal batteries even when the mixer is switched off and also powers the mixer.

The *APS* unit is provided with all the common mains supply tappings and its operation is indicated by two lamps located on one end. A red light shows that the mains supply is connected and a white light that current is being drawn from the power-supply/charger.

A three core plug-in lead is provided for the mains connection but (my being a typical engineer) only the line and neutral were connected. First results were a healthy shock when plugging the unit into the mixer and a red supply indicator going rather too bright for comfort. It was then decided to reduce mains voltage and use the 220V tap. Upon switch-on there was a flash, a loud report and smoke! Subsequent investigation showed that the neon mains supply indicator had exploded as a result of faulty insulation between the joints to the indicator lamp and the chassis of the power-supply/charger unit.

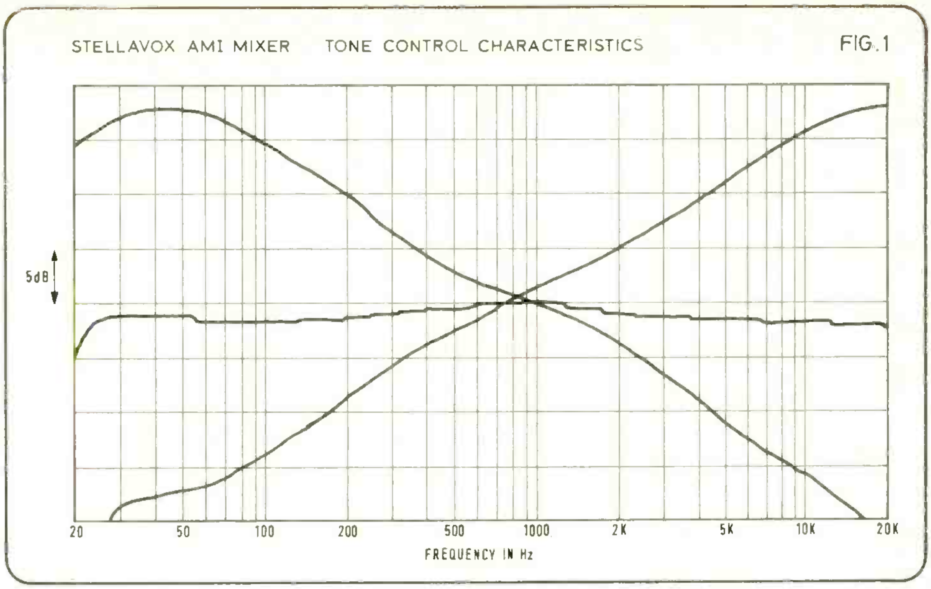
Yet further investigation of the mains power supply showed another dangerous aspect of its construction: there was a capacitor connected to the tagboard on the mains transformer, the leads of which could easily short to the metal case. The U.K. agents have been informed of these faults, and I understand that modifications will result. Meanwhile it is strongly recommended that the unit be earthed, at the risk of introducing hum loops.

While on the subject of power supplies, the opportunity was taken to check the current requirements of the mixer with the optional *S00* balanced output unit fitted. Under no-signal conditions, the standing current was 57 mA with the meter illumination off or 75 mA with the lamp on. This should provide a useful operational life in the order of 10 hours with nickel-cadmium accumulators.

'Beep'

Having got the mixer working, it was decided to investigate the metering and output levels from the mixer. Pressing the line-up tone generator 'beep' button produces a line-up tone at all the outputs of the mixer, irrespective of the fader settings, and also provides a zero level indication on the twin modulometer. It was immediately noticed that the 1 kHz 'beep' sounded rather distorted; in fact 2 per cent total harmonic distortion, with a substantial proportion of odd higher harmonics including 0.44 per cent seventh all based on a reasonably accurate fundamental frequency of 1.052 kHz.

While distortion of the line-up tone does not have any real practical effect upon performance, it is rather off-putting to find such a high percentage of distortion recorded on tape when dubbing.



