STUDIO SOUND

JUNE 1970 3s (15p)

tape recorder

TETRAHEDRAL AMBIOPHONY TOMORROW'S STEREO?

A HIGH QUALITY MIXER

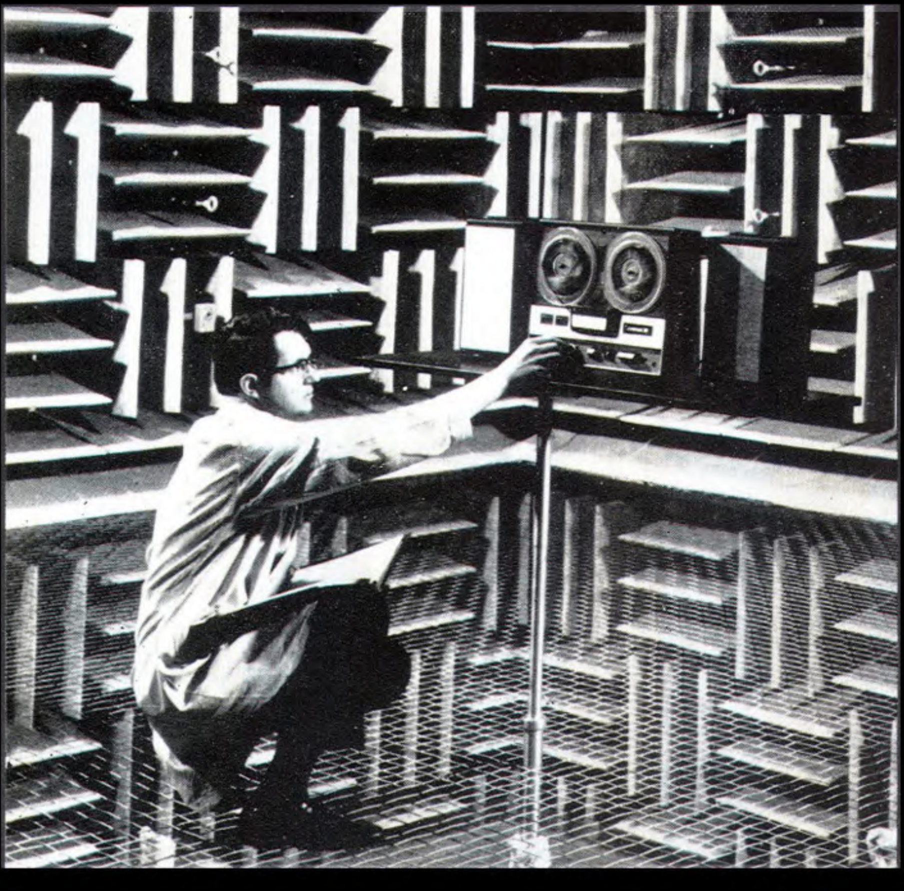
THE INCREDIBLE HAWAIIAN PIANO

A LOUDSPEAKER SUSPENSION

NEVE IN FOCUS

BALANCED AND UNBALANCED LINES

AROUND THE STUDIOS: MAXIMUM



Enter Sansui, and a new era in tape deck performance.

This is the all-new Sansui SD-7000. A quality 2-speed 4-track tape deck, it is the end product of three years of research and 10 prototype models. Our reputation as Japan's foremost audio-only specialist goes on the line with it. But we're confident that reputation is secure.

The SD-7000 compares favorably with the most expensive taping units on the market. Very favorably. Here's how it shapes up:

Three motors, four heads. Automatic Reverse, Repeat and Rewind either by recording 20Hz signals or attaching sensing strips. Special Tape Protection Circuit and built-in Tape Tension Switch. FET-equipped 3-stage equalizer circuit, separate Volume controls for two line inputs, plus a headphone Volume control. Automatic resetting reel clampers. Self-locking Pause switch. Sleep switch that automatically cuts off power. Tape travel direction lamps. All pushbutton operation. Rear illuminated VU meters. Source/Playback tape monitor switch and Stereo/Mono L+R recording mode switch. Six-position voltage selector. Output level attenuator switch.

The SD-7000 takes an honored place in the ranks of quality audio products, and presages a new era in tape deck performance.

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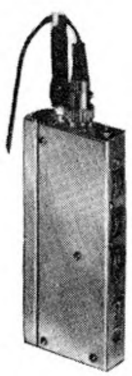
Hampstead High Fidelity for the highest quality microphones

Professional partners

STC MODEL 4038

The design patent for this microphone is held by the B.B.C. and is made to their demanding specification by STC.
Bi-directional substantially flat.
From 30-15,000 Hz.
Impedance 30 OHMS.



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This RMS7 radio microphone is a superb example of high quality miniaturisation and is the ideal microphone for location work in film and television studios, theatres and co-ordinating heavy civil engineering work.

AKG MODEL 202

Two-way dynamic cardioid microphone.
Two systems in one housing for high and low frequencies respectively. On-off switch and bass control swivel stand adaptor.
From 20-18,000 Hz.
Impedance 200 OHMS.

AS USED BY THE B.B.C.

FOR THE KEEN AMATEUR

STC MODEL 4119

Outstanding in performance, ideal for home recording or for public address where feedback is a serious problem.
Hyper cardioid polar response.
From 15-12,000 Hz.
Impedance 30 OHMS or 50,000 OHMS.

STC MODEL 4113

Superb cardioid microphone.
From 15-12,000 Hz.
Impedance 30 OHMS or 50,000 OHMS.

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5 good reasons to use an Ampex MM-1000 recorder.

1. It's expandable.
Our 8-channel MM-1000 Master Recorder costs only moderately more than other 8-channel recorders. But with the MM-1000 you can readily expand to 16 or 24 channels. Add-on 'kits' enable you to add channels as you need them. Or you can buy the MM-1000—16 or 24 channels completely checked out and ready-to-go.

2. It's designed to handle wide tape.
The MM-1000 tape transport is designed to handle 1" and 2" tape. It's the same transport that's now in use on over 3000 Ampex video tape recorders. When you go from 1" and 2" tape, you just change the tape guides and the plug-in head assembly. Lets you quickly change from 1" for 8-channels to 2" for 16 or 24 channels.

3. It's versatile.
The MM-1000 offers more standard and optional features than any other master recorder. Tape Motion Sensing for instance. Allows you to change modes without going into stop and without stretching or breaking tape. Automatic Tape Lifters? Yes, and with manual override Ping Ponging? Sure, Sel-Sync? Naturally. Also remote Sel-Sync. How about Variable Speed Motor Drive Amplifier? Yes again, plus an Electronic Timer with up to 4 remote read-outs for pinpoint accuracy. Versatile? You bet!

4. It's promotable.
"New Generation" recording capability is built into the MM-1000. The new groups demand this capability and record at the studios that offer it. The MM-1000 is capturing the imagination of these groups, and challenging their creativity. With the MM-1000 you'll have a promotable edge over studios with past generation equipment.

5. Its applications are totally unrestricted.
The wide range of applications for the MM-1000 mean its cost is amortized over a very short time. The MM-1000 may typically be used for Pop music recording, master Classic recordings, multi-language synchronised with videotape recordings, multi-channel sound source for film transcriptions. Or simply as a 'soul mate' in the search for innovation in music.

For further information on the Ampex MM-1000 recorder, send this coupon today.

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Company

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AMPEX

Studio Sound
& tape recorder

JUNE 1970 VOLUME 12 NUMBER 6

INCORPORATING
SOUND AND CINE

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COVER PICTURE

Some 3 000 glass-wool wedges line the 3M anechoic chamber at the company's Research Centre in St Paul, Minnesota. Here the chamber is being used to test the acoustic response of a small stereo recorder.

SUBSCRIPTION RATES

Annual subscription rates to *Studio Sound* and its associated journal *Hi-Fi News* are 36s. ($5 or equivalent, overseas) and 44s. ($5.60) respectively. Six-month home subscriptions are 18s. *(Studio Sound)* and 22s. *(Hi-Fi News)*, from Link House Publications Ltd., Dingwall Avenue, Croydon CR9 2TA

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

BEFORE DROPOUT dropped out of the April issue, he commented that with hard saving and/or HP 'your amateur can acquire a photographic rig to rival that of the professional—indeed both often tote the same gear'. He went on to claim that the situation is 'vastly different' in the recording world, either for financial reasons, copyright reasons, or both. We beg to take issue.

First, what is meant by 'professional'? In this case the small mobile studio, a competent engineer with a *Pro 20*, *A62*, *TR52* or high-speed Revox, who produces stereo masters for specialised full-price discs and less specialised cheap-labels—the low price being made possible by minimal overheads.

Competing with him offers a worthwhile challenge to the potential semi-pro. Whatever Dropout's view, any amateur prepared to read up basic microphone technique, the relevance of technical specifications, and the procedure for lining up a tape amplifier, is mentally equipped for serious (which means reasonably lifelike) recording.

Small matter then of choosing a tape machine. Forget ¼-track, forget 19 cm/s, forget unbranded tape, high-Z mikes, tiny spools, GPO headphones, and record/play heads. Look through the classifieds for an old 38 cm/s EMI, Leevers, Ferrograph or Vortexion, all of which have been advertised occasionally for under £100. If the budget is more flexible, form *Superbo Recordings*, assuming Peter Bastin hasn't already done so (headed notepaper carries considerable weight in this country), and take advantage of the tax relief offered to legitimate business — which is what *Superbo Recordings* is. Then buy your *HS77* Revox.

The machines we list are recommended not for their recording quality so much as the ease with which they can be lined up. This means more than a bias twiddler; the *HS77*, the *TR52* and the Ferrographs, in common with all studio designs, have accessible and clearly labelled presets on each channel governing record frequency response, replay response, meter calibration and bias, allowing the tape amplifier to be optimised to one or more speeds according to the standard line-up routine. The only difficulty is finding cash for a sinewave generator, VVM, distortion meter and reliable test-tape. The 'I'm only interested in making good recordings, not in the technicalities' attitude dies hard with some amateurs. Do these same people employ a chauffeur because they feel incapable of learning to drive?

Microphones raise a few problems. Professionals sometimes skimp on a relatively cheap recorder but rarely use cheap microphones unless they are multimiking, when a lot of moving-coils seem to crop up. A *C24* stereo capacitor costs over £300, two *4038* ribbons over £100, two Calrec cardioid capacitors about £70, two Film Industries ribbons (we all have to start somewhere) less than £20. You get what you pay for if you choose carefully. Otherwise you are stung.

As for copyright, musicians deserve every sympathy for the state of their profession. Performers should be approached on a personal and non-commercial basis: they welcome recording facilities as a means of self-analysis and, in tune or not, provide the ideal training material for recording engineers.

We have said this before, Alan Green (Folk Heritage Recordings) said it last month in the correspondence column, and we say it again: dedicated amateur recordists should affiliate themselves with local folk clubs, youth orchestras, jazz bands, and so on. Every musician you meet knows a couple of others and, provided their sights are set below a perfect *1812*, they may between them contribute something of lasting musical value. Whether they do or not, experience gained in this way allows Dropout's amateur to advertise his service confidently in the music papers. If the equipment starts to pay for itself, he is a semi-professional. If it doesn't, at least the hobby's more constructive than fishing.

FEATURE ARTICLES

REGULAR COLUMNS

EQUIPMENT REVIEWS

Ferrograph Series 7 tape recorder

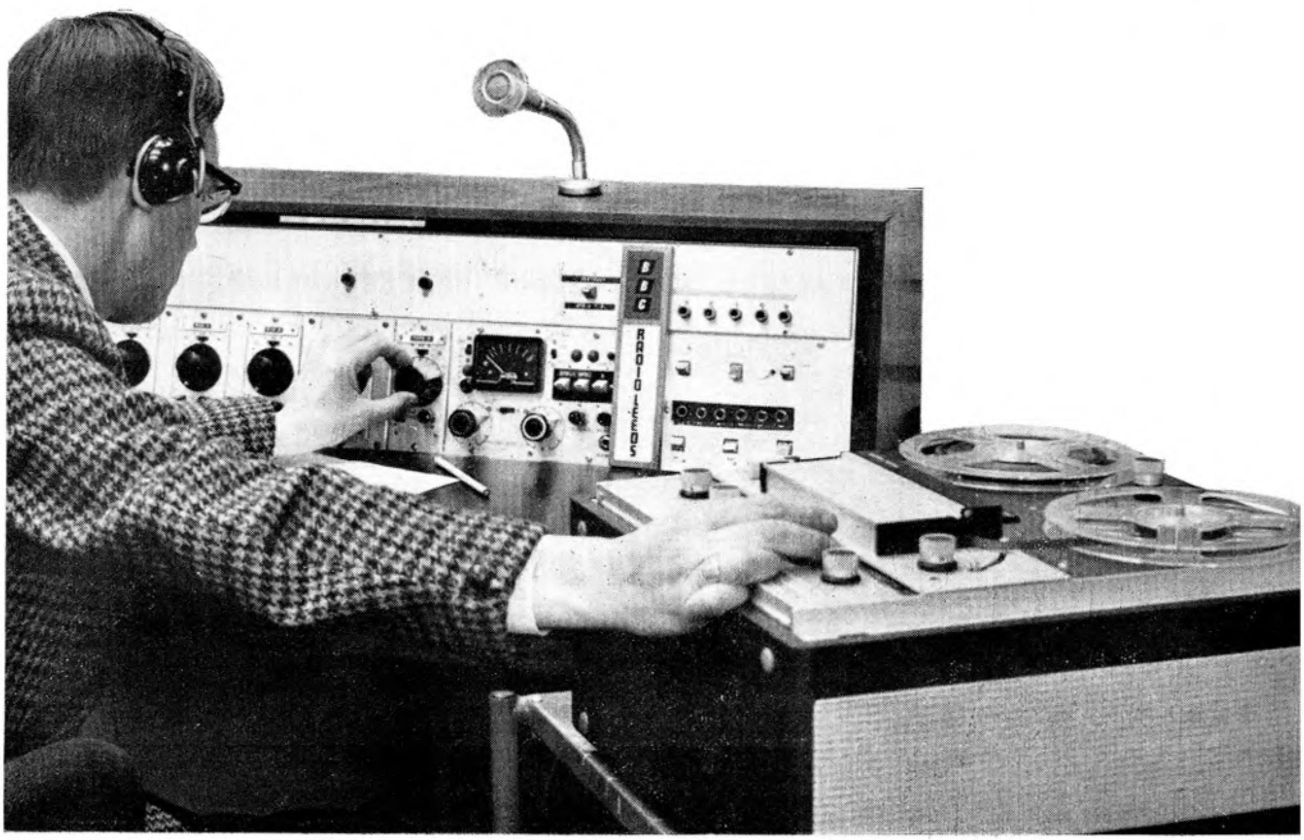
Where a tape recorder must be good and reliable, you'll find Ferrographs. In a radio station, for example, tape recorders are in constant use. Technical performance is all-important; absolute dependability and split-second control are essential. So Radio Leeds uses Ferrograph recorders.

Ferrograph Series 7 tape recorders are British made, available in mono and stereo, with and without end amplifiers. All instruments are solid state, three speeds. All incorporate an unrivalled range of facilities, including two inputs per channel with independent mixing, independent tone controls on each channel, endless loop, signal-level meters for each channel on playback and record, re-record on stereo models, and many others. The output is 10 watts per channel. Ferrograph recorders are available in elegant hardwood or in a vinyl case to suit any decor and method of use.

Follow the professionals; choose the recorder you know will serve you best at home and in your work: Ferrograph. Your local Ferrograph specialist will be pleased to demonstrate it to you. Alternatively, please write or ring for details and address of nearest stockist. The Ferrograph Co Ltd, The Hyde, Edgware Road, Colindale, London NW9
Tel: 01-205 2241, Telex: 27774

International Distributors:
Leroya Industries Pty, 266 Hay Street, Subiaco, Western Australia 6008, Australia;
Matelectric, Boulevard Leopold II, 199, 1080 Brussels, Belgium;
H Roy Gray Ltd, 14 Laidlaw Boulevard, Markham, Ontario, Canada;
Cineco, 72 Avenue des Champs Elysees, Paris 8e, France;
Henry Wells & Co. KG, 1040 Wien 4, Danhausergasse 3, Austria
Ferropilot GmbH & Co., KG, Hamburg 39, Sierichstrasse 43, West Germany;
Hi-Fi Installations, P.O. Box 2430, 276 Andries Street, Pretoria, South Africa;
Elpa Marketing Industries Inc, New York Park, New York 11040, New York, U.S.A.
There are Ferrograph Distributors also in most other countries. Please obtain details from the London office.

Ferrograph



studio diary

NEW STUDIOS

THE FIRST OF two new studios being built by Island Records in Basing Street was opened recently. A 20-channel 16-output Helios mixer has been installed by Dick Swettenham, plus mono and stereo Philips recorders and a 16-track 3M. The studio can accommodate up to 20 performers.

The same accommodation is provided in a £20,000 complex opened off Monmouth Street by Theatre Projects. Equipment comprises a 14-channel Electrosonic mixer and a 4-track Leevers-Rich. AKG *C12A* and *D19* microphones are being employed. Six 2-track Revox *A77* and four EMI *TR52* recorders are housed in a copying room, additional facilities being film projection and dubbing. Hourly session rates are £12 for 4-track, £8 for 2-track, and £7 for single-track. Four-track to 2-track reduction is £8, 2-track to single-track being £4.

LANSDOWNE GO 16-TRACK

A 24-CHANNEL Cadac is replacing the 16-channel hybrid mixer at Lansdowne Studios, W 11. Their 8-track Scully and 4-track Ampex will be joined later this year by a 16-track recorder. Existing equipment includes the Ampex *351*'s, two 2-track Telefunken *M10*'s, and two EMT plates.

CRAIGHALL ON WHEELS

CRAIGHALL STUDIOS have added a mobile recording van to their Edinburgh-based facilities. An acoustically insulated two-ton Morris incorporates a 12-channel Studer mixer plus 4-track Ampex *440* and 2-track Studer tape machines. Hourly charges are £18 for 4-track and £12 for 2-track, a further £10 being charged for initial setting up.

PYE ORDER AMPEX

AN AMPEX *MM 1000* 16-track has been ordered for Pye's Studio Two. The recorder will be used with a recently installed 16-channel Neve, replacing a mixer of Pye design.

RANK TO DISTRIBUTE WEIRCLIFFE

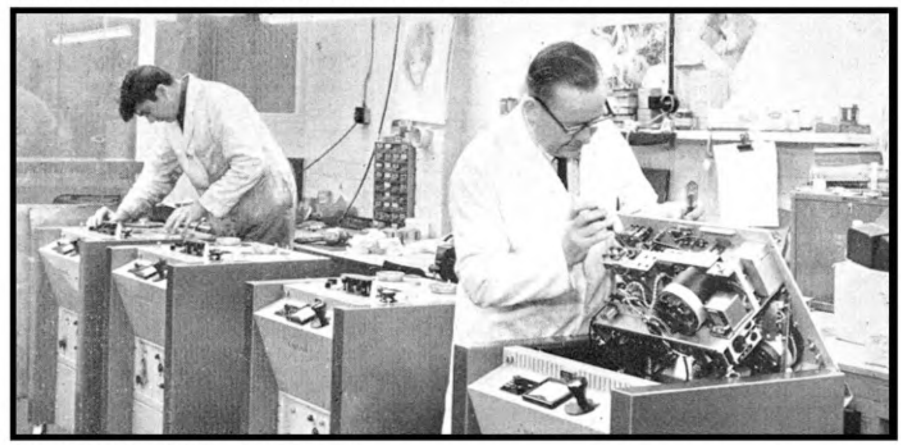
THE ENTIRE range of Weircliffe tape and film erasers is now being marketed by Rank Film Equipment, PO Box 70, Great West Road, Brentford, Middlesex (Tel. 01-568-9222). Model *9* accepts reels of up to 405 mm diameter, 35 mm maximum width, and erases 80 to 90 dB below saturated 1 kHz in eight to 16 seconds. Smaller versions are available: Model *8* taking 370 x 51 mm reels and Model *6* 210 x 25.4 mm.

APRS 70

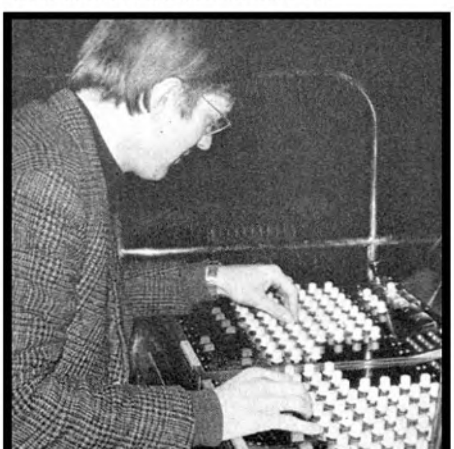
AEG, AKG, Ampex, Audio & Design (Recording), Audio Developments, Audio Engineering, Audix, BASF, F.W.O. Bauch, British Homophone, Cadac, Dolby Laboratories, EMI, Feldon Recording, Fraser-Peacock, Grampian, Hayden Laboratories, Helios Electronics, H/H Electronic, KEF, Leevers-Rich, Lennard Developments, Lockwood, 3M, Rupert Neve, Pye TVT, Richardson Electronics, Shure Electronics, *Studio Sound* and Unitrack—the line-up for APRS 70. The exhibition will occupy the Adelphi Suite in the Waldorf Hotel, Aldwych, having moved from last year's Hotel Russell venue. It promises to be the largest and most strongly supported to date and is being organised by the Association of Professional Recording Studios. The event will be open on Friday June 12 from 2.30 to 9 p.m. and Saturday June 13 from 10 a.m. to 7 p.m. Admission will again be restricted to members of the recording and broadcasting industry. A preview of APRS 70 will be published in our July issue, just before the exhibition opens. Further information is available from the APRS Secretary, John Borwick, 47 Wattendon Road, Kenley, Surrey (Tel. 01-668-1554).

TAPE RECORDER STATISTICS

A TOTAL OF 18 738 tape recorders were manufactured in November, and 20 497 in December, representing increases of 14% and 27% respectively on the corresponding months of 1968. Production in the final quarter of the year was 19% higher than in the fourth quarter of 1968. Value of delivered UK-made machines was 6% lower in November and 7% higher in December. Deliveries in the fourth quarter fell by 2%, and in 1969 as a whole by 15%. Export deliveries, however, showed an increase of 55% during 1969 and in the last quarter were approximately double those for similar periods in 1968. Deliveries of imported recorders showed increases of 29% in November, 2% in December, and 18% in the quarter as a whole. Deliveries during the year were 12% down.



Part of a batch of *E 141 M* single-channel recorders being constructed by Leevers-Rich for The British Forces Broadcasting Service.



Andy Bereza, the designer, at the controls of the AB mixer recently installed in the Prince of Wales Theatre. As reported in our last issue, AB Audio have now been taken over by Allen & Heath Ltd.

