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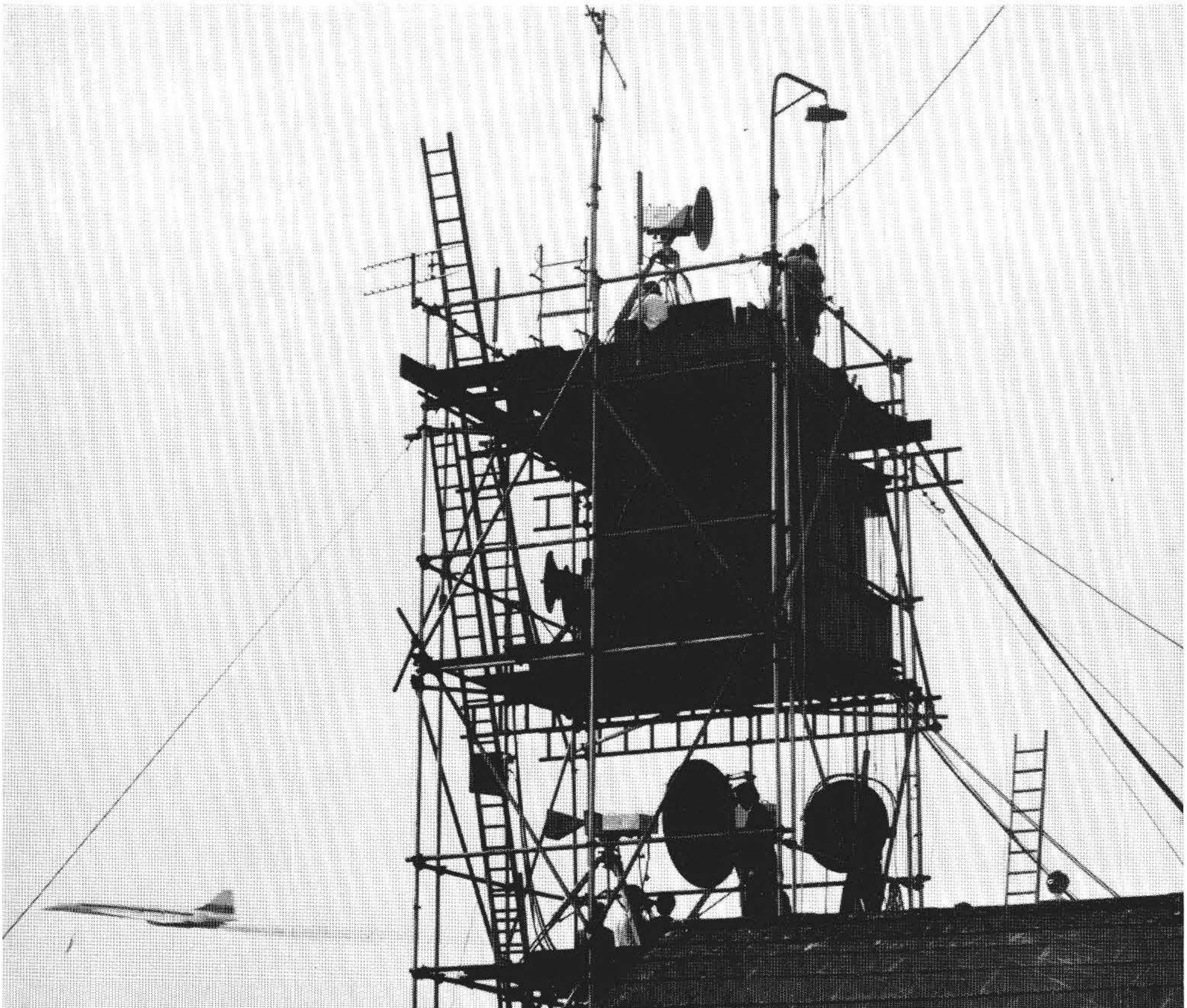

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The cover photograph shows a group of microwave aerials at the Farnborough Air Display, at the moment when a Concorde was flying past.

The major contributions are preceded by individual lists of contents.

Editorial

A New Plan for l.f./m.f. Broadcasting

In accordance with the views expressed by the majority of its members in the areas concerned, the Administrative Council of the International Telecommunications Union (ITU) has decided that a Regional Administrative LF/MF Broadcasting Conference shall be held to prepare a frequency plan for l.f./m.f. broadcasting embracing the whole of ITU Regions 1 and 3, which include the continents of Europe, Africa, Asia, and Oceania. The first session of the Conference will be held in Geneva in October 1974 and will be responsible for the preparation of technical data and operational criteria intended to serve as the basis for the preparation of a frequency assignment plan by the second session of the Conference, due to take place about a year later. If the aims of the Conference are achieved and a new Plan is agreed, it will probably be implemented around mid-1977 and will replace the existing 1948 Copenhagen Plan for the European Broadcasting Area and the 1966 Geneva Plan for Africa. There is no similar Plan for Region 3 which has evolved the present 10kHz spaced arrangement over many years. The transmitter density in Region 3 is much lower than in Region 1.

The Present Position

In Europe, the steady increase in l.f./m.f. broadcasting over the last twenty-five years has resulted in most countries contravening the Copenhagen Plan to a greater or lesser extent and there are now more than three times as many transmitters in operation than were provided for in the original Plan. Consequently, night-time reception on most frequencies is seriously degraded due to sky-wave interference from other stations, principally from those operating co-channel, although in some areas adjacent-channel interference has also assumed serious proportions. The problem is particularly acute in Central Europe where the transmitter density is at its highest. Moreover, in some countries, notably in East Africa and the Middle East, the situation is worsened by the fact that Region 1 (Europe and Africa) has adopted 9kHz channel spacing whilst in Region 3 (Asia and Oceania), 10kHz channelling is the rule. This leads to the production of inter-carrier heterodyne interference at multiples of 1 kHz throughout the m.f. band and is especially serious in those countries close to the borders between Regions 1 and 3. The adoption of a common channelling for both Regions would almost entirely eliminate this type of interference.