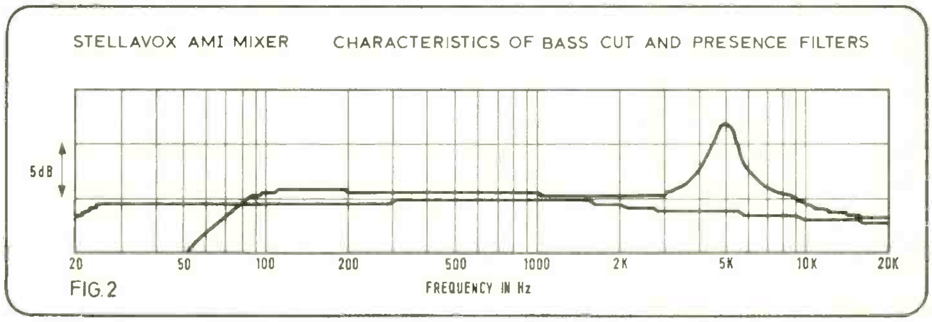


FIG.2

With the two master faders in the recommended open position, the modulometer indication was precisely zero level when 'beep' was pressed but the level was 1.5 dB low if the master faders were accidentally left shut; the same difference was found on all the mixer outputs which gave the following voltage levels:

	Left	Right
600 ohm line output at nominal 4·4V		
Unloaded	4·38V	4·41V
Loaded with 600 ohms	3·56V	3·60V
200 ohm line output at nominal 1·5V		
Unloaded	1·46V	1·50V
Extension output to recorder or other mixer at nominal 1·55V (200 ohms)		
Unloaded	1·54V	1·56V

The above figures demonstrate very great care in setting up the mixer and in design. Similarly, it was found that the output impedances were sensible for the rated load impedances.

There is some criticism of the monitoring headphone output; this does not have any separate level control and either feeds direct from the normal mixer 'extension output' or from it in series with 220 ohms in each channel. The monitoring headphones, therefore, work at a fixed level if the mixer is to deliver its nominal output and the resulting monitoring level will depend solely upon the type of headphones used.

Frequency response and equalisation

The overall response from the microphone inputs to the balanced line output could be readily set to within ± 1 dB from 20 Hz to 20 kHz, with a fall-off to -3 dB at 20 Hz; a very good performance almost exactly coinciding with the mid-positions of the tone controls. But frequency response measurement at zero level with the balanced output working into a high impedance gave a very odd and violent bass hiccough of 5 dB at 30 Hz (not present on the 'extension' output).

Fig. 1 shows the frequency response with the tone controls at the mechanically central position, and at their extremes. All five channels were found to have virtually identical performance and to be in close agreement with the manufacturers' specification.

It is suggested that the extreme bass-boost and treble-cut positions of the tone controls provide RIAA pickup equalisation. In the bass region, the plotted curve is very close to the RIAA characteristic above 50 Hz. The treble characteristic is within 2 to 3 dB on the excessive treble cut side which can be corrected with the treble control. There is, however, a snag. The input sensitivities and impedances are unsuitable for the common dynamic pickup cartridges which give around 3 to 5 mV and like into a load of 47 kΩ or thereabouts.

The effect of the presence bass-cut switches are shown in **fig. 2** from which it is to be seen that both facilities give a sensible characteristic

continued 65

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but that the turnover frequency of the bass-cut facility is somewhat lower than specified by the manufacturer.

Noise and distortion

The effective mixer noise with any single channel fully open and the master faders at maximum was found to be equivalent to -120 dBm (-128 dBm 'A' weighted) at the dynamic microphone inputs. The effective signal-to-noise ratio under maximum gain conditions was therefore 44 dB and this signal-to-noise ratio was increased to only 46.5 dB (56.5 dB(A)) when the line input was used.

If one compares the microphone input sensitivity (measured as 0.12 mV on channels one to three and some 3 dB less sensitive on channels four and five) with the line input sensitivity (measured as 32 mV on channels one to three and again 3 dB less on channels four and five) a far better signal-to-noise ratio should be expected on the line input.