SOUND 70 LECTURES

AT 2.30 p.m. each afternoon of the three-day exhibition of public address and allied equipment, organised by the Association of Public Address Engineers early in March at Camden Town Hall, lectures were presented by engineers well known in the fields of acoustics and public address.

Audio consultant James Moir presented the first paper which dealt mainly with the problems of room acoustics as regards the performance of a PA installation, with special emphasis on preventing acoustic feedback.

Consideration of the factors that determine the howl-round frequency when a PA system is operated in any enclosed space was then given by the lecturer. Any such space behaves as an assembly of three pipes mutually at right angles, the three pipes having lengths equal respectively to the length, width and height of

(continued on page 264)

NEXT MONTH

ANTHONY EDEN looks at Noise and its Measurement, Angus McKenzie reviews the Dolby *A301* Noise Reduction System and Keith Wicks interviews APRS Vice President, Jacques Levy. David Kirk field tests the Chilton 100S. This issue will also carry a preview of APRS 70.

The Grundig TK149 gives you the complete sound.

A tape recorder is only as good as it sounds. You know that and so do we. That's why we developed the Automatic TK149—to take the guesswork out of tape recording, to give you that distinct, clear sound for which a GRUNDIG is so justly famous. There's a lot of sophisticated engineering in the TK149 to bring it right up to Hi-Fi standards and, of course, it comes with more than £10 worth of quality accessories. But first things first.

The Features . . . Switchable automatic level setting without increase in distortion and using the unique GRUNDIG delay system. Illuminated recording level meter. Automatic stop at end of tape. Facilities for dual play and trick recordings. Heavy gauge plated steel chassis provides robust construction and perfect mechanical alignment. Handle unclips. GRUNDIG 'Easy-G' single dial control. Head cover unclips for easy access to heads and sound channel. Optional accessories available to give added facilities.

...and the Facts... Recording System: 4-track mono with dual-play facilities.
Level Adjustment: Automatic with the ingenious distortion-free Grundig delay system or manual override.
Tape Speed: $3\frac{3}{4}$ i.p.s. (9.5 cm/s).
Wow and Flutter: 0.2% r.m.s.
Maximum Playing Time: 6 hrs. (4 hrs. with the 1200 ft. of L.P. tape supplied).
Frequency Response: 40—12,500 Hz +3 —5dB
Signal to Noise Ratio: 45dB
Output Power: 2.5 Watts/5 Ohm
Input: Microphone/Universal 2mV/1.5MOhm
Outputs: High impedance 500mV/15kOhm, Earphone 11V/220kOhm, Ext. Loudspeaker 2.5W/5 Ohm. Monitor Output for synchronised recordings.
Loudspeaker: 6" x 4" high flux density unit.
Position Indicator: 4-figure digital with press button re-set.
Accessories Supplied: Moving coil stick microphone GDM 312, 1200' L.P. tape in library container, spare spool, connecting lead.

Grundig TK149. For the complete sound. Recommended Retail Price **£57.12.8.**
One of the incomparable range of Grundig Tape Recorders—
ask your dealer for a demonstration soon.

Hear it all on

GRUNDIG

Grundig (Great Britain) Ltd., London, S.E.26.

Tetrahedral Ambiophony

PART ONE

A reasoned approach to four-channel stereo, incorporating the vertical dimension. BY GRANVILLE COOPER

IT is a strange characteristic of human nature that as soon as we have achieved moderate success in performing some feat or other we lose interest, turn our attention to a more difficult problem, and let the remains of the original problem take care of themselves.

So it seems to be with stereo recording. After nearly 20 years of what we have come to call stereo recording we are just about able, with relative consistency, to turn out exciting and interesting results. But instead of sitting back and enjoying, we deviate from the proven format in two directions. On one hand we try to cheapen the system (multi-track recording to save studio time) and on the other hand we try to embellish the present system (hopefully) to improve it. The former change is in my opinion a retrograde step since it can give inferior results and, by reducing studio time, the quality of performance may well suffer. The second deviation, which is an attempt to improve the system, seems to me to be worthwhile provided the efforts do not get in the way of our enjoyment of the present system. The public and the industry has spent a great deal of money getting together music recorded in stereo. If we are to have a new system it will cost a great deal more money to re-record the whole repertoire again. For this reason alone the new system should if possible be an extension of the present one. We should be particularly wary of adopting any standard until the system has been well tried and we are sure that it offers good value for money. We should be particularly chary of adopting a system just because it happens to be in use: first assess it thoroughly. But why should we change the present system at all? It is capable of excellent results, particularly if coincident microphone technique is used. Good results are possible with multimicrophone techniques, for example the Solti *Götterdammerung*, but the results obtained are larger than life. Although they may be more exciting than would have been obtained with a coincident mike technique, it cannot be said that multi-miking allows realistic reproduction of what takes place in the concert hall. The difficulty with two-channel stereo is that all the sound appears to come from between the two loudspeakers, as though the listener were hearing the sounds from a doorway into the concert hall. An ideal system would reproduce the sound as it is heard by the listener in the hall.

If we are to assess the merits of a system we must write a specification for the ideal so that we have some yardstick with which to measure. Here we have a problem; there will most probably be disagreement between any two people as to what this yardstick should be. It would be presumptuous of me to try to define what 'we' require. I shall presume! What is required is, in my opinion, a system, and technique of operating the system, which will transport a seat in my home aurally into the best position (ideally my definition) in a concert hall of my choice so that I may listen to music of my choice in near ideal surroundings whenever I like. I want all this for a modest outlay and I wish to have the whole musical repertoire available, recorded by artists who are completely committed to their interpretation. In simple words my standard is the concert hall performance.

FIG. 1 CARDIOID PATTERN COMBINATION, AXIS INCLINATION 120°

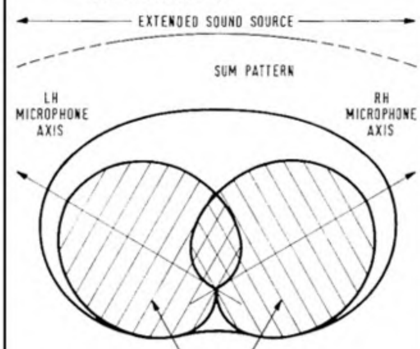
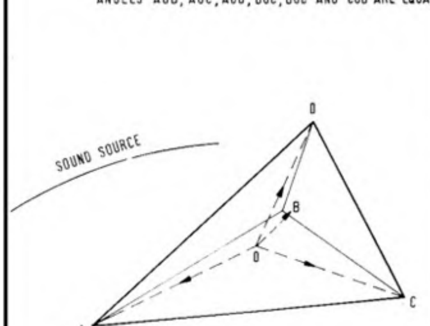


FIG. 2 TETRAHEDRAL MICROPHONE ARRANGEMENT WITH MICROPHONES PLACED AT O, WITH AXES ALONG OA, OB, OC AND OD. ANGLES $A\hat{O}B$, $A\hat{O}C$, $A\hat{O}D$, $B\hat{O}C$, $B\hat{O}D$ AND $C\hat{O}D$ ARE EQUAL



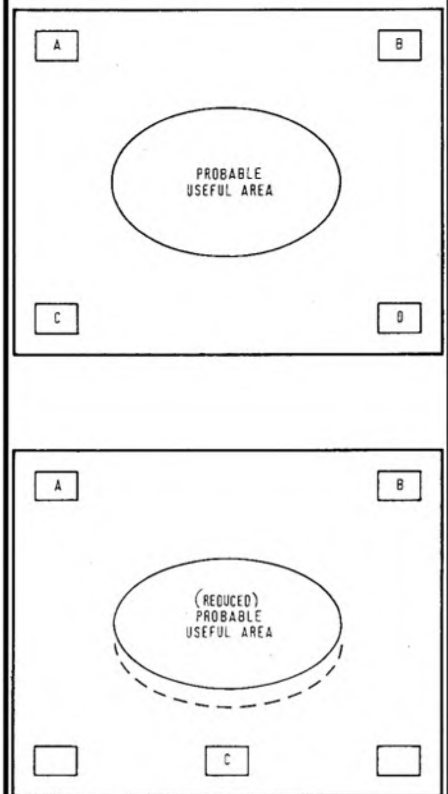
Music can be said to be a succession of stimuli applied to the brain and, for the majority of people, the sense which detects these stimuli is that of hearing—remember there are people who get considerable pleasure from reading a score without acoustic stimuli. Hearing works by detecting the instantaneous pressure changes which occur in the three dimensional air-filled space in which we live. These changes in air pressure at a point (our ear) are due to sound waves which are conducted by the air. A sound wave originates from a source and spreads out in three dimensions from the source. If there is more than one source of sound the pressure changes at the ear are the combination, instant by instant, of the pressure changes due to each source. Exactly what the effect is depends upon the relative loudness of the two sounds, how far away each is from the listener and the nature of the surroundings. In most practical situations some of the energy from each sound source will strike obstacles and be reflected. Some of this reflected energy will also arrive at the listener's ear and can be treated as though it were emanating from another source. Just what the combined effect will be depends on the exact position of the listener's ear in the three dimensional sound field. The majority of listeners have two useful ears and the resultant sound at each is different since the ears are not coincident and, speaking for myself, have a solid lining between them which modifies the sound field. Sound originating from the side will be detected differently by the two ears, allowing us to detect the direction from which the sound came.

Since the earth is by and large flat, excepting the odd mound here and there, we are normally only interested in detecting direction on a horizontal plane. Our two ears therefore work in a horizontal plane. That is to say that we have no ability to detect sounds in a vertical plane but this ability is mainly due to reflection effects that are suffered by sound waves travelling along the earth's surface.

In the first case, the first reflections occur very soon after the sound is generated by the source. The overall reflected sound contribution to the resultant sound at the ear is therefore complex. In the case of a sound arriving from above (in the open air) the reflections are from the ground and occur only after the direct sound is heard. In the concert hall, we have reflections from the distant walls and ceiling which come into this sort of category and I am sure no one, unless

(*continued overleaf*)

FIG. 3 COMPARISON EXPECTED USEFUL AREAS FOR 4 AND 3 CHANNEL TWO DIMENSIONAL RECORDING. USING TWO SPEAKERS D + E TO REPLACE C (CARRYING THE SAME SIGNAL) THE USEFUL AREA MIGHT BE EXTENDED TO THE DOTTED AREA.



his ears are of cloth, would disagree that sound in a concert hall is detected as coming from above and behind as well as from the sides and front. We do not need an ear on the top of our head to appreciate these sounds. We live in a three-dimensional sound field and that is what we must attempt to reproduce if we are to fulfil the aim set out above. There may be an economic argument against concerning ourselves with all three sets of directional information but let us leave that for the moment and turn our attention to methods of achieving our aim with the technology at our disposal.

We are able to record and reproduce the frequencies which are of interest in hearing and we can nearly cope with the dynamic range. We also have a system of recording and reproducing directional information in one and a bit dimensions, i.e. the present stereo system. One method of detecting the stereo information which has proved successful is to use two coincident microphones with cardioid characteristics. If these, having the appropriate pattern, are set with their axes at 120°, then the resultant vertical pattern in the plane through the axes will be similar to that shown in **fig. 1.**

If two microphones are added to the arrangement so that the angle between one microphone axis and every other is 120°, then a similar pattern to that shown will be formed for any pair of microphones and the integrated pattern for all four microphones will be practically spherical. The axes of the microphones will point out towards the vertices of a tetrahedron as shown in **fig. 2.** This microphone arrangement is basic to the system and so we might coin a name for the system based upon this tetrahedral microphone arrangement : *Tetraphonic* recording.

What do we do with the signals from the four microphones ? In principle the only requirement is to feed each signal to a loudspeaker placed in the listening room in a position corresponding to the 'line of sight' of the particular microphone. In order to make the most of practical microphones which have a frequency-dependent polar diagram, it is likely to be better to arrange two microphones to point to the primary sound source and to feed the signals from these microphones to two speakers placed in the conventional positions for stereo in front of the listener. The other two microphones will both be pointing to the rear, one upwards and one downwards. These two microphones drive speakers in corresponding positions. A listener sitting in the centre of the tetrahedron whose vertices are defined by the four speakers will experience a spherically symmetrical sound field similar (ideally identical) to that experienced by the microphone system in the concert hall. As with all attempts to reproduce a complete sound field, the ideal room in which to listen is one which does not contribute any colouration of the sound field itself. This room would, of course, be anechoic and it would need to be quite large—certainly not cheap.

If we compare this system with current systems we find some similarities and some important differences. Our system uses four recording channels, as does the system at present used by Vanguard. However, the two rear Vanguard channels can be collapsed into one (**fig. 3**) with no loss of information and only a small reduction in useful listening area. It seems very unlikely that either system will be effective for listeners near to any one loudspeaker since our hearing characteristics are such that our attention becomes captured by the loudest of several sounds. So three channels are likely to be just as effective as four used in this way. In tetraphonic recording, the fourth channel is used to carry additional information, i.e. the third dimension information. So in terms of information for money the tetraphonic system is superior. It is also a natural extension of the present stereo system to cover the other dimensions of depth and height.

Next month I shall discuss some of the practical problems of achieving satisfactory tetraphonic reproduction.

Bob Woolford, Granville Cooper, Peter Self and Sid O'Connell (left to right) setting up in the monitor room.

Four capacitor microphones (plus a separate pair) over the Salomon Orchestra during recent experiments at Surrey University.

LOUDSPEAKER suspension

A. J. Waldron describes an inexpensive method of combating Neighbour Effect

AFTER some months of worrying the neighbours with 'vibration' caused by the use of two rather large loudspeakers, admittedly at high volume, a system had to be developed to alleviate this problem. Many ideas were tried out, such as standing the cabinet on a foam rubber cushion. None was satisfactory until a suspension system was devised.

The system to be described is now in use at our Swan Street Studio at Torquay. It passes very little bass vibration through to the (wood) floor and it was possible to monitor at considerably higher level than with the speaker resting directly on the floor.

The Suspension System consists of a heavy base (a), a suspension frame (b), and two platforms (c) (**fig. 1**). The base for the prototype was made from an old table. The legs were cut down and the table turned upside down to form a heavy and stable base. Angled section metal was used to 'stress' the frame and to secure the springs that hold the platform in place.

Base

Cut the table down as shown. Measurements are not given as these depend on the size table available. Next, cut the angled metal as required. In the prototype, it was only necessary to drill four $\frac{1}{8}$ inch diameter holes for the springs. Large wood screws with washers were used to fix the angle pieces to the base. When the angle pieces have been drilled, screw them into place. Note that the base must be big enough to allow the suspension frame and platforms plenty of clearance. (At least 150 mm all the way round.)

Suspension Frame

The suspension frame is made entirely of metal angle. The dimensions of it will depend on the user's needs, but general terms will be used as follows:—

Sections A and A1 will be W+50 mm long and sections B, B^1 and B^{11} will be D+50 mm long where W=width of speaker cabinet and D=depth of speaker cabinet.

Before assembly, the two sections A and A^1 (see **fig. 3**) should be drilled to take the springs. The frame is assembled as shown in the figure and the angle pieces bolted together using the nuts and bolts usually supplied with such angle section metal packs.

Platforms

The platforms in the prototype were made from chipboard, but any convenient material may be used. When they have been cut to size and drilled in the appropriate places, both platforms are bolted into place using ordinary $\frac{1}{4}$ inch nuts and bolts.

If the unit is to be painted, now is the time to do it.

Springs

Old bed springs were used in the prototype. The type that the mesh is held on with. They were very strong and do the job extremely well. It is necessary to use powerful springs if the speaker is a heavy one. The platform suspension frame should hang horizontally and parallel to the base. With the speaker on, the springs should not drop the platform by more than about 50 mm.

The suspension system does not necessarily have to stand on the floor. It can be raised to any level required as long as whatever it is standing on is stable. In our Swan Street Studio, we have raised it so that the speaker 'looks' over the top of the mixing desk.

FIG. I

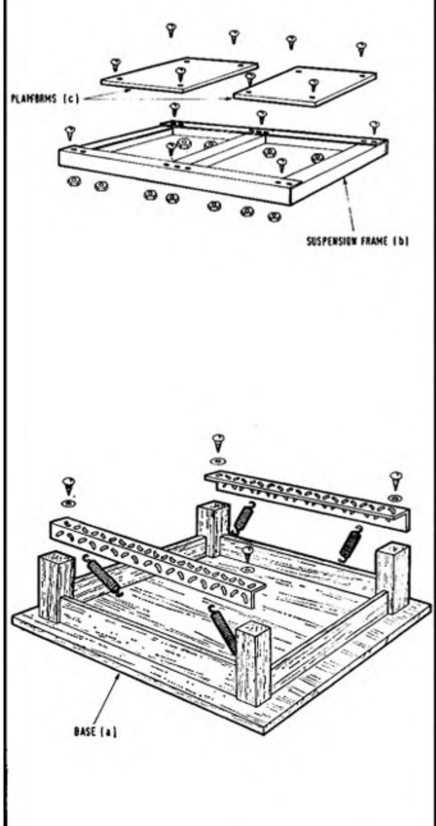


FIG. 2

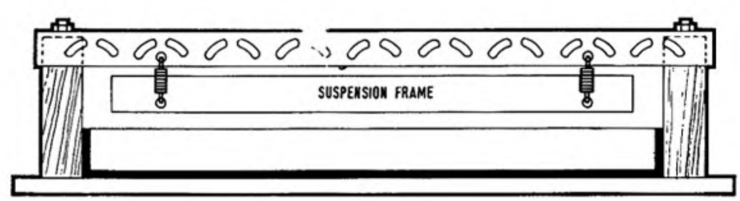
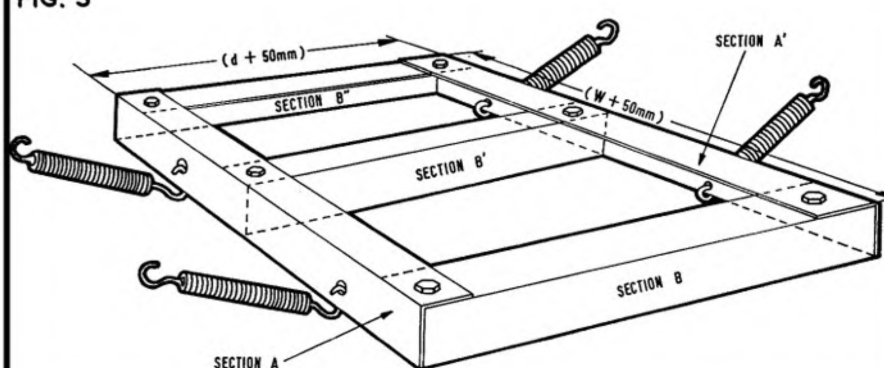


FIG. 3



around the studios

MAXIMUM SOUND

By Keith Wicks

MAXIMUM Sound Limited in the Old Kent Road is under the management of Dave Hadfield who designed the general studio layout and specified the facilities to be provided.

As you can see from **fig. 1**, it was necessary to build the control room perpendicular to the main part of the studio. Consequently very little of the studio is visible from the control room but Dave Hadfield told me that CCTV would soon be installed to overcome this problem.

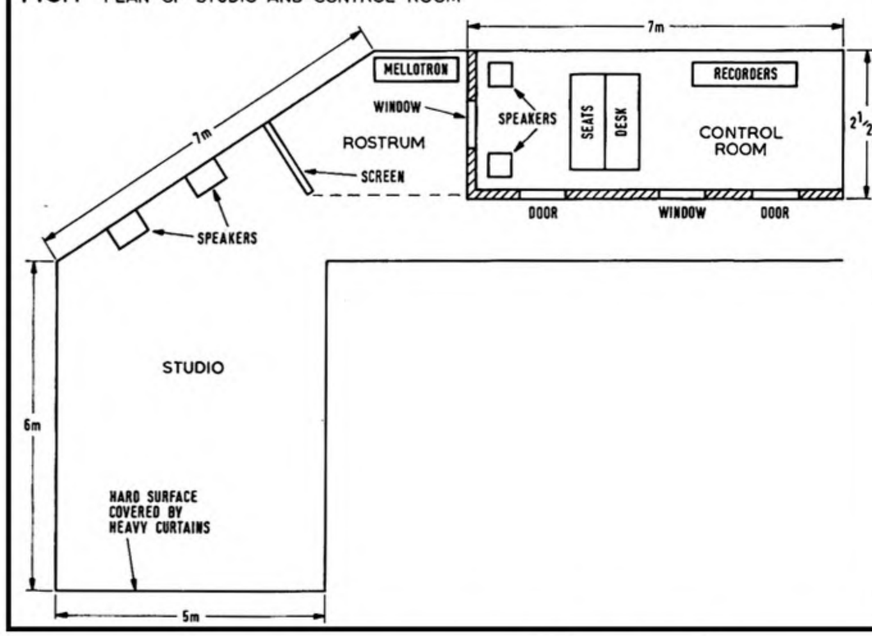
It is useful to be able to see the artists during a session, especially when directing them to alter their position relative to the microphone. Another advantage of the visual link is that it is often helpful in locating a source of unwanted noise. For example, at one studio I visited, occasional clicks were heard on top of the music being recorded. By observing the musicians, the clicks were traced to a noisy key on one of the clarinets. For these reasons, although not essential, it is certainly desirable for the engineer to be able to see the artists.

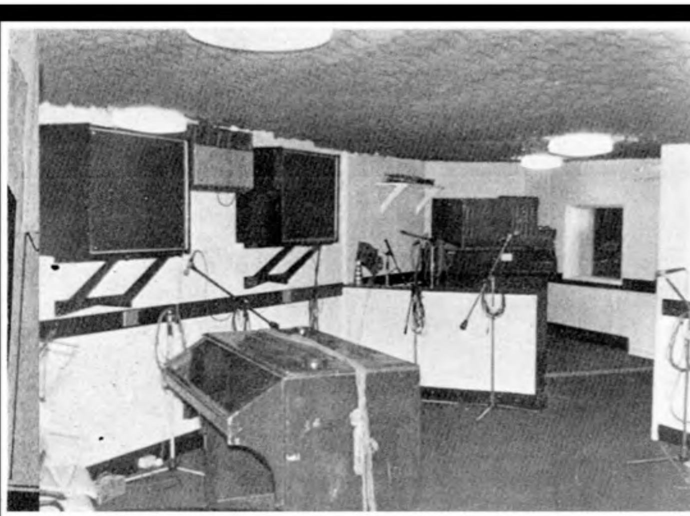
Maximum's studio is fairly dead, the ceiling being covered by a thick layer of foam rubber with an uneven surface. At one end of the studio, the wall, which is acoustically reflecting is usually covered by heavy curtains. When recording brass instruments, however, the curtains are drawn and the musicians play towards the wall. Microphones are positioned so as to pick up the reflected sound which results in a quality the engineers find preferable to that obtained by simply recording the direct sound from the instruments. This technique is sometimes used on vocals and it is interesting to note that, in spite of the development of electronic manipulation (filters, equalisers and so on), Maximum Sound makes full use of acoustic effects. Another example of this is at the other end of the studio. Dave Hadfield once played with a pop group and found that drums sounded better on a rostrum than on the floor; so when designing the studio, he had a rostrum built for the drummer. This is situated away from the main studio area and is partially screened in order to confine the sound and obtain good separation.