In preparation for the forthcoming Conference, the European Broadcasting Union (EBU) Technical Committee, in whose activities the BBC actively participates, has been studying for some years the possibilities of improving reception in the l.f./m.f. bands and has now reached a large measure of agreement on the best means to achieve this. Unfortunately, the problems are not entirely technical but also concern the programme planners since there is obviously a limit to the number of programmes which can be accommodated in the bands if reasonable standards of reception quality are to be maintained. From the technical standpoint, this limit has long been exceeded and only if countries agree to reduce their programme requirements in the l.f./m.f. bands is there any real prospect of a new frequency plan bringing about a substantial improvement in reception. The practice in many European countries of duplicating programmes on l.f./m.f. and v.h.f. is vulnerable to attack particularly by the underdeveloped countries of Africa and the Middle East where, for economic and coverage reasons, the radio services are radiated on m.f. only. The reasons for duplication are well known, i.e. v.h.f./f.m. receivers are more expensive and they need at least telescopic aerials which are inconvenient, they are difficult to tune, and the v.h.f. service is less effective for reception in motor cars. With the exception of car radio, these reasons are now less valid as the cost differential between modern v.h.f. transistor portables and their l.f./m.f. counterparts is now quite small and the tuning difficulties have been reduced by the introduction of automatic frequency control at very little additional cost. Many people are satisfied with the results they get on m.f. and do not wish to have the higher quality provided by v.h.f./f.m. because of its attendant disadvantages and, therefore, as long as duplication exists the public will continue to resist buying v.h.f. receivers. Only when popular programmes are available on v.h.f. but not on m.f. will there be an upsurge in sales.

New Transmission Systems

For a given number of transmitters, an increase in the number of available channels would permit a reduction in co-channel interference and a proposal, which received some attention in the EBU, was to change the present double-sideband (DSB) system of transmission to single-sideband (SSB), thereby doubling the number of channels almost overnight. This proposal is no longer in favour because listeners would need to equip themselves with relatively expensive SSB receivers but gain no substantial immediate benefit from so doing and a lengthy changeover period of perhaps ten to fifteen years would be required. An alternative to SSB, but which has the same changeover difficulty as SSB, is the transmission system which employs two independent sidebands (ISB). An important advantage of this system is that it would enable countries operating frequency assignments in accordance with a DSB plan to change progressively to ISB on a unilateral basis, the stimulus to change being provided by the possibility of radiating additional programmes without increasing interference to other countries' transmissions.

Reduced Channel Spacing with DSB

The Copenhagen Plan is based on 9kHz channel spacing and if this were reduced to 8kHz an additional two channels in the l.f. band and fourteen in the m.f. band would become available which, with a fixed number of transmitters, would, as mentioned earlier, permit a reduction of co-channel interference. Naturally, this would result in an increase in adjacent channel interference but an analysis of present transmitter operations in Europe shows that even with 8kHz channelling, adjacent channel interference would still be less serious than co-channel. Studies carried out by the EBU show that with current types of receiver a channel spacing of 8kHz is the optimum for achieving maximum transmitter area coverage taking into account co- and adjacent-channel interference. Channel spacings of less than 8kHz result in excessive adjacent-channel interference.

It would be desirable for 8kHz channelling to be used throughout the whole of ITU Regions 1 and 3 because this would reduce inter-carrier heterodyne interference in those countries situated close to the Regional border. However, if Region 1 changes to 8kHz channelling and Region 3 elects to retain 10kHz spacing the inter-carrier heterodyne interference would be less serious than with the present 9 and 10kHz combination.

Restriction of Audio Frequency Bandwidth and Signal Processing

Many European broadcasters, including the BBC, are now restricting the audio frequency bandwidth of their m.f./l.f. transmissions to about 4·5kHz resulting, with present-day receivers, in a small but useful reduction of adjacent channel interference without perceptibly degrading reception quality in other respects. If the receiver selectivity characteristics were specifically designed to match the transmission system, i.e. the inclusion of a low-pass filter with a sharp cut-off beyond 4·5kHz, adjacent-channel interference would be almost entirely eliminated. Clearly, if *all* transmitters in the new Plan are restricted to a radiated audio frequency of about 4·5kHz the receiver manufacturers will be encouraged to design receivers taking full advantage of the transmission system. For quantity production, the cost of the additional filter circuits in the receivers would not be great and would be more than outweighed by the benefits of improved reception.

Most broadcasters already apply a limited amount of modulation compression to their programmes to reduce the effect of interference to the wanted channel but with many types of programme material this could well be increased bearing in mind that the high quality v.h.f./f.m. transmissions are more suited to programmes with a wide dynamic range.

Frequency Planning Methods

The Copenhagen Plan was prepared using rudimentary methods of trial and error and failed to produce an equitable distribution of frequency assignments between the participating countries. Since 1948, planning methods have become much more sophisticated and computers are now available to check the work of the frequency planners rapidly and accurately. It is probable that some form of the 'lattice' planning method will be adopted at the forthcoming Conference. This method, which was successfully applied in the preparation of the 1961 Stockholm u.h.f. television Plan, makes use of a geometrical lattice of equilateral triangles having sides corresponding to agreed co-channel transmitter distances. It is a method by which medium- and high-power frequency assignments are made to countries according to their area rather than their population but this means that countries such as the United Kingdom, with a high population density but a relatively small area would not have their needs met and some modification of the basic method to take into account of population density and needs will have to be worked out.

Another EBU proposal of merit is that, to facilitate frequency planning, transmitters should be categorised according to the type of service they provide. For example, sky-wave services should be accommodated at the higher frequency end of the m.f. band, the propagation characteristics at the higher frequencies being more suited to this type of service. On the other hand, transmitters used to provide large area groundwave coverage are more effective if assigned frequencies in the l.f. band or at the lower frequency end of the m.f. band. Some authorities feel that with the increasing popularity of television, the future of radio lies in local broadcasting which, in the main, requires transmitters of low power. Having limited interference potential, these low-power transmitters could be accommodated on international low-power frequencies which would be similar in many respects to the present two International Common Frequencies but would need to considerably exceed them in number.

