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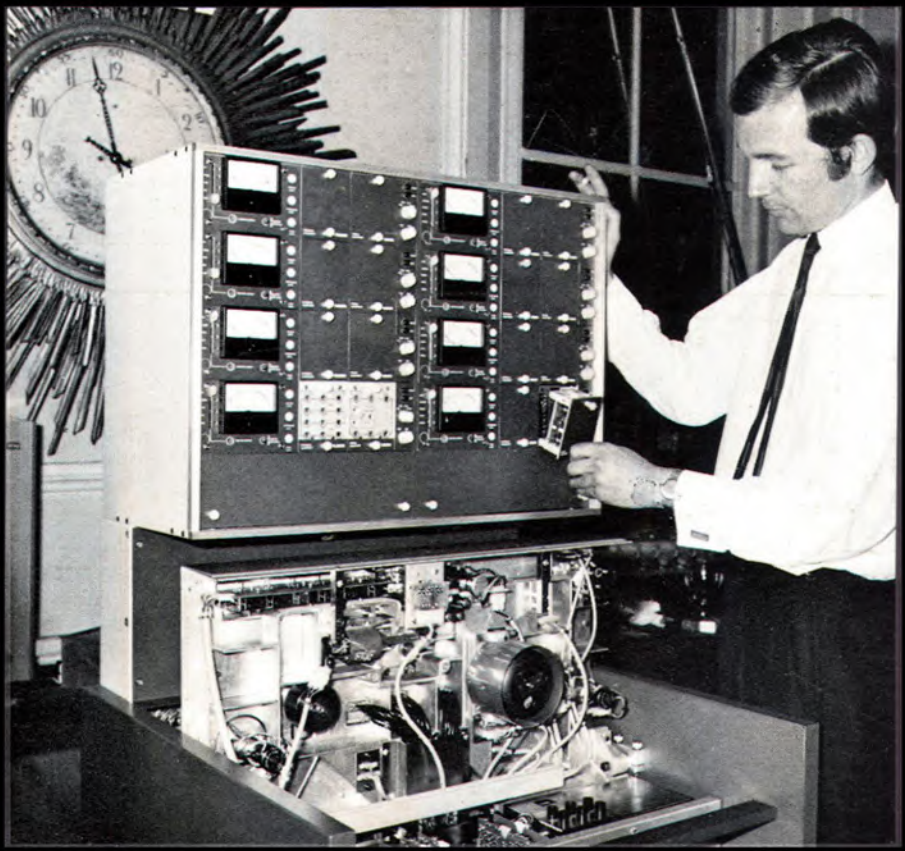

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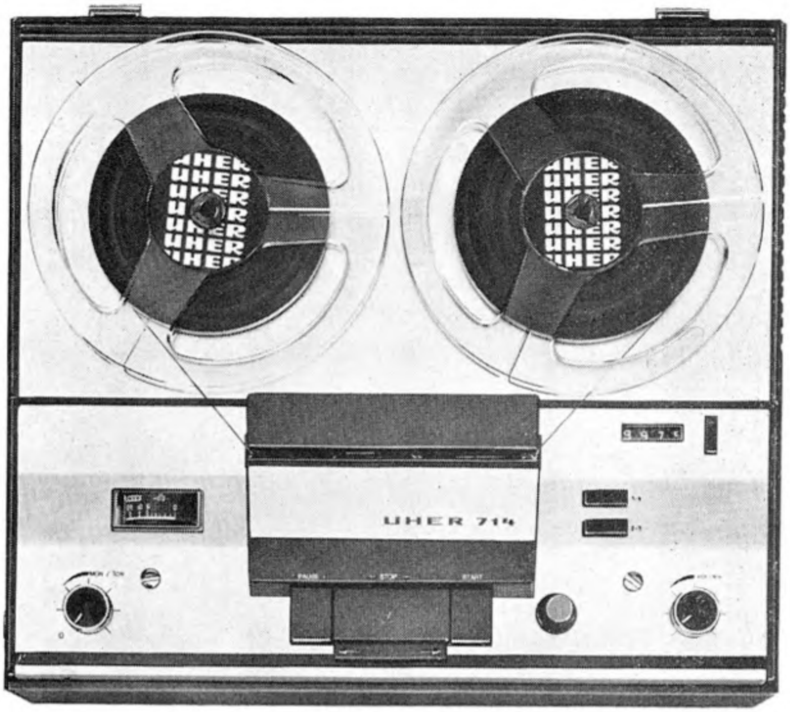

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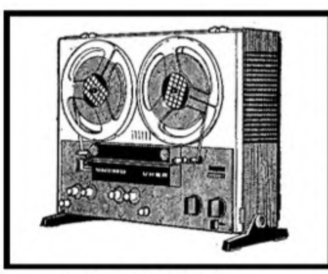

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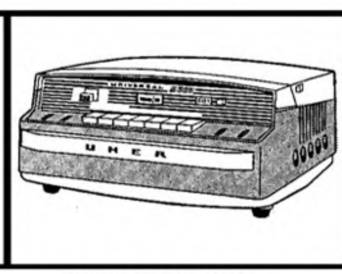

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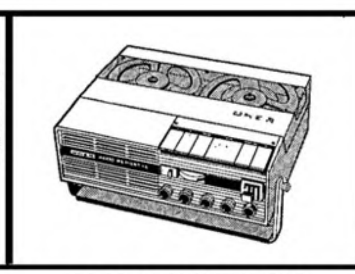

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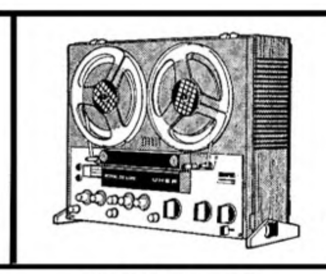

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AUGUST 1970 VOLUME 12 NUMBER 8

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COVER PICTURE

Michael Bauch (F.W.O. Bauch Ltd.) removes an amplifier panel from the Studer *A80* exhibited at this year's APRS Exhibition. A swivelling deck reveals the servo capstan motor.

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.

WHERE IS THE British recording industry going? To more and more tracks is the usual assumption. We came across a different answer recently in, believe it or not, the London headquarters of W. H. Smith's. This enterprising company has formed a division to study and eventually market television recordings. They are already promoting audio Musicassettes through their retail chain.

Study is a key word, as four different systems are being developed: *Videocassettes* by Philips and Sony, *Selectavision* by RCA, and *EVR* by CBS, ICI and CIBA. By the time this issue appears, the Teldec *Video Disc* system should have been unveiled to the daily press.

EVR was expected to be the first system on the market. It resembles conventional telecine in some respects, employing photo-sensitive film. Videocassettes are likely to follow, based on existing helical-scan VTR. Sony and Philips originally intended to standardise on an interchangeable 50.8 mm (0.5 inch) format, though Sony have since been reported as investigating a slightly wider tape.

Selectavision combines laser and holographic technologies in what could prove to be the cheapest of all the media. (Two companies, incidentally, are working on 'video discs'.)

Whichever system finally wins the day (Philips are the only company offering recording facilities), Smith's are gearing themselves to promote it by selling players and recordings. They are also considering the economics of a video library.

And where does this leave the studios? Thinking, perhaps, about the prospect of adding television production facilities to their present audio interests. Would the *Let It Be* LP have attracted greater sales if accompanied by telerecordings from the related Beatles film? Obviously, provided the price was within the consumer's reach. A small demand might eventually be created for telerecorded ballet and opera; even for closeups of Menuhin swinging into a Paganini solo.

If studios standardised on medium-price (£1 000 to £4 000) helical scan VTRs the move to television production could be within reach of all but the smallest two-track companies. On the other hand, if they opted for Quadruplex standards, extremely heavy investment would be required. A little matter of studio space would also have to be taken into account.

It is too early to consider the effect these developments may have on sound engineering. One can, however, reasonably wonder what the public will want to *watch*. Pop musicians miming? For a while, perhaps. Dramatic and documentary programmes? Probably, though inevitably on a hire or off-air recording basis; even an election broadcast would lose something after the fifth or sixth replay.

Since music is largely an abstract art, there might be some case for integrating synchronised visual patterns with the rhythm and mood of the sound. This was achieved with outstanding success in *Fantasia*, to Bach's *D Minor Toccata and Fugue*. The abstract sequence, like the more conventional film cartooning, must have been expensive and tedious to carry out, also perhaps to conceive. In 1968, the 'Cybernetic Serendipity' exhibition showed some of the attempts that have been made to automate visual art, though most were crude and of transient appeal. Fred Judd's 1963 *Chromatron* filtered audio into bass, middle and treble, activating three independent bulb groups: again of limited appeal but capable of development in several directions. For example, why not modulate the frequency of an oscillator with the audio signal, feeding the output tone into a second three-way filter and bulb arrangement?

A concluding point: television reproduction in its present state, even if one can afford wide-screen (?) colour, is quite inadequate for displaying pictorial art. Unless laser wall projection becomes commercially viable, synchronised optical pattern generators for home use could, in the short run, prove cheaper and more popular than canned television.

FEATURE ARTICLES

REGULAR COLUMNS

EQUIPMENT REVIEWS

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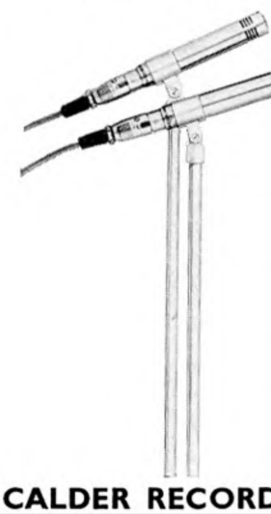
TRANSISTORISED CAPACITOR MICROPHONES



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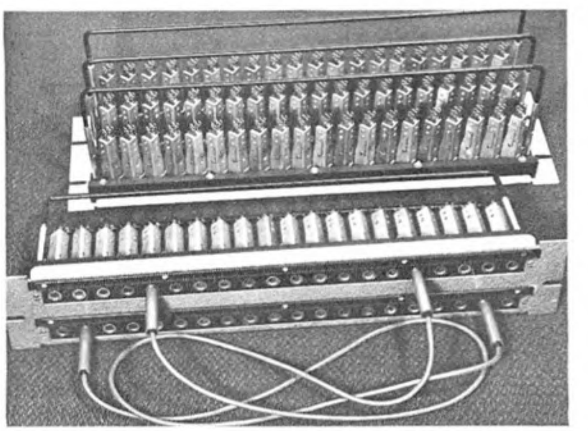

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studio diary

CAPACITOR MICROPHONE CAPSULES

THE RESPONSE to Trevor Attewell's capacitor microphone constructional articles, published in January, February and March *Studio Sound*, has proved larger than anticipated. STC have been able to meet only 140 of 300 orders to date and Mr Attewell is now making arrangements with alternative suppliers. Details to appear shortly.

FELDON IMPORTING MOOG

UK DISTRIBUTION of the Moog Electronic Music Synthesiser has changed hands from Audiotek Marketing to Feldon Recording Ltd., 126 Great Portland Street, London W.1. A new company, Audio-Europe Ltd., is being formed to handle the equipment in Europe and the Middle East. Directors will be Kevin Hibberd and Dag Fellner (Feldon) and two Scully executives. Feldon are also negotiating representation of Carl Countryman Associates (quadraphonic microphones), Orban/Parasound (stereo phase correction equipment), Spectrasonics and Metrotech.

BBC/PRECISION AGREEMENT

TAPES PRODUCED for BBC Radio Enterprises are to be marketed through Precision Tapes. These will include *Sir Malcolm Sargent, Music Maker*, an LP by The Young Generation, and a children's repertoire: *Jackanory* and *Listen with Mother*. The agreement was announced by PTs General Manager, Mr Walter Woyda.

MCPS AMATEUR FILM LICENCE

THREE CATEGORIES of licence have been introduced by the Mechanical Copyright Protection Society Ltd. for the benefit of amateur film makers. Class C 'to meet almost all amateur requirements' costs £5 10s per year, Class B at £3 10s per year allows 'a public showing once in each year'. Class A at £3 doesn't. Further details from the Administrative Secretary, Institute of Amateur Cinematographers, 63 Woodfield Lane, Ashtead, Surrey.

NEXT MONTH

KEITH WICKS visits Unitrack, Angus McKenzie reviews the Telefunken *M10* studio recorder, and John Fisher describes a variable-speed rewind system for three-motor tape decks. Exhibitions: a report on June APRS 70 and a preview of the International Broadcasting Convention to be held in London during September. John Shuttleworth will review the Spendor Monitor loudspeaker.

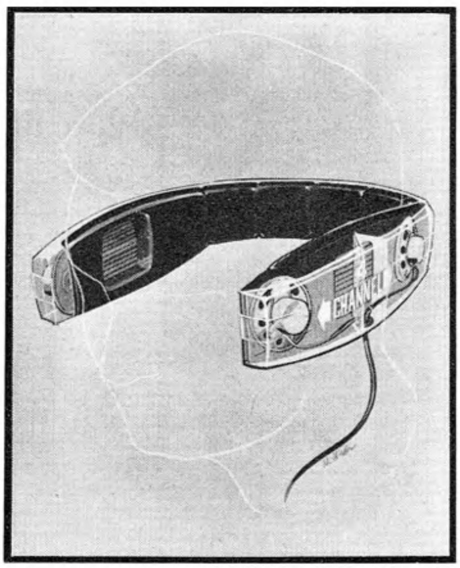
JANUARY TAPE RECORDER STATISTICS

IN JANUARY 1970, 21 428 tape recorders were manufactured in the UK, 32% more than in the previous January. Total deliveries of British made machines were 15% lower than in January 1969 but exports were 14% higher. 35% more foreign recorders were delivered in the month, compared with a year earlier, but re-exports were 54% lower. Stocks of British machines were 16% high on January 31 than at the end of the previous month. Stocks of foreign recorders fell by 6% over the month but were 27% higher than on January 31 1969.

Ministry of Technology Engineering Industries Statistics

FOUR-CHANNEL HEADPHONES

DEVELOPED IN the USA by Jon Fixler, the four-channel headphones illustrated were an obvious evolution of an earlier design intended to overcome in-the-head effects with two-channel material. Production samples are not yet available.



APRS 70 EXHIBITION

A REMARKABLE year for new equipment, the Studer *A80*, Unitrack *Uni-24*, Nagra *SN*, Ampex *ABR* recorders and Dolby *A360* single channel (three-band) and domestic *B* (single band) noise reducers all made their UK debut at the APRS 70 Exhibition in June. The eight track Studer (see cover) aroused considerable interest, not least because of its price, and incorporates a servo motor system on the lines of the Revox *77*. Its relatively low cost is made possible by simplified assembly techniques.

Stephan Kudelski arrived from Switzerland to demonstrate the miniature Nagra *SN* (see photo), a 9.5 and 4.75 cm/s pocket portable with capacitor microphone. He hinted, as he has hinted before, that a larger stereo Nagra will appear before long.

The servo-controlled Ampex *ABR* was seen for the first time, and will be available with any two speeds between 38 and 2.375 cm/s, and full, half or quarter track formats (two channels maximum). Bi-directional transport can be incorporated if required. The recorder is intended for general industrial use and logging.

Dolby's *A301* two-channel record or playback noise reduction unit was complemented

by the *360* single-channel unit (still three bands) at £240. Automatic changeover facilities are available for an extra £25 in the *361*.

A full report on the exhibition will be published in our next issue.

Matrix board channel routing is employed in the newly installed mixer at Hollick & Taylor studios, Birmingham 20. The system feeds an 8-track Leevers-Rich.

FIG. 1 Control Room

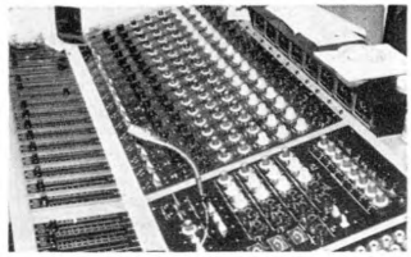


FIG. 3 Control desk

FIG. 4 Four-track Ampex

FIG. 5 Two-track Philips

around the studios

SPOT PRODUCTIONS

By Keith Wicks

GARY Levy (a cousin of Jacques Levy who was interviewed last month) is the managing director of Spot Productions in South Molton Street, London. The studio, which measures 7 x 5 m, accommodates up to 20 musicians. Most of the work at Spot is in the pop field. Such people as Billy Fury, Joe Brown, The Who, Robin Gibb, Family Dogg, Vikki Carr, Cleo Lane and Julie Felix, to name but a few, have recorded at the studio, and it was here that The Cream made their first LP, which was entitled *Fresh Cream.*

Bill Dyer, the studio's chief maintenance engineer, first showed me the control room, which is separated from the studio, and normally linked to it by CCTV. This is not shown in any of the illustrations as, at the time of my visit, they were awaiting a replacement tube for the monitor. However, they seemed to get on quite well without the visual link. **Fig. 1** shows the general layout of the control room.

The types of microphone used are Neumann *67*, *56*, *64*, *SM2*; AKG *C12*, *C28*, *D202*, and Sennheiser *MD 421N*. The mixing console was made by Tiros Electronics and has 16 output channels (**fig. 2**). The controls on the channel modules are, from top to bottom, channel gain, bass, middle, treble, echo send select (for spring, plate, or tape delay), echo send level, pan switch, and a switch to select the channel to the required output. At the bottom of each unit are two switches. One is to feed the channel to the loudspeaker foldback system or to either of the two headphone foldback groups, and the second lever switch is used to cut the channel when it is not required. Below each channel module is the associated slide fader, made by Dana and marketed by EMT.

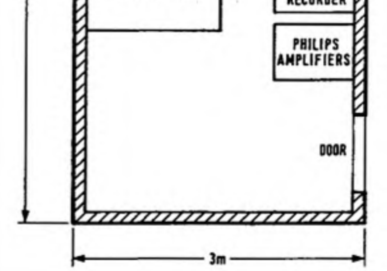
On the right of the desk is a panel containing, at the top, monitoring controls for the eight desk outputs. The first row are level controls, and the second row are for loudspeaker selection. The speakers are (once again) Tannoy Golds, housed in cabinets custom built for the studio. The monitoring amplifiers were also custom built and provide up to 40 W to each of the three speakers. Visual monitoring is provided by eight PPM's mounted on the top of the desk.

Other facilities in the right hand panel include echo return switching and equalisation, line in / line out switching, tape machine selection and talkback. The four small VU meters on the right were originally intended for echo send monitoring but are not used at the moment, neither are the two rows of jacks adjacent to this panel. There are (surprisingly) five main faders, three of these controlling one output, and the others controlling two and three outputs respectively.

A bay in the corner of the room contains the main jackfield, headphone volume controls, wide range equalisers, and limiters. Next to the bay is an Ampex mono machine on top of which is a Leevers Rich *A501* graphic equaliser. To the right is a fairly old Philips two-track machine and its large cabinet of associated amplifiers (**fig. 5**). Below the deck of this recorder is mounted a transcription turntable. The multitrack recorders are located below the speakers at the end of the control room: an eight track Leevers Rich (**fig. 6**), and a four track Ampex with its separate console of amplifiers (**fig. 4**).

Not long after I arrived at Spot Productions, a recording session began, featuring Tim Hollier and Richard Cuff with chief engineer John Hudson at the control desk. Hollier has been recording for about two years, and had some success in the USA and this country with his United Artists LP *Message to a Harlequin.* As a result of the interest this created, Tim Hollier Productions was formed by Tim with PR man Roger Fennings as his co-director. The company's first product was Tim's single *In This Room* released in March by Fontana.