Maximum Sound have a Mellotron as well as grand and upright pianos, so they are better equipped than most small studios in this respect. Microphones used are by Neumann, AKG, Beyer and Fi-Cord. The studio loudspeakers are wall mounted and can be seen in **fig. 2**. These are the popular Tannoy *Monitor Gold* units (38 cm variety) in cabinets of Tannoy design which, I am told, are much better than the Lockwood cabinets they used previously. Around the studio are a number of panels containing headphone sockets, four separate foldback groups being provided.

The control room at Maximum is different from others I have seen in that the desk is

FIG. 1 PLAN OF STUDIO AND CONTROL ROOM





used to divide the room into artists' and engineers' territories. There is a door to each section, which means that musicians can enter their own part of the room and listen to what they have recorded without getting in the way of engineers.

The desk (fig. 3) has 20 inputs and is intended to provide 16 outputs although, at the time of my visit, they were equipped for only 8-track, and some amplifiers were missing from the mixer. Pete Walsh, the studio maintenance engineer, designed the electronics.

Along the top of the desk is a row of 10 meters, the central pair being used for mono or stereo reduction, and the other eight for the desk outputs. When Maximum go 16-track, push buttons will be used to allow each of these eight meters to cover two outputs. (They use VU's in preference to PPM's simply because VU's are provided on their Scully recorders and they find it convenient to have all the meters the same.) Below the meters on either side of the desk are the line output amplifiers, microphone amplifiers equalisers, echo return units and channel faders. These are shown in fig. 4. The microphone amplifiers incorporate miniature lever switches for (1) high/low level inputs; (2) extra 10 dB gain if required; (3) bass cut; (4) treble cut. A rotary switch is provided to select the bass cut-off frequency (50 or 200 Hz) and there is a pre-set level control potentiometer. The equalisers have frequency-select and gain controls for bass, middle and treble regions, plus a pushbutton to switch the correction in or out. On the echo return modules, push-buttons are used for echo-select, and slide faders for level control. In addition to echo return facilities, these units allow the selection of channels to outputs by means of rotary switches.

The central area of the desk houses the complex headphone and loudspeaker monitoring systems, echo-send units, echo-send faders and main output faders.

There are three recorders in the control room: Scully 8-track, Scully 4-track, and a rack mounted single-track Teac. The head blocks on Scully machines are interchangeable, the 8-track model being used for 4 or 8-track recordings, and the 4-track model for single, 2 or 4-track work. The recorders can be seen in fig. 5 which shows engineer Roger Wilkinson editing 6.25 mm tape on the 4-track machine. A Fairchild spring reverberation system is mounted on the bay in the foreground. This is just below the Teac recorder which is used for echo, sometimes in conjunction with EMT plates. The plate output is fed to the recorder, tape-delayed, then mixed in with the original sound.

A Pye limiter is mounted below the spring reverb system, plus a patchboard and a number of headphone amplifiers.

On top of the control desk (fig. 3) is the following auxiliary equipment: on the left, an Astronic equaliser, and to its right, a limiter by Spectra Sonics which is used on most vocals, bass and drums.

Each of the four Tannoy Golds used for monitoring in the control room is fed from a 50 W Quad amplifier.

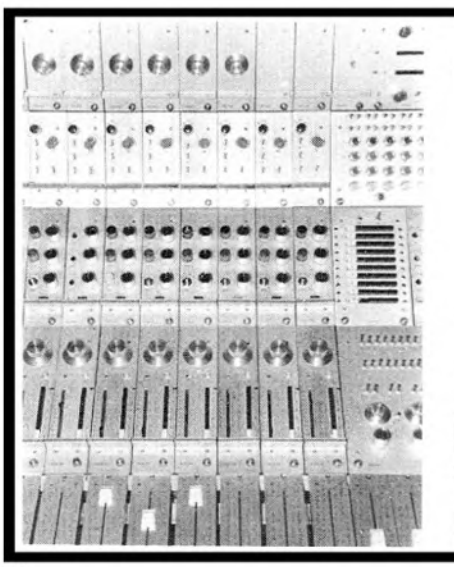
Large slide faders, mounted on the wall to the right of the desk are used for fading the multi-coloured lights in the control room. Dave Hadfield explained that it was worthwhile spending a little money in providing pleasant, colourful surroundings for customers and staff, and I'm sure he's right. Some sessions go on all day, and it must be rather depressing working for such long periods in some of the control rooms I have been to. I wonder which studio will be the first to have sound-modulated lighting, pulsating in time to the music.

My visit to Maximum Sound was timed to coincide with a Manfred Mann session. Manfred, with his ex-drummer Mike Hudd, has written many jingles for TV commercials, including BEA, *Dulux* and *Maxwell House*. Their music is now rather different from the fairly conventional pop they used to play and, like most sounds, rather difficult to explain in words. During my visit, Bernie Levin (who does *not* write for the *Daily Mail*) was adding an alto track to an 8-track tape of *Lady Ace*, one of the titles for the group's new LP (see fig. 6). This was a mixture of pop and rather modern jazz, which I found most enjoyable at times. Some of it was rather beyond my comprehension, however, leaving me with a mystified feeling and not much else—but that's my problem. Anyone interested in this sort of music might like to know that the LP will be issued by Philips in July, on the Vertigo label.

The basic hourly fees at Maximum Sound vary from £10 for single track recording, to £22 for 8-track. Editing single track costs £8 per hour, 8-track costs £16. When the 16-track equipment has been installed, the charges for this will be as follows:
Recording: £30 per hour
Reduction: £24 per hour
Editing: £22 per hour
Tape (50.8 mm): £25 per reel.

It seems that we are going to see a great boom in the recording business in the seventies with bigger and better equipped studios being built. As well as going 16-track, there are plans at Maximum to knock down walls and enlarge the studio. I look forward to a return visit when this has been done.

around the studios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5


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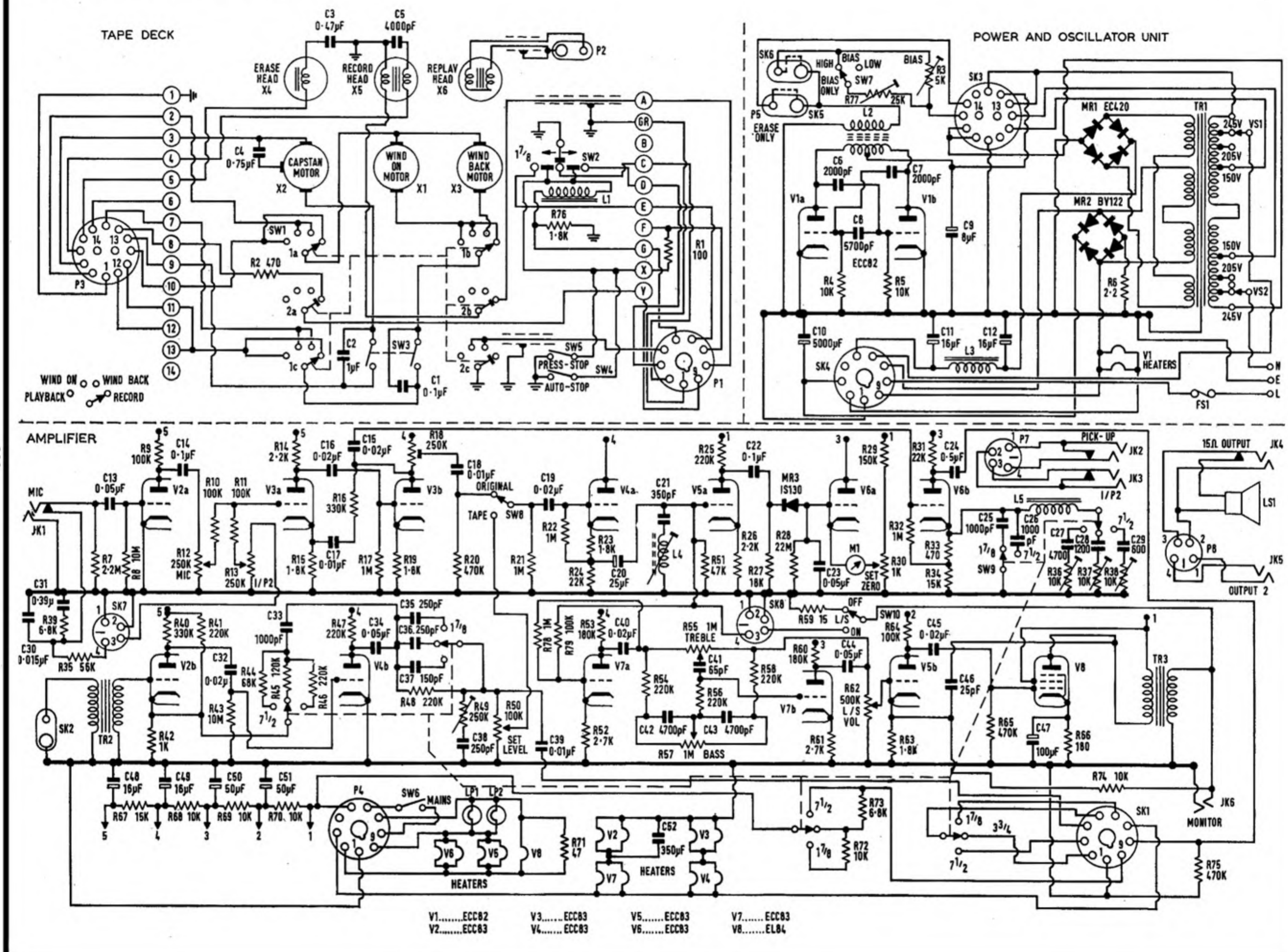
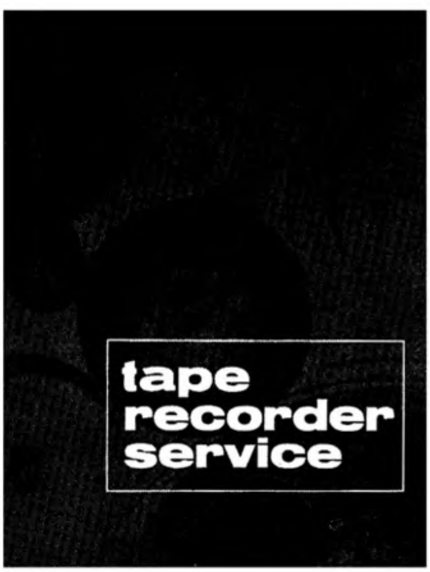
FIG. 1 FERROGRAPH 633 CIRCUIT DIAGRAM





FERROGRAPH SERIES 2 TO 5 TAPE DECKS

BY H. W. HELLYER

INDIVIDUAL differences between decks of the successive marques of Ferrographs may be significant to their owners, but we must take them in our stride and keep to the general case. First, to underline some previous remarks, a few illustrations of trouble spots.

Fig. 2 is a below-deck view of the selector switch and the Allen screw that retains it. This screw, in the collar under the eccentric cam, should be checked for security, as should its companion. A number of troubles, including incorrect movement of the pressure arm assembly, can stem from the small fault of a loose screw. When the usual causes of wow have been investigated, look for the clearances at the rocker arm of the head gate assembly and note that the linkage lever can rub on the deckplate if the mechanism has been strained at some time.

The sort of strain against which Ferrograph have always warned users, in upper-case print, is caused by attempting to turn the selector control when the tape is running, instead of pressing the stop button, as is correct. (This applies to all later models—one or two early versions operated differently). But the man who pits his wrist torque against the stiffness of a Ferrograph control deserves a breakdown, anyway. It is an old engineering maxim—'if it won't go, don't force it'—which comes to mind when thinking of the problems of operating these rocklike bits of machinery, removing reluctant knobs, starting Allen screws, etc.

The last digression becomes relevant when we think of the selector knob, also secured by Allen screws and the tightness of its own fit on the spindle. Most important rule is to use the right size of Allen key—packs are sold cheaply in practically every hardware store and the excuse to use a graunched end of the oldest screw-driver you can spare should never arise. I have found that the trick of removing self-tapping screws or steel-to-aluminium bond screws is also applicable to the overtight Allen screw. A steady application of force in the *tightening* direction, followed by a slight releasing jerk. The amount of force has to be gauged carefully, and some practice is needed to avoid snapped Allen keys—these, if they are any good, being hard enough to be brittle.

The top-deck assembly of brass pressure roller and rocker arm, and the linkage to the tape guide pin, the two pressure pad arms, and the autostop feeler, will be familiar to users and we have not wasted an illustration upon it. The obvious, general servicing notes apply: make sure the feeler can ride cleanly into the curved slot, that its pivot pin is secure and the barrel clean, that the linkage below to the switch, especially the angled rod, is free to operate correctly. Check that the start action, from the main starting bar and the cam and lever assembly, engages the switch fully and properly, and that it breaks cleanly when the solenoid is energised and the motor switch closed.

Note that the motor switch, lying diagonally between the rewind motor and the selector switch assembly, has a suppressor capacitor across it. Leakage at this point can cause a few problems. The other switches, that is, autostop and manual stop, do not have suppression. The greatest trouble I have found comes from the autostop switch and its bent rod, which can be seen at the bottom of **fig. 5.** On the *Series* 6 we find its position altered but its function remains the same. It is operated by a feeler arm on the later models, whose vertical plastic pin rests against the tape just after it has passed the right-hand bobbin guide. So the autostop now shorts out the hold-in solenoid after the tape has cleared the pressure roller, not with the tape still in the head gate as before. A small point no doubt, but the position of the autostop switch now makes servicing a little easier.

Coming to the subject of the hold-in solenoid, the armature of which is shown in **fig. 3** (once your eyes have become accustomed to the dappled view, for which I must apologise). Although the operation of the main bar, via the start key beside the selector switch, closes the main motor switch and imparts the action by providing power, it is the hold-in solenoid which then becomes energised and keeps the mechanism in the playing position. Power for the solenoid is derived from the HT line and between 25 and 90 mA DC are needed. In the earlier models, the solenoid winding was used as part of the smoothing system (see letters F and E of figs. 1 and 2 accompanying the April 1970 article). The *Series* 6 circuit, shown this month, represented by the 633 had a 100 ohm, 1W resistor from the HT line.

In parallel with the autostop switch is the push switch for manual stop action and these do no more than to short-circuit the solenoid.

(continued overleaf)

FIG. 2 Main selector switch, showing Allen key locking collar.

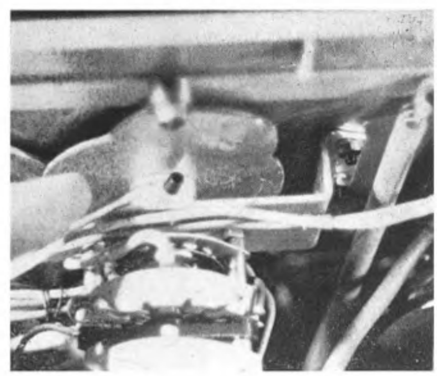
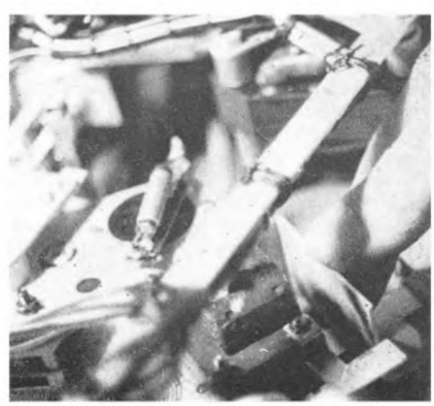


FIG. 3 Solenoid flap armature held off at its limit.

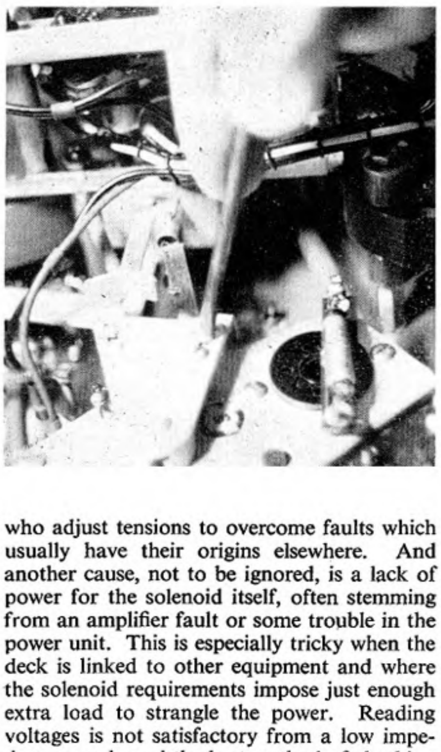


When the armature depicted in **fig. 3** is released, return spring action takes over and the locking for the pressure arm assembly is tripped as well as the brakes being applied.

Note in particular the action of the mechanism when the solenoid disengages to release the brake locking, because this consists not only of the main brakes, differential in their action so that the supplying motion receives greatest retarding force, but also a flywheel brake, a pad on an arm (see **fig. 5**), and a motor brake, which is a padded spring. These extras are not common to all models and, where they are fitted, can become an added cause of trouble. In particular, look for ingrained surfaces on the flywheel brake, a binding pivot, a motor brake too closely set, and binding levers of the main brakes where the slides have become impregnated with old grease that has gathered dirt.

Sluggish stopping and starting, often so bad that it looks like a faulty motor problem, can be caused by such minor troubles as thick lubricant between sliding parts. The long return springs for the lever system on the Ferrograph deck are easily damaged, especially by people who adjust tensions to overcome faults which usually have their origins elsewhere. And another cause, not to be ignored, is a lack of power for the solenoid itself, often stemming from an amplifier fault or some trouble in the power unit. This is especially tricky when the deck is linked to other equipment and where the solenoid requirements impose just enough extra load to strangle the power. Reading voltages is not satisfactory from a low impedance supply and the best method of checking is to insert a meter in the lead to the solenoid itself.

FIG. 4 Speed selector bracket with adjustment for limiting idler wheel lever travel.



The problem about checking voltages is that the applied voltage differs from machine to machine. In the earlier models, where the application was from the HT line, we would expect to read 278 V on play, 240 V on record at the 'hot' end of the solenoid coil, and 263 V and 227 V respectively at the output terminal (see fig. 2, April 1970). Later we find the solenoid tapped to the low-voltage supply for DC heating of the preamplifier valves, 12.6 V, as in the circuit shown this month. Supply is via the series resistor R1 (preventing shunting when the stop action applies a short-circuit). But now we find that the return line is via the equalisation and speed-change switches (SW2 and SW9) to prevent operation unless both are set at the same speed. Then it becomes necessary to insert another resistor, R76, 1.8 K, to prevent switch clicks and keep the loading constant. R76 is in parallel with the solenoid. In yet another version, the solenoid is in series with the power supply in the negative DC line and readings of –16 to –19 V are obtained. This is used in the *631* series, but the *632* and *634* are similar to **fig. 1** in this respect.

Earlier, we mentioned the start selector engagement and the inference may have been drawn that we were concerned with the function rather than the operation. These can be almost completely divorced and investigated separately. When the main switch is turned to record or play, the motor drive is applied to the flywheel via separate idlers for each speed. Speed selection is by the position of the bracket shown in **fig. 4,** which limits the travel of the arm on which the unwanted idler is mounted. Adjustment of the bracket is possible and the tightness of the locking screws indicated in the photo should be checked. Make sure also that there is absolute clearance from the idler wheel lever, to avoid irregular running problems. Check the idler bearings for congealed lubricant, and look for damaged washers and circlips. We have had several reported cases of wow which originated in the idler-to-motor engagement, and which were always aggravated by poor bearings on the idler itself. The slide arms, with their return springs, must not be bent (there is a temptation to bend up the limit end unnecessarily) and should be clean, with only the merest smear of medium grease where they pass through the guide slot.

Motors themselves will submit to removal and stripping, cleaning and reassembly, much more readily than those of some other machines I could name. Bearings can be run into alignment with a bit of patient tapping (with a mallet, not the nearest brick), and a touch of lubrication will not come amiss. By the time the characteristic Ferrograph resin starts to smell, you know you are getting a little late with your lubrication! It is even possible to reset the bearings on the spooling motors by a slight tap on the lower pillars while turning the spool carrier by hand.

The flywheel, however, is a different proposition. Its bearing is packed with silicone grease; oil simply washes this away. Upper and lower bearings are ballraces and the end rider devices need some care in their setting both to avoid unnecessary end play and to prevent irregular running. The capstan is bonded to the flywheel spindle, being a brass cylinder with a neoprene sleeve, and this is mounted in a bearing with a small spring as well as a ballrace. Taking the assembly to bits for cleaning and lubrication requires delicate handling. If the top cap is slackened off about 3 mm, the flywheel moved vertically very gently as a check for end-play and then the end cap replaced into the position that limits this most effectively, there should be little trouble. Make a final test by spinning the flywheel and then run it normally, apply the 'Stop' action

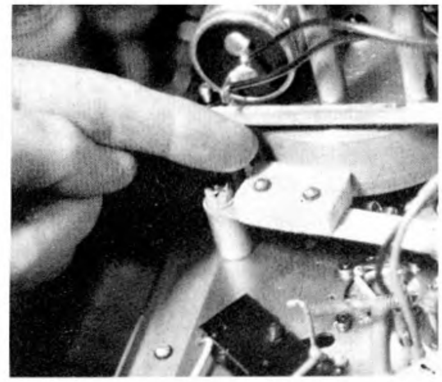


FIG. 5 Flywheel brake and autostop switch.

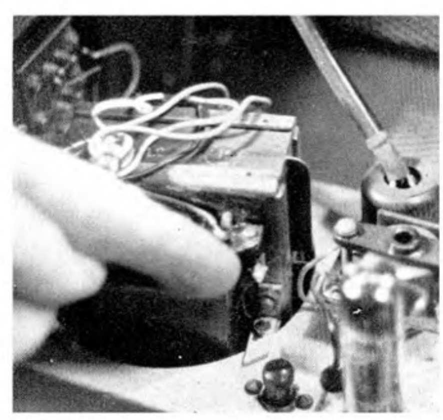
and note that it should run on for at least twelve seconds. (Flywheel speed at 19 cm/s is 177 RPM and correct capstan diameter is 20.5 mm.)

There have been several flywheel shape and mass alterations. When replacing a flywheel, neither a cheap nor an easy job, make sure you are ordering the right one and, while you are about it, that what you are ordering is the complete assembly, not just the flywheel alone. The parts list reads rather ambiguously here.