The Prospects

It is evident that the introduction of such technical innovations as reduced and uniform channel spacing, restricted audio frequency bandwidth and more sophisticated planning methods, would lead to some improvement in l.f./m.f. reception even if, taking the most pessimistic view, countries insist upon maintaining their present programme services. A substantial improvement in reception can be made only if there is a significant reduction in programme requirements in the m.f./l.f. band. At the 1974–5 Conference, the United Kingdom must expect to have to work hard to avoid a reduction in its present medium- and high-power l.f./m.f. assignments as frequency planning is likely to be biased on an area rather than a population basis. There will also be difficulties over l.f./m.f. assignments for those countries whose services are extensively duplicated on v.h.f., and those countries should continue to aim for reduction in duplication by attracting listeners to the v.h.f./f.m. band. The receiver industry can play its part towards this by producing attractive v.h.f./f.m. receivers—preferably with push-button selection of preset channels.

It must be accepted that the high-quality radio services are widely available on v.h.f. and reception quality on l.f./m.f. is inevitably much lower in nearly all respects. L.f./m.f. broadcasting will, for economic reasons, continue to be the principal means of providing radio programmes in the new and developing countries. In the highly developed countries the future role of l.f./m.f. may well lie in the provision of services for the motorist, the casual listener and for external broadcasting by the sky-wave, all of which are services where intelligibility and freedom from interference are more important than wide-band, low-distortion audio quality.

Cable and Radio Programme Links in the BBC*

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Director of Engineering

1 Introduction

Since the earliest days of broadcasting, use has been made of both cable and radio techniques in providing the point-to-point links which are essential to the operation of a broadcasting organisation. The choice of technique has always been dependent upon a number of considerations including performance requirements, levels of technological development, economics, and political environmental factors. Since all these vary with time it is not surprising that the optimum methods of link provision have changed over the years. The BBC's experience in this field during the past fifty years[1] is outlined below, together with a brief look at future possibilities.

2 Permanent Sound Links

Simultaneous sound broadcasting by the BBC started in 1923.[2] At this time up to six transmitters could be fed with a programme originating in London and ordinary telephone circuits were employed, the London to Glasgow link for example being derived on some 800km of '170kg/km' open wire line. Overhead wires were soon replaced by underground Post Office cables, and until recently the audio signal for domestic and overseas sound broadcasting and for the sound component of television, continued to be carried from point to point in the United Kingdom almost exclusively by direct audio transmission over copper wires. The BBC currently uses some 50000km of sound programme circuits of this type, the vast majority of which are rented from the Post Office.

Carrier transmission offers an alternative method of providing sound programme circuits but crosstalk and noise problems have been experienced in deriving such circuits on existing carrier telephony systems. They were never intended for this purpose. As a consequence, baseband transmission over balanced pair cable of the type used for telephone circuits is still the most widely employed method. A rough breakdown of analogue cable methods in use for distributing BBC radio programmes is shown in Table 1.

TABLE 1
Summary of analogue sound link provision methods

Method of Provision	*Proportion of Circuits*
	%
Carrier Phantom (Phantom on two multi-channel carrier cables, which is possible because the 50Hz–12kHz band on these cables is not used by the carrier system)	35
Loaded Cables (loaded with 16mH coils at a 0·91km spacing giving a cut-off frequency of about 10·2kHz	25
Mixed Construction	25
Unloaded Cables	10
Carrier Programme Circuits, also known as Music-in-Band. (A few of these are fitted with compandors to make them suitable)	5

Limited use is made of the rebroadcast link technique, for distribution purposes, i.e. feeding a transmitter with a baseband signal received 'off-air' from a broadcast transmitter situated earlier in the chain. This technique has the merit of extreme cheapness, but requires that the second transmitter be sited within the service area of the first if reasonable quality is to be obtained. This restriction can be overcome by setting

* This article is based upon the 1973 Mildner Lecture given by the author at University College London, under the auspices of The Department of Electronic and Electrical Engineering.

up a separate receiving site in the service area of the first transmitter and then extending the signal to the second by means of a cable or microwave link, but the economic advantages then become more questionable. All rebroadcast links are open to co-channel interference and their reliability in this respect becomes less as more and more broadcast transmitters are brought into service both in this country and abroad. The technique is of course extremely valuable in providing standby feeds to maintain a service during failure of the main feed.

Isolated pockets of listeners are served by low-power v.h.f. relay transmitters which receive their inputs 'off-air' from another transmitter. In these cases, however, the received signal is not demodulated but is frequency changed to the required carrier frequency, amplified and radiated. These installations are known as transposers.

The advent of stereo together with the growth in the number of home 'hi-fi' installations has recently resulted in a requirement for improved performance in many point-to-point sound links. Existing analogue transmission equipment does not appear to be capable of meeting this requirement on circuits of more than a few kilometres in length. Digital techniques on the other hand offer the possibility of substantially distortionless transmission which is independent of distance. These techniques are employed in pulse code modulation terminal equipment which has been developed by the BBC to provide thirteen high-quality, matched programme circuits on one bearer channel. This equipment was first put into service to feed the Wrotham f.m. transmitters in September 1972 and it is hoped that by the end of 1975 most of the BBC's high-power transmitters will be fed by this means. The audio performance of a channel is summarised in Table 2.

TABLE 2
Performance of 13-channel p.c.m. system

Frequency responses	40 Hz – 14·5 kHz ±0·2 dB w.r.t. 1 kHz Typically –1·5 dB at 30 Hz and 15 kHz
Signal/noise ratio	72 dB (peak signal/peak weighted noise)
Harmonic separation	66 dB at 1 kHz

The 13-channel equipment was designed for 6·336 Mbit/s bearer circuits. At present, standard 5·5 MHz analogue links designed for 625-line monochrome television transmission purposes are being used as bearers for the sound signal bit stream. These links may utilise either coaxial cable or s.h.f. radio techniques.