FIG. 2 SPOT PRODUCTIONS CONTROL ROOM



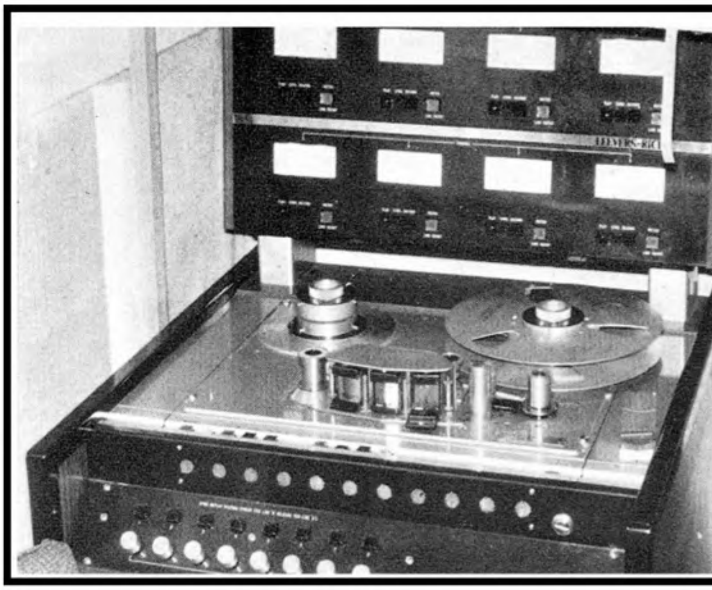


FIG. 6 Eight-track Leevers Rich

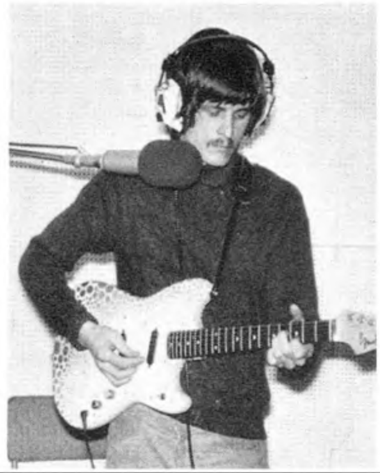


FIG. 7 Richard Cuff

FIG. 8 Tim Hollier

around the studios

Richard Cuff, a Canadian folk singer, is one of the artists signed up with Hollier's publishing company.

The four track Ampex was used for this session, and Richard and Tim are shown (**figs. 7** and **8**) providing vocal with guitar backing.

Before each take, a high level 40 Hz signal is put on the tape so that, when spooling a distinctive bleep allows the operator to locate particular items on the tape fairly quickly.

The studio use mainly BASF *LR56*, because, Mr Levy told me, it will take a higher level than other tapes, 4 dB more than Emitape for example.

At most sessions I have attended, something has gone wrong. This is because a lot of equipment is involved, and statistically, something almost *has* to go wrong. At Spot, some trouble was caused by interference from a fluorescent light being picked up on one of the electronic guitars, but the trouble was soon located and dealt with. This is the important thing as time, particularly in recording studios, is money. Another event was the appearance of a 'mystery voice' on the monitoring speakers, talking about taking someone to Paddington Station. When I asked one of the engineers if they always received taxicab radio, the reply was, 'No—it's sometimes the GPO!'

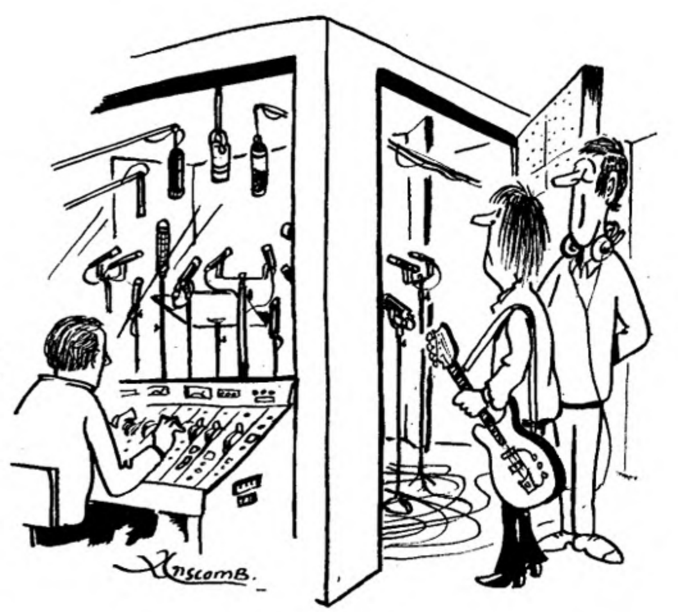
When a vehicle is transmitting directly outside your premises, it is certainly difficult *not* to pick up the signal, and Spot are in good company in this respect, as I know that the BBC have had similar problems at times. Fortunately, parking restrictions usually take care of this and, as in this case, the offending vehicle soon moves off.

Despite the problems, negligible recording time was lost. The operations and maintenance staff are most efficient, and I would not hesitate to recommend Spot Productions to anyone.

Their current hourly rates are as follows:

Eight track recording:	**19 gns.**
Four track recording:	**15 gns.**
Two track recording:	**12 gns.**
Full track mono recording:	**10 gns.**

As with most other studios, there are plans for expansion, although here they are fairly long term. Nevertheless, I look forward to another visit in two or three years time to see their enlarged studio and their 16 track equipment—or will it be 32 by then?



'With a voice like yours, you need all the help you can get.'

RECORDING STUDIO TECHNIQUES

FOR many people the use of bass and treble controls, and perhaps treble steep cut and rumble filters will be the only experience they have of using equalisation to alter the shape of an input signal frequency spectrum. In the recording studio, equalisation is very important for a number of reasons which may not at first appear obvious, but which I hope to explain clearly. There are a number of different types of equaliser, the best-known being the simple bass and treble control which these days is only very rarely used. Normal controls of this type tend to start boosting or cutting bass below 500 Hz or so, and treble above approximately 2 kHz. The maximum amount of lift or cut which is available on most equalisers of this type is approximately 16 dB at 50 Hz and 15 kHz respectively, although several cases exist where the manufacturers have unwisely provided the facility of more lift. Taking the bass end first, it should be remembered that most tape recordings made in the studio have to be cut on disc, which again has to be capable of being played satisfactorily on all gramophones. In many recordings, there are instruments capable of providing considerable power well below 100 Hz. For this reason, if such a tone control is used, there may be appreciable boosts at frequencies around 50 Hz which may sound effective in the studio but which the cutting engineer could take off again in order to make the signal suitable for disc. Cutting excessive amplitudes at bass frequencies would cause the playback stylus to jump. Most recording engineers prefer a type of lift circuit in the bass (if a lift is necessary rather than a cut) which boosts up to a certain amount and then flattens off or even falls back.

Very frequently bass cut is desirable and while this is usually done by means of a conventional series-capacitor roll-off circuit, often a slope of greater than 6 dB per octave is required. Whereas on consumer equipment, a rumble filter normally commences roll-off at 40 Hz or so to remove turntable rumble, the engineer's requirement is usually to remove traffic rumble picked up by mikes, or even to reduce the effect of soggy bass reverberation. Whereas high frequencies tend to be clearly directional from most musical instruments, lower frequencies spread in all directions and can be picked up by mikes at an appreciable distance from the source, and it is these that frequently have bass cut applied to them. Another example is the effect of a double bass with this roll-off applied, especially when the microphone is placed fairly near the bridge or even higher up. The twang and thump of a plucked note may be considered more exciting when recorded through a bass filter. If it is required to emphasise a very low register, the technique referred to of boosting the 100 Hz region by 6 dB or so, while hardly boosting the 40 Hz region, may well be considered safer.

Few stereo machines made over a decade ago had a response extending far below 50 Hz, though most professional machines made today frequently go well below 30 Hz without any appreciable roll-off. It is this bottom octave which can bring to life the sound of a cathedral organ, a large symphony orchestra, or indeed a pop group with huge loudspeakers for their bass guitars.

In general, when considering disc cutting, a high amplitude is permissible at lower frequencies in the lateral (sum) channel than in the vertical (difference) channel. It is frequently necessary, particularly when crossed bi-directional mikes are used in cathedrals, to introduce a bass cut in the difference channel only, thus cutting out reverberation at lower frequencies entering the stereo system from the out of phase lobes of the crossed pair. It is not unusual to find the LF difference channel on an untreated tape higher than the sum channel. The effect of such a cut will in fact usually increase the impression of presence in the lower register rather than decrease it.

Very similar criteria affect the application of high frequency boosts frequently necessary to increase the presence of an instrument. If such a boost is added to the output of a microphone in a particular recorded balance, the instrument will tend to sound louder and closer without its real amplitude increasing. This is because an instrument covered by a distant mike arrives with reverberation from the studio or concert hall surroundings and has its high frequency harmonics reduced by the characteristics of the studio. The ear tends to associate the intensity of harmonics in any sound with the distance of the sound source from the mike or listener. It has also been found that the air itself tends to reduce the HF proportion of distant sounds, even when little or no reverberation is present because of the sluggishness of molecular air movement. This is the reason incidentally that some capacitor microphones particularly liked by many engineers for recording in concert halls have a slight top boost actually present in the output of the mike. An example is the Neumann *SM2* stereo, now superseded by the *SM69*.

In the same way that high amplitudes of bass frequencies can cause mistracking on gramophone records, high amplitudes of HF causing high velocities to be present on the record can cause distortion on playback, since the playback stylus is physically unable to remain in clean contact with the groove wall. Magnetic tape also has an HF saturation point appreciably lower than at middle or lower frequencies, and once again distortion can be introduced if too much treble boost is added.

It is highly dangerous to apply pre-emphasis to singers who have a pronounced sibilant. A good example of this is the recent issue of Dana singing *All Kinds of Everything* in which the 8 kHz recorded velocity reaches approximately 34 cm/s at points corresponding precisely to the artist's sibilant. In order to overcome the disadvantages of a general HF boost, most studios employ a system by which an HF boost can be applied at several different frequencies with a further control on the bandwidth of the frequency boost in addition to a third control on the amount of lift. It is possible with such a device to introduce a boost at well below a sibilant frequency without any terrifying high cutting velocities at high frequencies, though any steep cut above this boost would tend to sound unnatural. An extremely versatile equaliser with this provision was made by Pultec in the USA some 15 years ago and is widely used in the UK. The same manufacturer also made some excellent passive low and high pass filters with a very steep cut-off, one knob controlling each end of the frequency spectrum. I well remember the effectiveness of providing with this latter unit a pass band extending only from 500 Hz to 2 kHz, the resultant sound being almost identical with the sound of telephone speech, and I used this technique for such imitations in a number of plays. Many studio control desks now incorporate the basic design of the first Pultec described. I am sure that it is not fully realised by the public that a balancing engineer should be, and usually is, just as much an artist with his controls as the musicians in the studio, particularly in the recording of pops. Before the days of the Dolby, it was most important to add as much HF equalisation as necessary *before* the initial tape recording since boosts applied when copying a tape could drastically increase hiss. There is now a tendency for equalisation to be applied to the output of multitrack recorders which almost always these days include Dolbys.

A further very useful tool which has been available for many years now is the graphic equaliser, with controls on each octave in the audio range. It will be seen that almost any curve is possible provided the slope required anywhere is not greater than about 12 dB per octave, though I understand some will give a greater figure if one octave is boosted and the next octave cut. The graphic equaliser is used to correct the overall proportion of different frequencies present but may not be sharp enough in boost and width for some requirements such as emphasising of cymbals or the rim shot of a drum. One of the first graphic equalisers made commercially in Great Britain was made by Astronic (AEE Ltd.); I have used two of them for many years and found them extremely versatile. The circuit includes several double triodes, each having a feedback circuit around it tuned to a particular octave. The inputs of all circuits are paralleled via isolation resistors and the outputs controlled by potentiometers, the outputs of which are mixed and then amplified in a line output stage, which also includes a master pot. I had to make a

(*continued on page 342*)

BY ANGUS McKENZIE (Roundabout Records)

PART 8

EQUALISERS

VIDEO-TAPE IN BROADCASTING

by Howard Dell

Video-Tape Editor, BBC Television

TWELVE years ago, BBC Television viewers saw the opening seconds of one of the *Panorama* programmes twice in succession —at first 'live', then followed almost immediately by a re-run. No, it wasn't a mistake in the studio, the date was April 14 1958 and the occasion was the now famous demonstration by the late Richard Dimbleby of a video recording machine, designed and built by the BBC, that used magnetic tape as the recording medium. Not surprisingly, it aroused considerable interest at the time for, although magnetic sound recording had been around for many years, the development of a video recorder had been hindered by the implications of the incredibly high head to tape velocities necessary to accommodate the bandwidth of the television picture.

The BBC's solution was the obvious one: running tape at a very high speed past a static head, but the limitations of this idea were soon apparent, not the least of which was the high tape consumption. It wasn't long, therefore, before the rotating head concept realising head to tape velocities of about 40 m/s, became the accepted solution to the problem and within a year the American Ampex machines employing this technique started to come into service in this country. Readers will probably be familiar with the principle of the quadruplex video recorder: 50.8 mm (two inch) tape running at about 38 cm/s is shaped by a concave guide around a rotating wheel. Four mini record-replay heads mounted at 90° intervals on the periphery of the wheel in turn lay a narrow stripe of video information 0.25 mm wide transversely across the tape where each stripe represents some 18 lines of the scanned 625-line TV picture.

So sound was the design of the original quadruplex machines that this is essentially still the method employed today in broadcast TV recorders, though of course system complexity has increased enormously, particularly since the introduction of colour.

Few people in 1958 can have realised the impact video-tape would have on television production techniques. So complete has the takeover been that today, apart from news and current events programmes, virtually every picture originated by a television camera hits the screen via the medium of tape. A quick glance at the *Radio Times* shows that, allowing for films and the occasional live programme, some six out of ten programmes of an evening's entertainment are on tape. However, the revolution has been so quiet that, to the general public, anything that isn't live must be a 'film'—a belief that extends to many newspaper TV critics who should know better; it can be very irksome to read a criticism beginning 'Last night's film of the Wednesday Play . . .'!

Why then has video-tape taken over from film as the primary recording medium in television broadcasting? For precisely the same reasons that 6.25 mm tape caused the demise of original disc sound recording: low cost, high quality and flexibility. Add to this the convenience of instant replay, and video-tape becomes ideal for the ephemeral world of TV.

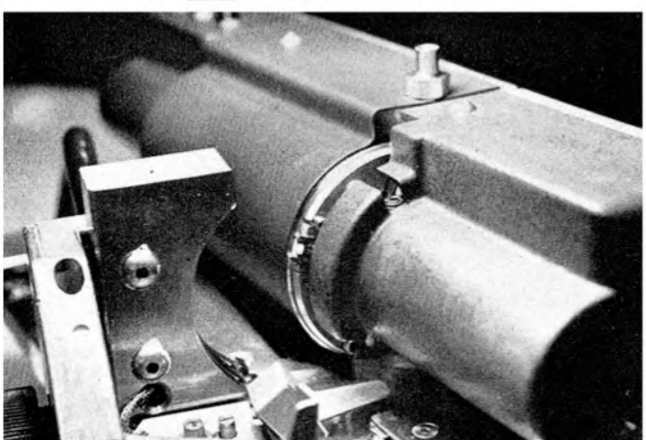
When talking of low tape cost, this is, of course, low only in proportion to a programme budget—a reel of 50.8 mm tape playing for 60 minutes costs around £100. And the price of a broadcast VT machine is shattering—whereas one has quite a wide choice of closed circuit helical scan machines for under £1 000, a fully equipped quadruplex recorder can set the buyer back by about £60 000.

Having spent a small fortune on such a machine, what do you get and how is it used? Well, to start with you get a tape deck of conventional layout that takes reels up to 36 cm diameter (that's 90 minutes playing time) with a normal capstan pinch-wheel arrangement and a recognisable sound head assembly, but there familiarity stops. The rotating vision head with its drive motor sits predominantly at the front and whines alarmingly on its air bearings when running at 15 000 RPM. Below the angled deck is a stack of plug-in printed circuit modules containing an uncountable number of transistors and integrated circuits—some to process the vision on record and playback, servo systems to drive and control the deck, elegant electronic devices to correct deficiencies of the tape-to-head transfer and, of course, the audio chain. Add to this a picture monitor, level meters, and two built-in oscilloscopes for constant monitoring of the circuitry, and you have a machine that will produce a standard colour recording that can be replayed on this, or any other similar machine, to give breathtaking colour pictures which can only be distinguished from 'live' by the trained eye of a video-tape engineer—and sometimes even he can be fooled.

Against the complexity of the vision circuitry, it comes as a pleasant surprise to find that the sound system is as conventional as on any audio recorder. In fact, the VT machine has two sound channels, the main one, used for the programme sound, runs along the top edge of the tape and the second, of slightly poorer quality and known as the cue track, lies near the bottom edge. The cue track can be used as a guide track for timing purposes, or might carry a second commentary (say in a different language).

The main audio track is 1.77 mm wide and with a tape speed of 39.5 cm/s ($15\frac{5}{8}$ i/s) the sound quality is, of course, quite good: a typical machine will have a flat record/replay response from 40 Hz to 15 kHz. Unfortunately, with the emphasis on picture quality, quadruplex video-tape is manufactured with the magnetic elements of the oxide coating aligned transversely which gives rise to a higher hiss level than with conventional tape. Wow and flutter presents no problem at all, as the tight

Close-up of a video head assembly. The shaped guide has been retracted, revealing the headwheel and one of its pole tips.



The BBC 1 Transmission Suite Control Room. One of the two remotely operated machines can be seen through the glass panel.

servo control of the tape transport demanded by the video department is more than good enough for the sound—the overall record/play wow and flutter can be held down at around 0.02% to 0.03%.

In order to keep the machines running at peak performance, routine maintenance is essential. Apart from a fortnightly major servicing period, maintenance staff keep a continuous watchful eye on all machines, particular attention being paid to the video head. Due to the high speed at which it operates and the fact that the pole tips are impressed slightly into the tape, the wear rate is very high and a head may well be worn out after about 100 hours running time. Rejected head assemblies (that is, head wheel plus drive motor) are returned to the manufacturer for refurbishing for further use.

Once set up, it is no more difficult to record a programme on a quadruplex machine than to make a normal sound recording. The operator has normal play, record, wind and stop buttons at hand and a kaleidoscope of 'fault' lights inform him of any malfunction while the recorder is running. What he cannot do, however, is check the picture off-tape as he is recording (as a three head audio recorder can monitor sound). There is always the danger therefore that the recording will unknowingly be marred by tape dropout, or perhaps a catastrophic loss of recording will occur due to the tape shedding oxide and clogging the heads, although 'clog' detectors are now becoming available.

Because of this uncertainty and the possibility of some other failure, programmes are nearly always recorded on two machines; obviously the chances of both recordings being faulty are pretty remote, but it does happen, though fortunately rarely in a manner that makes a show beyond salvage! Tapes are bulk erased before use, a purely precautionary measure since the machines have an efficient full width erase head. (It might be of interest to relate this to the editorial of the February 1970 issue of *Studio Sound*: the BBC-designed bulk erasers both rotate the tape *and* increase the magnetic erasing field from zero to a maximum then down again, automatically. Once per revolution 'thumps' just don't exist.)

After a playback of the whole recorded programme to check the technical quality, it is very likely that the tape will require editing before transmission. Probably 80% of taped programmes seen on the air have been edited to some degree. This may be just one join in a schools programme to cover a fluff or a break in the action, or upwards of 150 edits in a large drama where the play will have been shot deliberately in short sequences. Traditionally, video-tape has been edited by cutting and joining the tape at the required points with a special sticky tape, the butt ends being aligned in a microscope-fitted splicing jib. Technically, the cut is made in the field blanking period and with a positional accuracy on the tape of about ± 0.05 mm; exceed this and the picture as seen on the screen may well flash at the edit point, in extreme cases it could break up altogether and take several seconds to restabilise.

A more elegant method of editing has been taking over slowly in the last few years and with the introduction of colour has really come into its own: a process called electronic dub editing. Here the tape isn't cut at all—the whole programme is re-recorded from the master on to a clean tape using a pair of machines which can be locked together electronically in such a manner that the programme is built up from the required sequences. At the heart of the system lies an electronic editor, which controls picture edits to one frame accuracy at the turn of a knob and can effect speech and music edits that are quite undetectable—all without ever having to touch the tape. Combined with the wealth of television special effects possible these days: dissolve, colour distortions, freeze frames, etc., and as much sound tinkering as one can devise, there is potentially a creative programme editing service that is limited purely by the cost of time. In practice, the time spent editing a programme might be five hours for a thirty minute comedy show to two or three days for an important drama (like one episode of *Henry VIII's Wives*).

A tape costing around £100 when new will have an intrinsic value of perhaps tens of thousands of pounds by the time it is ready for transmission, so great care has to be taken both in its handling and storage. Before loading a programme tape, the VT machine is always cleaned with the solvent *Freon* and special paper tissues, and the heads are demagnetised with an ordinary hand degausser. Ingenious tape movement sensors on the decks prevent tape stretch by inhibiting selection of different modes (for example, play from rewind) until the tape has stopped, and the mechanical alignment of the tape transport is regularly checked to ensure even winds on the spools. Tapes are stored and transported on their edges in stout fibre or plastic boxes and the tape library has an air conditioning system with temperature and humidity maintained within very close limits.

So much for the predictable dangers—but what about the human element? Well, the machines have a record 'lock-out' switch which will prevent the record mode from being selected by accident, but of course it has to be set. Should someone actually manage to start recording on top of a programme, it is unlikely that much would be lost before the mistake were realised, and in this event a repair could be made from the second (backing) copy still held in reserve.