In next month's contribution, we shall illustrate the drive for the rev counter, one of the causes of take-up motor trouble. As the drive for the clock-type revolution counter is by cable from a gear on the take-up motor spindle, any tendency to bind can cause spillage by retarded take-up. There is a simple adjustment on the cross-pinion of the coupling gear and the trick here is to make the adjustment while the machine is running, not by trial and error when the machine is stopped. This is not easy, and sometimes means one must remove the deck, use lengthened coupling harnesses for the plugs and sockets and then rig up the deck to operate normally in some form of mounting jig. A lot of trouble for the owner-driver; normal routine for the full-time service agency.

In the next issue I hope to show how Ferrographs can be brought up to their electrical peak with the aid of a 20 kV meter, a valve-voltmeter and an audio generator.

FIG. 6 Rotating transformer to minimise hum.



VTR CIRCUITRY PART 7

BY HENRY MAXWELL

INTERRUPTION in the series of articles, due to illness, leaves us with this short conclusion, partly a summing-up, partly a brief skate over what was left unsaid.

Accurate registration depends very much on servo control and this is a major part of VTR circuitry. But the treatment of servo control and regulation is a much wider subject applicable to many audio tape recorders, so the detail of control circuitry can be dealt with separately and need not occupy us during this series.

Now that colour television is upon us, video tape recording of colour signals is necessary. This is not just a step up in frequency coverage. For the purpose of determining the extent of the problems encountered when trying to tape colour, we shall look at the additional factors that the system imposes. This implies that the usual determining factors have been taken care of: signal-to-noise ratio, transient response, freedom from dropout, frequency adjustment, deviation limits, servo control, and so on.

Each of the foregoing factors has its own possibility of error. Chromatic recording not only increases these possibilities, it adds to the list. Errors can affect the hue, the saturation, the chroma-luminance registration and, not least, the chroma-signal-to-noise-ratio.

Noise in colour systems lies in the 2 to 4.5 MHz area: system noise, including head-to-tape noise, and the moiré effect which is caused by the frequency modulation. Difference between colour and monochrome taping is that this impaired s/n ratio can cause random changes in hue and in saturation when the signal is decoded.

The same things that improve the true-noise figure give greater moiré effect. To reduce this needs higher frequencies. So high-band VTR machines are made with carrier frequencies in the 9 MHz region (the 525-line Ampex, for example). This results in a 46 dB s/n figure above the critical 1 MHz level and, by tailoring the slope of the pre-emphasis, this advantage can be retained.

The next problem is head loss when higher carrier frequencies are used, and this can only be overcome by better head design. Narrower gaps to handle the shorter wavelengths of the higher frequencies give less output. Gap depth becomes an even more significant factor. An amusing sidelight was revealed by C. H. Coleman of Ampex in a paper to the Royal Television Society which dealt with multiple generation dubbing of colour tapes. 'It appears that after years of trying to increase head outputs,' he said, 'one is now faced with having to deal with too much of a good thing.' The combination of large initial gap depth and high efficiency meant that, in the later stages of head life when the gap has worn down considerably, the output becomes so high that the playback amplifier has to be adjusted to compensate.

Something for nothing is not the usual way of the electronic world. So it is not surprising to learn that we are back to normal with chroma-luminance registration. This can cause the peculiar 'after-print' effect of the colour being displaced to one side. The effect is something like that of the colour printing in our schoolboy comics, with the added nuisance that the effect is not tolerable on a changing scene.

Good phase response is needed to overcome this trouble. Not least of the troubles associated with the later systems is a velocity difference between machines, which has led to some ingenious circuitry for keeping them in step.

At present, we are concerned with the ways in which the designer of a helical scan VTR can cope with colour, bearing in mind that he is going to meet the problems we have outlined. We can see that the main restrictions are going to be the smaller available bandwidth (than transverse scan systems) that is available and the timebase errors that can accrue. Two main systems are employed to reduce these problems. The simultaneous system uses three types of recording (a) Pilot Signal NTSC, (b) a modification of the foregoing and (c) separate track recording. The sequential system also has three methods of beating the error bogey: (1) tri-colour sequential recording, (2) chrominance signal line sequential recording, and (3) field sequential recording.

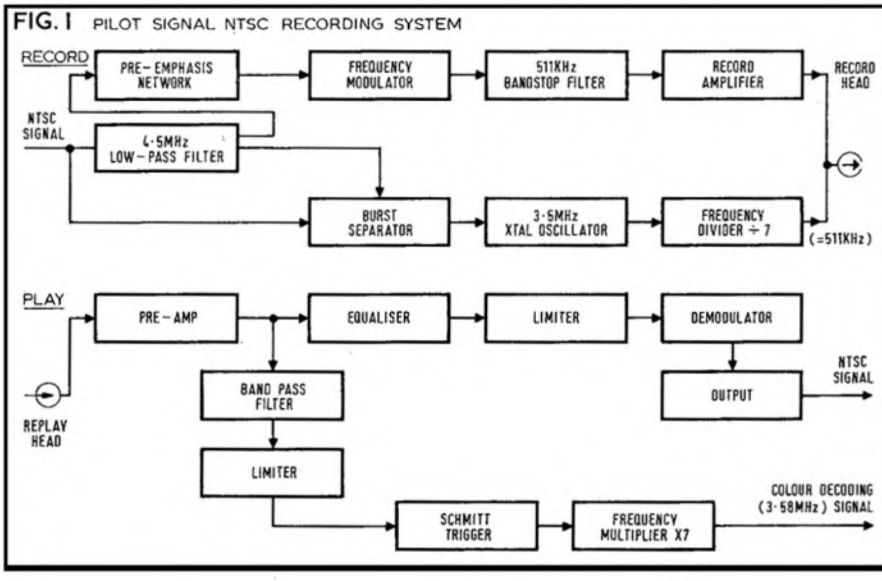
(a) A pilot signal frequency is chosen, synchronised to the burst signal of the NTSC signal being recorded, and an exact sub-multiple of a carrier frequency which must have the same phase-shift on replay. So we need a bandwidth capable of coping with the NTSC signal and good linearity in the FM chain. Circuitry does not have to be too complicated (compared with monochrome systems). The divided sub-carrier, following the Ampex system, is 3.58 MHz which, divided by 7, gives us a pilot signal of 511 kHz. The block diagram of **fig. 1** shows the basic circuit on record and play. Note that this is video-to-video recording and not via RF. The NTSC signal follows two paths, and is combined with the pilot tone when recorded. On replay, after preamplification, the 3.58 MHz signal for colour decoding is reconstituted by being picked off and fed through band pass and limiting circuits to a Schmitt trigger and a 7-times multiplier, while the video information is equalised, limited, demodulated from the FM sub-carrier, and amplified.

(b) The modified system is an attractive proposition for a cheap VTR set-up but is incapable of a high resolution. Its block diagram is a little more complicated (see **fig. 2**) and the details of frequencies have been omitted. This brings us into the sphere of the television engineer and the niceties of dividing and applying the composite colour information need not concern us.

The video signal is split into luminance and chrominance (colour-only) information. The luminance (Y-signal) goes via an adder to be applied as a tailored signal to the frequency modulator, while the chromaticity is first used to make up a pilot signal as before, and then used to control the modulating frequency.

On playback, frequency splitting again presents the Y signal and the pilot signal is multiplied, converted and compared with a standard 3.58 MHz oscillator before application to a converter. The whole adds and becomes the original Y + 3.58 MHz for video application. *(continued on page 244)*

FIG. 1 PILOT SIGNAL NTSC RECORDING SYSTEM



RECORDING STUDIO TECHNIQUES

FROM the early days of electrical sound recording and transmission, engineers have had to measure accurately the electrical levels of audio in amplifier circuits and land-lines. From the beginning of the 1930's, the BBC realised the importance of a logarithmic meter scale to simplify gain riding. In 1938 the desirability of a logarithmic meter reading peak programme level rather than the average volume became accepted. The BBC *OBA 8* amplifier was in fact the first mike amplifier fitted with a true PPM instrument, similar to the ones in use today. At the same time, the Americans developed the VU meter with an entirely different concept, that of reading average power or apparent loudness of the programme.

Before going any farther let me explain that the controversy over the use of PPM or VU is perhaps the biggest in the recording industry. Engineers trained to use one type of meter do not usually change their allegiance to the other type. I infinitely prefer the PPM type instrument but I hope that, by describing fully both types, the reader may be able to make up his own mind on this extremely important matter.

The BBC type is by far the most widely used PPM in the world, though the German type using a light source with a very small mirror movement is gaining popularity. The BBC PPM incorporates a scale from 1 to 7, dots being provided below 1 and above 7. Each division from 2 to 7 corresponds to a 4 dB difference in level, the difference between 1 and 2 corresponding to between 4 and 6 dB depending upon the age of the instrument, leaving of course an infinite range downwards below 1 and upwards above 7. Until recently all PPM's in use by the BBC were valved, having a right-hand current zero so that the meter needle, during switch-on, went from the extreme right of the scale to the left end. With the latest transistorised PPM's, the mechanical zero is at the left hand end ; the scale is identical. In the early days, BBC meters were very similar in scale appearance to PPM's but, as previously explained, were not peak reading instruments. Line-up tone corresponded to 7 on the scale, this being 8 dB below peak transmission or recording level. In practice, engineers peaked programmes at 7. Since the meter under-read the real peak of the programme by an average of 8 dB, this corresponded very roughly indeed to peak transmitting level. Very similar remarks therefore apply to these old type meters as apply to the operation of VU meters. The earliest *OBA 8* PPM used Mazda *AC/VP1 V051-μ* valves in conjunction with diodes, a stabiliser and a time-constant circuit to operate a meter whose specification was, and still is : that a 4 mS square pulse applied to the input circuit of the complete unit at peak level should raise the meter needle to within 80% (i.e. 2 dB) of the final meter position with a continuous tone of the same amplitude. Although peak level was originally set at seven and line up replay at five, during the '39 to 45 war, four became line-up level equivalent to zero dBm or 40% modulation of AM transmitters, and this therefore became 8 dB below peak transmission or recording level. An attack time of 2.5 mS was provided together with a 3 second decay time for 26 dB decay (i.e. from 7 to 1 on the scale). A calibration switch is provided to set the sensitivity of the meter accurately and in practice the PPM has proved extremely reliable.

'Volume units'

The VU meter was designed and developed by the American Bell Telephone Laboratories in conjunction with NBC: two independent scales, one graduated in percentages and the other in 'volume units', with zero VU at the same point as 100% (approximately 70% of full scale deflection). The meter was designed to have ballistics such that a pulse of 300 mS duration raised the needle to within 1% of the final position of a continuous tone at the same amplitude. Signals having peaks appreciably shorter than this therefore would be under-read on the meter by an amount varying with the duration of the pulse (measurements being given later). The VU meter consists of a 50 μA movement having an internal resistance of 3.9 K and is driven by a full-wave rectifier circuit originally consisting of copper oxide rectifiers but now of high quality germanium types. A meter connected directly across a 600 ohm line in which 1 mW is flowing would read zero VU. In order for the meter to read +4 dBm as zero VU, it is essential to add a further series resistance of 3.6 K, the normal resistor used with this type of meter. The designer reckoned that the meter would under read on average 4 dB but alas my own measurements have never been as hopeful as this, with the exception of loud organ music with its continuous peaks. NAB test tapes contain line up levels of 4 dB below 32 mM/mm so that such a tape will correspond to zero VU for a peak recording level of 32 mM/mm. Since this type of meter contains rectifiers, noticeable distortion may be introduced by connecting such a meter to a line having an impedance greatly in excess of 100 ohms, the distortion being easily noticeable on a 600 ohm line reaching 0.3% or more. If a VU is connected across a 600 ohm circuit before the recording amplifier, and the source impedance driving the circuit is not very low, then the distortion will be on the tape for ever more.

BY ANGUS McKENZIE (Roundabout Records)

PART 6

PEAK PROGRAMME AND VU METERS

Many engineers have scoffed at the use of magic eyes for measuring level but an accurately set up one can be surprisingly useful, certainly more so than a VU. Unfortunately, so few were accurately set up by the manufacturers. By far the most accurate means of measuring peak levels is to apply a bridged signal to the Y amplifier of an oscilloscope with the time base switched off and the scope calibrated with a tone such that peak tone corresponds to a known number of centimetres deflection. A double-beam scope enables the engineer to see both left and right simultaneously, provided the scope has independent dot positioning on the X axis. For experimental work, an extremely accurate indication of transients is possible and I would like to see such a device marketed with a scale on the cathode ray tube accurately calibrated and driven by a logarithmic amplifier, together with a peak storage circuit as used in conventional PPM's.

In practice the VU meter can under-read programmes having sharp transients such as pianos and human speech by up to 10 or even 12 dB, and orchestral triangles and bells by even more. An average VU under-reads classical music by 4 to 6 dB, and therefore it is probably safer to assume an under-reading factor of 8 dB. VU's do give a reasonable approximation of apparent loudness of a signal, however, but in my opinion this is unnecessary since a good engineer can judge this on his monitor loudspeakers.

In the last few days I have made some very interesting experiments on some different types of meter which I hope will alarm engineers not using PPM's and perhaps make them consider changing. A useful way of obtaining a 4 mS pulse is to record a reasonable level of 500 Hz tone at 76 cm/s for 20 seconds or so. 12 mm of this tone should then be cut out (right angle cuts) and edited on to two lengths of 20 second leader tape. The sequence on the tape will then be continuous level followed by leader followed by a 16 mS pulse followed by more leader. This tape should then be copied to 9.5 cm/s. When the latter dub is played at 38 cm/s, the pulse will be of 4 mS duration because of the speed-up effect, but the level will still be identical to the dubbed continuous level section. The level of the dubbing is immaterial. The meter under test should then be set up such that the continuous tone is at a noted scale division of the meter. If the meter has a long decay time it may be necessary to add bulk-erased tape in between the continuous level section and the pulse, to allow the meter to decay to its minimum reading before the pulse. The pulse on a good PPM should approach 5½ for a continuous tone reading of 6. On a tape which I made by this means, the PPM's on my TRD recorder did in fact correspond precisely to this specification with a 4 mS pulse. A VU meter, however, barely moved from the minimum position. A 32 mS pulse under-read by 12 dB, a 64 mS pulse by 8 dB, a 128 mS

(continued overleaf)

pulse by 4 dB, and only a 256 mS pulse approached a correct reading. Small wonder, then, that VU's are so inadequate for measuring the real peak level being transmitted or recorded on to tape. The same measurements were made using a Solartron AC millivoltmeter which under-read a 4 mS pulse by no less than 40 dB, whereas two models of Level millivolt meter under-read by 20 and 34 dB respectively. By this time I began to wonder if something had happened to my blip but, checking this on my oscilloscope, I found that it still came up to the same peak as the continuous tone. Since some engineers may be interested in copying this technique, I will also mention here a way in which the length of the blip can be checked with the aid of a frequency counter. The blip corresponds to 8 complete cycles of 500 Hz and, playing this into my Racal frequency counter, it counted 10 cycles. I then thought I would be clever and filter off the transients which probably caused the high reading, using a B & K third-octave filter. I was still more puzzled when the count became 13 cycles until I realised that the filter was ringing. I then tried an ancient Leak steep-cut filter at maximum top cut and at last obtained a consistent count of 8 cycles. I mention this lest any reader should fall in the same trap. I should be able to make available a short length of tape for any interested readers wishing to do the same checks.

And now to sum up the effect in practice of different recording level measurement devices. A laboratory test instrument such as a millivoltmeter or even Avo meter may be perfectly satisfactory for measuring continuous tone but would be completely useless for the measurement of even average loudness of a programme, and should therefore never be used for programme purposes. A VU meter can only be used satisfactorily for serious work if the engineer using it has been trained in the interpretation of its readings. If peak recording level on the tape is set up such that zero VU represents a level 4 dB below peak then I would suggest that speech, choral music and piano music should under no circumstances be allowed to peak more than –2 VU and better still –4, although the latter level is unfortunately not marked on the instrument. Loud pops without vocal, or orchestral music in which the brass and percussion section are not balanced too far forward, should not peak more than zero VU and only music with sustained peaks such as church organs should be allowed to peak higher than this. Many readers will have seen VU's in action with the needle almost wrapped round the end stop! In this case they will probably have been set up such that zero VU equals 4 dB below 32 mM/mm and the engineer is allowing the meter to indicate an appreciably higher level. In my own experience, however, when I had to use VU's for mobile recordings (not possessing a portable PPM 10 years ago), I frequently found my tapes slightly over recorded when played back on studio equipment including PPMs and large monitor speakers. It is sometimes difficult to hear slight tape distortion in temporary control rooms with unpredictable acoustics, in the heat of a mobile recording session, though it may be quite audible in the comfort and quiet of the base studio. In the case of the PPM, such a meter will always give the RMS value of the waveform peaks applied, or a reading very close to it in general practice, in which case the meter should be set up such that 6 on the scale represents the normal peak recording level. It is important to check that the PPM's characteristics are up to the tight BBC specification.

It is worth mentioning here that many PPM's have been designed to respond to peak values of the waveform and to register accurately all transits in the audio range. Such meters, although theoretically correct, would be likely to result in too low a recording level since even a 100 μS spit of large amplitude, which would be only just audible in the presence of music, might cause the meter to peak above the level of the music. It must also be remembered that the PPM will not give good consistent loudness comparisons, but here the good engineer will use his ears in selecting the gain required for different types of programme being recorded. The BBC, for instance, usually peak speech at 4½ whereas an orchestral concert should peak 6.

The BBC has now installed a number of transistorised PPM's, for instance in Maida Vale Studio 1 and Radio 3 continuity. Although the total decay time of these meters from 6 to zero is the same as the old type, the rate of decay is different and in fact has been criticised by a number of studio managers. Initially the new PPM falls far quicker than the old, and then begins to slow up near the zero as compared with the old type. Unless the engineer is actually looking at the PPM, therefore, a peak may be missed where in the old type it might just be caught. Also very quiet passages of music will hardly move the new PPM, causing the programme level to be brought up to coincide with the BBC's policy of not allowing more than 30 seconds of modulation to occur below one on the scale, with the exception of serious music on Radio 3, to ensure a good s/n ratio in quiet passages. This has frequently been overdone when the studio manager is confronted with the new instrument, although it happens less frequently with the old type. The PPM should also be checked at regular intervals to ensure that it is calibrated accurately and has not drifted. Incidentally it is possible to slug a PPM causing it to read average loudness rather than peaks by applying an electrolytic capacitor across the terminals of the meter instrument. The BBC do sometimes use this method for comparing loudness.

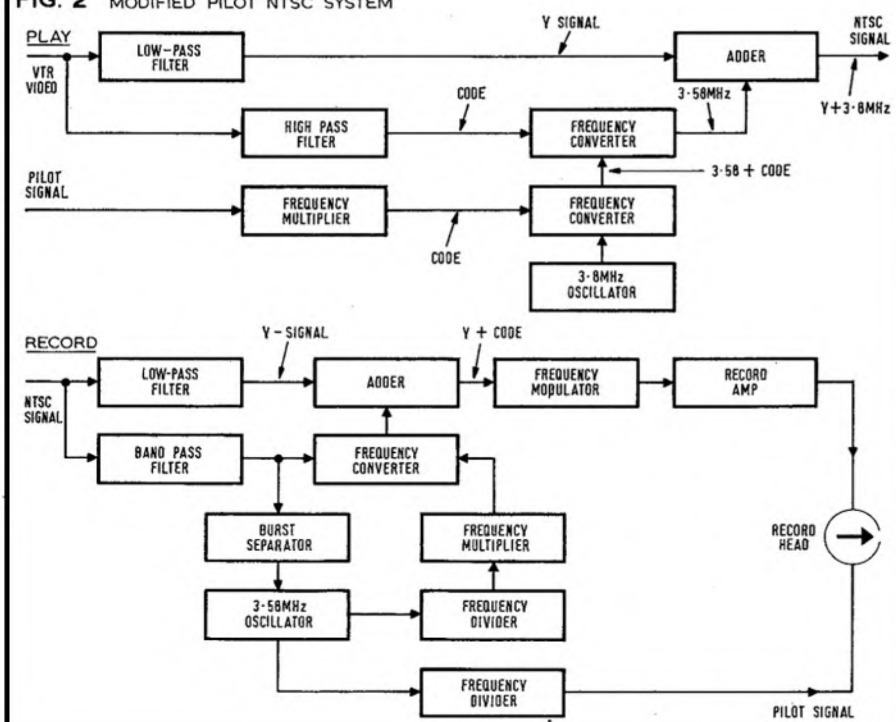
Finally I would suggest that the oscilloscope method previously referred to, with perhaps the addition of linear Z (brightness) modulation, whilst Y modulation is logarithmic, should give extremely good results allowing the trace to become brighter when the programme level nears its peak. Perhaps some enterprising manufacturer would like to make such a device and carry out experiments.

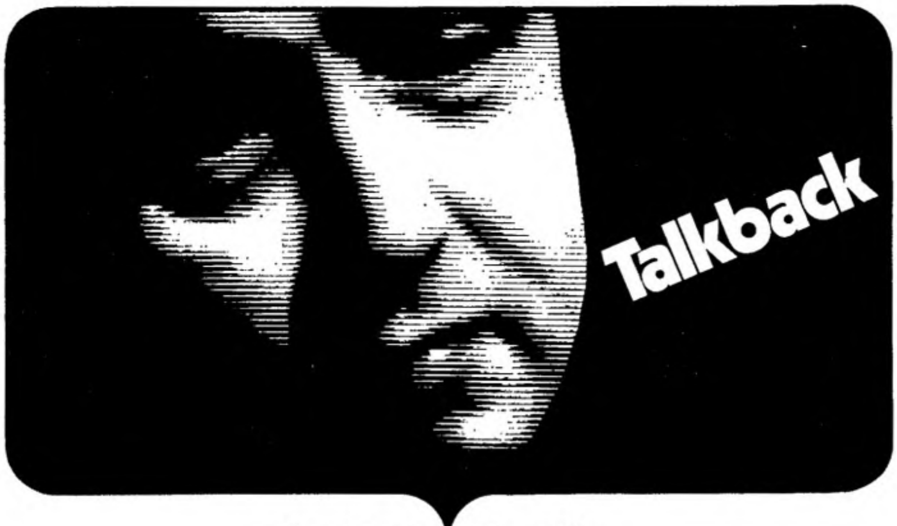
VTR CIRCUITRY CONTINUED

(c) Separate-track recording uses double modulation for chrominance and so we need twice the tape for the same amount of programme. The advantages, to offset this, are that a good picture quality can be obtained with minimal additional circuitry.

From the foregoing, it may be apparent that VTR circuitry has evolved from the simple development of tape recorder techniques into a field (if you will pardon the unintentional pun) all of its own.

FIG. 2 MODIFIED PILOT NTSC SYSTEM





by Peter Bastin

IF you set yourself up as *Superbo Recordings* you must register this horrible name with the Registrar of Business Names. That is, unless your name is John Superbo. The use of any name other than your own is an offence. By the same token, you cannot call yourself INTERNATIONAL SUPERBO RECORDINGS if all you do is record the local pop group on a Grundig. You must not describe yourself as being something you are not. So think about it.