This poor situation is brought about by the practice of potting down the line input to microphone level and still operating the microphone amplifier at full gain when the line input is used. Surely the line input could have been introduced at a higher level point in the amplifier or amplifier feedback increased so as to reduce the gain.

The next points of interest were the input overload limits; these were measured as the maximum input that could be applied with the main faders open, the pan pots in their central position, and zero level at the output of the mixer.

Under the above conditions, the line input clipped at the following inputs:

Channel	One	Two	Three	Four	Five
Input	1·6V	3·2V	3·2V	3·8V	1·6V

Such poor overload figures are quite useless if the mixer is fed from normal 600 ohm lines and are far below the specified input of up to 10V which would be entirely satisfactory.

Likewise the microphone clipping level of down to 6 mV for dynamic microphones is unsatisfactory and the 20 dB microphone attenuator is no palliative because of the effect upon signal-to-noise ratio.

Distortion was measured with a 100 mV line input, the main faders fully open, and the channel gain adjusted for the desired output level. Under these conditions the figures shown above in the next column were obtained:

All the above figures were similar on both output channels and are all far in excess of the 0.1 per cent maximum specified distortion at 0 dB output. Nevertheless, they may be acceptable but for the rapid increase in distortion with small overload outputs.

Any attempts to measure distortion at the optional balanced output with the mains power supply in use were a complete disaster because the hum level with even the master faders shut was only 36.5 dB below zero level output (4.4 V into 600 ohms) and mainly consisted of 100 Hz.

This complaint did not apply to the extension output, where the hum levels were satisfactory. It was therefore concluded that the *S00* output module must be fed from an unfiltered power line.

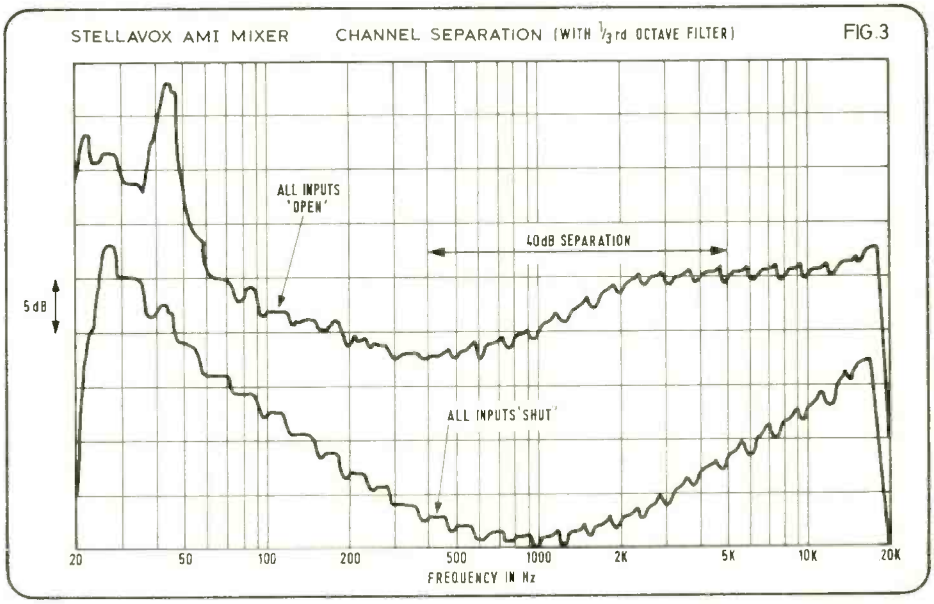
Output	Total harmonic distortion: Into high impedance	Loaded
Balanced line output		
+16 dBm	1·15%	2·5%
+15 dBm (4·4V rated output)	0·64%	0·56%
0 dBm	0·60%	0·37%
Extension output		
+8 dBm	1·50%	1·45%
+6 dBm (1·55V rated output)	0·91%	0·90%
0 dBm	0·23%	0·17%

It was concluded that the modulometers were generally similar in performance to the standard ppm.