Having edited the tape and stored it safely there comes the time when it is to be transmitted. Although most of the 20 machines at the Television Centre in London are capable of going on the air, in practice transmissions are usually originated from one of the 'Transmission Suites'. There are two installations so designated, one serving the BBC 1 Network and the other BBC 2, each containing two of the latest designs of VT machines in ultra clean surroundings.

The engineer responsible for preparing the tape for transmission lines up the machine controls on the two minutes of test signals recorded before the start of the programme. The most important of these will be the 'colour bars'—vertical stripes of different colours—which he will match to a standard. Satisfied that all is well, he thereafter operates the machine remotely through the glass window of the adjacent Control Room.

Final instructions to run the machine for playback on the network are given from elsewhere in the Television Centre by the Network Director who, at the end of the previous programme counts down the 10 second run-up of the VT machine (to ensure good stable pictures) then switches it on the air for, we hope, an

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The start of a transmission, as viewed by a Control Room engineer.

Front doors of a VT machine open to reveal most of the electronics.

THE PRINCIPLES OF QUADRAPHONIC RECORDING PART ONE

By Michael Gerzon

Are Four Channels Really Necessary?

THE recent growth of interest in quadraphony (i.e. sound reproduction via four loudspeakers) has encouraged the belief that four recording channels are necessary for a full quadraphonic effect. The author has recently published an account[1] of a method of four-speaker reproduction of ordinary two-channel stereo that can give a convincing all-round-sound effect, despite certain theoretical and practical limitations. This poses the problem of whether one really needs to record or transmit four channels of audio for four speaker quadraphonic reproduction.

The aim of this article is to give an elementary theoretical analysis which indicates that *three* recorded channels should be quite adequate for quadraphonic reproduction. The uses, advantages and limitations of this are discussed, and formulae are given which indicate how three-channel recordings can be reproduced via four loudspeakers, and how four-channel recordings can be reduced to three channels. The second part of this article is devoted to the use of these considerations in obtaining a system of *Periphonic* (Greek: *peri-*, around) sound reproduction, i.e. the reproduction of sound in all spatial directions, from in front, each side, behind, above and below.

While the author has used certain advanced mathematical techniques in deriving the material in this article, all the results are here stated only in terms of very elementary mathematics, and physical reasons are given for most of the phenomena. It is hoped that the information here will prove useful in designing multi-channel recording and reproducing systems, quadraphonic pan-pot circuits, and in other applications.

It is generally accepted that three speakers are not adequate for good surround sound, due to the limited listening area and the wide angle between the loudspeakers. This has led many people to assume that, because four loudspeakers are necessary for surround sound, therefore one needs to record four channels. The author has shown[1] that even two-channel recordings can be made to give a genuine surround sound (albeit with some defects), and this suggests that three channels might be quite sufficient to convey all the information required for quadraphonic reproduction.

There are strong arguments in support of deriving the sound for four loudspeakers from only three channels of recorded sound. The primary purpose of four speaker reproduction is to reproduce music realistically. It is generally recognised that the most natural recordings are obtained by coincident microphones, rather than spaced or multi-mike techniques. (It is true that the latter techniques may produce a more spectacular, 'pleasing' or analytic sound, but it is not the purpose of the present article to argue matters of taste.)

It may be thought that placing four coincident microphones with, say, cardioid directional characteristics pointing in different directions will give a reasonable four channel sound. For conventional quadraphony, which only conveys horizontal directional information, these microphones will normally have their axes pointing horizontally. However, it may not be generally known that, for microphones whose axes point horizontally, there are only three linearly independent microphone directional characteristics. Put another way, given four coincident microphones whose axes point horizontally, it is always possible to derive the audio output of at least one of the microphones by matrixing the outputs of the other three microphones together in suitable proportions. (Technically, this is expressed by saying that 'the space of horizontal microphone directional characteristics is three-dimensional'. This arises from the fact that all conventional microphone directional characteristics are linear combinations of zero and first order spherical harmonics. In future, high quality microphones may be developed whose directional characteristics involve second order spherical harmonics. Four such microphones would be capable of recording four independent channels for four speaker reproduction.)

This means that whenever a coincident microphone technique is used for sound to be reproduced over four loudspeakers arranged horizontally around the listener, as in **fig. 1** for example, only three microphones are actually needed to obtain all the audio information. The sound fed to each of the four speakers can be derived by suitable matrixing of the three

FIG. 1 QUADRAPHONIC SPEAKER LAYOUT

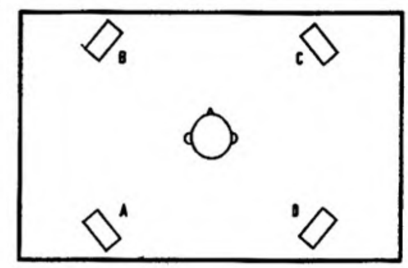
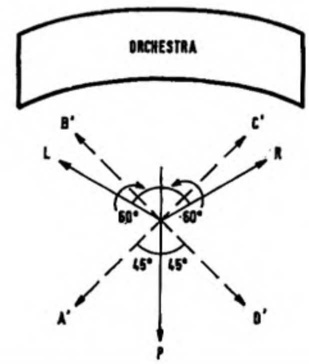


FIG. 2 COINCIDENT MICROPHONE ORIENTATION



microphone signals. Thus, for many purposes, only three recorded channels are needed to convey all the information reproduced by the four loudspeakers.

In order to give this assertion concrete form, it is first necessary to describe the layout of the reproducing loudspeakers. In the standard quadraphonic system, the four loudspeakers are to the rear left, front left, front right, and

THE PRINCIPLES OF QUADRAPHONIC RECORDING PART ONE

By Michael Gerzon

rear right of the listener, in a square as illustrated. It is convenient to label these loudspeakers A, B, C and D respectively, and to use these letters to indicate the four audio signals which must be transmitted to the four loudspeakers.

Consider three identical coincident microphones with, say, cardioid directional characteristics pointing in the three directions indicated by solid arrows in **fig. 2.** Thus, one microphone points 60° to the left (giving an output L), one points 60° to the right (giving an output R), and one points backward (giving an output P). According to what was said above, it is possible to derive all other horizontally-pointing microphone outputs from L, R and P by matrixing. Also note that the signals L and R form a good stereo signal. The signals A′, B′, C′, and D′ fed to the four loudspeakers A, B, C and D could well be the outputs that would be given by cardioid microphones pointing in the four directions (broken arrows) labelled A′, B′, C′ and D′ in fig. 2, i.e. 135° to the left, 45° to the left, 45° to the right, and 135° to the right. Rather messy trigonometric computations show that A′, B′, C′ and D′ may be obtained from L, R and P by the matrixing described in **Table 1.**

If the microphones pointing in the directions L, R and P of fig. 2, have a given identical hypercardioid directional characteristic, then the signals A′, B′, C′ and D′ obtained by the matrixing of Table 1 will be the signals which would be obtained from identical hypercardioid microphones pointing in the directions A′, B′, C′, and D′ of fig. 2,

Thus, for coincident microphone recordings, we need only record the signals L, R and P given out by identical cardioid or hypercardioid microphones pointing in the directions of the three solid arrows in fig. 2, The signals fed to the four loudspeakers can be obtained by the matrixing of Table 1. This illustrates the principle that good quadraphony can be recorded using only three channels, although four loudspeakers are needed to reproduce it.

Having shown that coincident microphone recordings need only three channels, the question naturally arises whether other types of four-speaker audio can be recorded or transmitted using only three channels. Most of the four-channel material recorded at the moment consists of more or less independent sound on each channel, due to the use of widely spaced microphones. There is one obvious way of reducing genuine four-channel recordings to three channels. This is to derive the three signals L, R, and P that would be picked up by imagined microphones pointing along the solid arrows in fig. 2 if the sound A, B, C, and D of the four channels were played through loudspeakers in the four directions A′, B′, C′, and D′ in fig. 2. While we are only imagining 'make-believe' microphones picking up imaginary loudspeakers, the computation of the signals L, R, and P picked up by these microphones does give us a prescription for reducing four channels to three. Unfortunately, the signals L, R, and P thus obtained depend on the choice of directional characteristic of these imagined microphones. This illustrates the fact that there is no unique way of reducing four-channel material to three channels.

Table 1

Converting three channels L, R, P to four channels (see fig. 2)

$$A' = 0.506\,L - 0.311\,R + 0.805\,P$$
$$B' = 0.977\,L + 0.161\,R - 0.138\,P$$
$$C' = 0.161\,L + 0.977\,R - 0.138\,P$$
$$D' = -0.311\,L + 0.506\,R + 0.805\,P$$

Table 2

Converting four channels A, B, C, D to three channels (in the manner of 135°-null hypercardioid microphones).

$$L = 0.418\,A + 0.724\,B + 0.194\,C - 0.112\,D$$
$$R = -0.112\,A + 0.194\,B + 0.724\,C + 0.418\,D$$
$$P = 0.612\,A + 0.612\,D$$

Table 3

Reproduction of four channels transmitted via three channels (as in Tables II and I)

$$A' = 0.739\,A + 0.306\,B - 0.127\,C + 0.306\,D$$
$$B' = 0.306\,A + 0.739\,B + 0.306\,C - 0.127\,D$$
$$C' = -0.127\,A + 0.306\,B + 0.739\,C + 0.306\,D$$
$$D' = 0.306\,A - 0.127\,B + 0.306\,C + 0.739\,D$$

Table 4

Converting four channels A, B, C, D, to three channels (in the manner of cardioid microphones).

$$L = 0.445\,A + 0.695\,B + 0.262\,C + 0.012\,D$$
$$R = 0.012\,A + 0.262\,B + 0.695\,C + 0.445\,D$$
$$P = 0.604\,A + 0.104\,B + 0.104\,C + 0.604\,D$$

Table 2 gives the matrixing that converts four channels to three assuming that our fictional microphones are hypercardioids with a null response 135° off axis, i.e. with a 15.31 dB front-to-back ratio.

The four-channel material, with signals A, B, C, and D, can be transmitted or recorded via three channels, L, R, and P by the recipe of Table 2. The signals A′, B′, C′, and D′ for the four loudspeakers can be rederived by the recipe of Table 1. Of course, something is lost in the process of reducing four channels to three. **Table 3** gives the signals A′, B′, C′, and D′ emerging from the loudspeakers in terms of the original signals A, B, C, and D, after these have been reduced to three channels L, R, P and been reconstituted according to Tables 2 and 1. It will be seen that the signal B′ (say) emerging from loudspeaker B consists mainly of the signal B, plus the signals A and C each attenuated by 7.66 dB, plus an out-of-phase crosstalk of the signal D attenuated by

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15.31 dB. However, this crosstalk should not seriously affect the directional characteristics of the reproduced sound, as satisfactory results are obtained with the much higher degree of crosstalk obtained when two-channel stereo is reproduced via four loudspeakers.[1]

Thus, while the three-channel transmission or recording of four-channel material causes little loss of directional effect, the sound does tend to spread out among the loudspeakers. The reconstituted sound may be thought of as a spatially blurred version of the original. This is illustrated by what happens to coincident microphone recordings. Suppose that A, B, C, and D were picked up by coincident hypercardioid microphones with nulls 135° off axis pointing along the broken arrows of fig. 2. Then the reproduced signals A′, B′, C′, and D′ obtained by reducing to three channels and reconstituting as in tables 2 and 1 will be the sound that would be picked up by *cardioid* microphones pointing along the broken arrows of fig. 2.

Thus a certain amount of information is lost even with coincident microphones in the process of converting from four channels to three, and back to four again. For both spaced and coincident microphone recordings, the degree of loss depends on the chosen imaginary microphone characteristic used to convert four channels to three. **Table 4** gives the reduction from four channels to three when the fictional microphones are cardioids. The reproduced channels are then as in **Table 5**, in which the degree of sound spreading on to adjacent channels is greater than in Table 3. However, crosstalk on to the opposite channel is eliminated, which is a desirable requirement with spaced microphone recordings.

Table 6 gives the reduction from four channels to three when the fictional microphones are hypercardioids with a null 120° off axis (i.e. with a 9.54 dB front-to-back ratio). Such a three channel signal will not be very suitable for reconversion to four channels by the recipe of Table 1 in many cases, as the reproduced signals will be as in Table 7, in which the crosstalk on each channel from the opposite channel is a rather excessive −9.54 dB. However, if the signals A, B, C, and D originate from a coincident microphone system, then the reproduced signals A′, B′, C′, and D′ will be the same as A, B, C, and D.

This shows that, if four channels are reduced to three, conversion using an imaginary 120°-null hypercardioid (Table 6) works best with coincident microphone recordings, conversion using a fictitious cardioid (Table 4) has desirable properties for spaced microphone recording in which no pan-potting is used, and conversion using an imaginary 135°-null hypercardioid (Table 2) is a good intermediate compromise between these conflicting requirements.

Sounds which are pan-potted exactly halfway between two adjacent speakers can be conveyed without loss via three channels, as long as the conversion to three channels uses a fictitious 120°-null hypercardioid (Table 6). For example, consider an audio signal X pan-potted halfway between speakers A and B. Then the four-channel signal is $A = 0.707X$, $B = 0.707X$, $C = 0$, $D = 0$. After conversion to three channels via Table 6, one has $L = 0.787X$, $R = -0.079X$, $P = 0.354X$. The matrixing of Table 1 reconstitutes the signals $A' = 0.707X$, $B' = 0.707X$, $C' = D' = 0$. Similarly, a signal pan-potted half-way between speakers B and C, C and D, or D and A can be transmitted via three channels by putting, respectively, $L = 0.604X$, $R = 0.604X$, $P = -0.146X$ or $L = -0.079X$, $R = 0.787X$, $P = 0.354X$ or $L = 0.104X$, $R = 0.104X$, $P = 0.854X$.

All the above indicates that reasonable quadraphonic sound can be conveyed via three channels. It is therefore worthwhile to examine the various domestic recording and transmission media to see what advantages three-channel recording might have over four-channel recording.

Take the problem of transmitting four-speaker sound via FM radio. The author has recently proposed[2] a system of broadcasting three channels which involves the use of no subcarrier frequencies not already used for stereo. This system inherently has a much better noise performance, and causes less adjacent-station interference than any four-channel FM multiplex system. In the case of FM broadcasting, significant improvements in technical quality can thus be obtained if quadraphonic sound is conveyed via only three channels.

The use of only three channels also has significant advantages in domestic tape recording. Current proposals for quadraphony involve recording four channels side-by-side on 6.25 mm (quarter inch) tape. The width of each tape track on the best four channel tape heads is only about 1 mm. This means that the outer tracks are badly affected by dropout, and the hiss level is rather high. If only three tracks had to be recorded, the track width would increase to 1.6 mm, which would improve the signal-to-noise ratio by at least 2 dB, and would dramatically reduce drop-out. The three-track format would also be compatible with half-track stereo recordings. Furthermore, the cost of reasonable quality multitrack heads is rather high, and three track heads would only cost about half as much as four-track heads of comparable quality.

Alternatively, it would be economic to record quadraphonic tapes using three tracks in each direction. The quality loss involved in this would be substantially less than when four tracks each way are used. Indeed, experience with 8-track cartridges indicates that eight tracks on 6.25 mm tape cause many severe problems, due to the difficulty of accurate track alignment. The wider track widths and guard bands possible with tapes using three channels each way should reduce these problems.

It is more difficult to see whether the use of only three channels gives any advantages with gramophone records, as one first has to consider how multi-channel gramophone records might be manufactured. To preserve the low cost of gramophone records, it is essential that any multichannel disc should be manufactured by the simple process of pressing a blob of vinyl, and this rules out adding channels by modulating the colour, the dielectric constant, the magnetisation, or other esoteric properties of the disc. Conceivably, two channels could be added to ordinary stereo by modulating the slope of each of the two groove walls, but there are numerous difficulties in designing a pickup to recover this information. In practice, it seems certain that additional channels will be added by modulating an ultrasonic subcarrier with frequency between 30 and 40 kHz.

Several companies are known to be working on multichannel discs using subcarriers, with reasonably promising results. Despite the very low amplitudes of such subcarriers (about the wavelength of light . . . it appears that a fairly low noise level can be achieved, thanks to the ability of heated record cutting styli to reduce noise at high frequencies. By recording two modulated subcarriers, each causing a direction of stylus motion 90° from that caused by the other, four channels can be carried on one disc. The crosstalk between the subcarriers will be poor, because of the poor channel separation of pickups at high frequencies, but even this can be minimised by recording the two subcarriers 90° out-of-phase with respect to one another.

Back to seventy-eights?

However, at the current 33⅓ RPM rotation rate there is one severe problem with the subcarrier method. When the record is tracked, the pickup will produce fairly large amounts of harmonic distortion above 5 kHz. As the usual stereo tracks will be recorded at a much higher level than the subcarriers can be, distortion products of the audio will interfere badly with the subcarrier modulations. This problem can be partly overcome by using higher rotation

(continued overleaf)

Table 5

Reproduction of four channels transmitted via three channels (as in tables 4 and I)

$$A' = 0.707\,A + 0.354\,B + 0.000\,C + 0.354\,D$$
$$B' = 0.354\,A + 0.707\,B + 0.354\,C + 0.000\,D$$
$$C' = 0.000\,A + 0.354\,B + 0.707\,C + 0.354\,D$$
$$D' = 0.354\,A + 0.000\,B + 0.354\,C + 0.707\,D$$

Table 6

Converting four channels A, B, C, D, to three channels (in the manner of 120° -null hypercardioid microphones).

$$L = 0.379\,A + 0.733\,B + 0.121\,C - 0.233\,D$$
$$R = -0.233\,A + 0.121\,B + 0.733\,C + 0.379\,D$$
$$P = 0.604\,A - 0.104\,B - 0.104\,C + 0.604\,D$$

Table 7

Reproduction of four channels transmitted via three channels (as in tables 6 and I)

$$A' = 0.750\,A + 0.250\,B - 0.250\,C + 0.250\,D$$
$$B' = 0.250\,A + 0.750\,B + 0.250\,C - 0.250\,D$$
$$C' = -0.250\,A + 0.250\,B + 0.750\,C + 0.250\,D$$
$$D' = 0.250\,A - 0.250\,B + 0.250\,C + 0.750\,D$$

QUADRAPHONIC RECORDING CONTINUED

rates, e.g. 45 or 78 RPM, but it is known that the distortion level in the vertical component of the stylus motion is larger than in the horizontal component. Thus the easiest way of reducing the distortion's interference with the subcarriers is to record the subcarriers horizontally only. But if this is done only one subcarrier can be recorded.

Thus the technical problems associated with using modulated carriers may mean that only three channels may be available on gramophone records. In such a case, it would again be desirable to convey quadraphony via three channels. Care will be needed to ensure that the polarity, phase and frequency response of the subcarrier channel matches that of the stereo channels, at least in the mid-frequency audio range, so that the matrixing of Table 1 can be performed accurately.

While quadraphony can be conveyed over three channels, we have seen that four channels are capable of conveying sound with less spatial spreading of the sound onto adjacent channels. What, then, is the precise nature of the additional information conveyed by four channels?

In a four-channel recording, there is one audio signal essentially independent of the three audio signals L, R, and P of Tables 2, 4 or 6, which conveys no directional information whatsoever. This is the 'focus' signal F defined by

$$F = \tfrac{1}{2}A - \tfrac{1}{2}B + \tfrac{1}{2}C - \tfrac{1}{2}D.$$

This is the only combination of the four signals A, B, C, and D which is always zero for coincident microphone recordings. For any four channel recording, given the signals L, R, P, and F, it is always possible to rederive the original signals A, B, C, and D by means of matrixing. When the signal F is suppressed (i.e. when only L, R, and P are transmitted), a reasonable facsimile of the original directional effect can still be obtained by means of the matrixing of Table 1. Thus the essential difference between three and four channel quadraphony is the addition in the latter of the 'focus' signal, which conveys no directional information, but only information about how widely a sound appearing to come from a given direction is spread out among the four loudspeakers.

The question of when a four channel recording with signals A, B, C, and D is capable of being passed through three channels without alteration has a simple answer; this can be done if and only if the focus signal is zero, and the matrixing that achieves this is that of Tables 6 and 1.

Despite the fact that three channels are sufficient for quadraphony, commercial pressures make it likely that practical quadraphonic media will in fact convey four channels. However, we have seen that 'focus' information can be dispensed with without excessive losses of directional information. This prompts the thought that, in four-channel recordings, perhaps the focus information can be discarded, and other information smuggled into its place. The next part of this article will describe how, by this means, 'conventional' (!) quadraphonic recordings can be used to reproduce height information via suitable reproducing equipment.