THE SLOUGH Arts Festival is in its Silver Jubilee Year and offered to cine, photographic and audio enthusiasts the opportunity to enter competitions in the amateur film, transparency and tape-recording fields. The judging locale (impressively): Pinewood Studios. Several categories for tapes and a maximum time limit of five minutes. I received a letter and five sheets of information about the Festival on February 28. The closing date was March 1. I received a duplicate wad of information on March 6, *five days after the closing date*. Now this is not nearly good enough. If the organisers want the public to respond, they should be at great pains to make sure that the public know all about it in time to do something about it; not send the information out the day before the contest closes and describe it in the accompanying letter as 'advance notice'.

ONE OF THE tape magazines has been twittering about amateur participation in international video recording contests. Haven't they noticed that, for all domestic hobby purposes, VTR's don't exist? Try buying one of the small Japanese models and you'll see what I mean. How the Sony/Philips Videocassette system will fare, I shall be interested to see. Between them, these enterprising two companies have conquered just about every barrier between videotape recording and the general public, to be finally defeated by the Chancellor of the Exchequer. Edward Heath turned up at the Verdi *Requiem* recording session, which may imply some enthusiasm for the subject. We managed to squeeze Phonopost out of Labour but by heaven we paid for it. Thirty-three-and-a-third per cent.

A FEW COLUMNS ago, I made some candid remarks about stereo which upset the Editor's digestion for a day or two. I suggested that this form of audio is a heaven-sent sales-gimmick: that the real hard commercial value of stereo is the golden opportunity for dealers to flog good, bad and very indifferent 'hi-fi' to good, bad and frequently indifferent customers. To me, stereo, like communism, is an ideal which is rarely, if ever, implemented in the manner in which it was intended. In my limited experience, the only way to listen to stereo is on a headset. To sit like a stuffed owl, immobile between two carefully positioned speakers, is insulting. I like to hear my music whether I am doing yoga in a corner or half-cut under the table. Stereo, one might say, is often an excuse to excuse bad mono. Mind you, I speak as a listener and would in no way wish to deprecate the magnificent efforts of the technicians who produce stereo. My argument is that there are far too many people who accept stereo as (a) the Right Thing and (b) as the Ultimate Sound. I once knew a very peculiar chap who built a bungalow with a hyperbolic parabaloid roof. His lounge ceiling was not the flat sort; it followed the lunatic contours of the roof. He claimed that it was for listening to stereo and that a flat ceiling would ruin everything.

It has been suggested that, in its present form, stereo is not the definitive method for high quality reproduction in domestic circles. Why assume that the concert hall style of presentation, which stereo seeks to emulate, is the most appropriate for the drawing room? Very big names have cast doubts on this point. Leopold Stokowski, on BBC, re-stated an opinion that music should come from all four corners of a room, in effect placing the listeners right in the centre of the performance. This is a very good suggestion, based, I feel sure, on the technique of all-round sound first brought to public light at the Festival of Britain, nearly twenty years ago. This way of listening to music must surely be an improvement upon two fidgety little speakers balanced on the sideboard and the whatnot.

IT'S BEEN with us for quite a time now, to the disgust of some, the amazement of others, and delight of the majority. The mini-skirt. But according to the Americans, not *all* men delight in this tiny garment—especially conductors (and they don't mean the double-decker type). Apparently it's not the legs which disturb them, but their acoustics, if you follow me. Dr Vern O. Knudsen told the German magazine *Neue Revue* that mini-skirted ladies can disturb the balance of acoustics in a concert hall, thus putting to shame all the well-laid plans of the architect. He said 'Clothing is sound-absorbing. A girl in a mini-skirt thus (?) reflects the sound much more strongly than if she was wearing a normal dress'. Knudsen carried out an experiment by putting ten mini-skirted females in a sound chamber and firing a pistol. The same number of girls in 'normal clothing' in the same chamber with the same pistol absorbed over 50% more of the sound than the minis. Conductor Robert Stolz (who apparently does not want to renounce the mini-skirt at his concerts) said 'The difference will just have to be compensated for by special upholstery for the seats!' This opens up enormous possibilities, especially on television. It could well encourage young avante-garde producers into spectaculars where *absolutely no sound-absorption is required.*

THE STANDARD Radio Corporation of Japan (of course) has introduced a television set with an 80 mm screen. Known as the *TV3AUK*, it will work from AC mains, car-battery or nine cells. It weighs 2 kg and measures 180 by 170 by 89 mm. Presumably for small minds and beady eyes.

LETTER FROM J.D.D. of Llanidloes, in *Electrical and Radio Trading*: 'We sold a National stereo compact unit (5 W + 5 W output) and in two days it was returned because the barrack room engineer who lived next door thought that 10 W was going to wear records out ten times as fast as a 3 W.

ADVERTISEMENT: City of Cape Town. Tenders are invited for the supply and delivery of a contra-bassoon. Official tender forms are available from the Orchestral Director, Room 5, Ground Floor, Administrative Building, Longmarket Street, Cape Town.

If you've got an old, spare, contra-bassoon lurking about upstairs, you're in business. But don't forget the tender forms, if you can spare the odd half-hour to write out the address.

IN 1964 I published a series of articles entitled *A Studio Quality Audio Mixer* which outlined the theory and construction of a modular six channel unit which I built for serious amateur recording. The response to the articles was amazing; in correspondence alone I have filled four lever-arch files. This interest was apparent soon after the series started and a set of printed circuit cards was designed to meet some of the demands. From the orders of these alone I can say that at least 100 similar mixers have been built, and there seems to be no end to the number of demands for printed circuits.

Two of the recurrent questions are: 'Have there been any transistor improvements since 1964?' and 'In retrospect has the mixer concept been successful, or would you make any changes if you started again?' It appears, therefore, that the time has come to re-write the series to incorporate all improvements, though the original articles will be referred to. In particular, several of the photographs will be used again, since many of the changes are elaborations on the original concept. There may be consequently slight differences between the new drawings and the old photographs. It is not necessary for existing users to change their units—the concept is the same, and the improvements are often marginal.

The most important change is that silicon planar transistors are now available at a much lower cost than the germanium devices used initially. These transistors have several advantages: the gains are higher, allowing one to design more stable circuits with lower distortion figures as a by-product; the noise figures are significantly better, with the obvious advantages. On the other hand, because the frequency response is several orders better than the germanium versions, R.F. pick-up or oscillations may occur unless the mixer is designed to keep the input leads short and away from the outputs.

The concept has proved very successful; the modular construction allows separate parts to be altered to keep pace with developments in studio practice, and caters for recent trends in the industry. Today multichannel recorders are frequently used both for popular and classical recordings. Pop studios often use 16-track machines, with 24-track in the wings, classical recording is usually 4-track but certain sessions have already used 8-track. Studio flexibility is the reason behind these trends. This reflects on mixer design; a few years ago the mixer was designed to accept a variety of sources, amplify these to a standard level and mix the result into two channels. Each of these channels then passed through limiters, equalisers, filters and tone controls before passing to the recorder. For most applications, two equalisers pluggable to any channel are sufficient, and

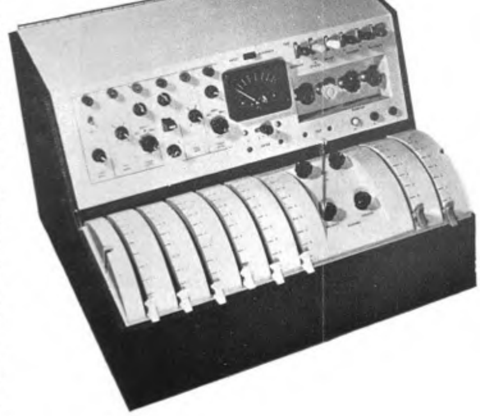
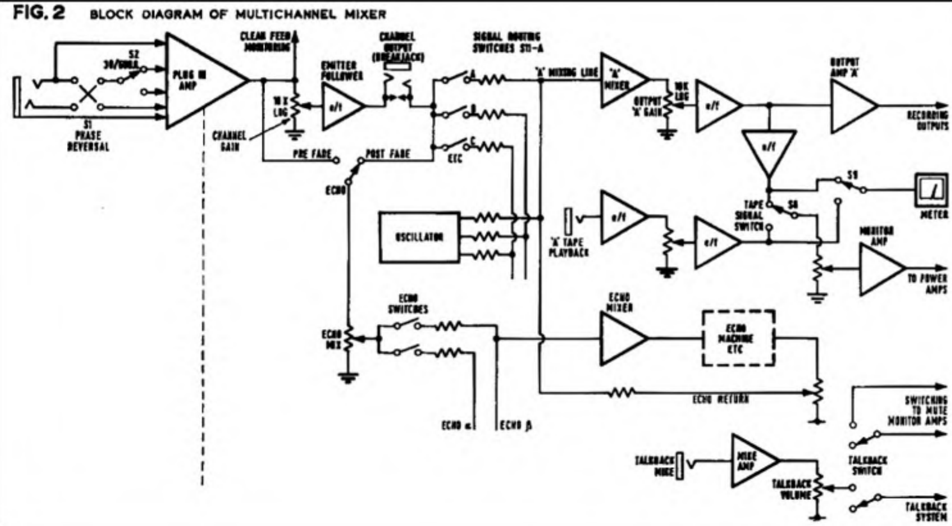


FIG. 1

David Robinson details an updated version of his 1964 design

FIG. 2 BLOCK DIAGRAM OF MULTICHANNEL MIXER



A HIGH QUALITY MIXER

these can be provided easily in the mixer framework. More than two independent outputs from the unit should be provided, however, particularly with quadraphonics around the corner.

This series then will show the **many stages in the design and construction of a sound mixer. A feature of the design is that—being planned on a modular basis—any of the 'building bricks' can be used independently** of the whole **design,** when they would form a **foundation for a simpler unit. Thus, while** the articles describe the complete mixer, any section which is suited to a particular need can be abstracted in the confidence that it will work as well as the full design.

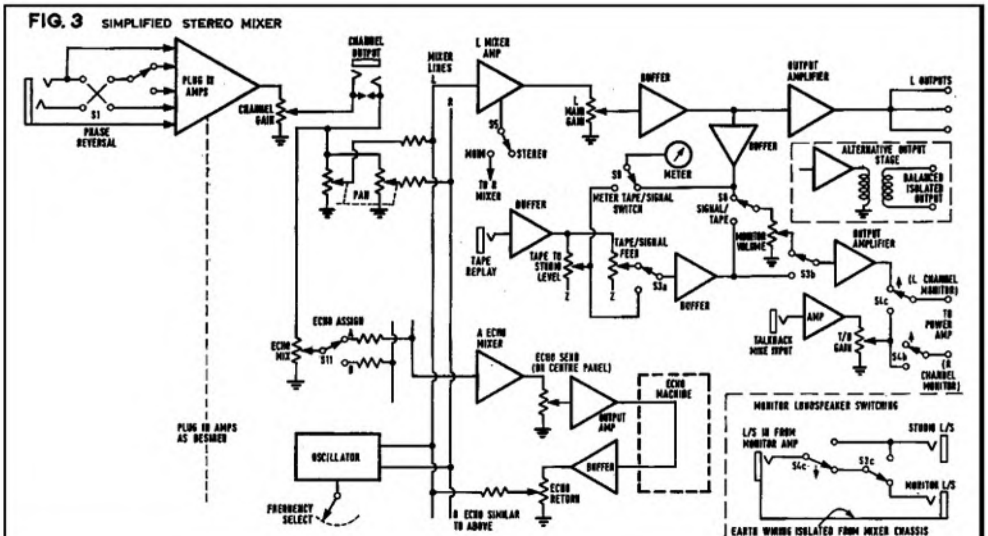
The planning of a studio sound mixer is a slow project, since of necessity the device is fairly complex. Many hours spent with paper and pencil sketching block diagrams and 'logic circuits' will be repaid by the ease of operation of the completed unit. With so many different functions to perform, it is most important to position all the controls so that they fall naturally to hand, and to separate them into groups to make it almost impossible to select the wrong control even in dim light, or, more important, under the strain of that sudden crisis which always seems to arrive. Here colour-coding is of real value, together with position.

One design requirement was that the mixer should be portable so that, for recording away from the studio, in churches and halls for example, the whole equipment could be transported easily. The circuits are very reliable. In general, two types of transistors are used, both of which are cheap and easily obtainable. A mains power pack will be described, which is fully stabilised and can also be used as a bench power supply. Since consumption is low, the mixer can be operated from a battery supply; in these conditions the signal lamps are disconnected.

The first step in design is to draw a block diagram showing the mixer logic. The original mixer had six channels but this has proved inadequate on several occasions; the new design has been expanded to twelve channels. (There is no reason why for large installations this number should not be trebled.) Referring to the **fig. 2** block diagram, the output from the preamplifiers can be switched to any channel; this allows the faders for a stereo pair always to be adjacent to one another.

(continued overleaf)

FIG. 3 SIMPLIFIED STEREO MIXER



With a conventional two-channel output mixer, the switching is somewhat simpler and monitoring facilities can be economised. **Fig. 3** shows this. The actual mixer to be described will be based on this block diagram, but obviously with a modular system it can be expanded to fill any need. (**Fig. 2** is only an extension of **fig. 3,** for example.) Here one often has to split the output from a single channel input in varying proportions in the two channels by means of a pan-pot so that the sound can be placed at any position in the stereo image plane; in the four channel version this facility is not required, or it can be derived by suitable cross plugging. The output of every channel is also available on a break-jack to allow equalisers, filters, etc to be inserted at will. The mixing amplifiers follow, and then an output amplifier to each channel. There are comprehensive facilities for monitoring the outgoing signal and the off-tape signal from the tape recorder. This replay signal can also be switched to the studio to play back to the artists. Talkback facilities are also incorporated.

Each recording session demands a different set of equipment and there are two ways of overcoming this problem. In a fixed installation, or in a professional arrangement, all sources are processed to make the actual mixer stage see the same signal from each. Thus the signal from a microphone is simply amplified, while that from a tuner might be attenuated, and that from a pickup both equalised and amplified. So far, units have been constructed to suit magnetic (either 30 or 600 ohm) and capacitor microphones or line. Also a magnetic pickup cartridge equaliser, tape head preamplifier, tone controls, and a high input impedance module to accept the output from valve gear which might have an output impedance of 100 K or so. This shows the advantages of the plug-in system: the basic mixer is never outmoded because of other equipment changes, and any non-standard equipment can be accommodated into the system merely by building a new unit to plug in the space provided.

A Second Feature

After the preamplifiers, the signal passed to the fader bank, the second important feature in the design, as shown in the photograph **fig. 1.** It was decided to have six separate channels, and a master for each output, which is a total of eight attenuators. If knobs are used on rotary or conventional potentiometers, they should be large so that several channels can be operated simultaneously with ease. With practice, a rolling motion between fingers will allow about three faders to be operated by each hand. However, eight large knobs occupy a large amount of panel space, and this is clearly unacceptable. The solution was to use quadrant faders, which occupy about 38 mm of space but still retain a large travel for accurate level setting, and stereo matching. It is also easy to use several at once, since each requires only one finger. In

FIG. 4 BACK PANEL

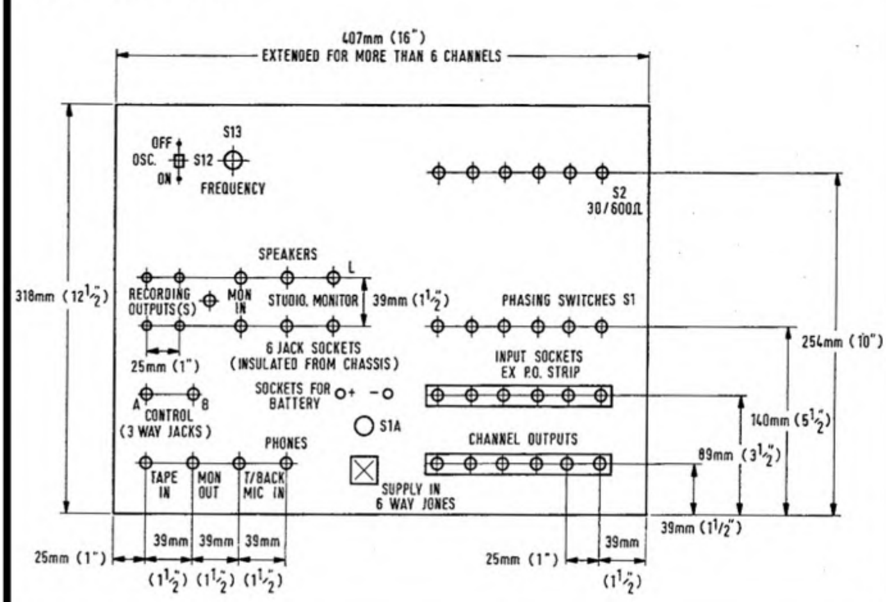


FIG. 5 FRONT PANEL MAIN DIMENSIONS

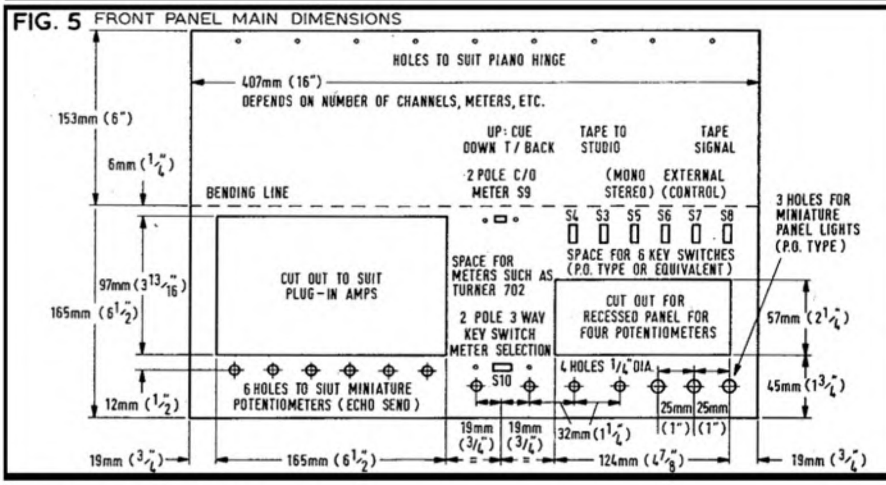
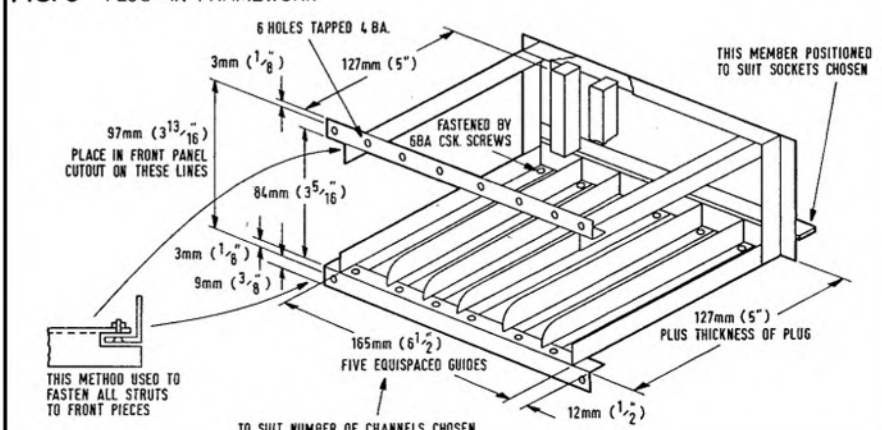


FIG. 6 PLUG-IN FRAMEWORK



practice it has been found very easy to control stereo microphones on adjacent channels, and this is preferable to ganged channels.

In 1963 the only commercially available quadrant or linear faders were prohibitively expensive, so a home-made solution was devised using gear wheels and normal carbon potentiometers. Now there are several inexpensive linear faders which can be used in preference to the home-made variety but, for those who like to construct as much as possible themselves, details of the original design are included.

The signal passes to the mixer amplifiers as previously mentioned and then returns to the fader bank and the main gain controls, grouped together at one end of the bank under the right hand. At this point the monitoring takes place, and is designed so that any change to the monitor controls will have no effect on the output signal which is being recorded ; these controls are recessed into a small panel to prevent accidental movement. An amplifier identical to the output amplifier is used to amplify the signal so that any (external) power amplifier can be used for monitoring. The output from this power amplifier is returned to the mixer so that it may normally be switched to the monitor speaker. For talkback, and playback to the studio, a loudspeaker in the studio is connected to the amplifier output.

The last feature of the monitoring is the metering equipment. In the prototype this was based on a single PPM switched to read either the left channel, or right channel, or the greater of the two. While this method has proved very satisfactory, many people prefer separate meters for each channel and they should consider this when planning their metalwork. Therefore S10 may or may not be required. Since the PPM is expensive, a circuit for VU meters will also be given.

There are two further circuits in the mixer which should be briefly mentioned at this stage. The first is a simple oscillator which can be switched to four spot frequencies of 1, 10, 12 and 15 kHz. These are very useful for lining up a recorder and for setting the outputs to identical levels for stereo. It is a good idea to record 30 seconds or so of peak level (32 mM/mm for PPMs and 18.5 mM/mm for VU's) of 1 kHz at the start of any important tape which might later be copied or used as the master tape in the cutting of a record. The second circuit is a flasher system to operate a red panel light in the event of a fault or danger condition ; in this case it is operated by either the oscillator, the talkback switch, or the tape-to-studio switch, since in all these cases a false signal would be recorded. The flashing light attracts attention more effectively than a steady one.

The first step in the construction is the metalwork for the case to contain the electronics. 16 SWG aluminium is suitable. With low impedance transistor circuitry, there is no need to have extensive screening in the mixer, and the wiring can be bell flex, which simplifies the work. As with any high gain system, there is always the danger that the output signal may be coupled into the input, and this can cause oscillations. The gain from the 600 ohm input to the output is normally set at about 80 dB, or 10 000 times. Thus if only 1/1 000 of the output signal is fed back in phase to the input, the circuits will oscillate. If all the wiring is held together in one cableform, there will be some capacity between these wires. With modern transistors, the circuits can still be flat to 1 dB at 500 kHz, and with an 80 dB gain, there has to be less than 0.5 pF capacity between input and output to cause oscillation.

The solution is obvious. Firstly, the circuit response is made to fall in a controlled manner from 25 kHz. Secondly, the wiring harness should be designed to keep all input wires together at one side of the mixer (the left hand side in the prototype), the mixer wires in the centre, and output wires on the other side. The front panel is hinged at the back so that the inside is readily accessible for maintenance and wiring. **Fig. 4** shows the back of the mixer. It was decided at an early stage to use jack plugs throughout as standard, with the three-contact version for stereo (two signal and earth). Since many people use two separate leads, provision was also made for this. In addition, each recording amplifier has five independent outputs. Thus more than one machine can be safely driven, since shorting any output makes no difference to the other outputs. A version using a balanced transformer output will also be described.