It is convenient to use television-type links as bearers, since most v.h.f. f.m. transmitters are co-sited with 625-line television transmitters and a well-integrated transmission system thus results. The use of television links is not mandatory, however, and in the long run it may well be more economical to use other types of bearer, e.g. 4 MHz carrier cable circuits, or 6·336 Mbit/s channels in a future p.c.m. 'digital highway'.

Digital techniques can also be applied to the distribution of sound programmes over routes where as few as two channels may be required. In such cases the bearer would probably be provided by the first level in the proposed hierarchy for p.c.m. systems, i.e. a 2·048 Mbit/s channel (the BBC is currently investigating techniques which may enable up to six high-quality circuits to be squeezed into 2·048 Mbit/s).

The sound component of the television signal is now distributed to most of the BBC's 625-line transmitters by means of a p.c.m. signal which is sent during the line-synchronising pulse interval of the video signal.[3]

The sound is sampled at 30 kHz and the digital numbers representing these samples have to be packed into the sync pulses which occur at the rate of about 15 kHz. Alternate samples are delayed by half a line period, and are inverted and interlaced to attenuate the d.c. component. Each sample is described by an 11-bit number and pilot-tone-controlled companding is used to achieve the approximate equivalent of 13-bit sampling.

This BBC-developed technique results in a high-quality sound channel, with a performance substantially independent of circuit length, at a cost in bandwidth no greater than that of the video channel. The audio performance of a sound-in-sync channel is summarised in Table 3.

TABLE 3
Summary of Sound-in-Syncs Audio Performance

Frequency response	50 Hz – 13·5 kHz ±0·7 dB w.r.t. 1 kHz
Signal/noise ratio	64 dB (peak signal/peak weighted noise)
Harmonic separation	50 dB at 1 kHz

With sound-in-syncs, there is a greater chance of the sound failing at the same time as the vision, but reserve arrangements such as rebroadcast standby receivers are provided to cover emergencies.

Sound for the more distant 625-line transmitters and for the 405-line transmitters is in the main provided by rebroadcasting the analogue signal received from an 'up-stream' transmitter.

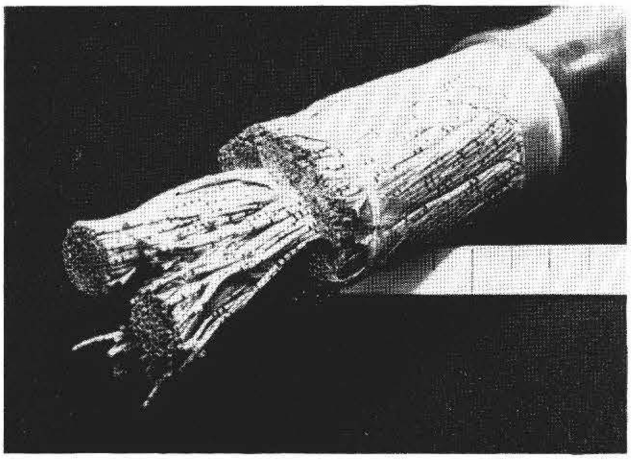
3 Temporary Sound Links

The vast majority of temporary sound links were originally provided on telephone circuits, but it is of interest to note that a radio link was first used for an outside broadcast in 1923 and that 1925 saw the use of m.f. radio for outside broadcast from a moving train and from an aeroplane.

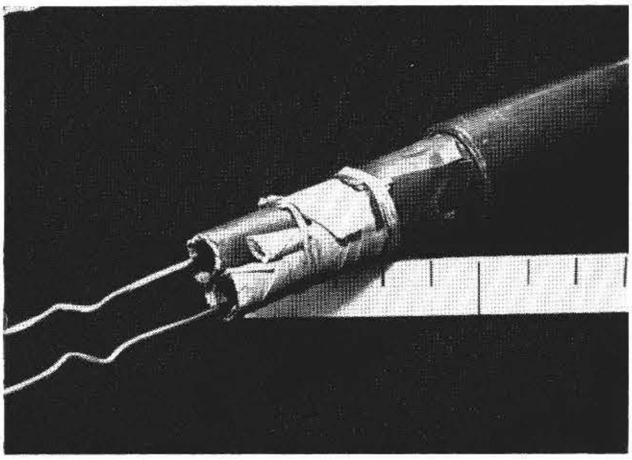
Most of the sound circuits which are set up for outside broadcast purposes these days are derived on telephone cables, using specially equalised pairs. The performance required of the circuit is to some extent dependent upon the nature of the programme material to be transmitted and the difficulties involved in setting up. If loaded cables only are available it may be necessary to adopt phantom or bunching*

* In bunching the effect of loading is cancelled out by connecting together the two wires of a pair and using it to form one leg of the circuit, the other leg being formed from the other part in a cable quad.