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1) M. A. Gerzon: 'Surround Sound—From 2-Channel Stereo!' Hi-Fi News, August.

2) M. A. Gerzon: 'QUART—A System of Multichannel FM Multiplex'. Hi-Fi News, May 1970.

VIDEO-TAPE IN BROADCASTING CONTINUED

uneventful transmission.

After transmission the tape is sent back to the library and this raises the question of storage of old programmes with its biggest attendant problem—space. Simple mental arithmetic will tell you how much room is needed to store any quantity of boxes 380 by 380 mm, and 77 mm thick. Not only that, but the shelves have to be pretty strong too—the tapes weigh around 9 kg each.

For this and other reasons the BBC has decided to limit the number of tapes that should be in store at any time, so there is a constant review of the value of recorded programmes. (Is a two year old *Z Cars* really worth keeping?) Finally, therefore, after a couple of repeats and with no possibility of further overseas sales, an executive may decide that a certain programme is no longer required and then the tortuous process of programme wiping begins; check and cross check, there *must* be no error here.

Tapes released in this manner may well go back into service for re-use. If they have been edited (a history card kept with every tape details this) it will be necessary to check how many joins there are along its length. An excessive number will destine the tape to the dustbin. If there are only half a dozen in 90 minutes, well spaced, the tape will be inspected, joins remade if necessary, and then re-issued, although it is unlikely that such a tape would be used for major colour programmes as these would always use uncut tapes. It has been found in practice that the life of a tape is determined by the number of joins it receives—they are never around long enough to wear out!

Finally, it might be of interest to readers of *Studio Sound* to look at the position of helical scan machines in broadcasting. Until now the quality of these cheaper recorders has been such that they were only suitable for industrial closed circuit applications but recent designs have a performance that is attracting the attention of broadcasting engineers. At the rate modern technology advances, it can't be too long before a helical scan machine rivals a 50.8 mm quadruplex machine in quality ; whether or not it will take over as the standard TV recorder is something we will have to wait and see. In view of the worldwide standardisation and investment in the quadruplex machine, the larger breed may well remain in command until the advent of a completely new method of vision recording. One thing seems certain—the manufacturers of tape have a very rosy future before them!

The author would like to thank the British Broadcasting Corporation for permission to publish this article.

RECORDING STUDIO TECHNIQUES CONTINUED

number of modifications on my own units in order to improve the hum level, but I understand that their latest transistorised units are greatly superior in this respect. An extremely versatile graphic equaliser unit is made by Leevers-Rich for just under £200, which also incorporates a high pass filter for removing rumble. I hope to review one of these shortly.

When considering stereo equalisation, the engineer must bear in mind that an equal amount and type of equalisation must be used in each channel, particularly when stereo pair mikes are used. If this is not done, stereo spread will wander from side to side, and can sound quite intolerable. For this reason also, it is important for the equalisation presets in the tape recorder to be adjusted carefully giving as near a flat response on each channel as possible. If it is found, however, that one channel has perhaps an uncontrollable bump at 8 kHz, it is best for the other channel to have an equal bump. Otherwise, when the tape recording is played back on another machine, there would be a swing of stereo image at high frequencies.

It may appear from the above comments that equalisation is in use all the time on almost everything but this is most certainly not so. Engineers must remember that in real life we hear most sounds 'flat' and anything with attenuated top sounds that way because of its distance. Depth should always be present on a recording in addition to width and this can be ruined by over-equalisation. This tends to happen on many American recordings. Some of the cleanest sound I have heard has been recorded from the output of a single stereo capacitor pair without any equalisation, giving startling realism. This technique, however, usually only applies when the acoustics of the studio or concert hall are excellent. Equalisation should not be regarded as the be-all-and-end-all but as a cure for deficiencies of one kind or another, and as a spotlight for a particular instrument. There is nothing more horrible than a dramatically over-equalised highly limited sound which bashes the eardrums continually, On the otherhand, perhaps that is the type of sound that the teenagers now demand, alas.

Let me again remind readers that what may sound remarkable on a stereo master tape may be impossible for disc. The engineer should realise that he is then at the mercy of the cutting engineer who may change the perspective of the master, quite apart from the inherent changes due to the new medium. Often the poor cutting engineer gets the blame, but more often this should be laid on the recording engineer responsible for the original balance or reduction. Since the advent of the Dolby *A301*, lower tape peak recording levels are in general use, allowing the tape to handle higher amplitudes at high frequencies in relation to low ones. Caution please; remember the cutting stylus has a limited HF performance, based on the angles between the faces of the cutter.

SOUND balancing

PART FOUR
LIMITERS AND COMPRESSORS

BY BOB AUGER

WE now turn to the question of controlling dynamic range, In the classical field, the practice of tampering with the composer's dynamic markings by either manually or electronically interfering with the sound level is generally frowned upon and this accounts for the very wide range of many of the classical records available. The introduction during the last few years of low-noise transistor equipment into the mixing chain, together with such devices as the Dolby system, have allowed the recording engineer to cope with extremely wide dynamic range and it is no problem today to accept the loudest fortissimo passages down to the quietest pianissimo which can be produced by an orchestra, without the engineer having to adjust the main gain control feeding the signal to the tape recorder. Considerable experience and skill, however, is required to get the best from modern disc-cutting lathes, and the lack of such skilled personnel results in many a disc becoming a travesty of the original tape.

I have digressed into the disc field for the moment in order to illustrate why some of the major companies introduce various forms of volume compression at the time of cutting the disc. Squashing at a higher average level allows the music to be heard clearly both through an indifferent pressing and also on poor reproducing equipment.

It is the realisation of these kind of problems that has led the pop side of the industry into developing highly compressed or limited discs which are guaranteed to have a dynamic range of at least 2 dB from beginning to end and furthermore are cut at the maximum possible level so that competitors will not shout them down when played on a juke box. No balance engineer can be considered fully worth his salt today (commercially speaking at least!) unless he has a good working knowledge of the various types of limiters and compressors which are available and the experience which enables him to use them to their best effect.

Although every make of compressor or limiter has its own inherent characteristics, the following is a brief guide to some of the more general uses for these pieces of apparatus.

The Limiter

The main purpose of the limiter is to 'chop' volume peaks and a typical use of this device in classical music recording would be to reduce the volume of one timpani roll in an otherwise mezzo-forte recording. In this instance it can be seen that, by holding down one otherwise troublesome peak, the average level of the whole recording on tape or disc can be raised slightly with obvious benefits in the signal-to-noise ratio. The use of the limiter in classical music is somewhat rare and usually only a last resort in the kind of example outlined above. However, limiters are sometimes used in vocal microphone circuits when operas are being recorded, since troublesome peaks are encountered particularly in the soprano register of the human voice. Once again, all attempts to control the volume by natural means such as careful microphone placements in relation to the singer, and just as careful placing of the singer in the studio, should be tried before resorting to electronic means.

In popular music, limiters are widely used not only in individual microphone circuits but also to control the overall level of a group of microphone channels and, inevitably, in the final mix of multi-track tapes down to mono and stereo masters. It is usual practice to limit the overall mixed sound to a very narrow dynamic range.

The various types of limiters available to the professional engineer have the following input and output level controls usually adjusted for unity gain when the limiting section of the amplifier is not in operation. A *threshold* control sets the point at which the limiter commences its operation. As the control is brought further into use, the level at which the limiter operates increases, usually by 2 dB steps. After the threshold has been set to decrease the volume peaks by the correct amount, it may be necessary to increase the output level of the limiter in order to set the overall programme level to its new higher average. A *release time* control, usually calibrated in fractions up to approximately 1.25 seconds, controls the amount of time the limiter takes in restoring the programme level to normal following a high volume peak which has been sensed at the threshold section of the circuit. In normal use, the release time is set to recover quite quickly, something in the order of 200 to 300 mS.

Various types of meters are installed by different limiter manufacturers, the most common being some form of gain reduction meter indicating the amount of peak 'chopping' being carried out by the limiter at any moment. In the case of the heavy timpani roll mentioned earlier, the meter would only read the amount of gain reduction of this particular roll throughout the whole of the performance of the piece of music, at all other times during the performance the meter would read zero.

The Compressor

The operation of this piece of equipment differs to the limiter in that not only is the peak level of the programme monitored and controlled, but the lowest levels of the programme are brought up to a preset minimum. This is rarely encountered in the recording and broadcasting of classical music but has its place in the production of light and popular music, A typical use of the compressor would be to enhance one section of a light music ensemble. Feeding all the microphone channels on the rhythm section of the ensemble (drums, bass, guitar) into a single compressor results in a smooth rhythm sound with each instrument clearly discernible. If the same rhythm section is not compressed, the various instruments would probably mask each other from moment to moment. If a compressor is used at the output of the mixing desk to control the dynamic range of the entire programme, it can often help to produce a 'glossy' sound which seems to emanate from an area larger than the loudspeaker enclosure. When two compressors are used in stereo working, the apparently widened sound image can be most impressive.

One point to be borne in mind when working in stereo is that the two compressors should be linked electronically so that a high peak in one channel will trigger both compressors simultaneously. Failure to observe this method of working will shift any central instrument over to the opposite side to that which has received the momentary peak.

A selection of ratios will be found on most compressors, typically 30:15, and 10:1. 30:15 indicates that a 30 dB variation in input level would be restricted to 15 dB variation in output. When the most severe 10:1 setting is chosen, the threshold is set so acutely as to produce virtually a limiter/compressor.

When the compressor is used on the human voice, the resultant sharp wave-front produced in the circuit often results in an audible sibilance. The *de-essing* control introduces a steep trough in the higher middle frequencies to combat this problem. Increasing the de-ess setting produces a deeper trough in the frequency response, the control being so adjusted that only the sibilance is reduced in level. A meter similar to that in a limiter indicates the amount of overall compression which is being applied to the programme.

The judicious use of limiters and compressors is probably one of the professional engineer's greatest skills. To control both the individual volume levels and overall programme levels in such a manner as to satisfy both the musicians and other engineers who have to work with the tapes is no mean feat. Perhaps the greatest praise an engineer can receive from a knowledgeable member of the public, or indeed an artist, is that his recordings always sound gimmick-free when in fact a considerable amount of electronic equipment has been utilised in the production of the work.

A PROFESSIONAL HOME RECORDING STUDIO

By Arthur Garratt

AS a broadcaster I have found it essential to have facilities for recording with professional quality both on location and in a studio. This is in addition to the editing and other ancillary requirements which we will come to in due course.

There are certain important features of a professional's equipment. The studio must be available at any time to make a high-quality recording—it is out of the question to connect up apparatus every time you want to use it and then break it down again until another recording session is on top of you; this means a permanent set-up. Then every recording made must satisfy certain quite stringent conditions. The recording levels must line up within a decibel, noise and distortion must meet the requirements of FM broadcasting, the frequency characteristics must be flat from 50 Hz to at least 15 kHz, and the ambient noise of the studio must be inaudible. Additional desirable features include the ability to play in both tape and disc inserts, some form of cueing and talkback between the control cubicle and the studio and means of listening to inserts on headphones (or 'cans' if you prefer).

These requirements demand good equipment and regular maintenance. Unfortunately not all the equipment needed is available off the shelf; a substantial amount of it has to be designed and built for the job. Of course, there are specialist firms who can do this, but of necessity their prices are uneconomic if it is possible to do the work for oneself.

Before I go on to describe my own studio, let me explain what I use it for. Nearly all my work is for broadcasting, although I occasionally record tapes for visual aids and copy tapes for special purposes. I do very little music recording. In addition to making tapes for the BBC, I record a regular weekly programme for the Voice of Kenya, interview authors for American radio networks and prepare features for the Australian Broadcasting Commission. Most of my recordings are of a scientific or technical nature, but sometimes I break away and interview someone like Twiggy.

Well, that's a bit of background, now to the equipment and set-up. I started with a small Fi-Cord *1A* for field work. This was an excellent machine but with certain obvious disadvantages, such as small spool size and a level indicator that left much to be desired. My second acquisition was a Ferrograph *4AN*. This has been a virtually trouble-free stalwart. I use it for most of my editing, for copying and for playing in tape inserts for features. I then bought a second-hand stereo Ferrograph *88* which is used almost entirely for editing at 38 cm/s. Next came a Uher *4000-S*. This machine has flown nearly half a million miles with me. It has had one new motor, otherwise I have done nothing to it except replace a drive belt (of course it broke in Paris when I was on a job!) and keep the starting switch contacts liberally covered with Electrolube.

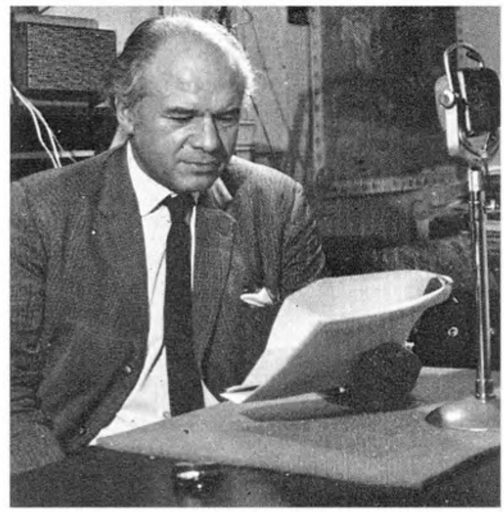
I still needed a professional studio machine. For one thing I wanted to record full-track to meet the broadcasting organisations' needs, for another I wanted off-tape monitoring so that my studio manager (my wife) could check recording quality during the actual performance. So I acquired an EMI *TR51* in very good condition. With certain modifications, this has done me proud for several years.

My set-up at home is to use my study as a studio and the adjoining room, once a butler's pantry, as the control cubicle. Now it is not feasible to have a fully insulated studio. This means separate flooring, no windows, quiet air-conditioning and all the other things that only broadcasting organisations or full-scale disc recording companies can afford. So I have gone about half way. The room has double doors, the inner one padded, double glazing, and an acoustically treated ceiling and walls—where there aren't masses of books which do a pretty good job of absorbing sound. There is really only one thing I can't keep out, the roar of a low flying jet. As I live some ten miles from Heathrow, I like to do any long recording sessions at night!

In the studio I have a cue-light and talkback speaker and a jack point for cans. I use a microphone suspended in rubber on a desk mounting. To avoid bounce-back from the desk (most people ignore this quite serious factor) I cover it with a piece of expanded nylon foam, designed to put on kitchen draining boards.

In the sound cubicle are the two Ferrographs, the *TR51* and a disc player side by side, two on a table and two on a Lektrokit trolley, the under part of which carries amplifiers, a tone generator, a PPM, a mixer and an FM tuner. One side of the room has a large rack carrying test gear and components. There is also a table for servicing and construction.

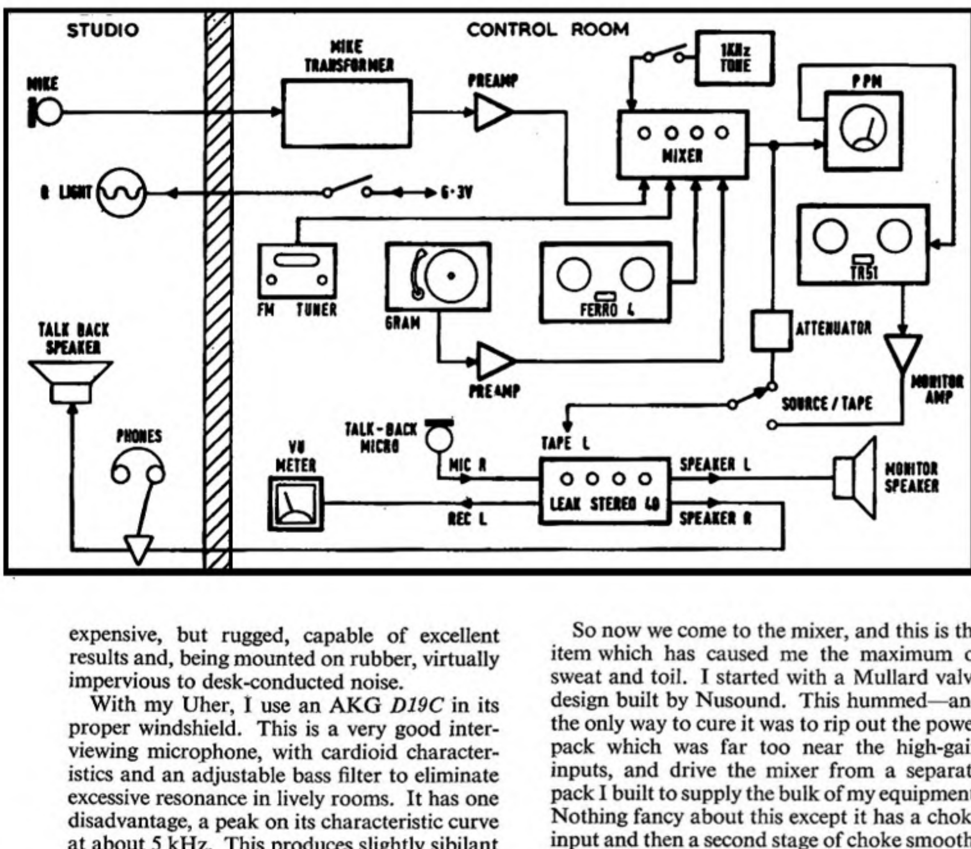
Now to the equipment in more detail, tracing it through the chain from voice to tape. The first link is obviously the microphone. Here a word of warning: don't pick a microphone on its published characteristics. No microphone is perfect, at least at any price I can afford, so the selection is a compromise. For my studio I have settled on an AKG *D25* moving-coil cardioid. For some years I used a Reslo ribbon though this had several disadvantages including low output and limited HF response. The major problem was not due to the microphone itself but to its bi-polar characteristic. My studio is small, so I have the problem of standing waves—Eigentones if you prefer. The best way to reduce these is to use a cardioid. With its suppressed rear sensitivity, most of the standing wave trouble disappears and I no longer need a vicious bass-cut to make male speech sound realistic. The *D25* is fairly



The author in his studio. Note talkback speaker and desk foam. Cue light and phones jack are on panel behind the *D25* microphone.

EMI jointing block mounted to operate Ferrograph pause control.

A PROFESSIONAL HOME RECORDING STUDIO



expensive, but rugged, capable of excellent results and, being mounted on rubber, virtually impervious to desk-conducted noise.

With my Uher, I use an AKG *D19C* in its proper windshield. This is a very good interviewing microphone, with cardioid characteristics and an adjustable bass filter to eliminate excessive resonance in lively rooms. It has one disadvantage, a peak on its characteristic curve at about 5 kHz. This produces slightly sibilant speech, which can be an advantage in noisy conditions such as the flight-deck of an airliner, as it improves intelligibility. I cannot speak too highly of the AKG matching windshield, which is superb—the only snag is the price!

The next link is the microphone matching transformer. Both my ribbon and the *D25* are 30 ohm microphones and I have not succeeded in matching these direct to transistors; more about that later. I don't believe in having an expensive microphone and then using a cheap transformer, so I use a Gardner professional model with a switch so that I can offer it either 30 ohms or 300 ohms. The transformer, which fits an international octal base, is enclosed in a die-cast box—how useful those die-cast boxes are—mounted on to the side panel of the Lektrokit trolley. Incidentally, although I have put the transformer well away from AC fields, it is so superbly shielded that a de-fluxer held alongside it doesn't seem to worry it at all.

So now we come to the mixer, and this is the item which has caused me the maximum of sweat and toil. I started with a Mullard valve design built by Nusound. This hummed—and the only way to cure it was to rip out the power pack which was far too near the high-gain inputs, and drive the mixer from a separate pack I built to supply the bulk of my equipment. Nothing fancy about this except it has a choke input and then a second stage of choke smoothing, so the ripple from it is negligible and the regulation very good. But the mixer still didn't meet my very stringent demands. There was still a trace of hum which disappeared when I put some extra screening round the grid inputs to the two high-gain stages. The next problem was noise. This wasn't bad, about -50 dB, but still not good enough. The original Mullard design used leaky-grid bias, obviously to cope with various input levels. This came out, wire-wound grid and anode resistors went in and the noise dropped to about -58 dB. But I could still hear it, so I moved into the world of transistors. This started with the BBC microphone amplifier, designed in 1963 and using germanium transistors. The first stage was a *GET 106*, followed by two *GET 104*, all with massive feedback. (The circuit is given in BBC Engineering Monograph, No 46. This is well worth having because it contains other useful circuits including the BBC PPM meter that I'll be talking about later.) This little preamplifier caused me more trouble than any other piece of equipment I have ever built. I spent weeks trying to get rid of flicker noise. I changed all the transistors, resistors and capacitors half a dozen times and spent half my time prowling round shops trying to get *GET 106* which are obsolescent. Eventually I gave it up, convinced that transistors just weren't on, despite the fact that the BBC panels using the amplifier were perfectly satisfactory. I think the answer is that the BBC select their *GET 106* very carefully; my six samples either wasn't big enough or it wasn't representative!