The crosstalk between channels is shown in **fig. 3** which demonstrates a satisfactory crosstalk performance for stereo recording and is unlikely to cause any embarrassment in practice.

A somewhat abbreviated investigation into the automatic gain control facility on channels 4 and 5 showed that this function operated on the maximum signal applied to either channel and made appropriate reductions in gain to both channels. This will naturally prevent image wobble when recording in stereo. The attack time of the system was not particularly fast which is a sensible compromise with the recovery rate of approximately 2.5 dB per second after the removal of an overload.

However some 300 mV of input were required to reach nominal zero output level and the gain did not stabilise until the input reached 480 mV which is associated with an output 4 dB above zero level. The input would then accept over 12V input without overload.



The remainder

Checking the various input and output impedances with a 1.592 kHz bridge, all was sensible and to specification including the 205 ohms input for 200 ohm microphones! While this input impedance is suitable for 30 ohm microphones, it is too low for dynamic microphones with higher rated impedance.

The facility for switching between phantom capacitor microphones and parallel fed microphones did not present any problems, except that the applied voltage was found to be 10.5V instead of the rated 12V.

Measurements on the speed of the modulometers showed that they had fairly similar characteristics to the normal ppm and that tone bursts in the order of 15 ms would under-read by 4 dB on maximum deflection. Applying a 10 dB overload of 1.5 ms 3 kHz tonebursts at a rate of 15 per second did not have any influence on the application of a steady tone at zero level.

Summary

The mechanical design of the Stellavox *AMI* mixer is to a high standard, as are the layout of controls and appearance. The concept of the facilities is good and the potential of the unit considerable. However, this new design contains too many shortcomings and faults to be recommended at this stage.

Throughout the review considerable co-operation was received from the UK agents for Stellavox and it is understood that some of the matters criticised in this review were in hand while the review was being done.

It is to be hoped that Stellavox will take due note of these and other criticisms, in which case it will be a pleasure to review a modified version of the mixer when the 'bugs' have been removed. **Hugh Ford**

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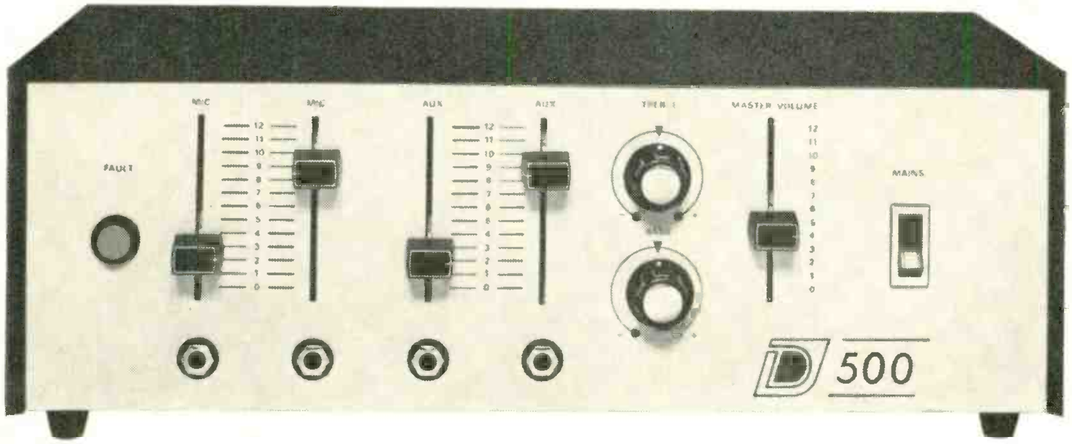

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Harmonic Distortion: less than 0.5 per cent from 10 Hz to 100 kHz, less than 2 per cent from 1 Hz to 1 MHz.
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Amplitude stability: less than ±2 per cent from 10 Hz to 100 kHz, less than ±5 per cent from 1 Hz to 1 MHz.
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Input Supply: 110V or 240V ac 50/60 Hz, or 27V dc from three self-contained *PP9* batteries.
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THE FARNELL *LFM2* is a very well constructed instrument with inbuilt facilities for mains or battery operation and covers a wide frequency range with sine or square waves.