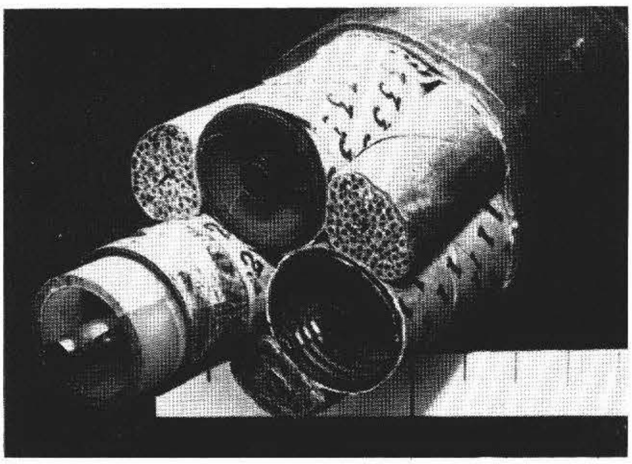
Fig. 1 Cables—past, present, and future



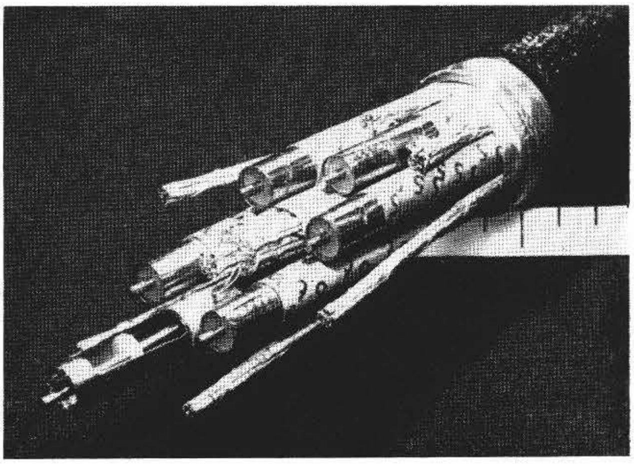
(*a*) 816 pair telephone cable (1936)



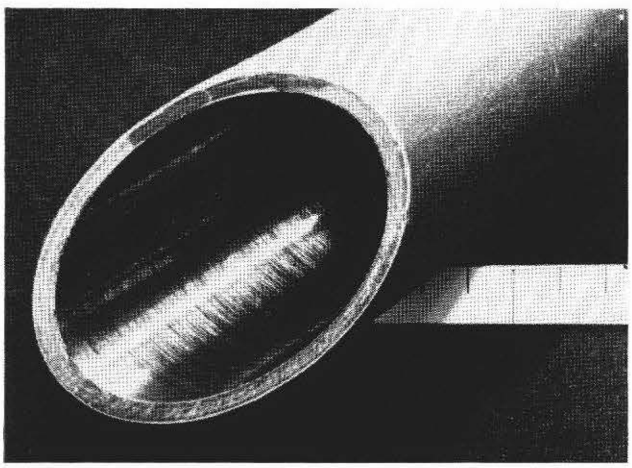
(*b*) 25 mm television screened balanced pair (1936)



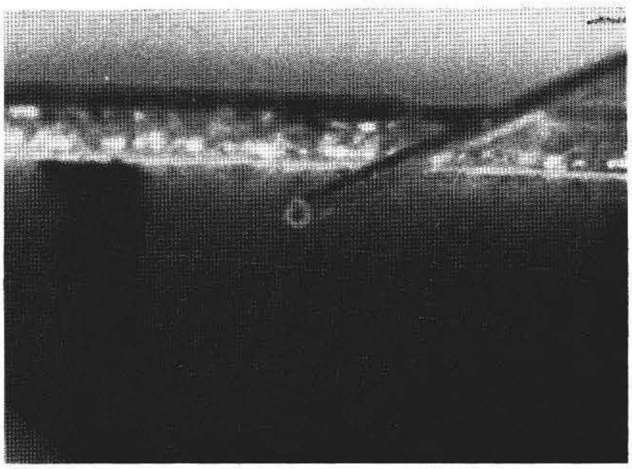
(*c*) 25 mm coaxial pair cable (1947)



(*d*) 9·5 mm coaxial pair cable (1947)



(*e*) 50 mm experimental waveguide (1972)



(*f*) 60 μm experimental liquid filled optical guide (1973)

The vertical stripe on (*f*) is part of one of the dividing lines on the centimetre scale shown with the other cables techniques. When circuits are required at very short notice, use is made of the public telephone network or of mobile v.h.f. transmitters, but in the former case the results are of lower than normal broadcast quality and reliability. V.H.F. links are also used when the provision of a line would be particularly difficult or costly.

The sound-in-syncs method (in some cases in a so-called 'ruggedised' form) is also used for the sound component of some television outside broadcasts, although many existing mobile television radio links contain an analogue sound channel which is still often used.

4 Permanent Television Links

When the BBC began its high-definition television service in November 1936, the only source of programme and the only transmitter were located immediately adjacent to each other in Alexandra Palace. No interconnecting transmission link was therefore required. However, spurred on by the desire to televise the 1937 Coronation the BBC arranged with EMI and the Post Office to construct and lay a special balanced pair cable (see Fig. 1b) from some of the more important ceremonial sites in London (e.g. Westminster and Hyde Park Corner) through Broadcasting House to the Alexandra Palace transmitter.

The attenuation of this cable was 5dB/km at 3MHz, but with the equalising and amplifying equipment then available a bandwidth of only 2MHz could be achieved over the 13km to Alexandra Palace. Up to 2MHz, the spread of response was ±0·5dB.[4] The provision of this pioneering circuit marked the beginning of a long and continuing period of co-operation in the field of point-to-point television transmission between the Post Office and the BBC.

On restarting the Television Service in 1946, the pre-war London cables were repaired; the repeater equipment at Broadcasting House, which had been destroyed by bombing, was replaced, and the special balanced pair cable extended westwards to the important OB area in West London (Royal Albert Hall, Olympia, etc.). Until the introduction of the 635-line standard in 1965 (when it was replaced by a 9·5mm coaxial), this cable and associated equipment gave excellent service as part of permanent OB collection facilities.

Soon after the end of the war plans were made by the Post Office, in conjunction with the BBC, to link Alexandra Palace to the site of the Birmingham area transmitter by means of a cable. This London–Birmingham system was originally planned on the basis of a 1000-line picture standard, requiring the cable characteristics to be controlled up to 26MHz! A special 25mm diameter coaxial pair was developed and a composite cable produced containing two of the special 25mm pairs, four 9·5mm coaxials (the 9·5mm being a standard for multichannel telephony), and a number of simple twisted pairs for control and supervision purposes.

By installing repeaters at 3-mile intervals it was considered that an adequate noise performance could be obtained over a 26MHz bandwidth. It soon became obvious, however, that a 1000-line system was not practicable with the limited television camera resolution which was possible then.