Then came those very good articles by Frank Jones in *Hi-Fi News*, and I started all over again. I bought an integrated circuit, which just didn't meet its noise specs and which was a devil to keep stable. Then, almost in desperation, I built up the microphone amplifier in the recent Mullard book, *Transistor Audio and Radio Cicruits*. The first *BC108* was noisy—maybe I'm fated—but the second one was quiet and the amplifier is excellent. I use it feeding the old Mullard valve circuit for convenience, with about 20 dB of gain—all my troubles are over, noise and hum are quite undetectable and well below studio ambience.

I decided to standardise all my high level inputs and outputs at 600 ohms. This meant redesigning the rear end of the mixer with a beefy cathode follower which didn't clip when fed direct into 600 ohms, but this didn't present too much trouble. I have also built a cathode follower output into my Ferrograph *4AN*, fed from Jack 2, again with enough beef to look straight at 600 ohms. One problem here was the coupling capacitors, I needed 18 μF for each output. Electrolytics just aren't on because of leakage, so I have incorporated some large paper capacitors into the Ferrograph and the mixer output line.

There is one great advantage in having low impedance circuits, in addition to less hum pick-up. This is that one doesn't have to take care about the self-capacitance of connecting lines. Those useful prefabricated links you can buy are diabolical if used with high impedance circuits and I often wonder how many treble decibels disappear from amateur equipment because they are used. At 600 ohms, of course, you can laugh at a few puffs of self-capacitance and the prefabricated links come into their own.

The next problem was a good level meter. The *TR51* has a VU meter and let me say straight away that, in the teeth of all the sound-film boys and the whole of American radio, I think these are virtually useless. What one really needs to know for tape is one thing and one thing only—peak recording level. Matching levels of speech and music can only be done by ear anyhow, so one needs a means of preventing overmodulation. A secondary requirement is to have a meter which will let you record a piece on Monday and an extra sentence on Friday

(continued on page 347)

which can be cut in with no level change. And this demands a proper PPM; it also needs a logarithmic scale if most of the programme modulation is too low to read. So I built the BBC *ME12/5* PPM and, despite spending nearly £10 on the meter itself, I have never regretted it. On the side I have four jack sockets so that I can use any input or output I like and parallel them if necessary. The PPM feeds directly into the 600 ohm line input of the *TR51*, or into the Ferrograph *4AN* or the Uher. All these have been lined up on test tone and the gain controls marked to give correct tape modulation. At any time, I can record or copy on any of the three machines and I know my levels are perfect.

I also have a small transistorised 1 kHz tone generator which is set at 8 dB below peak. Any tape which I am going to copy has a leader with 1 kHz tone at this level recorded on it, so I can set levels and copy without checking the actual mod on the tape.

The *TR51* has a monitoring head and a curious circuit which produces a travesty of the original out of the speaker while recording. I decided the simplest way out of this one was to purchase a *TR52* monitoring amplifier. I have added a switch so that I can go from 'direct' to 'monitor' immediately. I have built a cheap and nasty level meter on my monitoring pre-amplifier output so that I can line up on tone and ensure that 'direct' and 'monitor' are at the same level.

Well, this is the recording chain, but there is another chain as well, an amplifier for aural monitoring and a suitable speaker, The amplifier I use is a Leak *Stereo 30* feeding a pair of *Paralines*. During recording, I push in a jack which squelches the righthand speaker and feeds the speaker in the studio instead. With the amplifier set to stereo and the input from the studio into the left channel, nothing, of course, comes out of the studio speaker. But here comes the cunning part. In the control cubicle a microphone is connected to the Leak microphone *right* input. So to talk back to the studio, all that is necessary is to turn the function switch to 'microphone'. This squelches the studio output on the cubicle speaker and automatically avoids howl-round.

The microphone used for talkback is a Grampian *DP4*, far too good for the job. but it came with the Fi-Cord and is a good emergency stand-by in case anything happens to the AKG *D19C* when I have to do a job on the Uher (a professional has to have belt, braces and a bit of string!). The talkback speaker is a Goodman *Maxim*; I need a good speaker here in case I want to monitor inserts in the studio.

If I want to play in discs, I use a Deccadec with a Deram pickup. This feeds into a 2 M load and is much better than some people will ever believe!

Well, that is the recording and playback set-up. The results are very good, quite acceptable to the BBC, who have relatively high standards. Needless to say it only works because it is well maintained, and I have a lot of supporting equipment for this purpose.

I can't overstress the importance of regular maintenance. It is only too easy to have a gradual falling off in performance which goes undetected—only regular checks *using instruments* can avoid this. The ear, coupled with the memory just isn't good enough.

The basic equipment needed for regular testing is fairly straight-forward. First one needs proper test-tapes. I use two CCIR test-tapes, one at 19 and the other 38 cm/s, both made by EMI, and one NARTB tape recorded at 19 cm/s, made by Ampex. I bought the Ampex tape in New York at a price so elevated that even the man who sold it to me was shaken! But these tapes are individually made and must cost money. Then you need a means of measuring output, ideally a valve voltmeter and an oscilloscope to ensure that the waveform is correct and you are not measuring noise or hum. This is the basic equipment. Also an audio oscillator to line up the record channel. Additional essential equipment is a head defluxer and, if possible, a bulk eraser so that one can produce a clean tape for noise tests.

Test procedure is simple. First you clean and deflux all the heads. I use a *Bib* cleaning kit which works very well. In fact I clean and deflux at least once a week, whether I am doing full maintenance or not. After this, run the test tape and check the replay head alignment. If you are selling tapes, the head alignment must be spot-on. If you have a separate record head and facilities for downstream monitoring, the record head can be aligned at about 8 kHz. When all is well with both heads, move on to the replay equalisation, then the bias and record equalisation. This is where those excellent service articles by H. W. Hellyer are so valuable. When he writes about one of your machines, cut out the article and paste it on the wall.

The next step is to check whether the erase heads are functioning properly. Record a sinewave at about 1 kHz, peak level, erase with the gain at zero, and then play back with the replay gain right up. This gives a qualitative check; you can of course use the valve voltmeter or oscilloscope to get quantitative readings, but this is one test that the ear can do for you.

If you change the make of tape you are using, you must set up the bias and record equalisation all over again or you may have increased noise and distortion.

The usual way of finding the optimum bias level is to record a sinewave at about 1 kHz with various levels of bias. The output rises with increasing bias to a peak and then starts to drop again. The correct bias setting is when it has dropped 1 or 2 dB below peak. If you have a separate playback head, bias setting is simple and quick. Put in tone, at a level about 8 or 10 dB below peak, set the bias to zero, start to record, gradually increase the bias, and watch the output until it is just over the hump. If you have only one machine with a common record/replay head, you will have to calibrate the bias control, or measure the actual bias with a valve voltmeter or oscilloscope, and record a series of tone bursts at different bias levels. On playback choose the bias level which is just over the hump. If you have two machines, you can record on one and play back on the other, leaving a reasonable loop between the two.

The Uher *4000* can't be set up this way. Here the trick is to record a tone at 1 kHz and at 10 kHz and set the bias to give the same output on each, making sure your playback equalisers are correct for DIN recording, which the Uher ones are not!

These are the basic checks. One other check is for wow and flutter. To do this properly you need a wow and flutter meter; prices start at about £50. Luckily this is one test an experienced ear can do very well. Record 1 kHz tone and play it back. Wow is very obvious to the ear while flutter gives a roughness to the tone, seen on the oscilloscope as a series of spikes on the trace. Serious wow and flutter on a respectable machine can be cured. Check that the capstan and guides are clean, pressure pads in good order, the tensioning spring on the pinch wheel is correct, and so on. Ferrographs will wow appallingly if the flywheel bearing needs lubrication, so check this if things sound really bad. Some machines, like the *TR51*, wow towards the end of the reel. Here the answer is to use the largest reel the machine will carry and don't record to the end of it.

By the way, if you haven't an oscillator, you can record 440 Hz tone from the BBC test tone transmissions on Wednesdays and Saturdays at 23.30 on Radio Three. Played back at double speed, this works well as a qualitative wow test.

Now recording levels. Here you can probably rely on your machine to set a figure. I use BBC levels, having got a friend to record some 1 kHz tone at the standard level of 8 dB below peak. This ensures that my recordings are at standard BBC level which has been chosen so that different makes of tape can be used without bias adjustment. Not all makes, of course, but most of the better known brands will work at one bias setting giving constant output so that tapes can be mixed without audible level changes.

To end, a few tricks which might be useful. As I said, I use my Ferrograph *4AN* to play tapes in for features. One problem here is to avoid an audible click on starting the Ferrograph. This I have decided is unavoidable unless the mixer fader is at zero and then

(*continued on page 359*)

A HIGH QUALITY MIXER

Part Three—Input Amplifiers

David Robinson

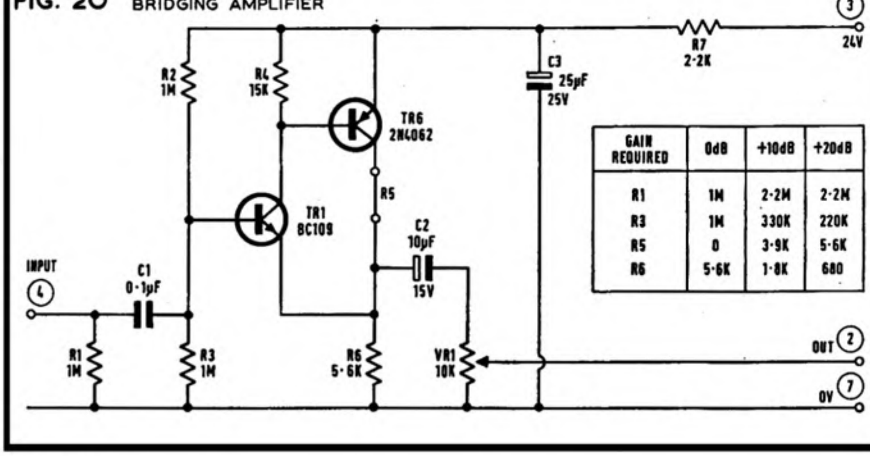
Fig. 19

PARTS One and Two of this series described the basic idea behind this portable mixer, and the metalwork required for a typical realisation. Part 3 discusses the signal requirements of the input stages, and describes a few suitable stages.

In the interests of economy, plug-in amplifiers were chosen to form the input stages. Every module thus fulfils one function, and the complication of a series of multipurpose input stages is avoided. Inputs can be split into high and low level (flat or equalised) signals; each requires different treatment.

Before becoming involved in the detailed circuitry, the module itself (**fig. 18**) should be described. The construction is very simple, and requires the minimum of tools. All are identical

FIG. 20 BRIDGING AMPLIFIER



GAIN REQUIRED	0dB	+10dB	+20dB
R1	1M	2·2M	2·2M
R3	1M	330K	220K
R5	0	3·9K	5·6K
R6	5·6K	1·8K	680

FIG. 21 LINE AMPLIFIER

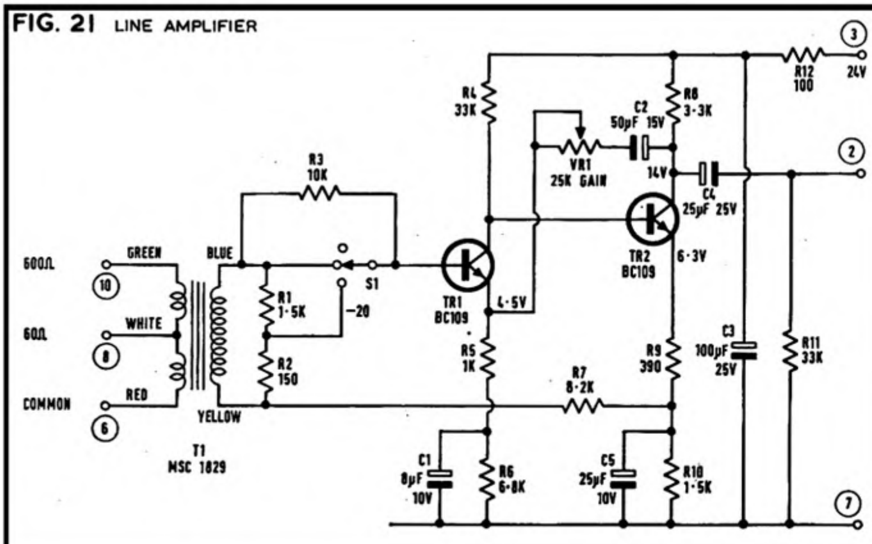


FIG.18 CONSTRUCTION OF PLUG-IN MODULE

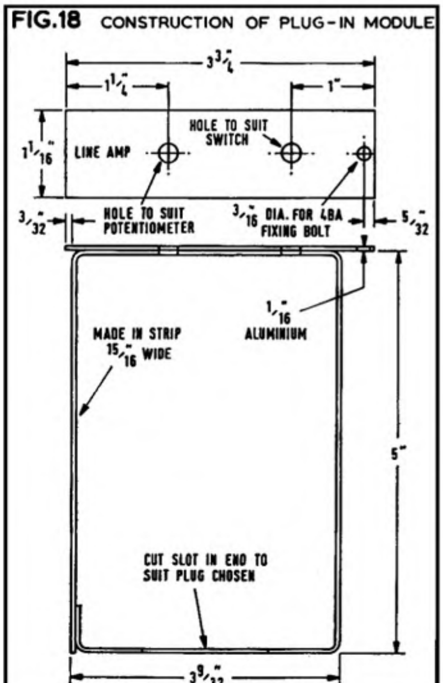


FIG. 22 MICROPHONE AMPLIFIER (BASED ON A BBC DESIGN)

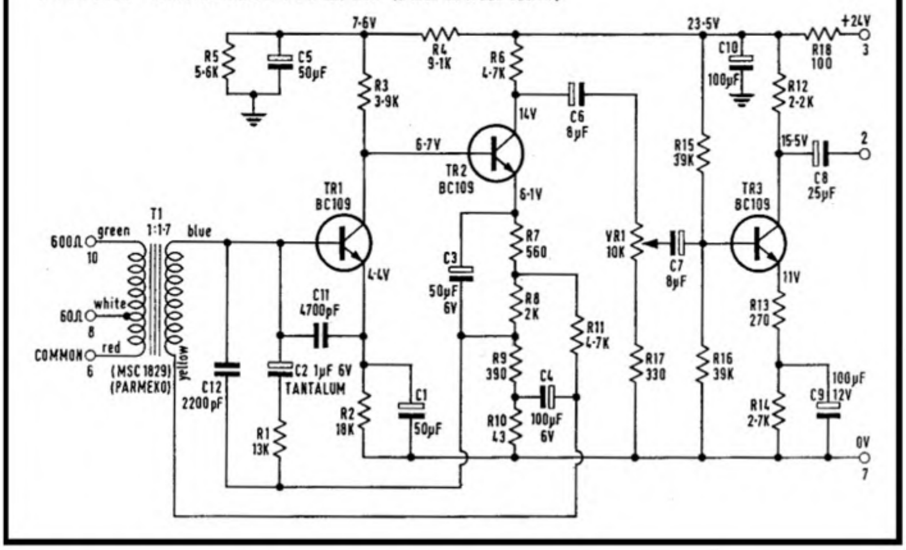


FIG. 23 MAGNETIC CARTRIDGE INPUT STAGE

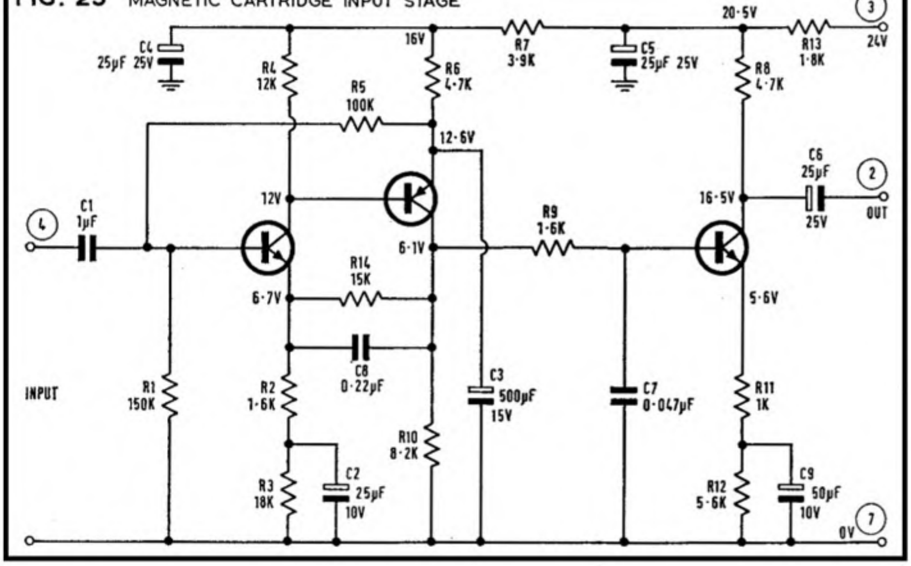
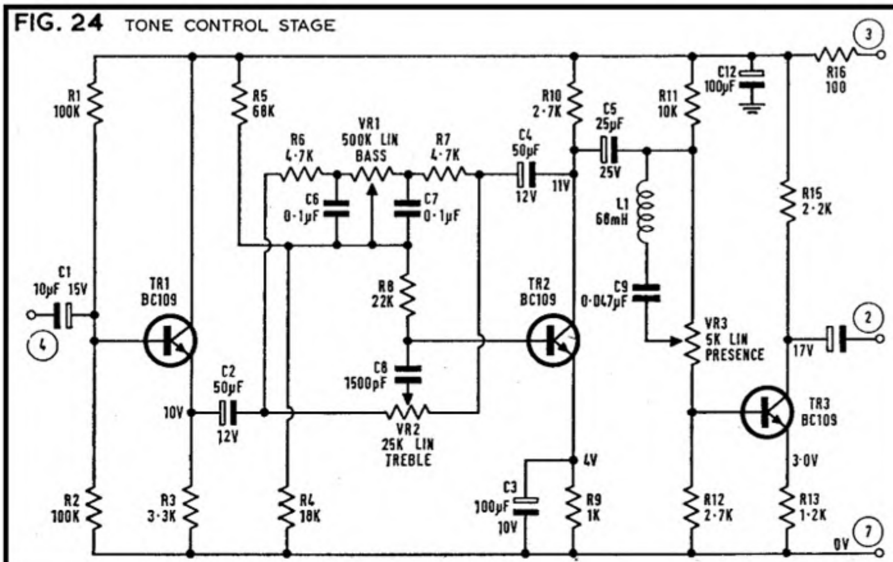


FIG. 24 TONE CONTROL STAGE



in overall dimensions, and differ externally only in the front panel.

The units are made from one strip 15/16 inch wide, which is bolted to the 1 1/16 inch front panel. The rear is cut to take the multiway plug chosen. Details are given for the line amplifier, and in this case the front panel can be held neatly to the chassis by the fixing nuts of the two controls. Units with one or no controls will use nuts and bolts for this purpose Some care should be taken on the construction of the chassis, since the ease with which the units can be plugged in and out will depend on the accuracy.

The actual components are mounted on printed circuits which are held in position by bolting to the bottom of four aluminium pillars which are in turn fastened to the chassis frame. A metal plate is then bolted to the top of the pillars to provide shielding between units, and to protect the components when the unit is removed from the mixer. The photograph (**fig. 19**) shows a completed unit with this metal top-plate removed, in this case a mike amp.

The simplest amplifier is the bridging amplifier, which accepts a signal already fairly high in level and merely acts as a buffer. It should have a high input impedence to avoid loading the signal source, and a low output impedence so that the ouput is immune to hum pick-up or interference from other signals in the interior of the mixer. **Fig. 20** shows a suitable circuit. The input impedance is 330 K, (if R1, 2 and 3 are all made 5.5 M, then it is over 1 M and the circuit can be used for crystal or ceramic cartridges). The circuit can also be modified if necessary to provide gain, determined by R5 and R6. Varying the ratio of these will alter the gain. However, the biasing of Tr1 must also be altered to keep Tr2 collector at about half the supply voltage, which then gives the maximum output capability.