All operational controls are mounted on a clearly laid-out front panel, which is provided with an output meter which also functions as a battery check meter when a slide switch at the rear of the instrument is operated. Unfortunately the battery check operates without the oscillator being switched on, and can therefore give misleading results by checking the batteries under open circuit conditions. Accompanying the battery check switch at the rear of the instrument is the mains/battery slide switch, and the mains input socket which accepts a miniature Bulgin mains connector.

Reverting to the front panel there are two controls, a four position attenuator with 20 dB steps and a continuously variable control. The output is fed to standard hollow terminal posts on 19 mm spacing, the low potential one of which may be linked to the earth terminal post. A 'sync' output terminal post is also provided.

Frequency is determined by six pushbuttons which are frequency multipliers in a ten to one ratio and a clearly labelled dial equipped with a slow motion drive. Dial calibration is from one to ten, without the provision of any overlap between decades. A further pushbutton selects sine or square waves, and a slide type power on/off switch is provided.

The general standard of construction is high, all internal components being mounted on one of two good quality printed boards which are tidily wired to the front panel controls. High grade components are used throughout and the cabinet is of substantial construction and provides protection for the front panel controls as a result of the provision of two solid carrying handles.

The process of replacing batteries is rather tiresome because, firstly, the instruction manual fails to tell which of the two covers one must remove to gain access to the batteries and, having found the right cover, one has to remove two Philips head screws and two slotted head screws, none of which is captive.

In other respects the instruction manual is a useful document because it is provided with full setting-up instructions, a clear circuit diagram and parts lists.

Unusual calibration

Now to the actual performance of the instrument; we started by checking the accuracy of the output voltage after unravelling the rather unusual meter calibration. The meter bears two scales: a dBm scale calibrated -6, 0, +3, +6 and +9 dBm (the steps between markings are too large) which relates to the output voltage when the instrument is loaded with 600 ohms—so far so good—but the second scale is calibrated in volts peak-to-peak when the instrument is loaded with 600 ohms; of course the idea is to give a useful indication of the square wave output. As there is not a switched internal 600 ohm load, and not many of us want square waves into 600 ohms, this calibration is all rather confusing. Just to add to the difficulties the zero dBm mark is right in the cramped part of the scale and there is no +10 dBm mark, so it is just about impossible to set zero dBm with any respectable accuracy.

Attenuator accuracy and metering accuracy are not noted in the specification but we found the accuracy of the attenuator at 1 kHz to be within ±0.5 per cent, but the meter accuracy to be only mediocre at full scale with a virtually constant error of +4 per cent falling to a +1.5 per cent error at half scale deflection. The 'sync' output provided a 2.217V (rms) sine wave at 1 kHz and was independent of the output settings.

The output voltage stability was excellent, being 1.2 parts in 10^3 over a period of four hours after warm-up, and reducing the mains supply from 240V to as low as 180V had very little effect. Frequency calibration accuracy was initially checked at three points on the scale for each frequency decade and was found to be well within the specification of ±2 per cent, with a maximum error of +1.5 per cent and 0.03 Hz below 10 Hz. However, investigation of the calibration accuracy at intermediate points gave errors of +3 per cent at 40 kHz and 25 kHz. Switching between sine and square waves had negligible effect on frequency, as did loading the output or lowering the mains supply from 240V to 180V. Investigation of the frequency stability over a period of four hours after warm-up showed a drift of only 4 parts in 10^4, which is substantially better than the specification would suggest.