Fig. 5 shows the main dimensions of the front panel. The meter cut-out is for the Turner *Type 702*, and these dimensions must be modified to suit any other meter. The large cut-out on the left-hand side is for the plug-in units, and here again can be modified to suit other requirements. The author's units are 28 x 95 mm, and use Painton miniature ten-way plugs, with the sockets in the framework. These have proved entirely reliable and can be recommended. There are other types, such as the McMurdo red range, on the general market at about half the cost, but the dimensions may have to be modified. Any connector with six or more ways is suitable.

The runners for the plug-in units were made from pieces of aluminium angle, shaped to suit, and the sockets are mounted on a framework made from the same material. The framework is held together with 6 BA nuts and bolts although, if a material other than aluminium is used, the holes could be tapped to save using nuts (**fig. 4**).

The recessed panel is made as a box in aluminium sheet, and is bolted to the rear of the cut-out. To finish this off, fibreglass is pasted over the join and the whole smoothed off with a file when dry.

The prototype used surplus telephone key switches for the bank on the top right of the unit. Nowadays you can buy modern key switches from ex-government stores, but even their new price is reasonable. The modern ones are preferable both for reliability and appearance.

This finishes the construction of the framework. Next month's article will give full details of the fader bank, and start the electronics with the design and building of the mains power pack so that the testing of the other units can be carried out easily.

Front panel switches

Ref.	Function	Type	TMC type	Keyswitch type	Knob
S3	Type to studio	Key, 60 c/o	S.590656	MLK 26	Yellow/black insert
S4	up: cue studio down: talkback	Key, 2p make 4p c/o	S.527617	MLK 09	Red/white
S5	Mono/stereo	Key, 4p c/o	S.527658	MLK 04	White/black
S6, 7	Control keys (for ext. equip.)	Key, 2p c/o	S.527656	MLK 01	Black/white
S8	Tape/signal monitor	Key, 4p c/o	S.527658	MLK 04	Green/white
S9	Tape/signal meter	slider, ¼p c/o			
S10 (see text)	Meter select	Key, 2p 3-way	S.527648	MLK 14	Black/white

Rear panel switches

S1	Phase reversal	2p c/o
S2	Mike select 30/600	1p c/o
S12	oscillator on/off	2p c/o
S13	frequency select	3p 4w

Fader bank switches

S11	Echo assign	1p c/o

Component suppliers

Keyswitch controls from: Edmundsons Electric Ltd., 2 Tulse Hill, London S.W.2.
TMC controls from: TMC Ltd., Roper Road, Canterbury, Kent (minimum order £5)
(average price of each key about 10s.)

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*Ferrograph 713
*Ferrograph 702/4
*Ferrograph 722/4
Fidelity Studio 4 Tr. Mono
Fidelity 'Braemar' 2 or 4 Tr. Mono
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Grundig 144 4 Tr. Mono
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*Philips 4408 Prof. 3 sp. 3 Tr. Stereo
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*Revox 77 Stereo Transistor
Sanyo 929 4 Tr. 2 sp. Stereo
Sanyo 939 4 Tr. 2 sp. Stereo
*Sanyo 990 3 sp. 4 Tr. Stereo
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*Tandberg Series 12X 3 sp. 2/4 Tr. St.
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*Telefunken M207 2 sp. 4 Tr. Stereo
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Neve in focus

Fig. 1 The Factory

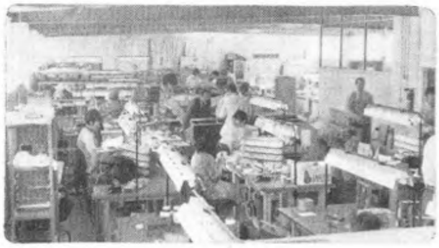


Fig. 2 The Wiring Shop

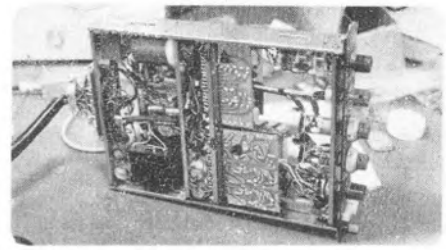


Fig. 6 Module under Test

A pictorial visit to Neve Electronic Laboratories

by
KEITH WICKS

Fig. 3 Design stage

Fig. 7 Assembled Desk (less side panel)

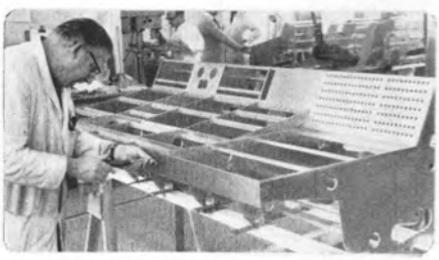


Fig. 4 Console Framework

Fig. 5 Testing Printed Circuits

Fig. 8 Dismantling desk prior to dispatch

RUPERT Neve and Company Limited have supplied sound control consoles, audio switching matrices, and associated distribution equipment to major companies in the television, music recording, radio and film industries throughout the world. Their market includes Canada, the USA, Singapore, Turkey, Yugoslavia, Spain, France, Norway, Southern Ireland and of course the UK.

The organisation was formed in 1961 and in 1964 the Neves moved to an attractive former rectory at Little Shelford, near Cambridge. A marketing and sales organisation was set up during 1967 and the 1968 sales were such that, in the autumn of that year, work started on a new factory at Melbourn (near Royston, Hertfordshire) and was in operation by February 1969. The offices were completed shortly afterwards. A second factory, located at Kelso, will be responsible for producing many of the modules, and repetitive work from the autumn of 1970.

The Neve organisation consists of two companies, Rupert Neve & Co. Ltd. being the marketing and sales company for equipment, which is designed and developed by Neve Electronic Laboratories Ltd. The directors of both companies are committed Christians and the importance of this was indicated by Mr Neve when he said 'Our businesses are dedicated to serving the Lord and we seek his guidance in all things'.

When I talked to Mrs Neve about this, the absolute sincerity of the firm's policy became apparent. She explained that not all employees of the companies are committed Christians, although it is preferred that they be sympathetic towards the companies' declared policy of the support of Christian outreach by means of radio.

Continuous close liaison is maintained between the sales division and their customers during all stages of a contract. On planning a console the first step is to establish exactly what the client requires and to decide the best way of providing these facilities in the space available. Neve have supplied a wide range of desks with inputs varying from six to 48 and output groups varying from two to 24.

Some designs allow the main console to be augmented by auxiliary units to ensure maximum flexibility of the system. Examples of these are Thames Television where a 24-channel console can take either one or two additional 12-channel auxiliaries. At Pye Records, a similar philosophy has been applied to an 8-output music recording desk.

Neve claim they can design consoles to suit virtually any requirements and deliver such equipment within six to nine months, depending on its complexity.

With the exception of the modules to be manufactured at the new factory at Kelso, all stages of production will continue to be carried out at the factory at Melbourn.

When the console is complete and in full working order the test results are shown to the customer for his approval and, if he should so wish, random tests may be carried out in his presence at this time. The console is then dismantled, packed and despatched. Wherever the equipment goes, an engineer is sent to supervise the installation and to check out the equipment with the customer to ensure that it meets the specifications previously accepted by the client.

SOUND balancing

PART THREE
MULTITRACK
APPLICATIONS
BY BOB AUGER

THE next item in the mixing chain is usually the echo return gain which is the point at which the signal from the various artificial reverberation devices arrives back at the desk. Some desks have an equaliser module in the circuit allowing tone correction in the echo circuit, which can be very useful in emphasising the low frequency 'boom' of a large cathedral. The combined signal from the selected channel and echo return circuits are then fed to a group fader which is usually the last control before the signal leaves the desk for the tape recorder. Important features to remember are the auxiliary equipment insertion points which are always to be found in the channels themselves and in the group outputs. These points allow the insertion of such devices as limiters, compressors, artificial echo tape-loops, etc. I have yet to see even the most up-to-date mixing desk in a pop studio that did not have extra equipment jacked into it within a few hours of its being commissioned. One can usually tell how long the desk has been installed by the number of 'blisters' it has attached to it. No matter how comprehensively the desk has been planned, one can always find something that needs to be added by the time the unit has been delivered and installed (even if only a coffee percolator!) It is quite usual to find desks accommodating 36 channels feeding into sixteen groups, which means that the operator is faced with a bank of 52 faders for a start. Such a desk would also probably accommodate six to eight artificial echo devices and up to four foldback lines. As if this formidable combination were not enough, we have yet to consider the loudspeaker monitoring network, when one takes into account a 16-track recording, means that we now also have to provide a complete sixteen-input two- or four-output mixer for monitoring purposes. Since the advent of the 16-track recording, most engineers record all the tracks 'dry' (i.e. without artificial echo) and it is usual to provide echo facilities on the monitor mixer, giving the producer and artist some idea how the finished product will sound. In fact it can today take as long to set up the monitoring side of the business at the commencement of the session as it did ten years ago to set up the whole desk for a stereo session. Where the engineer has to keep a very clear head is, of course, in knowing which effect he is creating on the monitoring side as distinct from the permanent recording he is making. Only too often one hears tales where producers have been told by the engineer at a recording session, 'Don't worry about that problem, it's only the monitors. It'll be easy to put right in the dubbing session', only to find out later that this was not true and that the missed cue was in fact missed on the mixer and not at the monitoring stage. The tendency now of course, is to just store information at the recording session itself, sorting it out into a presentable product when the musicians have gone home. The engineer and producer can rebalance in the comparative peace of the dubbing room.

First commercial use

At this point I would like to look back for a moment over the history of multitrack recording during the last decade or so. Probably the first commercial use of the technique was made by the American guitarist Les Paul and his wife Mary Ford when they produced a series of mono recordings for Capitol records. During the late 1950's this remarkable duo produced records featuring multitrack guitar and solo voice using a specially constructed mono tape recorder built by Ampex. The tape heads of this machine were arranged in the unusual format of play, erase, and record. The basic guitar track was recorded and subsequently played back on headphones. The original track and the second guitar part were re-recorded together as the tape passed the record head a split-second later, the first recording being erased as it passed the erase head (**fig. 1**). The technique was repeated many times, adding many guitar and vocal parts, until the finished result was achieved.

There were two major problems with this method of operation:

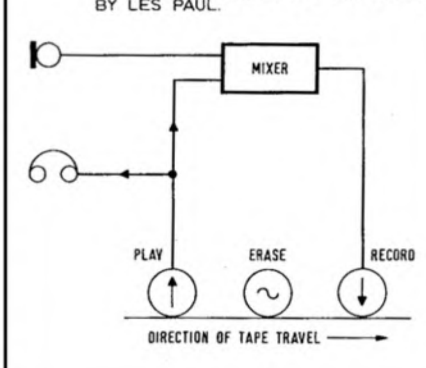
(a) As the original recording was being transferred time and time again, a steady deterioration in quality and a considerable accumulation of tape hiss occurred with each succeeding recording.

(b) A mistake by either of the artists anywhere along the line negated the whole recording and the project had to be recommenced right from the basic track. A simple way to overcome this problem would have been to have had two mono recorders transferring the sound from one recorder to the other as each part of the music was added, but it would appear that the artists' funds did not run to this!

A major step forward was achieved when Ampex produced an 8-track recorder using 25.4 mm tape in the early 1960's and Les Paul was one of the first artists to use this innovation. The machine was a logical development from the standard Ampex 2-track which had been developed in the 50's and was based on the normal *300* tape transport. The bank of eight valve amplifiers was so bulky that they had to stand apart from the tape console in a separate rack. When constructing this machine, Ampex also incorporated their selsync system which had been developed from earlier 3-track equipment. This system allowed the record head of any particular track to be switched to the input of the playback amplifier, permitting music previously recorded on the first track to be monitored by the artist as he or she recorded the next section of the music on Track 2. Obviously as Track 1 playback and track two record were in the same vertical plain, the final recording was perfectly synchronised vertically across the tape. When all eight tracks were completed, the finished tape was reduced down to mono or stereo. This method of operation was a vast improvement over the earlier single track method in that, should the artist make a mistake, only the current part of the work had to be recommenced.

Very soon after 8-track equipment was provided for Les Paul, Atlantic Studios in New York re-equipped with similar facilities. I was fortunate enough to visit Atlantic in 1960 and was amazed to find their technical thinking was far ahead of its time. The mixing console was arranged in two sections: the first had microphone stages in the usual manner, but was only for controlling the microphone levels into the tape recorder. It was apparently rare to use more than eight microphones and any multitrack recording consisted of simply one microphone feed per track, each track being controlled by the engineer to 100% modulation. The outputs from the tape recorder were fed back to Section 2 on the mixing desk where all the normal equalisation, echo and stereo panning facilities were to be found. The eight tracks

FIG. 1 NO-MISTAKES-PLEASE MULTITRACK TECHNIQUE EMPLOYED BY LES PAUL.



were mixed to two outputs and these were monitored in stereo. In other words, the tape recorder was being used simply to store elementary information from the studio, the engineer at the session experimenting with various means of equalisation etc., at the record producer's request, without interfering with the basic recording. When I add that, besides this revolutionary thinking, both the studio and the control room had multi-coloured lighting [*Keith Wicks please note !*—**Ed**], each with its own light mixing panel, and that a second control room specially for reduction had been constructed with identical dimensions to the studio control room (in order that the acoustics of the two rooms should be as similar as possible), we may appreciate what a competent engineer Tom Dowd is. No wonder the Atlantic label has had such tremendous commercial success in the pop field during the last few years. During the latter part of the 1960's, the majority of American and English pop recording studios advanced to 8-track recording. During the last three or four years, transistors have finally exploded into the professional audio scene to the extent that it is quite unusual today to come across multitrack recorders with valve amplifiers. Also, of course, the pop industry seems to have decided that eight tracks are no longer enough and the move is towards 16, 24 and even 32-track equipment. The interesting thing about this development is, however, that the original Atlantic philosophy has now become almost universal: as far as possible engineers are trying to maintain the principal of one microphone per track. It is remarkable though that, when the average engineer is faced with recording 'straight-stereo' (i.e. attaining the finished sound at the time of recording or compromising with three or four tracks, he is quite happy to mix many microphones together and commit himself at the session. Once the move is made to 8-track recording, which often calls for only two microphones to be mixed together on a given track, this gives the engineer the willies since he wishes that this kind of commitment did not have to be made at the time. Consequently, once the first step is made on the uphill (or downhill?) path to multitrack recording, the logical end seems to be to keep the master tape as a simple means of storing information directly from the microphones. Since monitoring all these tracks is a formidable problem in itself, requiring a comprehensive mixer to get all the tracks down to two (or at most four) loudspeakers, possible future studio installations will consist simply of variable gain microphone amplifiers feeding directly into the recorder, the whole of the studio mixing desk being used for the dual purposes of providing the monitor feed during the sessions, and reduction dubbing purposes thereafter.

Severe competition

Naturally, although Ampex were the pioneers in this field, they nowadays face severe competition (see last month's survey). Ampex, 3M and Scully made, I suppose, some 90% of multitrack equipment found in studios today. Studer are just about to bring an 8-track machine on the market with a 16-track recorder to follow very shortly. Also, the English Unitrack machine has now just become available in 16-track form, but is not yet in general use. Perhaps this is the machine we have been waiting for and at last the great British Tape Recorder has arrived on the international scene. It's been a long time since the *BTR 2*! All these machines have the same switching facility as the selsync system but we have to be careful of our use of words since Ampex very sensibly made *Sel-Sync* their trademark many years ago, as in fact they did also with the word *Videotape*.

Artificial Reverberation

Many years ago, a friend of mine was making a pop record which featured the dulcet tones of one of our present day disc jockeys (it was such a long time ago I daren't mention his name, but I believe the song was called *Too Young*!) in mono of course. The engineer suggested that a small amount of artificial echo added to the voice was now the done thing. This he explained would help to round out the sound. Also, if done carefully, it would help make the voice stand out from the orchestral accompaniment. As the recording was being carried out in a comparatively small studio and the string section sounded a little cramped and dry, the producer suggested that perhaps some of this artificial echo stuff might be nice on the violins. Whereupon the engineer threw up his hands in horror. 'You can't do that—it will make them all swimmy', so the echo was never added to the strings. A year later the Mantovani sound was born. Ah well, it is sometimes difficult to be a pioneer.

The first time I was aware of artificial reverberation on a pop recording was in the late 1940's when some really excellent engineering was apparent on some of Stan Kenton's 78 RPM discs. I presumed at that time that the recordings were made in a large reverberant dance hall or somewhere similar, but was amazed at the clarity and presence of the rhythm section. Now, of course, we know how it was done.

Echo Chamber. This is probably the oldest system and consists of a small tiled or plaster-surfaced room housing a loudspeaker fed through the echo send panel on the studio mixer plus an omni-directional microphone which is connected to the echo return circuit. The main problems with this system are that the quality of the sound from the chamber is largely a matter of luck depending upon the inherent 'colour' in the chamber, also the decay period of the echo is hard to plan in advance. A modern studio complex may require something between eight and twelve echo devices and this means possibly the whole basement of a building would have to be set aside for the purpose, which is not very practical.

Tape Delays. This system has the merit of using a very small amount of space and in fact being portable. The system is well known and consists of using the record and play heads of a tape recorder simultaneously, the delay time being decided by the gap between the two heads and tape speed used. Simple repeat was common effect on the early rock-and-roll records and is quite useful for producing a very gimmicky voice from time to time, particularly if one is trying to simulate the effect of a public address system. If the signal from the replay head is mixed back to the record head, the resultant blurred effect is much more pleasing. Most studios refer to this as the 'tape-spin' ffeect. Although it is customary to use a standard tape recorder for this purpose, a much better system is to construct a recorder using a loop of tape with three or four replay heads plus some form of simple mixing in order to achieve the right amount of signal from each replay head.

Metal Plates and Springs. Without doubt, the most widely used artificial echo system to be found in professional circles today is the EMT, a 2.7 x 1.2 m steel plate 12 mm thick, which is suspended by springs from a metal frame. This system is without doubt the most flexible available in that the plate may be tuned when it is installed, by adjusting the suspension to produce a smooth peak-free response giving little coloration. A mechanical damping system allows reverberation to be varied from zero to six seconds. The system works by the echo send signal being amplified and exciting a drive unit at one end of the plate, the sound vibration passing along the plate and being monitored at the other end by a ceramic pickup which is fed back to the console. The reverberation period is adjusted by means of a special damping system which is mechanically operated but which can be driven by remote control from a mixing desk. It may, of course, be hundreds of metres away on a different floor of the building. A stereo version of the plate is available which, although retaining a single drive unit, has two pickups fitted, the shifting phase between the two pickups giving a life-like spatial effect. One of the greatest advantages of the unit, apart from its flexibility, is that the complete system in its mounting case is little over 30 cm thick, thereby allowing several plates to be stored adjacent to each other in the space that would be required for a single echo chamber. It can hardly be considered portable though. A simplified version of this principle is the reverberation spring. This was pioneered by the Hammond Organ Company as a means of providing a reasonable reverb facility on their electronic organs. The short length of spring is suspended between two posts, one post housing the exciter unit and the other housing the pickup. The problem with this system is that the majority of springs have a lot of coloration and that no suitable damping system has been produced commercially for adjusting the decay period. However, the Grampian system, which is available at moderate cost and is quite portable, is a very useful system to have on hand and if used carefully can be considered for professional work. A large number of these units are used today even in the most hallowed walls. Many a first class recording and (dare I say?) broadcast has gone out enhanced by the Grampian unit.

One of the problems in using any artificial reverberation system is that, as the amount of echo is increased, the performer appears to recede into the reverberation. Since it is customary in pop recording to retain 'presence' at all costs, some form of compromise has to be achieved. The most common method of combating this problem is to delay the echo send line via a tape recorder operating at 76 or 38 cm/s. Even at the higher speed, the time taken for the tape to pass from the record to the play head and thence to the echo device is sufficient to register the initial impact of the sound upon the ear before the echo becomes apparent.

To be continued

commercial BREAK

adrian hope visits sounds aquarian

AT first sight most television commercials seem an insult to the intelligence of anyone who is trapped into watching them. But it would be naive to think that what they are is anything other than by intention. In fact, when looked at a little more closely, most commercials show themselves to be skilfully made mini-films directed towards a clearly defined audience. To bear this in mind can make at least some of them almost endurable for anyone interested in music, recording or film techniques.

More often than not TV commercials rely heavily on their sound as well as their visual content and the two are often welded together with a fair degree of skill.

Over the past year or so, more and more commercials have been using curious sounds in a musical way to interesting effect and I eventually became curious about how some of the sounds were made. I thus set about finding out who was responsible for the sound on some of these commercials and, at the end of a fairly tortuous trail, found two men and a small recording studio in Soho. The recording studio is Sounds Aquarian and the two men are John Hodge and Mick Berg. The studio is small, has 4-track facilities, and is cheap to hire by today's standards.

Hodge and Berg are employed by a company to run the studio but, by renting the studio themselves, they together produce a lot of curious-sounding commercials. Virtually all the commercials are multitracked by what might best be called the 'Hodge/Berg Duo' and, although it is hard to say exactly who does what, a fair summing up would be that Hodge functions as musician, composer and ideas man, with Berg as musician and electronics wizard.

They explained to me one afternoon recently that the whole thing started as a deliberate attack by them on the fairly tight closed-shop commercial market. It is no secret that a lot of money can be made in this market and commissions are not easily come by. Sensibly they realised that their best chance of entry was by providing something different and this they have done for an increasing number of television commercials over the past year or so including ones for *Crunchie* (chocolate), *Sure* (deodorant), *Mazola* (cooking oil), *Kraft* (cheese) and more recently *Dulux* (paint). At the time of writing they are working on one for Nestle's *Ideal Milk*.

They played me several of these past sound productions over studio monitors and this served to make me even more curious about how various things were done. Each production was wholly different but all had in common the same scheme of things. Namely an interesting but fairly simple melody buried amongst, or created by, a large number of different and largely unidentifiable sounds. I anticipated that they would be cagey about revealing how a lot of it was done but fortunately their view is that ideas are always changing and they seldom bother to use the same sound source idea twice. Thus if anyone wants to copy what they have done to date, they are more likely to be flattered than offended. Only once or twice did they smile inscrutably at a question and change the subject.

One of the first sounds that intrigued me was a curious percussion effect that sounded far too high pitched to be an ordinary drum—it turned out to be a closely miked metronome with added artificial echo. Another was obviously a guitar but one which somehow managed to sound like the Glenn Miller brass section—with the musicians moving their hands over and away from the instrument bells to produce a wah-wah effect. Also a bit puzzling were the low notes obtained. A bass guitar used here, not for rhythm but as a solo instrument. Berg was playing and had plugged direct into the main recording console via a variable bandpass filter which Hodge fluctuated to get the wah-wah result. They use a similar filter on microphones to get the same type of effect on voice and wind instruments.