The second circuit requirement was for an amplifier which would accept an 0 dBm input signal to match a programme line or standard sound distribution system. The input must be balanced and isolated from earth, which means a transformer has to be used. If the amplifier can be designed to accept a slightly smaller signal as well, it can be used for high-output microphones such as the capacitor variety which usually have a built-in preamplifier to match the capsule and to raise the very low output signal-level to something more manageable. **Fig. 21** shows the feedback amplifier that was designed to accept a signal of 0 dBm without distortion in the maximum gain position, and much higher in the normal or line position.

The input is balanced to earth by the capacitance in the windings; the input sockets are arranged so that one side is automatically earthed if an unbalanced line is used. The transformer secondary passes direct to the base of the first transistor and is amplified in the first stage which has some local feedback from the undecoupled part of the emitter resistor. The first transistor should be of the low-noise type, such as the *BC109*, for best results. Tr2 forms another amplifying stage, again with some local feedback, The junction of R10 and 9 is a suitable point from which to stabilise the DC conditions, and C5 is added to decouple the chain.

The AC output is taken from the collector of

(continued overleaf)

the second transistor and at the same time variable feedback is applied to the first stage by C2 and VR1, which raises the input impedence so that the transformer is correctly loaded by R1 only. VR1 appears as a front panel control, and is used to present roughly the amount of gain required so that the quadrant faders, which are the main controls per channel, are all approximately in the same position; this makes for easier mixing.

The most important amplifier in the complete mixer is undoubtedly that for the microphone. This has to accept in one extreme a very low signal from a quiet passage in a recording, and provide an amplified signal without significantly increasing the noise level; at other times it must be capable of accepting a large signal without over-loading. The design of such a stage is difficult and must not be undertaken without much time for experiment. The circuit shown in **fig. 22** is one solution, which has been developed from a circuit of the Designs Department of the BBC; the author is indebted to the Corporation for their permission to publish the circuit and its details.

The original circuit is fully described, together with the theoretical and practical considerations which led to the final design, in the BBC Engineering Monograph No. 46 (*The Application of Transistors to Sound Broadcasting*), which also includes several other interesting circuits; this is a most useful publication for the audio engineer.

The noise figure for the amplifier is typically 3 dB, which means that the noise at the output terminals is 3 dB greater than, or about 1.4 times, the noise from the thermal movement of electrons (Johnson noise) in a resistor of the same value as the input impedance, followed by a noiseless amplifier of the same gain. Put another way, this is an excellent amplifier. For a 600 ohm resistor, the inherent noise is -127 dB, or about 500 μV; the amplifier makes this appear as -124 dB, which means that extremely small signals from microphones can be handled successfully.

At the other end of the scale, an input level of -25 dB can be accepted before the onset of serious distortion, and this represents a very loud noise for most microphones, almost at the threshold of pain. For louder situations a pad can be used on the input, or the line amplifier used, if by then the microphone output itself is still undistorted.

The amplifier input is transformer-fed to provide both a balanced input and also facilities for 600 or 60 ohm microphones, this covers all commonly encountered types. Two transistors are used in a feedback amplifier with both DC and AC feedback for stability and to determine correctly the input impedance which is essential for low-noise operation. The first transistor is set at the optimum collector-emitter voltage and collector current, and is DC-coupled into the next stage. The emitter of the second transistor is in a long chain of resistances which are used in the feedback circuits. R11 is in the DC path, and the AC feedback is both shunt (R1, C2) and series (C4). The transformer is included in the loop; it is a Parmeko *MSC 1829*, the same type as is used in the line amplifier described earlier. C11 and 12 form an RF filter to prevent

FIG. 25 FREQUENCY RESPONSES TONE CONTROL

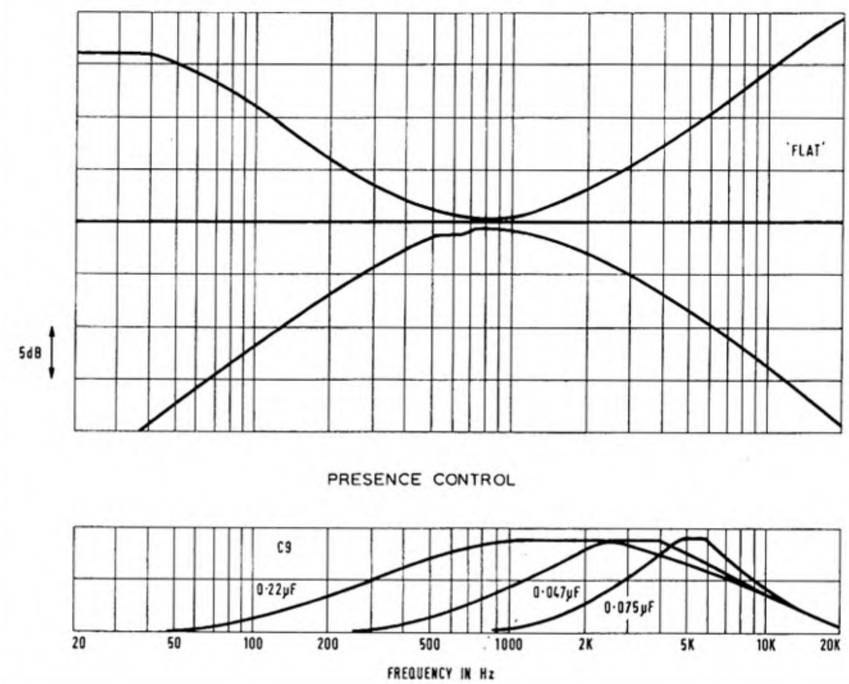
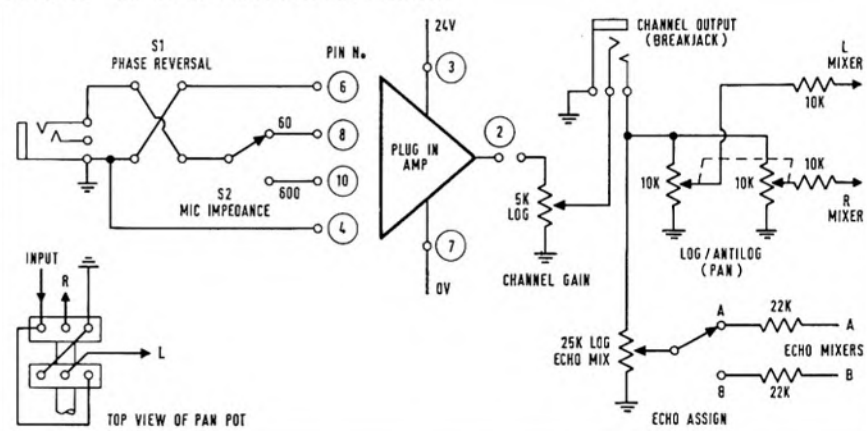


FIG. 26 WIRING DIAGRAM FOR INPUT STAGES



radio interference from being picked up and passed to later stages.

The output from this stage passes to a variable 30 dB attenuator mounted on the front panel, and then to an output stage which has a controlled gain of some 17 dB with feedback from the undecoupled part of the emitter resistance. The output is taken from the collector and passes eventually to the channel fader in the rack at the front of the mixer.

Other input amplifiers described here are magnetic cartridge input, and a tone control stage. These are shown in **figs. 23** and **24** and need very little description.

Fig. 26 shows the wiring of the modules in detail, and is derived from **fig. 3** in the first part of this series. Next month's article will describe the remaining input stage preamplifiers, mixing and output amplifiers, together with wiring details for these stages.

COMPONENT SUPPLIERS

Audio transformer *MSC 1829:* Parmeko Limited, Barking Division, 30 Thames Road, Barking, Essex —about 30s each.

Printed Circuit Cards—from the author, c/o *Studio Sound*

Ref. 214	Bridging amplifier	5s
Ref. 113/2	Line amplifier	7s
Ref. 114/4	Microphone amplifier	7s
Ref. 166/2	Magnetic pickup input	7s
Ref. 149/2	Tone control stage	7s

Coil LI 68 mH 430 turns of 38 SWG wire, on 18 mm core

Mullard Parts: core LA 2500, bobbin DT 2178, board DT 2221, clip DT 2222 } total cost 9s 1d plus 3s p.p.

Sasco Limited, P.O. Box 20, Gatwick Road, Crawley, Sussex

Preh faders (see last month)
G.E. Electronics (London) Ltd., Eardley House, 182 Campden Hill Road, London W.8

tape recorder service

FERROGRAPH 713

By H. W. Hellyer

LIKE the good fairy, if not quite so pretty, Stanley Kelly comes to my aid with a long and detailed description of the Ferrograph *Series 7* and lifts the most onerous part of these servicing articles from my shoulders. Interested parties are referred to the review on pages 215 to 219 of the May *Studio Sound*.

Early in his article, Mr Kelly mentions that the layout is logical and maintenance should present no difficulties. It's a funny thing about logic—somehow its very precision can defeat its own object. We all know that, despite the claims, tape recorders are not really built with servicing in mind. That would imply that servicing was occasionally needed, wouldn't it?

So let's begin with the points that Mr Kelly had no space to mention. First, dismantling. If access to the electronics alone is required, it is better to leave all face plates on and merely remove the cabinet from the chassis.

'Merely'—that's a laugh. The first time one disrobes a *7* can be the occasion for much sweat and some tears. After that it is easy, like riding a bike. Before undoing any screws, make sure there are no external connections to the panel at the top (machine standing vertically). Then remove the two fuses. Don't worry about mixing them up: they are both 1 A, size *00*. After this, get the handle down below the level of the cabinet cutout. The easiest way of doing this is to raise it to the fullest extent then push one side down first, allowing the wire slides to carry the whole handle sideways so that the bolt-heads which secure the cheek-pieces of the aperture are cleared by the descending end of the handle. Then the other end will slide down and under fairly easily. I make it a practice at this point to tie a loop of thin cord around the handle. This facilitates reassembly, when the process of getting the handle back out of its cosy recess has been the cause of all those aforementioned scratches.

The cabinet can be removed from the chassis in the time-honoured way by laying the machine on its face on a soft pad—vital for 23 kg or so of machinery. As it is often necessary to disconnect loudspeaker plugs and the earth link to the cabinet lining (neither of which is easy to perform with the Ferrograph lying on its face), the best approach is to 'rock' the chassis from the cabinet, with the machine in its vertical position.

Removal of the face plates for complete access to the machinery (the view of **fig. 3**) can only be done with the correct Allen keys. These are 3, 4 and 6 BA sizes and a set should be supplied with the recorder.

First step is to remove the knobs: on earlier models, the on-off knob could be left. In most cases this knob comes off, as does the speed-change. After this, the four very tight screws holding the upper plate are removed and the plate eased off, top end first. It is sometimes simpler at this stage, if doing a full dismantling exercise, to loosen the lower plate before lifting off the upper. When the upper plate is lifted a connection to it will be noted: the earth return line of the autostop circuit. Although the plate can be swung aside to the extent of this wire, this is not good practice and unsoldering the joint is recommended. Later models have improved on this system, and there is no link wire to bother about.

For the lower plate, the control panel, we again commence by removing knobs. After which the braid-wire contact to the right-hand autostop arm has to be removed. Again, this is not necessary on the later models. Four screws secure the plate and beneath these, under the plate, there are small brass distance pieces.

We now have the situation shown in **figs. 2** and **3**. Some servicing can be carried out without going to this extent, and access to vital parts can be gained by part removal of the foregoing assemblies, to achieve the condition shown in **figs. 5** and **6**. Work around the head gate is made easy by the extensive flap provided, for which Ferrograph are to be praised. Not so praiseworthy, however, is the way the pressure arm engagement is made, with the necessity to close the loading gate and switch the machine to 'Run' before complete panel removal.

Even more fun is head removal, if it is necessary to change one. The head plate is a very natty block and, at first sight, work on it looks easy. After unplugging the phono connections to the amplifier support assembly, and unsoldering the two bottom connections on each replay board (red sleeved wire for mono, red and white sleeves for stereo, with the red sleeve to the upper board), one comes up against a capacitor clip and some unreachable screws. The clip around the 0.75 μF capacitor clamp can be removed, allowing the head cables to be pulled through when the securing screws for the block are undone. Note which chassis holes these cables are fed through: it will be important on reassembly. Before you can undo the four clamping screws, the gate mechanism has to be swung back to the loading position.

Having released the head block, access can be gained to the heads but removal is still a bit tricky. The erase and record heads cannot be removed from the block until the adjacent guides are taken off. These are held by screws accessible from beneath the block. The heads are mounted on sub-plates pivoted on pins that are thrust through from rear to front and tensioned by the upward thrust of mounting springs. It is necessary to apply a considerable force to remove the pressure on the pivot sufficiently to push it through from front to back before the head can be released. Then one must take care that the springs under the mounting, and adjustment screws, do not get lost, going back in the right place on reassembly. At that stage the difficulty of pushing the pivot pin back into place will occupy your attention so much that you will probably forget to refit the thin brass shims that you may find have been fitted for additional packing and spacing beneath the head.

Having taken the head from the plate, we are now faced with its final removal and find that there is a thin Paxolin plate fitted over the pin-holes. To remove this without breaking it needs a lot of care, even movement, and perfectly cleaned pins. This is quite easily done with the aid of a solder extractor gun—what my apprentice calls a 'sucker-pop'. Without such an aid, much patient unsoldering and scraping may be needed. As one who is against the application of unnecessary heat—or mechanical shock—to recording heads, I recommend the former method, using the aspirator.

When refitting heads, take care that the leads do not foul the pivot pin. Joints should be as neat as possible, sleeved up to the pin. The playback live lead especially is very near the pivot and should be dressed carefully around and under, allowing full spring action with no chance of a pinched casing. It is presumed that a note will have been taken when dismantling and no problem of working out which lead goes to where will bother us, but as a general rule, the outer screen goes to the left pin of a pair, and the leads to the front pins of the stereo heads are in the following colour code: green, yellow, white, reading from left to right.

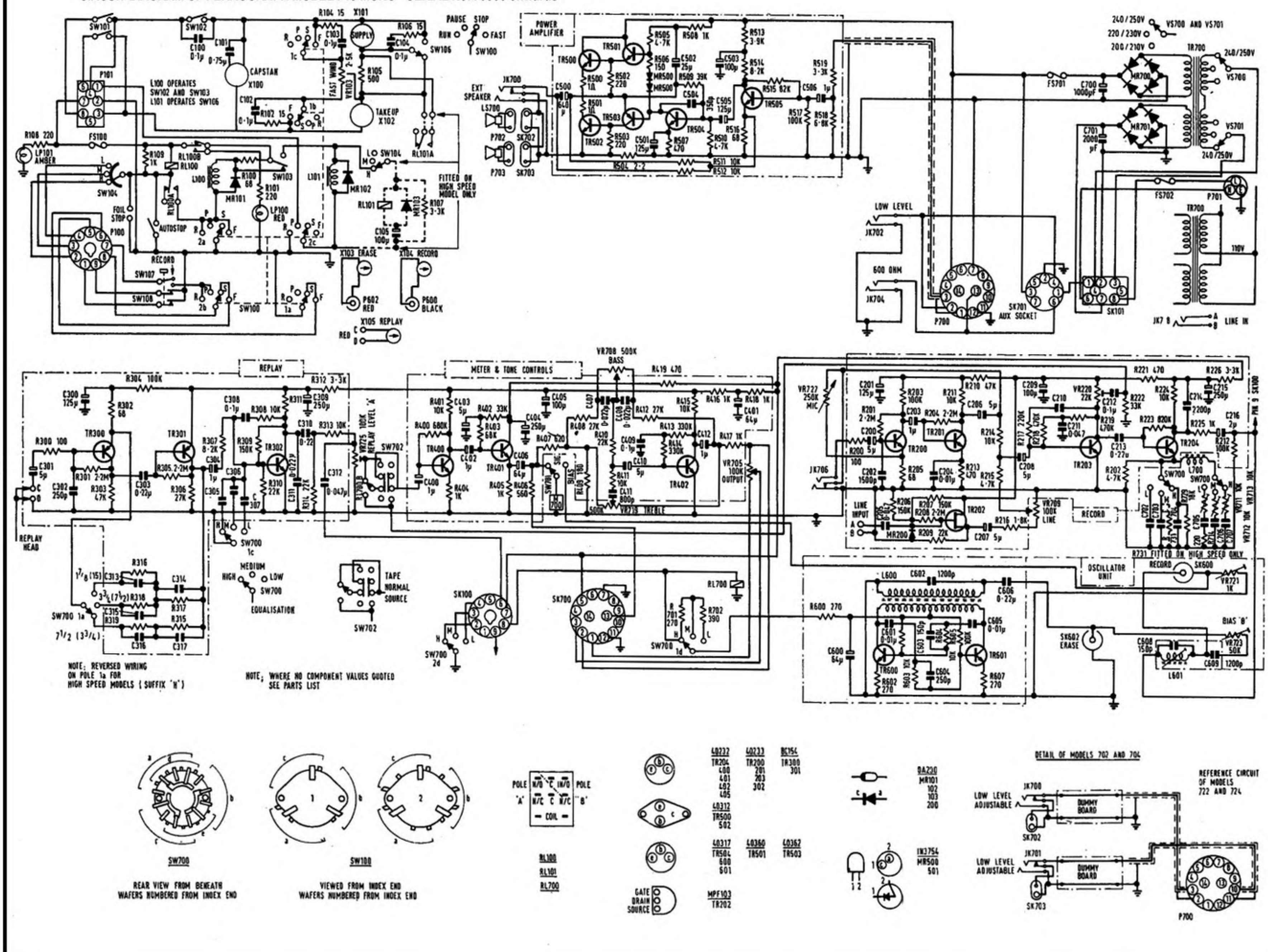
Springs are again a problem when refitting. The important ones on the record and play heads are the adjustment compensating springs under the front left socket-head screw. Springs under the right pair of screws keep the head firmly against its pivot.

I do not intend to waste space on head fitting and alignment. Setting up follows the normal procedure but I would add one very definite requirement to the instructions given by the makers and that is clean and soften the pressure pads before any attempt at adjustment is made. On practically every 'private' machine that has come to our workshop, the flutter performance has been degraded by an accumulation of dirt on the pressure pads. Hardening of pads will cause losses at the HF end of the spectrum.

While around the head channel, observe **figs. 4, 5** and **6** and check the adjustment of the pinch roller pressure arm and the loading assembly. Clearances are important. With a 3/16 inch feeler inserted between the armature and the solenoid (play solenoid is on the right side of deck, see **fig. 3**) and the function control switched to play, the pinch roller should just touch the capstan. Two screws hold the bracket, and the pinch roller can be adjusted with the mechanism stopped so that a touch on the edge of the flywheel with a finger just

(continued on page 353)

CIRCUIT DIAGRAM OF FERROGRAPH MODEL 713 MONO - SERIAL No. 70500 onwards



succeeds in moving the pinch roller. Remove the feeler.

Behind the roller bracket is a small plastic ramp, with obvious adjustment slots beneath its boltheads. Too obvious, for machines inevitably come in with this ramp adjusted too tightly up to the roller bracket. There should be about a sixteenth of an inch clearance. (If the Editor insists, I can transpose these figures to mm. The makers, well aware of international requirements, give both standards of measurement. Taking note of D.K's recent article on metrication, I may even convert to half-Kirks, or milli-Crabbes.) [*The Editor does insist*—**Ed.**]

Going back to the armature of the solenoid, where the pinch roller arm can be seen re-appearing from under the transverse plate, note that there is a nut adjustment. This setting is again important and almost invariably done wrongly. The aim is to get a clearance between the armature and the arm so that the spring does the work on the roller, not the holding magnetic force. If this is working properly, there should be 0.5 mm clearance of the pinch roller from the capstan when the function lever selects pause. And in the play position,

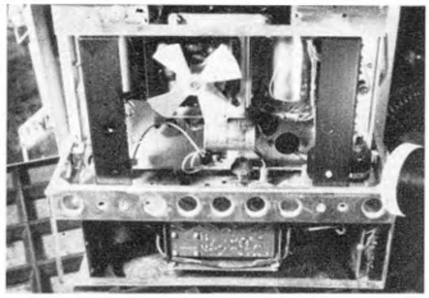


Fig. 2 The bare inside. View of the 713 (mono) with cabinet removed.

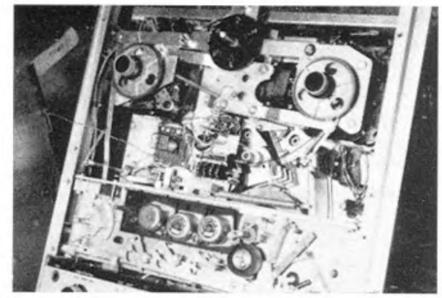


Fig. 3 Front view of mechanism with all covers removed.

Fig. 4 Head gate and pressure pad arrangements. Note guide positions.