The stability of output against frequency was good, being measured as ±0.1 dB between 10

continued over

continued

Hz and 100 Hz, or ±0.3 dB from 1 Hz to 1 MHz, and there was no undue amplitude bounce when operating the variable frequency control. Frequency setting was quite good, and it was possible to set chosen frequencies within one part in 10^4 with ease, but there was some backlash present in the slow motion drive. For audio measurements the specified audio frequency distortion of better than 0.5 per cent is rather high and, in spite of the instrument being far better than specification, distortion is a limiting factor, as will be seen from the following measurements:

Frequency	Total Harmonic Distortion
5 Hz	0·50%
10 Hz	0·43%
100 Hz	0·34%
1 kHz	0·32%
10 kHz	0·32%
100 kHz	0·31%
500 kHz	0·57%

It is to be seen from the above that the distortion outside the frequency band 10 Hz to 100 kHz where the specification is better than 0.5 per cent is exceptionally low and very much better than the specification limit of two per cent maximum.

Analysis of the distortion components at 1 kHz showed a rather large odd harmonic constant of 0.31 per cent third harmonic and 0.085 per cent fifth harmonic. However, noise and hum were less than −70 dB . . . a good standard.

All the above distortion measurements were obtained at an output of +12 dBm into a 600 ohm load, which one might expect to be the worst condition, but investigation at lower outputs had negligible effect, as did lowering the mains supply voltage to 180V. As previously stated, hum on mains operation was at least 70 dB below the output signal, and the performance of the oscillator was found to be identical with mains or battery operation.

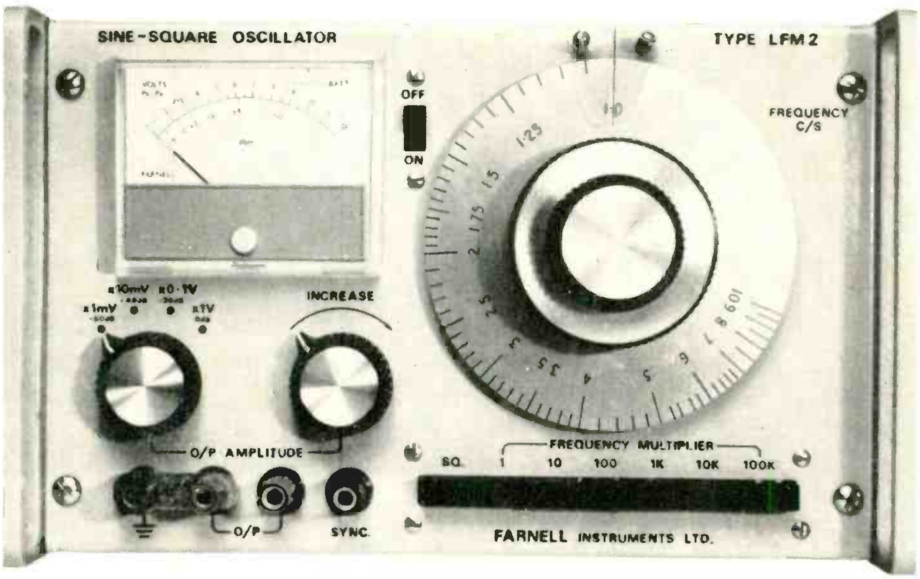
The square wave performance of the oscillator was excellent with a constant rise time in the order of 50 ns irrespective of oscillator frequency, and negligible droop even at 1 Hz.

The manufacturer suggests that the 'sync' output can also be used to lock the oscillator to an external source with a locking range of ±0.75 per cent for a 1V peak-to-peak input into the 'sync' terminal. We only managed to obtain a lock-in range of ±0.5 per cent at 1 kHz, but appropriately larger locking ranges can be obtained if higher inputs are applied.

Summarising, the Farnell *LFM2* is an excellent source of tone for frequency response checking over a wide frequency range, and also produces very good square waves. However, in spite of meeting its specification for distortion, the distortion is too high for many audio frequency measurements and the odd harmonic components are particularly embarrassing. The output voltage metering is reasonably accurate for many purposes, but the meter scaling is indequate and confusing, and the minimum readable output of about 0.75 mV is too high for use with sensitive amplifier inputs such as microphone inputs and pickup inputs.

This oscillator is one of the relatively few that can be conveniently rack mounted, and it should find application for the frequency response testing of studio equipment, but in other respects its application is rather limited from an audio engineer's point of view.

Hugh Ford



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