Producers of TV commercials generally create their visuals first and their sound next. The usual routine has been for them to show the script and a rough cut version of the final film to the man responsible for producing the soundtrack. He has then gone away and written the music for eventual recording at a short session, usually in a studio with projection facilities. This is necessary if there is to be any real synchronisation between the visuals and the soundtrack. Those that rely on larger orchestral sounds tend to be much less rigidly synchronised with the action than those which have been recorded by a smaller group.

It is here that the Sounds Aquarian technique scores. Over a period of days, they claim they can turn out a composite track with music, pointing and effects timed with a stopwatch down to something in the region of 100 mS.

They showed me a typical breakdown for an ad:

Time	*Visual*	*Musical mood and effects*	
0.0	CU Headlights	Effects	
1.2	CU Flashlights on		
2.4	MS Front-on E-type travelling		FAST CHASE TENSE
6.9	CU Eyes	*(1) effect	
8.1	MS Mercedes		
10.1	CU Lipstick		
13.4	MS E-type		
15.8	CU Mirror and eyes	*(2) effect	
17.00	LS E-type pulls up by boat		
19.3	MS Mercedes		
20.3	LS Girl into cabin		
21.7	MS Girl in cabin looking at watch	*(3) effect	
22.9	LS Mercedes arriving		
25.7	CU Girl applying cream in cabin		VERY TENSE
27.2	MS Merc. girl with compact		
31.4	CU Girl in cabin reaching for gun		
32.3	MS 2 Girls in door with gun	*(4) effect	
32.9	CU 2 Girls proffering products		RELAX
33.8	CU Hand with cream		
34.9	CU Hand with cream		
36.7	3 shot girls		
38.7	End pack shot		

When you bear in mind that the above calls for a fast-moving melody between the 2.4 and 22.9 second marks, a very tense theme between the 2.4 and 22.9 second marks, a very tense theme between the 22.9 and 32.3 second marks and a gentle feminine melody to associate with the (cosmetics) product for the last 6.4 seconds, you get some idea of the headaches involved in writing something musically acceptable for a 40 second advert. Then, add to that the difficulty of incorporating half a dozen spot-on pointing effects without the final result sounding absurdly disjointed, and you have something approaching the full problem.

All rules have an exception, of course, and sometimes they do use outside help. For instance, early in one commercial I heard what sounded like a French Horn and later in the commercial heard what seemed to be two rather different French Horns. The first 'Horn' turned out to be a guitar imitating a French

(continued on page 258)

FIFTEEN, THIRTY-EIGHT, OR A POULSEN?

DAVID KIRK LOOKS AT METRICATION AND ITS ALTERNATIVES

TWO thousand years ago, an average Roman walked 1 000 (double) paces and invented the mile. In the centuries that followed, Britain has blundered blindly through furlongs, rods, poles and perches, to our present yards, feet, inches and mils (mils ?). The Roman mile changed its length on the way, from 1 611 to 1 760 yards.

This evolution served its purpose (keeping the peasants on the farms) but, as life became faster and more complicated, the Imperial system of measurement grew increasingly absurd. Five different multiples are employed in distance definition alone:

1 mile =8 furlongs
1 furlong=10 chains
1 chain =4 rods
1 rod =1 pole (useful)
1 pole =1 perch (equally useful)
1 perch =5.5 yards
1 yard =3 feet
1 foot =12 inches

During the 19th Century, a Convention was held in Paris to select a more convenient set of standards. This was to be based on 10 (binary types please read 'ten') and would encompass distance, area, weight, power, capacity and, for France at least, currency. A basic unit, the metre, was determined by dividing the earth's quadrant through Paris by ten million. The measurement of the quadrant was later found to be incorrect but this had no practical bearing on the chosen unit.

Metric values have since been adopted by many European countries, though for many decades no British government dared inconvenience living voters for the benefit of future generations. The change to metric in an otherwise well-developed country is obviously not worth the bother and cost in the short term, though in the long term it offers the advantages of simplicity and international standardisation. The fact that a future Paris Convention may opt for a duodecimal or binary-based system is neither here nor there:

1 kilometre =1 000 metres
1 metre =1 000 millimetres
1 millimetre=1 000 micrometres

TABLE 1
Playing Time (Hours and Minutes)

Feet	15 i/s	7½ i/s
900	12min	24min
1 200	16min	32min
1 800	24min	48min
2 400	32min	1h 4min
3 600	48min	1h36 min
4 800	1h 4min	2h 8min

TABLE 2
Nominal Tape Lengths (Feet)

Spool Diameter	SP	LP	DP
5¾ inch	900	1 200	1 800
7 inch	1 200	1 800	2 400
8¼ inch	1 800	2 400	3 600
10½ inch	2 400	3 600	4 800

TABLE 3
Playing Time (Hours and Minutes)

Metres	38 cm/s	19 cm/s
270	12min	24min
360	16min	32min
550	24min	48min
730	32min	1h 4min
1 090	48min	1h36min
1 460	1h 4min	2h 8min

TABLE 4
Nominal Tape Lengths (Metres)

Spool Diameter	SP	LP	DP
15 cm	270	360	550
18 cm	360	550	730
21 cm	550	730	1 090
27 cm	730	1 090	1 460

Not the best of arithmeticians, I needed four seconds to realise the number of millimetres in a kilometre (1 000 x 1 000). Determining the number of inches in a mile took

(continued overleaf)

FIFTEEN, THIRTY-EIGHT, OR A POULSEN? CONTINUED

30 seconds (12 x 3 x 1760). What happened to the centimetre? It drowned on the way across the Channel, in the same boat as the decimetre, decametre and hectametre. Officially, Britain is adopting the SI (System International) version of metrication, based on multipliers of 1 000 and ignoring anything in between. But which official? The British Standards Institute, in BS 1568/Pt 1/1970, deigns to use centimetres when expressing tape speeds.

Sound recordists everywhere will be thrilled to know that:

$\frac{12f}{60s} =$ tape playing time in minutes when f is the tape length in feet and s the speed in inches per second.

Working in metric, this becomes:

$\frac{100\ m}{60\ c}$ where m is tape length in metres and c the speed in centimetres per second.

My mind does not work this way when determining the playing time of any particular reel. It is easier to remember that a 10½ inch NAB normally holds 2 400 feet of Standard Play and runs for just over 30 minutes (32 minutes if machine and mains are right), at 15 i/s. The other common beast, a 7 inch cine reel of 1 200 ft Standard Play, runs for just over 15 minutes. If most of your recording is on 3 inch reels at 1⅞ i/s, then you simply remember a different set of figures.

In the changeover to metric, which worries me as much as it does Messrs Alcock, Arbib and Bastin (see recent letters), I suggest we simply remember that a 27 cm NAB holds 730 metres of tape and runs for 30 minutes at 38 cm/s. The cine reel becomes 18 cm in diameter and holds 365 metres. If you like mnemonics (can never remember that word), 365 is the number of days in a year. The alternative is to try and ignore metric altogether, in the way the printing industry may attempt. Some time ago, I sent a photo off for blockmaking, specifying a width of 50 mm. It came back more than 8 inches wide: the mm had been interpreted as 'ems'. A standard printer's em equals one sixth of an inch. We could do the same in tape recording: 15 i/s equals one Poulsen, 7½ i/s equals one Kelly (or Half-Poulsen), 3¾ i/s equals a Bastin, and so on. Why stop there? Shall we make 0.25 inch into a Spratt? Half-inch tape then becomes Duo-Spratt, the multitrack boys using Quadra-Spratt and Octo-Spratt. Try converting into Arbibs—the 45° cutting distance across 1 Spratt. Or simply stick to feet and inches. But for how long, when the companies manufacturing tape recorders will be forced to deal with raw-material supplies produced in rounded-off metric dimensions? In five or ten years the industry would (deservedly) be a laughing-stock.

If you are worried about minor discrepancies in tape length occurring during the move from Imperial to metric measurement, you needn't be. Much of the tape used in the country is produced on the Continent (typically BASF and Agfa) under metric conditions. In any case, all tape lengths are nominal and a reel that is going to run a minute short of another at 38 or 19 cm/s will have visibly less windings. If you want to be on the safe side, there's always Double Play.

Tables 1 and **2** show playing times at 15 and 7½ i/s of the common tape lengths. The two are obviously complementary. **Tables 3** and **4** are the metric equivalents. The fact that BS 1568/Pt 1/70 specifies 30.1, 10.05 and 4.76 cm/s (I had to look them up) is immaterial since the error incurred in rounding off to 38 and 19 cm/s is less than 0.3%, which is within their recommended tolerance of ± 0.5% for professional equipment. If anyone faces problems it is the tape recorder manufacturers, who have only each other to blame for dropping the early AEG 1 metre/s.

TABLE 5

Tape thickness

	SP	LP	DP	TP	QP
Thou' (inches)	2.2	1.37	1.1	0.73	0.55
Micrometres	55	36.7	27.5	18.35	13.75

Width
0.246 ±0.002 inches = 6.25 ±0.05 mm

COMMERCIAL BREAK CONTINUED

Horn and the two French Horns proved to be conventional horns played by two session men brought in for an hour. It will be interesting to see whether this attention to detail comes through when the final product is heard over tinny television-set loudspeakers.

In their Dulux commercial track, there were several fascinating bursts of sound or 'sound flares' which I would defy anyone to tie down. A good guess would be an electronically doctored harpsichord bursting into a massive chord. The sound source was eventually revealed as a curious instrument called an organ guitar. This appeared a few years ago but even now is not very widely used. It looks very similar to an ordinary electric guitar but is actually quite different. The body houses the electronics of a small organ but, instead of a keyboard, the guitar strings serve to make and break various electrical contacts according to which fret they are pressed against. Even then nothing happens until a plectrum, which serves as an electric contact, is touched on a string. This way, in theory at least, a guitarist can sustain a note or chord ad infinitum. Even with organ guitars at about £300 each, Berg and Hodge would not reveal exactly how they coax their own particular brand of sound flare. 'It took us about three days of experiment' Hodge mused.

While on the subject of curious musical instruments, I was shown a Heath Robinson modification to an ordinary guitar which basically involved winding piano wire round the guitar strings to provide a vaguely 'Indian' guitar effect comparable to an old jangle piano. Doubtless most people will be familiar with this piano effect and know that it can be produced by putting drawing pins in the piano hammers. Aquarian do the same kind of thing with brass pins and felt pads.

When demonstrating their jangle guitar, Berg explained that usually if the sound track called for a large resonant chord he would retune his guitar so that he could play the necessary chord with all strings open. In this way far greater volume, resonance and sustaining power can be obtained.

By about this time I had started to realise why Hodge and Berg are not too concerned about preserving trade secrets. For a start, nobody can ever tie in every sound with every explanation and, as often as not, the effect obtained can never be accurately repeated, even by themselves.

Particularly interesting was some work they were doing on the various ways of obtaining emphasised crescendos. Ironically, having gone through a long series of experiments with tapes of piano chords played backwards, they eventually settled on a conventional organ chord played forwards. They are quite happy to discard ideas that on paper looked good but in practice just didn't sound right. For instance, after a long time spent recording the sounds of bubbles blown through water with a straw, reproducing them at faster and slower speeds to alter their pitch and rhythm, they discarded the idea and Berg used an adapted Hammond organ to do the job.

Probably their most dramatic effect is the 'misuse' of reverberation springs. For a long time guitar amplifiers and organs have incorporated artificial reverberation gadgets which take the form of coiled metal springs, like loose electric fire elements, strung between a pair of electro-mechanical transducers—loudspeakers without cones. Into one transducer an audio signal is fed and the resultant vibrations are fed down the spring to be picked up by the other transducer and thereby converted back into an audio signal. The result of passing the signal down the springs is the well known 'reverb' effect and what many pop musicians have found by accident is that if you kick, drop or fall over an amplifier equipped with such a gadget you can instantly produce a sound described as a cross between a mammoth explosion and a house falling down. There is nothing new in all this but, through studio monitors, a scientifically kicked reverberation unit is quite a sound.

As explained earlier it has until now been usual practice to create visuals first and sound next. For what may probably be the first time ever, Hodge and Berg are now being commissioned to create 30 or 45 second tracks from which it is hoped an idea for a film will eventually emerge. It will be interesting to see what the movie men make of a concerto for signal generator, jangle piano, watch and clock ticking, sitar, flute and Chinese bell recorded with tape echo.

balanced and unbalanced lines

An investigation into their effectiveness in microphone signal transmission
by Anthony Eden

THIS article sets out to explain the difference between balanced and unbalanced lines, as well as the advantages and disadvantages of using one type of line rather than another.

One method of conveying information between two points is to make use of a pair of wires. If the information being transmitted consists of alternating voltages such as audio signals, the pair of wires will have the following features:

(a) The wires will possess resistance.
(b) Both wires will set up electric and magnetic fields and will affect each other.
(c) Both wires will be affected by electric and magnetic fields from external sources.

Let us examine the effects of each feature on the information being transmitted along the wires:

(a) All conductors possess resistance, one factor limiting the distance over which signals can be conveyed. If the source of this information is of low output level, the resistance of the conductors must be very low to avoid significant reduction in level. This presents little practical difficulty since it is unusual to require more than 100 m of cable between a microphone and mixer or recorder. The loop resistance of such a length is very low, typically about 1.5 ohms, and the signal loss is negligible.

(b) When an alternating signal passes along a conductor, two separate fields are set up—an electric field and a magnetic field. The electric field, set up as a result of a *voltage* difference between the two conductors, creates a *capacitance*. The magnitude of this capacitance depends upon the diameter of the conductors, the distance between them, and their length. If a microphone has an impedance Z_m (and we consider it to be a pure resistance, a reasonable approximation) and we connect it to a length of coaxial cable having a capacitance of C pF per metre, we can represent the signal arriving at the amplifier by means of **fig. 1.**

For the moment we will ignore the fact that the cable has other losses as well. Now it can easily be shown that the frequency response at V_a will be 3 dB down at f where

$$f = \frac{1}{2\pi C.R_m}$$

If we use a long length of cable, C will increase and hence reduce the frequency response of the generated signal (V_m). However, the most important variable is R_m, for if this is high and C is large, the HF response will be curtailed considerably. If R_m is in the order of 20 ohms to 1 K, the frequency response will normally be curtailed above the audio range. If R_m is high (say 50 K), then the value of C (and hence the length of the cable) must be kept low in order to maintain a good HF response. Let us take a practical example. If we have a microphone of 50 K impedance and a cable of 100 pF metre (a typical figure) and we do not wish the response to be more than 3 dB down at 10 kHz, then the maximum cable length (x) we can have, using our basic theory, is:

$$x = \frac{1}{6.28 \times 10\,000 \times 50\,000 \times 100 \times 10^{-12}} \text{ metres}$$
$$= 3.1 \text{ metres.}$$

It can be readily appreciated from the above why high-impedance microphones cannot be used with long connecting leads. If we had used a 200 ohm microphone in place of the 50 K model, the cable could have been extended to at least 100 m and would still meet our requirements. There are a number of cables available with considerably less than the quoted capacitance value (that quoted being common to television coaxial lead), particularly balanced screened cables.

FIG. I

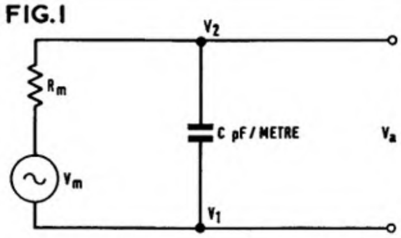
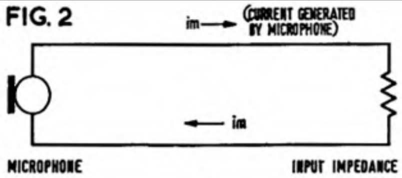


FIG. 2



Some readers may be conversant with the concept of 'the characteristic impedance of a cable' and the above simple calculation may seem to violate the rules for line theory. A typical cable may be quoted as having a characteristic impedance of 75 ohms and the attenuation loss, according to line theory, should be independent of frequency. Unfortunately line theory does not apply to short lengths of cable (less than about 5 km) and so the cable cannot be considered as having a characteristic impedance and its attenuation is controlled by the dominant characteristic which is the capacitance. In practice the frequency loss is not quite so severe as our simple theory would suggest since the capacitance of the line will also be shunted by the matching load.

(c) A magnetic field is created round a conductor when a current is passed through and gives the conductor *inductance*. This has little effect on the signal causing the field but influences neighbouring conductors. The effect due to this interference is *crosstalk*. Although electric fields create crosstalk problems in the HF and VHF bands, we shall concern ourselves here with the problems of magnetic field crosstalk which is the important problem at audio frequencies.

If we have two wires forming one circuit, one wire will produce a magnetic field due to the conducted current and the other wire will produce an equal and opposite field (since the same current must flow through both wires in opposite directions, **fig. 2**).

Now the object of the exercise is to make the field of one wire cancel that of the other, giving a zero external field and therefore causing no trouble to neighbouring circuits. The first requirement is to place the two wires close together for maximum interaction between the fields. This is improved considerably by twisting the wires round each other. This is the reason why heater wires in valve amplifiers are always twisted together, to eliminate the fields from the AC heater supply. The technique is used by the GPO to prevent one telephone circuit from interfering with another —which could be a major problem when there are four thousand circuits in one cable! It is usual to wrap an earthed outer braid round the conductors to prevent the magnetic fields spreading from the twisted leads.

The above considerations are important

(*continued on page 261*)

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where several circuits are to be run side by side, particularly if some of the circuits have low voltage sources such as microphones. We encounter much more serious problems, however, if large external fields are present. An example of this would be if a microphone cable ran alongside a current-carrying mains cable. The magnetic field set up by the mains cable can be very large and will induce a current into the low-voltage microphone cable unless special precautions are taken. If we use a twisted pair of wires for the microphone circuit, the magnetic field induced into the cable will usually be of the same magnitude and direction in each wire. There is no harm in this so long as two conditions are fulfilled:

(a) The effect of the induced currents is cancelled out at the far end.

(b) The current induced from the mains circuit is the same in each microphone wire.

When an external field induces a current into each wire of a cable, as far as microphone circuits are concerned, it is usually true to say that condition (b) is fulfilled, particularly if the wires are twisted together. Condition (a) is a more serious problem and accounts for many hum difficulties occurring during recording sessions. In **fig. 3** we see that the external field has introduced a current into each of the microphone wires. Using a simple tape recorder or mixer input stage, where the microphone is fed direct to the first-stage transistor at medium impedance, the current *i* will produce a corresponding voltage *V* in each wire. However, one wire is at earth potential and must be fixed at 0 V. Thus we have successfully introduced this external mains field into the input circuit of the recorder. Such a circuit is known as *unbalanced* as is the type of cable. In an attempt to counteract this state of affairs, we wrap the earthed wire round the inner conductor to reduce the possibility of a voltage being induced in the latter wire. For this reason, cheap TV feeder cable, where the braid is very loosely wound, should be avoided for microphone work.

A better method of overcoming induced hum effects is the use of a *balanced* cable. This incorporates two conductors to carry the microphone circuit, plus a separate earthed shield. Most of the induced hum is absorbed by the shield and anything penetrating is induced equally into the two microphone conductors. At the mixer or recorder input end, these two conductors are fed into a transformer which has an earthed centre tap. The induced field is fed away to earth without generating any resultant voltage in the secondary winding —see **fig. 4.**

This arrangement is obviously much more satisfactory when strong external fields are present. In practice it is impossible to make both transformer windings identical but a degree of match is obtainable for reducing hum fields to an adequate level. The normal method of balancing transformers is to wind, say, 200 turns, tap at this point, and then wind 200 more. At low frequencies the balance is quite adequate but at higher audio frequencies the mismatch can be quite severe because the capacitance of the windings differs in the two halves since the windings are normally made one on top of the other. It is for this reason that radio interference can be experienced in areas of high-strength radio transmission.

There are a number of alternative methods of winding transformer primaries to cancel out hum effects. One method does not use a centre tap at all but relies on the fact that the primary winding may be uniformly wound so the two induced voltages appearing in the primary winding will always exactly counteract each other anywhere in the winding and therefore generate no voltage in the secondary.

This principle has been adopted by the BBC in the design of a transistorised microphone amplifier (described by David Robinson in August 1964) and has been incorporated in the design of the recently introduced Unimixer *4S* microphone mixer. This type of input has the advantage that either balanced or unbalanced input connections can be accommodated without any modification. Transformer inputs have a few disadvantages in relation to unbalanced direct inputs and the pros and cons are listed in **Table 1.**

If you are using an unbalanced transformer it is normally quite easy to adapt it for use in the balanced condition. **Fig. 5** shows the normal unbalanced arrangement for a transformer such as the Grampian *G7/LH*. Lead 2 is disconnected and wired to the ring terminal of a three-pin jack socket. Lead 1 is connected to the tip. Across leads 1 and 2, a couple of 1 K 5% series resistors are connected with the centre point wired to the sleeve, **fig. 6.** This allows balanced or unbalanced 30 ohm microphones to be used as the effect of shorting out the lower 1 K resistor with an inbalanced tip-and-sleeve plug is very small. It is important that the two resistors closely match each other since this decides how effective the cancellation will be. Theoretically, and often in practice, a better degree of matching can be obtained with two well-matched resistors than by using a winding centre-tap. A centre-tapped transformer using two resistors is a better proposition if radio interference is experienced, since the balance is maintained over a much wider frequency range.

Finally, let us consider the occasions when it is necessary to use a balanced cable and when it is safe to work unbalanced.

From a cost point of view, unbalanced coaxial cable costs about two-thirds the price of screened balanced cable. Unbalanced input stages are also a cheaper proposition since there may be no need for an input transformer.

If short leads are to be used (up to 10 m), unbalanced leads will normally give no trouble at all. It is always advisable to earth the screen however, since this prevents stray voltages in the screen finding their way into the input stage.