The London–Birmingham cable circuit was therefore redesigned for 405-line working, using a carrier system employing a 6·2MHz carrier frequency with amplitude modulation, full lower sideband and a 0·88MHz vestigial upper sideband (see Fig. 2b). Carrier operation was employed since techniques for transmitting baseband video signals over long coaxial cables had not been developed at that time. As the bandwidth requirement on the cable had dropped from 26MHz to 7MHz, it was found that only every fourth repeater need be installed; the resulting 12-mile spacing gave a quite adequate noise performance.[5] Distortion figures for this link were typically within ±1·0dB for gain over a 3MHz video bandwidth, 6 per cent non-linearity and 56dB peak signal to r.m.s. noise ratio.

The majority of the equipment for this system was designed and built at the Post Office Research Station. It was a stable, reliable system which provided a link in each direction for nearly twenty years, and of which the designers could be justly proud.

The next step in the expansion of the television service was to extend the distribution network from Birmingham to the north of England, for a transmitter in the Manchester area, and from London to South Wales, for a transmitter near Cardiff. To achieve this, and to enable contributions from these areas to be made to London, the BBC asked the Post Office to provide additional two-way links capable of carrying 405-line television signals.

It would have been very advantageous technically if these two circuits had been engineered in the same way as the London–Birmingham route, that is using an identical carrier system, so that non-demodulating working could have been used. Unfortunately the expense of the special 25mm coaxial pairs was not justified now that the dream of a 1000-line system had faded. Standard 9·5mm coaxial cables therefore had to be used, and with these the 7MHz upper frequency limit of the London–Birmingham system could not be transmitted without prohibitively close repeater spacing. A new system was therefore designed with an upper frequency limit of 4MHz, which would enable an adequate noise performance to be attained with repeaters spaced at 9·6km intervals.[6] However, an upper frequency limit of 4MHz meant that a simple single modulation arrangement was not now possible, the input video signal spectrum of 3MHz and the line carrier spectrum limited to 4MHz would of necessity overlap (see Fig. 2c).

The great failing of this system was that harmonics of the 1MHz carrier fell within the transmitted band of 0·5–4MHz. Any non-linearity in the line equipment produced 2, 3, and 4MHz components which were demodulated at the receiving terminal into 1, 2, and 3MHz interference patterns. For many years these patterns were a constant reminder of the price paid for economy; no tears were shed when the circuits using the system were withdrawn from service in the late 1960s.

Further expansion of the distribution system was limited by lack of cable duct space so the Post Office decided that a two-way multiple-hop s.h.f. radio link would be the most economical method of extending the network from Manchester to the first Scottish transmitter. This resulted in the first s.h.f. link in the United Kingdom to be designed exclusively for long-distance television transmission.[7] The transmitted frequencies were in the 4GHz band, non-demodulating repeaters with an intermediate-frequency of 70MHz being used. The modulation system was f.m. Typical performance figures were within ±1·5dB gain/frequency response, 25 per cent non-linearity and 58dB peak picture to r.m.s. noise.

The main shortcomings of this link were poor linearity (not at all uncommon on early s.h.f. circuits) and fading, the latter due to some very long hops (over 65 km) between repeaters sited near the north-east coast—when a sea mist blew inland prolonged fading occurred! In spite of these failings, however, this trail-blazing link opened a path followed by virtually all later circuits, that of multihop, frequency-modulated s.h.f. transmissions with non-demodulating repeaters, for point-to-point television.

Further expansion of the 405-line distribution network was by means of the rebroadcast link techniques previously described.

The disadvantages of these techniques are that the link can be one way only; the signal received at the second transmitter has already been through the distortion-prone 'transmitter/receiver mill' once; and that it is open to all the types of interference experienced at the edge of transmitter service areas. A number of transmitters covering medium population areas were fed in this manner, but the rising level of interference over the subsequent years from newly introduced broadcast transmitters forced the BBC to have many of these links converted into fully engineered s.h.f. circuits.

The subsequent widespread use of s.h.f. radio channels for point-to-point transmission has relegated cable circuits into the background for long-distance inter-city television circuits. However, a very important step forward in the field of short-distance television transmission took place in the mid-1950s, with the development of direct video working over coaxial cables.

At about the same time the BBC developed a simple short-distance system using a 15 MHz amplitude-modulated carrier which could be transmitted on the same coaxial circuit as the direct video signal. This permitted two vision circuits to be obtained over one coaxial cable (see Fig. 2d). These develop-