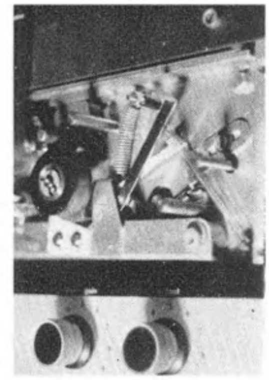
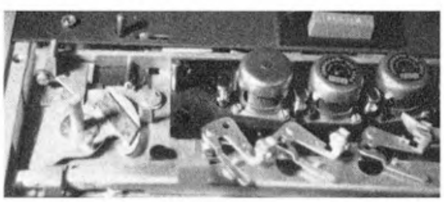
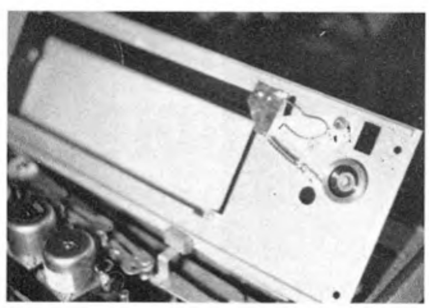


Fig. 5 Lever and spring assembly at right of head gate.

Fig. 6 Head gate seen from above, showing also the important switch mechanism and lever assembly at the left head gate.





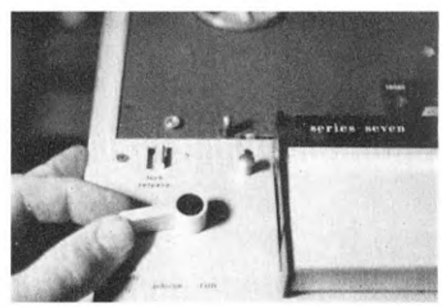


Fig. 7 (right) The head covers have been removed and lower face plate pivoted up to show the very simple pulley and flap device discussed in the text.

Fig. 8 (far right) On early models, the function selector tended to work loose. A set of BA Allen keys are an essential for work on this machine. Flatting the spindle may prevent the trouble re-occurring.

the pinch roller tension should be between 1.2 and 1.4 kg. This can be set by the aforementioned nut on the adjusting arm by the solenoid.

Around this area there are one or two vital factors still to be checked. On top of our solenoid you will see a micro-switch with a screw on the armature operating it. Make sure it switches out on pause and, more important, 'in' on run. This switch interrupts power to the left hand motor, which is not energised during play or record.

The loading arm, which withdraws the pressure pads to ensure clearance for lacing up the tape, can cause a bit of bother. It is not a precise movement and neglect can cause it to bind, affecting the pressure pad settings. Look out for looseness of the tongue under which the left-hand end slids, and the spring on which it rides. This is secured by a single screw and is susceptible to damage if not properly set up.

Another similar spring that can become buckled during attempts at service, and has more than once been refitted wrongly, is the lock release spring at the left side. This must slide neatly under the pin protruding from the transverse flange to let the lock release V-piece contact the raised portion of the cam. (See **fig. 6.**)

Two other points that tie in with our previous investigations are the play braking and the main brakes. To stop the annoyance of tape flop when starting, a retarding brake is fitted to the supply motor assembly, operated by the cam system of the function selector. There is no direct adjustment of this brake arm, but the pad softness must be observed, especially if the machine has had a lot of use, and the freedom of the pivot point and the elbow joint should be checked. The combination of power switching, via the aforementioned microswitch, and the play and pause positioning of this auxiliary brake should be one of the first things to be confirmed.

Main brakes offer not only service adjustment, but also user adjustment. Whether or not this is a good thing depends on the user, need I add? Page 54 of the user's handbook tells us that 'the amount of braking torque applied to each reel is adjusted by sliding a locking screw in a slot.' In theory the fast wind variable facility obviates the need for sharp braking but in practice the machine is often stopped from a maximum torque.

These brakes are wrap-around types, giving a very effective servo action. The main adjustment is by two bolts in each brake arm, passing through slots which then allow the arm to be moved so that the brake achieves greater or lesser clearance. The most important check is that the clearance between brake and drum is even when the start solenoid (the one in the middle) is energised, and pause is engaged.

(continued on page 361)

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A.K.G. K60	£13 10 0	£11 16 0
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BEYER M67	£29 6 0	£24 18 0
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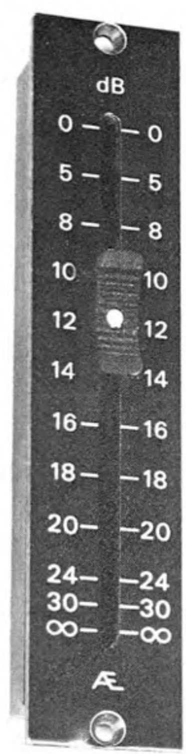

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equipment reviews

CALREC CAPACITOR MICROPHONES

MANUFACTURER'S SPECIFICATION.

Calrec 1050: Studio cardioid capacitor microphone. **Frequency response:** 30 Hz to 20 kHz ±4 dB. **Front-to-back ratio:** 20 dB. **Source impedance:** 50 ohms maximum. **Load impedance:** 200 ohms (or 30 ohms with *CL 1081* matching lead). **Equivalent self-noise:** 20 phons. **Maximum SPL for 0.5% total harmonic distortion:** 400 μB (125 dB). **Operating voltage:** 45 to 50 V. **Current consumption:** 400 μA. **Battery life:** 400 hours. **Length:** 136 mm. **Diameter:** 22 mm. **Weight:** 113 gm. **Price:** £48 16s. (Omni *1000* available at £44 6s.)

Calrec 850: Miniature studio cardioid capacitor microphone. **Frequency response:** 40 Hz to 20 kHz ±7 dB. **Front-to-back ratio:** 20 dB. **Length:** 120 mm. **Diameter:** 19 mm. **Weight:** 90 gm. **Price:** £46. (Omni *800* available at £43.) Other details as above.

Calrec 652: Cardioid capacitor microphone. **Frequency response:** 40 Hz to 20 kHz ±4 dB. **Front-to-back ratio:** 15 dB. **Maximum SPL for 0.5% total harmonic distortion:** 200 μB (120 dB). **Current consumption:** 2.5 mA. **Battery life:** 100 hours. **Length:** 136 mm. **Diameter:** 22 mm. **Weight:** 113 gm. **Price:** £28 11s. (Cardioid *654* with bass roll-off available at £29 6s. Omni *600* at £23 4s.) Other details as above.

Battery Power Units: £7 10s, £9 4s and £12 14s respectively for *600*, *800* and *1000* series.

Mains Power Units: Eleven versions, for one, two, four or six microphones, from £8 4s to £59 10s. Complementary cables, attenuators, windshield and stand adaptors available.

DURING the last quarter century, capacitor microphones have been more and more used in broadcasting and recording in place of moving-coil and ribbon microphones, although the latter types have not been completely superseded.

It is extremely difficult to make a good capacitor capsule and in the past these have been very expensive to produce, some mono capacitor mikes costing over £150 and stereo ones over £300. Because of the very high impedances involved, special valve circuits had to be used having an input impedance of around 200 M in addition to having very low noise components. The ceramic Nuvistor was slightly more convenient but it was not until the advent of the high input impedance, medium output impedance FET, that it was possible to consider making capacitor mikes with a performance better than the earlier valve ones, and yet with an even simpler circuit.

I have in the past heard many attempts at making capacitor mikes cheaply; most of these have been omnidirectional, therefore of no use for stereo, and have had a poor noise level. Quite recently, when the three Calrecs arrived for review I expected them to be greatly inferior to the acceptable expensive models. Let me make it clear from the start that, with a considerable shock, I realised they were in many respects as good as much dearer capacitors.

The first one tried was the *CM 652*, an inexpensive cardioid capacitor available with mains or battery power supplies. The mains supply for 240 V 50 Hz has a pilot lamp to show it is on, a fuse holder with a 2 A fuse, a captive three-core mains lead and a captive 8 m lead on to which the microphone is plugged. The audio output is on an unbalanced jack socket and also has a 1 K potentiometer to reduce the output level for sensitive mike stages. The first power supply tested had an audible 100 Hz hum, particularly noticeable during quiet speech, and when I commented on this to the manufacturer another power supply arrived very quickly, with a slightly modified internal cable arrangement which I understood avoided an earth loop and kept the audio conductors away from mains wiring. This second power supply had no audible hum whatsoever, and further units will have the same modification. The battery supply was then checked and found to give an identical performance from the mike; it became operative as soon as the mike was plugged on to the 2 m cable. The audio ouput was again on an unbalanced jack socket and the battery power supply takes five *PP3* 9 V batteries which are connected in series to give 45 V supply. Since the only other component in the unit is a small audio output pot similar to the one found in the mains power supply, I feel that many users will prefer to make their own, especially as the jack socket provided may not, in any case, be convenient for present day equipment. The mains power supply is very small since, with modern zener diodes and other components, it is possible to get very low ripple voltages within a small space. Some years ago it was necessary in some instances even to include a miniature accumulator in the power line to improve smoothing. For this reason modern power supplies are also considerably cheaper than before.

Now I would like to describe the mike itself and comment on its performance. The mike is only 136 mm long and 22 mm in diameter, having a cable socket at one end with the cardioid capsule at the other, and is cylindrical in shape. The mike cable has only three connections to the mike, an earth return for the DC which is also the earthy side of the audio output which becomes a screen on the mike cable, and two other wires inside this screen, one carrying the live DC from the battery or mains power supply and the other the live audio output. Hence, the mike is essentially unbalanced. The screening was so effective that, despite the main output being unbalanced, no ticky hum or other breakthrough was experienced, despite also the Brookmans Park transmitter being only approximately 10 miles away. I would recommend, however, that an unbalanced-to-balanced transformer of approximately 600 ohms impedance be connected to the power unit audio output if longer cable runs are necessary, and the potentiometer taken out of circuit. The microphone was found to be exceptionally low in hiss level, measuring 3 dB better than another commercially available professional FET front end mike. which was previously considered excellent. The noise level also measured 6 to 8 dB better than a good example of a valve capacitor mike. Suffice to say that, even for recording very quiet speech, the hiss from this mike would be almost unnoticeable. This excellent noise-free performance would make the mike ideal for recording very quiet sound effects. I found that it fitted neatly in to the AKG *D19* windshield which, although fairly expensive, was very effective. The *CM 652* appeared to have a remarkably flat bass response, but a rising extreme top response, which in any case is not uncommon with capacitor mikes. When the AKG windshield was used, however, the mike sounded flatter as the shield tended to cut extreme top. The microphone response was well extended into the deep bass and extreme top. The microphone has been in use for some while and has

(continued overleaf)


1050

652

850

CALREC REVIEW CONTINUED

always been completely silent in operation without any demoisturising or warming up being needed. The cardioid response was well maintained although it was found that at extreme HF this response became more hyper-cardioid in characteristic. There is no doubt whatsoever that this microphone can be very safely recommended to semi-professionals and will give superb realism in stereo. Users of Uher battery tape recorders may be interested to know that this mike performs excellently when plugged straight in to the socket on the front of the machine, though the output is about order of 15 dB higher than from the average moving-coil. The combination of the Uher and the *CM 652* would be superb for recording the very quietest of wild life sounds in the still of the night.

Two other models in the Calrec range were tested, both appreciably more expensive. The first was the *CM 850*. This was specifically designed for recording fairly close speech and has therefore a built in bass cut of 10 dB at 50 Hz. It was tested with the mains power supply which was found to hum, but presumably this could have been put right with a similar modification to that carried out to the *652*. The output is balanced and appears on a locking three-pin DIN chassis-mounting plug on the power supply. I would personally have preferred a more convenient three-pin DIN socket. The power supply for this mike has a captive mains lead but the mike lead plugs in on another three-pin DIN socket and is 8 m long. The *850* is 120 mm long and 19 mm in diameter, again very smart in appearance. The top response was found to be considerably flatter than the *652* and the noise level appears the same but for the aforementioned hum.

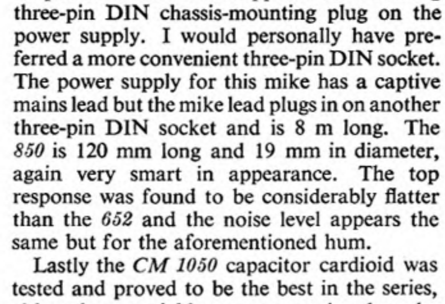
Lastly the *CM 1050* capacitor cardioid was tested and proved to be the best in the series, although appreciably more expensive than the *652*. I compared the performance of this mike very carefully with a professional capacitor costing much more and, surprisingly, the Calrec consistently had a better noise level.

The lack of distortion was very noticeable and the sound was exceptionally smooth, with an excellent bass response. Although the *1050* is the most expensive in the range, it is still appreciably cheaper than almost every other professional capacitor and yet audibly sounds as good. I therefore highly recommended it if it can be produced consistently. The power supply for the *1050* had no audible hum and the hiss level measured the same as the *CM652* with the weighted dBa curve. I understand from the manufacturers that the noise level of the average sample measures between 18 and 21 phons, and all the samples under review would certainly have been as good as this.

Finally a word of warning about capacitor microphones. These should always be stored in a dry medium-temperature atmosphere when not in use. Although they are fairly robust, they will not stand being eaten by pop singers and for close speech recording windshields should always be used to protect against breath moisture and LF wind noise. Capacitor microphones are also very sensitive if their stand is touched and even a slight jolt will give a thumping noise in the audio output. This should be borne in mind in the choice of a microphone stand and also in placing the stand as the same effect will be obtained if it is placed on a long loose floor board. **Angus McKenzie**

UHER 1000 Mk.2

MANUFACTURER'S SPECIFICATION (19 cm/s). Single-speed professional battery portable recorder with synchropulse head. **Wow and flutter:** ±0.15% unweighted (0.05% DIN weighted). **Absolute speed deviation:** ±.5%. **Frequency response:** 40 Hz to 15 kHz within 2 dB. **Signal-to-noise ratio:** better than 52 dB unweighted. **Equalisation:** CCIR 70 μS. (Can be altered to NAB 50-3180 μS response by rewiring four soldered links on printed-circuit cards.) **Peak level distortion:** less than 2%. **Microphone input:** 0.3 mV at 200 ohms, balanced. **Line output:** 4.4 V at 600 ohms. **Power supply:** Five *HPU2* cells or 6 V storage battery. **Dimensions:** 270 by 215 by 85 mm. **Weight:** 3 kg. **Price:** £331 15s 4d.
Distributor: Bosch Ltd., Radlett Road, Watford, Herts.



THIS is a slightly newer version of the machine reviewed by Terence Long in October 1969. The circuit is almost identical to that published then. Wow and flutter have been significantly improved by altering the tyre on the speed reduction disc, and harmonic distortion seems to be lower, but this may be due to a change in tape issued with the recorder.

Although the record level meter looks like a standard VU, it is in fact connected as a peak level meter, with 0 dB corresponding to 32 mM/mm reference tape level. It is connected to the playback amplifier so that it monitors the actual level *on the tape*. The slight delay in the kick of the meter can be rather disconcerting when monitoring one's own voice, but provision is made for it to monitor *record* level whenever the pause key is pressed.

The automatic record level system is interesting as it uses a photo-resistive cell as an attenuator element; this is illuminated by a light bulb which is fed from a rectified and filtered signal from the record amplifier output. It is not a true limiter but more an AGC circuit with a short attack time and relatively long (10 to 15 seconds) recovery, so that a consistently high input will simply reduce the gain and hold it at a lower level until the input changes.

The pen traces of **fig. 1** show that the total cumulative wow and flutter does not exceed 0.06% RMS. The speed imperfections consist of almost equal proportions of high frequency tape friction noise and very low level 7 Hz capstan wow which only shows when record and play wows are in phase. To keep the machine insensitive to normal portable recorder movement, the speed reduction disc can have little flywheel effect and the compromise adopted can be considered very satisfactory, with wow and flutter down near static recorder level.

The top curve of **fig. 2** shows the response at line output when playing the 19 cm/s DIN 45513 70 μS test tape. Playback distortion when playing the 32 mM/mm peak level band on this tape was 0.75%.

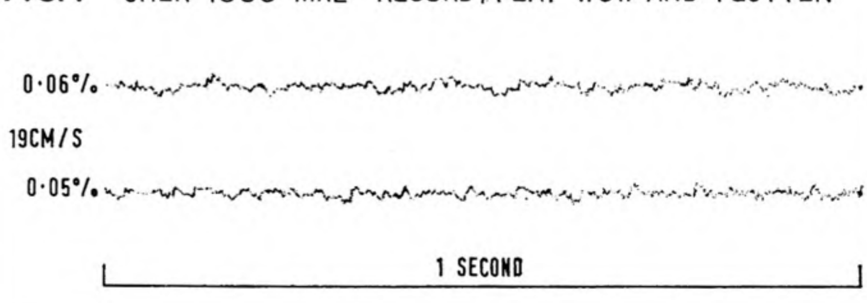
System noise with no tape was 56 dB below reference level and weighting to the IEC 'A' response reduced the reading to −67 dB.

Peak recording level erased on the machine gave a weighted reading of 62 dB and bulk-erased tapes showed a noise level of −65 dB.

The overall record-play response at 20 dB below reference level is shown by the lower curve of **fig. 2.** This is well within the specification limits.

The pilot tone head may be unfamiliar to some readers but it deserves a brief description as it allows two signals to be recorded across the full width of the tape with a separation of better than 45 dB. The 1 kHz pilot tone is interrupted at frame frequency by a contact in a cine camera so that the taped sound can be transferred to film with exact synchronization between picture and sound.

FIG. 1 UHER 1000 MK2 RECORD / PLAY WOW AND FLUTTER



The pilot head comprises two V shaped gaps forming, in effect, two 45° gaps which are electrically in phase opposition so that they record a signal which can be replayed on the pilot tone head but cannot be detected by the vertical playback head gap. Both the audio record head and the pilot tone head are fed with bias when the controls are set to record. A 2V 1 kHz signal fed to the pilot head records peak recording level on the tape and, on replay, a signal of 0.4 mV is available from the pilot head. Although 45 dB sounds an adequate audio/tone ratio the pulsed tone can just be heard if recorded at peak recording level. To reduce crosstalk to vanishing point, the pilot tone level is reduced by 20 dB so that the pilot tone playback level falls to 0.04 mV. A tuned amplifier can be used to improve the signal to noise ratio if required.

The acoustic response of **fig. 3** was measured by recording one-third octave bands of filtered white noise and then measuring the sound output on the speaker axis during playback. This shows that the response is sensibly level from 250 Hz to 6 kHz which is adequate for judging speech quality and general sound effects.

The *1000 Report Pilot* is a thoroughly professional battery portable with very low mechanical noise, extremely good signal-to-noise ratio, exact equalisation and wide frequency response. Distortion is remarkably low and the controls can all be reached by the fingers of the hand holding the recorder. It is one of the few machines available where the speed can be varied and set to exactly 19 cm/s by means of the built-in strobe facility. The pilot tone facility makes it capable of exact 'lip sync' sound recording for film work.

I notice that the line output is now available on a pair of Continental 2 mm plug sockets which is a vast improvement on the original six-way socket that also carried the pilot tone head connections.