If you use your equipment frequently for

(continued overleaf)

FIG.5 UNBALANCED TRANSFORMER

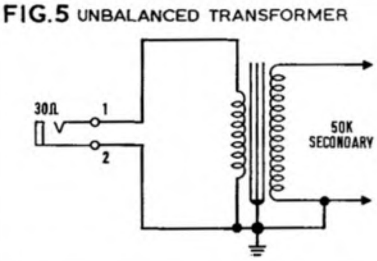


FIG.3

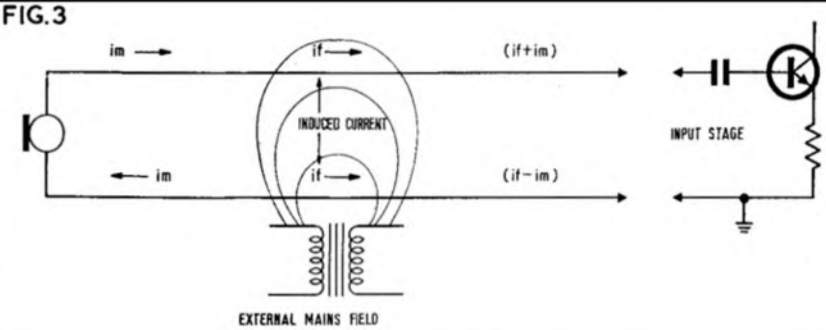


FIG. 4 BALANCED TRANSFORMER

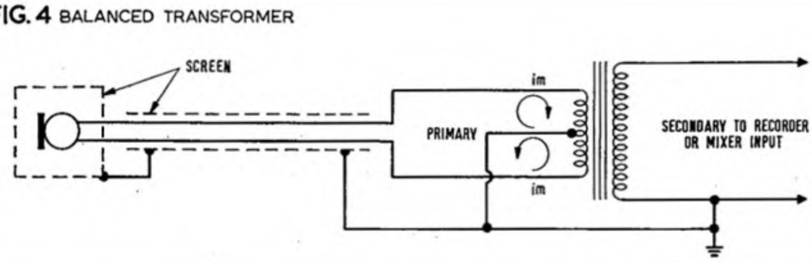
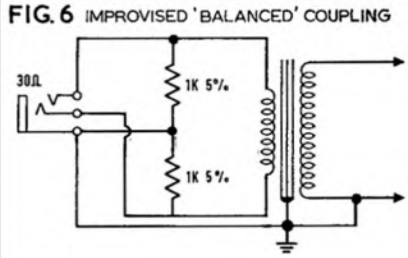


FIG. 6 IMPROVISED 'BALANCED' COUPLING



location recordings, it is a wise precaution to use balanced leads since it is all too easy to be caught unawares by stray magnetic fields. Avoid running microphone cables close to current-carrying mains lead, keep away from concert hall or theatre step-down transformers (usually out of the way in the basement) and avoid fluorescent lighting. The latter involves strong magnetic fields.

This concludes our investigation into microphone interference effects. The author is always willing to try and help readers with problems related to these matters.

Transformer Input

1 Can accommodate balanced lines.

2 Transformer requires good screening to eliminate hum.

3 Primary load must not exceed a certain value or frequency response will be impaired.

4 Easy way to achieve almost noiseless voltage gain.

5 Different microphone impedances can be accommodated by altering turns ratio at primary.

Direct Transistor Input

1 Accommodates only unbalanced lines.

2 Hum pick-up in transistors is much less.

3 Much wider tolerance to input stage impedance.

4 More noise introduced by using a transistor preamplifier.

5 Although possible to increase gain, in practice most designers only give one sensitivity setting for the input stage.

equipment reviews

TANDBERG 16X STEREO TAPE UNIT

MANUFACTURER'S SPECIFICATION (19 cm/s). Quarter-track domestic stereo tape unit. **Frequency response:** 40 Hz to 20 kHz ±2 dB. **Signal-to-noise ratio:** 60 dB (IEC weighted). **Distortion:** 5% maximum, at peak level. **Inputs:** 5 mV at 57 K or 100 mV at 1 M (auxiliary). **Outputs:** 0.9 V at 200 ohms. **Bias frequency:** 85.5 kHz. **Spool capacity:** 18 cm. **Tape speeds:** 19, 9.5 and 4.75 cm/s. **Heads:** Erase, record/playback and back-bias. **Wow and flutter:** 0.07% RMS. **Crosstalk:** 50 dB (stereo); 60 dB (mono). **Weight:** 8.7 kg. **Dimensions:** 390 x 300 x 170 mm. **Price:** £89 10s including tax. **Manufacturer:** Tandbergs Radio Fabrikk A-S, PO Box 9, Korsvoll, Oslo 8, Norway. **Distributor: Farnell-Tandberg Ltd., Hereford House, North Court, Vicar Lane, Leeds LS2 7NS.**

THE Series *16X* tape unit is designed to provide tape facilities in a domestic audio installation. The output level is fixed. A simple three-position playback switch provides stereo, or either right or left mono channels into both power amplifiers. Two central silver push buttons switch either or both channels to record. A coaxial pair of record gain controls allow mixing of any combination microphone or line inputs for mono operation, or independent control of each channel on stereo.

The record level meters are of the peak level type with a quick upswing and slow return which enables recording levels to be set with confidence and high accuracy.

Two DIN microphone sockets are provided on the deck plate, high and low level line input phone sockets being fitted at the rear of the unit. A standard DIN socket is also provided for alternative radio input and output connections.

The absolute tape speeds were measured with a lightweight strobe wheel and found to be within 2% limits at all speeds and at all parts of an 18 cm reel of LP tape. The same reel was wound or rewound in just over 2.5 minutes in either direction.

The four-digit tape position indicator clocked up exactly one digit per revolution of the RH take-up reel.

Short-term tape speed fluctuations were commendably low with cumulative record and play wobble not exceeding 0.04% RMS at 19 cm/s, 0.08% at 9.5 cm/s and 0.85% at the lowest speed of 4.75 cm/s.

Fig. 1 shows the high speed pen recordings at each speed. It will be seen that the traces are free of high frequency flutter at all speeds and this is almost certainly due to the lack of pressure pads on the heads. A single pad and plate is mounted to the left of the erase head to provide a constant tape tension across the heads.

The play-only responses of **fig. 2** show that the 19 cm/s equalisation is exactly to the CCIR 70 μS time constant and that the 9.5 cm/s equalisation is to the NAB 90+3180 μS characteristic.

System noise was very low at 66 dB below peak recording level when weighted to the IEC 'A' curve.

Reference level of 32 mM/mm was recorded on BASF *LGS35* low noise tape with the peak record level meter reading 0 dB. Distortion at this level was 1.3% and 3% distortion was recorded with the meter reading just above 2 dB. This means that peak recording level can be allowed to kick into the red occasionally with no audible tape distortion.

Bulk-erased tape gave an RMS IEC weighted reading 64 dB below reference tape level or 66 dB below 3% peak recording level but machine-erased tape noise ranged from barely 60 dB on slightly worn tape to 64 dB on brand new low-noise tape. A one-octave band analysis of the noise showed a marked rise at low frequencies and this noise is thought to be a function of crossfield biasing which penetrated to the full depth of the tape oxide and is modulated by minute variations in thickness of the oxide or base material. Very slight second harmonic distortion in the bias waveform can bring this low frequency rumble to the threshold of audibility on wide-range reproducers. To put things into perspective, I should emphasise that the noise on this review machine is well below the danger level, even on old tape. The purpose of mentioning it at all is that the effect has been noticed at varying levels on other crossfield biased recorders, and a change of tape, or a thorough degaussing of the erase and recording heads, is required if rumble creeps up from the depths.

The overall record-play responses at the three tape speeds are shown in **fig. 3** for *LGS35* reference tape, but similar responses were obtained on samples of three brands of low-noise tape. The 4 to 6 dB rise in HF response could indicate over-emphasis of high frequencies during recording or slight under-biasing. We know from **fig. 2** that playback

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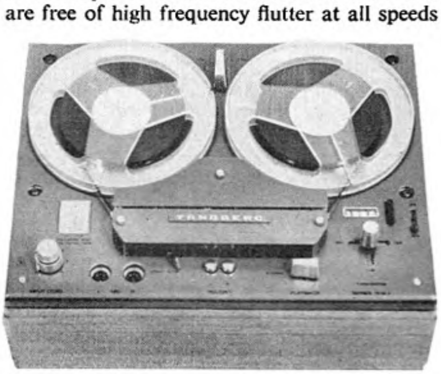


FIG. 1 TANDBERG 1600X RECORD / PLAY WOW AND FLUTTER

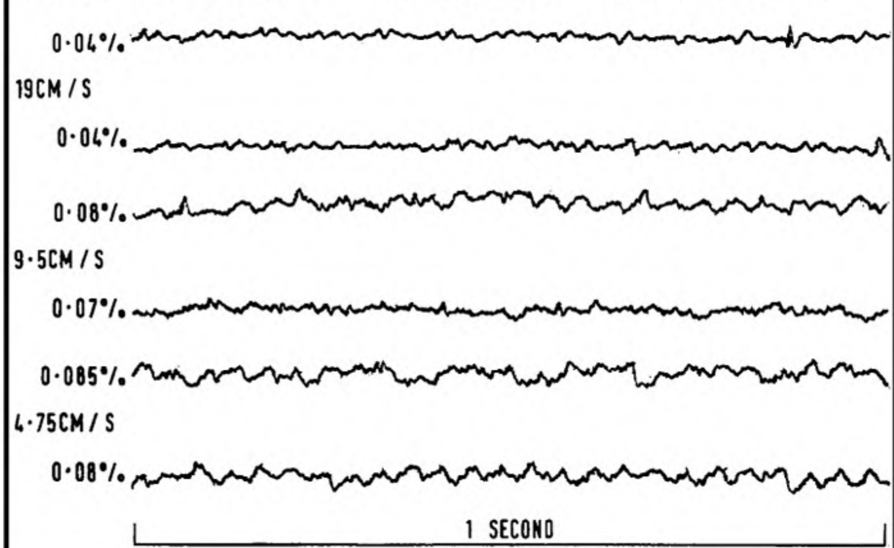
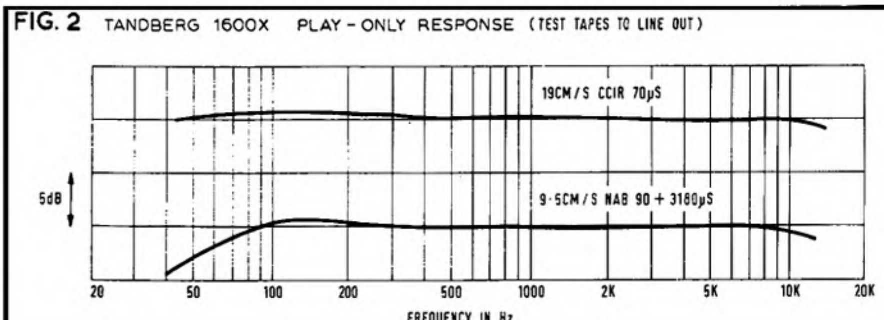


FIG. 2 TANDBERG 1600X PLAY - ONLY RESPONSE (TEST TAPES TO LINE OUT)



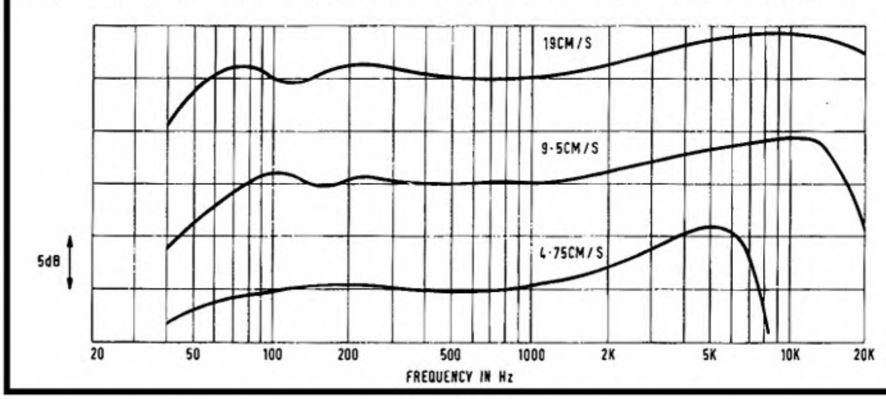
TANDBERG 16X REVIEW CONTINUED

equalisation is correct, and the tape distortion is at a low level, so my guess is that pre-emphasis is rather overdone for modern high-coercivity low noise tapes.

COMMENT

I like the separate record and play switching which is very flexible and almost foolproof and has that natural 'feel' which is characteristic of well-designed equipment. The peak-level meters are easy on the eye and use the dynamic range of the new tapes to the limit with little risk of overload. Playback equalisation is exact and system noise remarkably low. Recorded noise is adequate at better than 60 dB below peak recording level, but care should be taken in the choice of tape if low frequency rumble is to be avoided. The overall record-play response may be a bit 'bright' for some tastes, but it is better to have slightly too much HF response than not quite enough; it can always be tamed by judicious use of a tone control on the associated equipment. **A. Tutchings**

FIG. 3 TANDBERG 1600X RECORD-PLAY FREQUENCY RESPONSE (LINE IN, BASF LG535, LINE OUT)



STUDIO DIARY CONTINUED

the auditorium. Each pipe has characteristic resonances not only at the frequency which makes the length equal to one half-wave length, but also at all its harmonics.

After an outline of the absorption characteristics of common building and furnishing materials (from carpets, curtains and thin sheet materials to resonant absorbers), the Sabine's formula for calculating the reverberation time of a hall was quoted.

Amplifier tone controls allow system gain to be reduced in those regions of the spectrum where feedback is occurring. This technique can be taken to its logical limit by inserting frequency response equalizing networks that permit the overall gain to be reduced only at those frequencies where the room resonances provide tight coupling between microphone and loudspeaker. In favourable circumstances these can be extremely effective, an increase of at least 10 dB having been achieved in some installations.

When planning the position and mounting of loudspeakers, the polar response of the unit must be taken into consideration. It is almost impossible to use column and line source types vertically, because the beam will either be so high above the listeners' ears as to be useless or to be too low thus absorbed in the first few rows. Correct siting of columns and line source speakers is best performed by simple geometry on the drawing board. A basic rule is that the whole of the front of the loudspeaker should be visible from any audience seat, which means that the base of the speaker cabinet is sited at least 2 m from the floor. Phasing of microphones and loudspeakers is very important for optimum performance, and there are various methods for checking this condition. With the advent of the *Phaserite* tester (marketed by Keith Monks Audio), Mr Goodwin suggested that this device be investigated before considering the older methods.

The last lecture covered balancing and using a sound system, ably and amusingly presented by Hugh Brittain, audio engineer and this year's President of the APAE, with demonstrations supplied by Haydon Warren (PA engineer and APAE Hon. Librarian). Mr Brittain tackled the question of frequencies and their wavelengths in a novel manner. A wave in any medium is upset by any obstacle of about its own size or bigger. If the obstacle is much smaller than the wavelength a certain amount of scatter is set up. Applying these ideas to a column loudspeaker, and referring only to the top and bottom units in it, Mr Brittain produced a handy supply of paper with perforations at 2 kHz units. A toilet roll used for this demonstration to show a 2 kHz sine wave was very well received! This device was also pressed into service to show the directionality or beaming effect of column loudspeakers.

Mr Brittain has a pawky Scottish humour, which he employed to good purpose in presenting a lot of useful information in a palatable form. In mentioning the application of 'gun' microphones to performers who will not stand near the microphone, he was sometimes sorry that this type of microphone did not live up to its name! A cheaper device can be made from a pressure microphone and an old radar parabolic reflector. Manufacturers should be investigating the design of a really directional microphone. To be good at low frequencies it would have to be folding.

Referring to the different levels of vocal approach to the microphone he explained that one cannot shout an 'sss'—either you spit or your dentures drop out! In some church installations, he has found what he calls the Verger Effect, in which the volume is turned up to some point and left there, the human speaker may be near or far, loud or soft, but no control is done.

The incredible Hawaiian Piano

by David Kirk

SOME weeks ago, I dragged a recorder over to the piano (an upright, we can't afford the real thing), dangled a ribbon at the back, plugged in headphones, switched to 'echo', and proceeded to make unremarkable noises. Having tried several echo levels on chords and staccato notes, I decided to cut the speed from 38 to 19 cm/s to experiment with the increased-duration echo. Normally one does this by stopping the recorder first—it is polite to the capstan motor. The motor in this case was a servo-controlled type rather than the usual hysteresis-synchronous, so it seemed safe enough to cut straight through 'off' down to the slow speed. I didn't think this in so many words; I just did it.

The result was unlike any sound I have heard before, being a prolonged discord between the incoming signal and the reduced-pitch echoes. Only one problem: when the tape is played after the event, it sounds nothing like the original effect. An approximation can be achieved by trying to cut the replay speed in synchronisation with the original speed-change but this is not satisfactory.

I tried again a few days later, this time recording the output of the first recorder (live signal plus echo) on a second machine. In this way, the problem of synchronising with the original speeds is avoided. Then came another idea: slurring the tape by (left) hand and playing small ditties with the right. The combination of pitch variation and echo (one without the other was not effective) produced the unique sound of, for want of a better description, a Hawaiian piano. By pinching the tape gently in time with each note or chord (sustaining for maximum effect), it is possible to achieve a closely controlled glissando which almost totally disguises the identity of the original instrument. The pinching is undertaken, obviously, at the point between the feed reel and the head channel.

An intriguing aspect of this technique is that the 'master' tape on which the effect is produced is of little musical interest and would normally be erased.

Postscript: Is nothing original? Within days of writing the above, I heard exactly this effect being used extensively in *The Zodiac Cosmic Sounds*, a remarkable LP issued by Elektra (EKS 74009 Stereo). The real thing's answer to *Switched-on Bach*.

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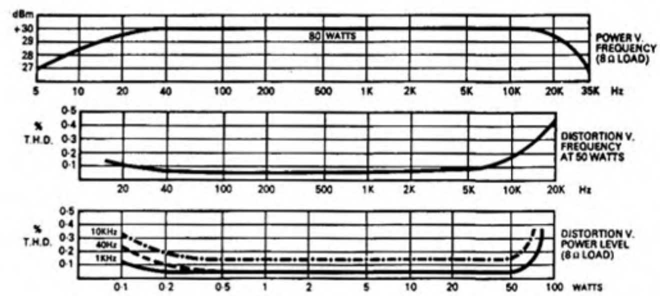

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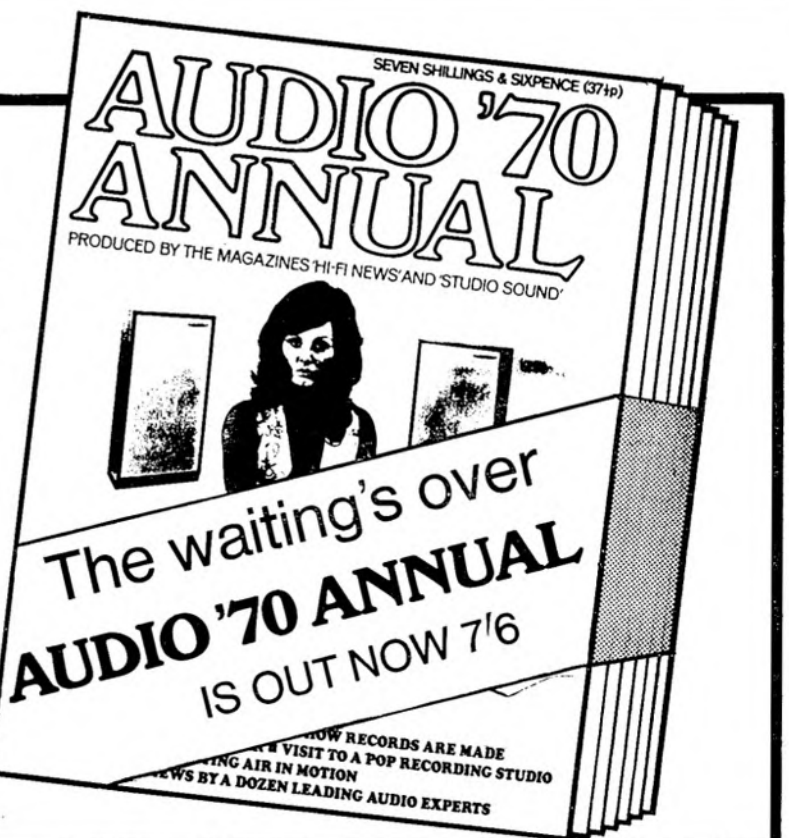

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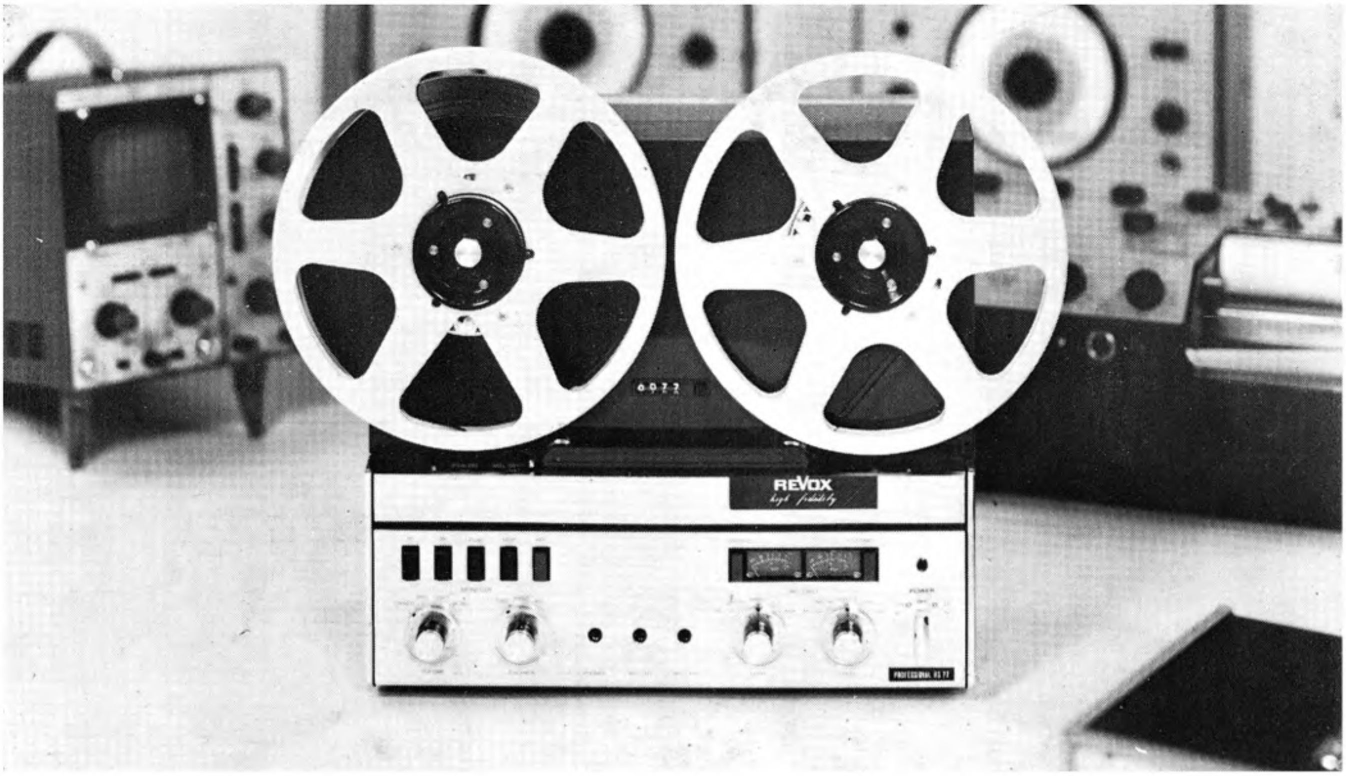

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