Fig. 2 Cable System Spectra

(*a*) Baseband for 405-line signals

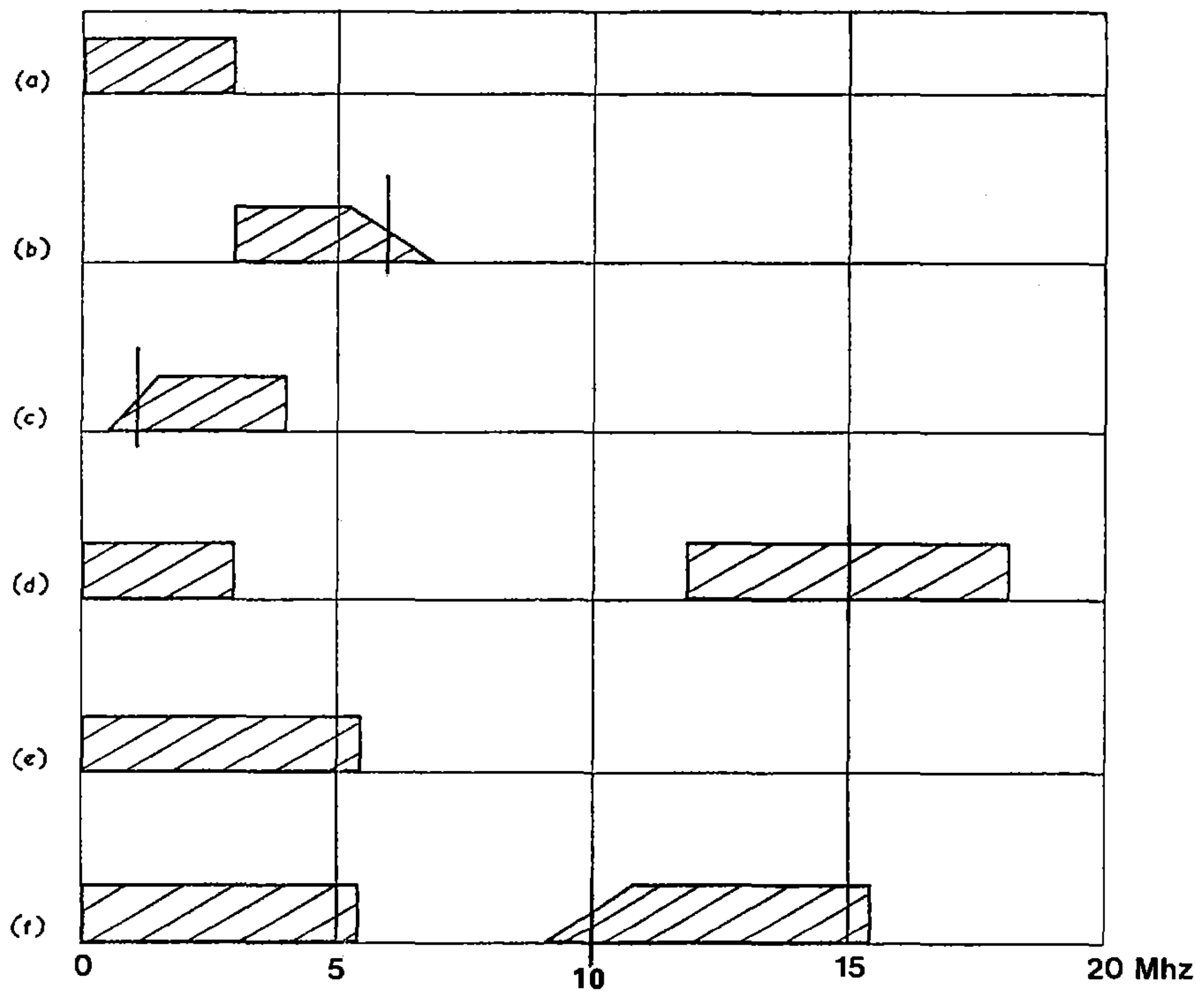
(*b*) 3–7 MHz carrier for 405-line signals

(*c*) 0·5–4 MHz carrier for 405-line signals

(*d*) Baseband plus 15 MHz carrier for 405-line signals

(*e*) Baseband for 625-line signals

(*f*) Baseband plus 10 MHz carrier for 625-line signals



ments enabled considerable economies to be made in both terminal equipment and inter-connecting coaxial circuits over distances up to about 16km. For example the BBC uses about twenty permanent vision circuits between its Television Centre in West London and the Television Network Switching Centre at Broadcasting House.

While these developments were taking place, a considerable advance was made in the method of testing and specifying vision circuit performance. This was the now almost universally adopted use of sine squared pulses and the 'K' rating system of assessing the linear performance of a circuit, developed by Dr Lewis and his colleagues at the Post Office Research Station.[8]

Both the direct video circuit and the 15MHz carrier circuit were capable of 'K' rating performance of 0·5 per cent and the non-linearity performance was 3 per cent. The signal-to-noise ratio of the direct video circuit at around 60dB was generally some 10dB better than that of the carrier circuit, because of the lower cable loss at video frequencies.

Apart from minor extensions and improvements, the 3 MHz 405-line, point-to-point network had reached its full coverage by 1960, providing a reliable high-quality distribution service to virtually all the populated areas of the United Kingdom; and allowing all major sources of programmes to contribute to the network.

By the early 1960s development of the 405-line picture system had reached its performance limit and it was felt that the provision of any new service, including the possibility of colour, warranted a change to a higher line-scanning standard and the European 625-line standard was adopted. This required a 5·5MHz bandwidth instead of the 3MHz of the 405-line standard, and demanded a much higher linearity and better noise performance for colour. It meant that the whole of the BBC's permanent point-to-point network would either have to be replaced or modified to meet the more exacting standard required. Furthermore, the Government's agreement to the setting up of the BBC's second television service (BBC-2) at this time, meant that another new network would be required to distribute this programme throughout the United Kingdom.

Development of wideband frequency-modulated s.h.f. channels had now reached a point where the various design parameters (e.g. degree of pre-emphasis, intermediate frequency, s.h.f. amplification and demodulation at intermediate repeaters) could be established. A high degree of standardisation was therefore achieved in the provision by several manufacturers of the large number of new links which were required. These modern s.h.f. links are capable of very good television transmission performance, 'K' rating of better than 1 per cent, monochrome non-linearity of 1 per cent, differential gain of 2 per cent, differential phase of 1° and a signal to r.m.s. noise ratio of 48dB, being readily achieved.

The replacement of the vast majority of the BBC's 405-line network with 625-line colour standard links, was completed in only five years—a most notable achievement on the part of the Post Office. 405-line signals to feed the existing BBC-1 v.h.f. transmitters are now obtained from BBC-developed line-store standard converters. The 625-line inputs for these converters are obtained by off-air reception of the nearest BBC-1 u.h.f. transmitter when this is not co-sited with the v.h.f. 405-line transmitter.

One disadvantage that has resulted from the integration of the Post Office vision links with their telephony wideband systems has been that 'through-intermediate frequency' working has not been possible at network terminal points such as Birmingham, Manchester, Bristol, etc. This is due to the switching requirements of the telephone links, and the use of common equipment. 'Through-i.f.' working could reduce the distortion which inevitably results from the demodulation and remodulation processes.