A. Tutchings

FIG. 2 UHER 1000 MK2 PLAY-ONLY AND RECORD-PLAY RESPONSES TO LINE OUTPUT

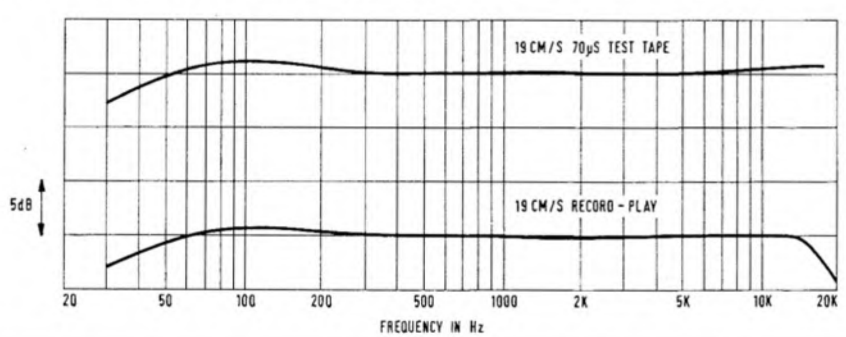
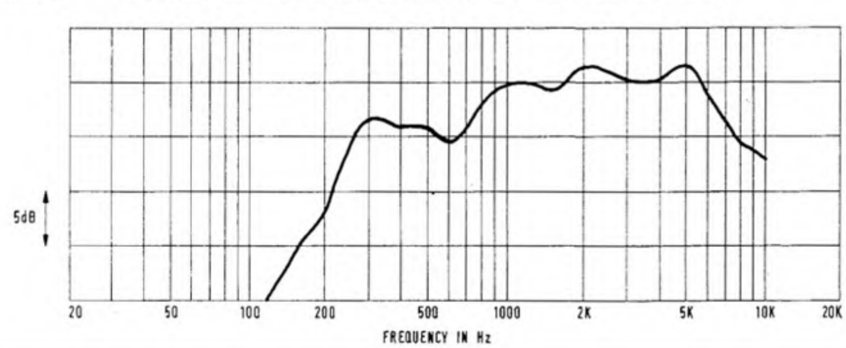


FIG. 3 UHER 1000 MK2 ACOUSTIC RESPONSE LINE IN TO SOUND LEVEL ON LS AXIS



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BANG & OLUFSEN BM5 RIBBON MICROPHONE

MANUFACTURER'S SPECIFICATION (19 cm/s). Figure-of-8 stereo ribbon microphone, coaxial up to 90°. **Frequency response:** 30 Hz to 13 kHz ±2.5 dB. **Sensitivity:** 85 dB below 1 V/μB. **Hum sensitivity:** 146 dB. Bass attenuator switch and phase-change. **Output:** 200 ohms at 1 kHz via 3 m lead, 5-pin DIN plug. Supplied in rosewood cas ewith table stand. **Price:** £30 9s. **Manufacturer:** Bang & Olufsen A/S, Struer, Denmark. **Distributor: Bang & Olufsen UK Division, Eastbrook Road, Gloucester.**

THE Bang & Olufsen *BM5* stereo microphone is a pressure-gradient ribbon unit for balanced or unbalanced inputs, with a nominal impedance of 150-200 ohms. It can be bought as one unit or made up by combining the *BM6* (mono) microphone and the *BM7* add-on unit to form a composite co-axial stereo assembly.

First impressions, as I drew the beautifully polished wooden case from the layers of cardboard which protected it in the post, were very favourable. At last! A domestic microphone in a strong and durable case that would withstand years of use and knocking around on recording sessions, instead of the more common cardboard and polystyrene packaging that soon becomes as useless as it is tatty. Alas! The wooden case is not enough. Despite the flocked board and padding, the microphone had been able to knock against the end of the box too much, and it was broken. Perhaps a trip through the post is *not* the gentlest treatment (I learn from correspondence with B & O that they have known this happen before and do not send equipment out except by their own van); but a microphone *should*, surely, be able to stand up to this treatment in its case? So black mark number one—a little more firm padding could have kept the mike in one piece.

Black mark number two goes to the thin and brittle plastic coupling between the upper and lower halves of the microphone, which enables the axes of the two halves to be swivelled relative to one another. It was this that had broken. The plastic castellated moulding is part of the miniature 9-pin plug which allows the top half to be unplugged from the bottom or—black mark number three—in an extreme case could allow the plugged-in half to fall free if the mike was suspended and the plug had worked loose in the course of time. Very unlikely, perhaps, but that would hardly console the relatives of any unfortunate sitting below. Perhaps it would be a good idea if B & O were to supply a warning with the mike, saying that it should only be used on a stand and not slung—the alternative would be to improve on the swivel-and-plug arrangement in line with the excellent finish of the rest of the microphone. To be fair to them, B & O—like most domestic mike manufacturers—make no adequate provision for slinging; though the excellently made locking 5-pin plug/socket arrangement for plugging the lead into the base of the mike makes it rather tempting to sling by the cable—never very satisfactory (apart from being unsafe) because with temperature change the lead is almost bound to twist.

Having started off on this dismal note, let me hasten to say there are excellent features to this microphone and, with certain reservations which will emerge as this test proceeds, it is worthy of serious consideration by anyone with about £30 to spend on a stereo microphone.

The *BM5* has a long, slender tubular housing finished satin chrome, and its appearance is fairly unobtrusive although it could catch the light from bright spots. It has a black plastic cap at the top and small plastic caps on the speech/music and in-out-of-phase/off switches; the latter are easily lost, but fortunately the switch lever is a neat black serrated-edge plastic arm and looks as well without. The switches look rather fragile altogether, but in

domestic use I doubt whether they would be operated a great deal. The mike is provided with rather a short multi-core lead, and slips into a friction-fit collapsible desk stand which comes in the case. When the mike is to be used on a floor stand, the feet are left off, and the central portion of the stand then acts as a threaded adaptor. The English translation in the booklet with the mike implies that it is threaded to fit the ⅝-inch standard, but in fact it is a coarse ⅜ inch thread and will require a further adaptor for many stands: these are available from B & O.

So to performance. First impression here was inevitably that the microphone was rather insensitive—ribbons always are, unless you have a *4038*, but it seemed unduly lacking in sensitivity. The nominal output impedance of the mike is given as 150-200 ohms, and I understand from B & O that this was arrived at to suit the 200-ohm inputs of their recorders. It seems the old standard input impedances of 60 and 600 ohms have gone by the board and one is faced with mikes of all manner of source and load impedances as standard, with inevitable matching problems. The mike was first fed to 600 ohm input and subsequently to 60 ohm. It is not good practice to feed inputs of lower impedance than the mike as the frequency response suffers; in this case the results did not seem to be too bad. I have one American transformer with which it is possible, by changing soldered connections, to obtain almost any impedance input and ratio. This gave a small improvement in sensitivity over the 600 ohm input which would probably have been worthwhile in practice. However, the mike seemed still to be relatively insensitive by comparison with, say, the Lustraphone stereo ribbon or the Film Industries ribbon. Since the Film Industries is about 30 ohms against the *BM5* 150 ohms, I could not do a direct comparison: I shall be interested to see whether measurements by Alec Tutchings bear me out on this.

No sibilant splash

Having added yet another criticism to my list, let me say straight away that the sounds obtained with this mike are as realistic as any I have heard from a domestic mike, and considerably better than most. There is no trace of sibilant splashing on speech, or accentuation of 'hissy' sounds; the Film Industries is inclined to give a little prominence to consonants, probably because of the slight rise in output around 5 kHz (shown in the recent review) which makes it particularly useful on choruses where words tend to get a little lost, or for recording in an acoustic with a bass heavy echo or reverberation—the *BM5* by contrast gives no trace of this 'presence' and it seems to be nearly perfect for one or two voices, solo instruments, or a small group, and for natural speech. For speech within about 60 cm of the mike it is worth using the internal bass-cut switch, which lowers the output by a nominal 10 dB at 100 Hz. This is naturally a compromise correction, but seems to give acceptable results down to about 9 inches. Even speaking closer than this, there is no problem of sibilants 'blasting' or 'popping'. (Somewhere I now have a recording of 'Peter Piper picked a peck of pickled pepper' recorded at every 5 cm down to about 10 cm, a distance at which no one would dream of using the mike, with no trouble from popping.)

Used to record folksinging, the *BM5* gave a very good account of voices and guitar, and very good stereo. Guitars seem to show up in a subtle way any LF and lower-mid frequency response shortcomings of a microphone, becoming tubby or boomy in sound if all is not well. In this case it was quite natural, and fingering and picking noises were most realistic. Just occasionally speaking voices sounded a little 'full', but this may have been due to mike positioning and the room.

Recording quiet singing, and particularly recording the spinet, required full gain setting on a Brenell *STB-1* with the transformers supplied to match the Film Industries to 100 K inputs, which seemed to confirm that the mike is relatively insensitive, although this is inevitably something of a mismatch. But the sound quality is very pleasing. Given quiet mike amps, suitable matching transformers or amplifier input impedance, and a fairly loud sound, the results can be excellent.

The magnetic system inside the mike seems to be U-shaped, duplicated above and below the ribbon (instead of the familiar cup magnets behind the ribbon, used in the Reslo, Film Industries and Lustraphone designs), with the pole pieces cut away at the centre of the ribbon. This probably contributes to the smoothness of the frequency response and would I imagine give a better and more symmetrical polar diagram at all frequencies than the cup magnet designs. It is probably electrically less efficient than the cup magnet designs, which would account for the low sensitivity.

The ribbons are fairly long by comparison with the other designs I have mentioned, which no doubt helps the sensitivity, but probably at the expense of the high frequency polar response in the vertical plane. However, I noticed no particular deficiencies on this score, and in practice it is surprising what a small vertical angle one uses even for an orchestra and chorus, which perhaps explains how it was ever possible to use long-ribbon designs like the old *AXB* broadcast ribbon mike.

Phase Switch

Despite its fragile appearance, the phase switch is a useful facility, particularly if (like myself) you happen to be using a combination of DIN-jack adaptors, extension leads and transformers that somewhere ends up with one channel out of phase with the other! By paralleling the output at the tape recorder and/or amplifier and listening to the result, it is easy to tell when the two halves are in phase with each other—when out of phase there is a pronounced drop in level, especially in the bass. This is particularly easy to do if the top half of the mike is swivelled so that the axes are in the same direction while one is phasing. Another useful feature of the switch is that it extends the effective range over which the axes can be rotated relative to one another. Normally, with the channels in phase, the mike case is designed to allow the top section to swivel so that the angle between the axes is

within the range 0° to 90°. If the channels are put in anti-phase, however, it is the front (signified by the B & O symbol) of the lower half and the back (plain) face of the top section that are in phase, and the mike can be then swung round to make use of the angle between these two faces, which can lie between 90° and 180°. It is not often that one in fact uses anything other than 90° with ribbons, but it does mean the facility is there when required. It is only practicable to do this with a design where—as in this case—the polar diagram is fairly symmetrical.

One curious and unfortunate feature of the replacement mike sent for review was that in assembly it had been put together with the phase switch opposite the case symbol for the bass cut switch and vice versa! This gave me some interesting moments trying to phase the mike with the bass-cut switch; it also meant that the *plain* face of the top half was the one *in* phase with the principal face of the lower half. I suppose this shows one good thing—the review sample was not specially picked!

The overall impression? B & O made strenuous efforts to contact me and to get a replacement mike to me when they heard that the review sample had been damaged. Although it was some while before the second arrived (the Christmas rush did after all coincide), I suspect that a purchaser could expect helpful and interested treatment. (Initial tests were done using the damaged mike strapped together with insulating tape which is acoustically fairly dead and neutral, and also sticks such things fairly securely.) It was interesting subsequently to compare the two nominally identical mikes side by side under virtually identical controlled conditions. The results were virtually indistinguishable, and the two halves of both mikes seemed to be well matched. This contrasted with an earlier experience of the mono version converted to stereo with the plug-in additional unit, where the two halves were so dissimilar that we decided not to use the mike on that occasion: it is possible that one or the other unit had been damaged, I suppose, but the review units were much closer matched. Personally, if I were even contemplating stereo when buying the mono version, I would go the whole hog and buy the stereo mike which only costs about 50% extra, to be absolutely sure of matching.

Inevitably, the B & O invites comparison with the Lustraphone stereo ribbon; in basic form, without the frills of phase switching, the latter costs some £10-£12 less than the B & O, and seems to be more sensitive. I would not think there is a great deal to choose in terms of frequency response, and I know a lot of people with experienced ears have a high regard for the Lustraphone. So what does one get for the extra money? Possibly a slightly better horizontal polar response and stereo image, though any improvement is subtle; a stronger case, which nevertheless will not allow the mike to be handled too roughly; the doubtful advantage of being able to buy the two halves separately; a token desk-stand/adaptor; phase switching (available in the Lustraphone at considerably increased cost) and a bass-cut switch. Totted up, this seems to give fair value for money, *if* you want all the frills, certainly if you want to feed directly into a B & O recorder; and you are sure of a very pleasing sound quality. I may have overstated the apparent lack of sensitivity, but it remains my principal reservation apart from the fragile coupling, to return to my earlier points. Nevertheless, if you have about £30 to spend and would prefer a coaxial stereo mike to two separate units, the *BM5* is definitely worth shortlisting.

My impression is that the extreme top is a bit down (in this respect the Film Industries seems to be better) and I *suspect*, though I cannot be absolutely sure, that the bass end is marginally more realistic than the Lustraphone: my impressions conflicted a little on this. But the general impression is of a wide, smooth and uncoloured response.

Recording my octavina spinet, the *BM5* gave very natural and realistic results. The spinet played through the *BM5* and a pair of Goodmans *Maxims* was extraordinarily realistic, and the thought came to me that anyone who must indulge in amplifying harpsichords to sound through a modern orchestra could do a lot worse than this combination as an accurate and unobtrusive set of transducers.

John Fisher

PROFESSIONAL HOME STUDIO CONTINUED

quickly opened as the tape starts, which is inconvenient. Why I say unavoidable is because I have built elaborate mains filters into the Ferrograph which keeps the starting click out of the other equipment but is still audible on the Ferrograph output. If you look at the inductance of the three motors, you can see that the job of suppression is hopeless before you start! Later Ferrographs have an excellent pause control; mine has a rather primitive one with a small operating button on the tape-head cover. You can attach a Leica cable release to this, but this needs considerable wrestling by the Studio Manager. I have attached an EMI editing block to the deck of the Ferrograph in such a way that, to close the tape cover lid, you have to depress the pause button. With the machine running and the cover closed, the tape is stationary. To play in, you lift the cover which releases the pause button. To stop you close the cover, easing the button behind the editing block with your thumb—all very simple, yet effective.

(continued on page 361)

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PROFESSIONAL HOME STUDIO CONTINUED

Another source of clicks was the cue-light circuit. This is fed from the 6.3 V heater winding of my main power pack. A pair of chokes and capacitors in the cue-light circuit cured this. Clicks from domestic refrigerators and central heating ignition can be eliminated if a good mains filter and a good earth are used—not the mains earth. One problem I have is that the pump driving the small-bore central heating system introduces slight acoustic hum into the studio—so I turn off the central heating during a recording session.

With good HT smoothing and balanced feed leads from microphones, I have no hum problems at all. One good rule is keep all transformers and chokes well away from the early audio stages. Magnetic hum is very difficult to eliminate, electrostatic hum is easy.

Of course, because I design and build a lot of my own equipment, I necessarily have other instruments, like valve and transistor testers, a C-R bridge and so on. One thing I always do is to check the values of capacitors and resistors before I use them ; they are sometimes marked incorrectly or are faulty, in which case the trouble can take quite a time to locate. I spend a bit extra to buy really low noise resistors, preferably wirewound, for the early audio stages —I think these few shillings are well spent.

I would like to end by saying that I think the service given by many manufacturers, notably Ferrograph and Leak, is superb. Queries are answered rapidly and intelligently, and faults corrected without charge long after the guarantee has expired. Service like this always prejudices me in favour of buying future equipment from such companies. I have another list of firms with whom I shall never deal again, but this one is confidential.

TAPE RECORDER SERVICE CONTINUED

Clearance should be about 0.8 mm. If any adjustment is needed, double-check that the brake is fully engaging when stop is once more selected.

For smooth operation of the brakes—once this positive action has been ensured—the tension of the spring in the arm that is in line with the end of the brake arc has to be adjusted. Some models have an Allen key adjustment, others will answer to attention from a screwdriver. Extend the spring and reclamp to increase brake pull against the reel torque. Always go back and check the stopping from full fast wind in each direction after making any of these adjustments.

Of the other adjustments that may be needed, the start switch setting causes the most confusion. There are two micro-switches mounted together in the middle of the deck and two locknutted screw heads that connect with them. The lower switch, which had changed its function since the earlier models, has to be set carefully so that it just makes, and no more, when start is selected. The only way to do this is by operating the mechanism by hand with the power disconnected, then trying for correct action with the power on again, and setting for more positive action if the relay chatters.

The above-mentioned change of function: earlier models used this switch to put a resistor in series with the *L100* coil, auto-stop relay, as an economy device; later models had the switch in the 24 V supply line to the coil, again inserting a resistor, and also completing the voltage supply to the run solenoid. Either way, its action matters for positive starting, and some care in setting is justified.

by Peter Bastin

ON APRIL 17, the national press announced that Mr Brian Young, a former Charterhouse headmaster, had been appointed Director-General of the Independent Television Authority. Since 1964 he has been director of the Nuffield Foundation. Mr Young admitted that he had never been inside a television studio. He also admitted that he knew nothing about television production. Which makes me wonder even more about this mixed-up world we live in. Would a large manufacturing industry take on a man at £18 200 a year who publicly admitted that he knew nothing whatsoever about the industry, the processes, or the business in general? I think not. Mr Young's annual salary is roughly equivalent to the salaries drawn by eight men who *know* their job.

'A UNIQUE tape recorder with features unknown outside the professional field', announces a recent REW advertisement. And what are these remarkable features? Automatic threading and electronic reversing. I suggest these are almost unknown outside the *domestic* field. On two occasions now, I have had to wrestle obnoxious little auto-threads on cine projectors ; even when they are working properly, which isn't often, they are more a hindrance than a help. Actually the Ampex *1100*, which REW were advertising, has an excellent auto-thread arrangement which could be applied to all domestic recorders if spool manufacturers would only wake up. The takeup spool is slotted right across the hub to take the leader. No mechanism to go wrong.

HAVE YOU ever heard or recorded a ghost? A friend of mine, who is an archivist, is in contact with a Mr Carlson of Massachusetts USA. Mr Carlson's wife 'is a medium' and, through her, he is in touch with a lady born in the 17th Century, who claims to be Shakespeare's niece. He has sent three ¼-track tapes, recorded at a speed of 4.75 cm/s, of interviews with the lady. Most of the material is of very poor quality, although Mr Carlson apologises for this. The 'ghost' (via Mrs Carlson) speaks in a strange accent, apparently regarded as an old English accent. To me, it is distinctly Irish. Whether Mrs Carlson is Irish or not, I don't know, but there is certainly no American brogue evident. However, a peculiar contradiction appears to exist. Shakespeare's niece quotes her mother as saying that 'Bill Shakespeare is a lazy bum'.

NO TRUTH in the rumour that the Hong Kong Electronic Trash Company, unable to sell their budget-hi-fi amplifier, are to market it as a speech scrambler.

TELEPHONE CALL a little while back from a reader who had discovered that a licence is needed to make tape recordings in Hyde Park and on Speakers' Corner. This appears to result from a ban on the ubiquitous 'transistor', aimed at squeaky little radios. I have since discovered that you almost need a licence for *yourself* in Hyde Park; a visitor was recently fined when he refused to leave. If I lived in London I'd complain. Fortunately I don't.

SOME THINGS surely try the patience of the audiophile. In a recent edition of BBC TV *A Man Called Ironside* a woman was being subjected to the old-hat psycho treatment—strange noises, threatening voices coming out of the woodwork, etc. The voice in this case was a half-speed growl, doctored with heavy reverberation, repeat-echo etc. The source of this terrifying manifestation was a throat microphone and transmitter, worn by the villain. It was not, of course, explained how the villain managed to record at one speed and transmit at another all in one action.

NEXT TIME you buy a stethoscope earphone set for your Grundig dictation machine, it will come in a box labelled *Studio Quality Accessory*.

THE JUNE editorial devoted a lot of space to urging the serious amateur to think professional. Good. The sooner the Revox/AKG amateur learns to use his equipment in the way for which it is designed, the better. A lot of low-quality competition entries from amateurs submerges a small amount of near-professional recording from the same sources. However, I take issue with some of the editorial recommendations. Forget quarter-track, 19 cm/s speed, unbranded tape, high-Z microphones, tiny spools, GPO headphones and record/replay heads—these are the recommendations. Unbranded tapes, tiny spools, GPO headphones, I agree with. 19 cm/s speed, I don't. A good deal of broadcast material is recorded at this speed and who can argue with that? [*We can*—**Ed.**]

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Editorial assistant required for *Hi-Fi News*. Person under 20 preferred, with a particular knowledge of and enthusiasm for music and records, and preferably an interest in audio. Some journalistic experience an advantage. Apply in writing giving full details to the Editor, Hi-Fi News, Link House, Dingwall Ave., Croydon CR9 2TA.

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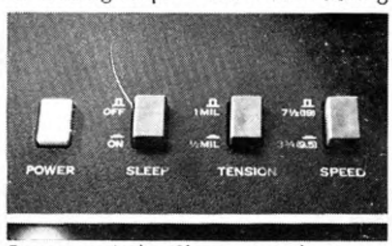

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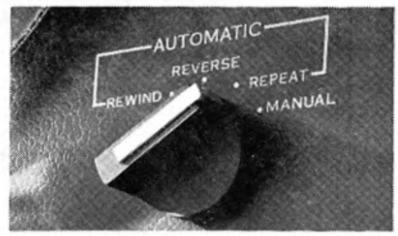

The SD-7000 will be available soon at authorized Sansui dealers, and it will pay you to check it out more fully. Unless of course you want to go on playing patch-up stereo.

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