The new radio links, with the exception of a few short spurs in the 2GHz and, latterly, 11GHz bands, operate almost exclusively between 6 and 7GHz; with the main repeater amplification at an intermediate frequency of 70MHz, and standard CCIR television pre-emphasis in the modulators. Originally, the equipment used valves, but in recent years considerable replacement has been made with solid-state equipment.

Although the backbone of the BBC's vision distribution and contribution network is Post Office installed, and maintained, the BBC provides a number of its own s.h.f. links to fill in sections which the Post Office do not provide. These conform in general to the same technical details as the modern Post Office links, and meet the same performance specifications. The total length of 625-line colour capable point-to-point links is now some 8000km. Some 1600km of u.h.f. rebroadcast links are also used on spurs from the main backbone while small pockets of viewers are served by transposers. Local links, particularly intra-city ones, continue to be derived on co-axial cables in the main. Baseband transmission is used with, in some cases, the addition of a 10MHz channel (see Fig. 2f).

Table 4 shows distortion limits for some 625-line television signal parameters. The figures in the right-hand column are for a system including an RGB Coder, a Studio Centre, a third generation video tape, a chain of point-to-point links, two rebroadcast links, and a final transmitter.

TABLE 4

Distortions Having a High Probability of not being Exceeded

Parameter	*Chain of point-to-point links (e.g. London–Belfast)*	*Camera Output to farthest Transmitter Output*
Random noise unweighted	−49dB	−36dB
Non-linearity luminance	6%	27%
Differential gain	± 8%	±26%
Differential phase	± 8°	±25°
Chrom. lum. crosstalk	± 9%	±21%
K (bar)	3·5%	9%
K (2T pulse)	2·5%	11%
K (2T pulse/bar ratio)	2·5%	8%
Gain inequality	±14%	±24%
Delay inequality	±40ns	±109ns

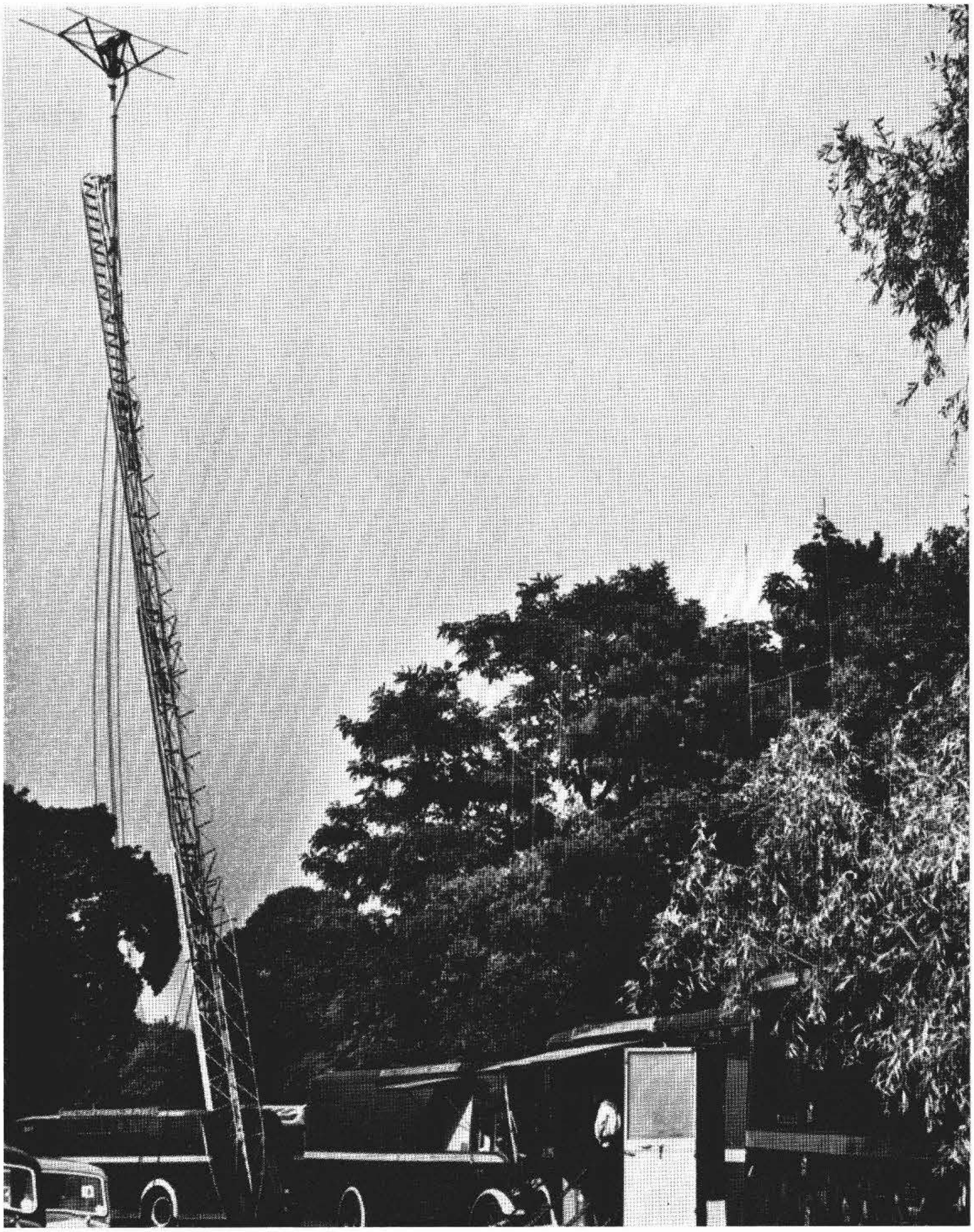


Fig. 3 Outside Broadcast Unit (circa 1947) comprising, from left to right, Aerial Vehicle, Transmitter Van, Mobile Control Room, and Power Van

5 Temporary Television Links

Mention has been made previously of the balanced pair television cable laid in London in 1937. To extend the 'catchment area' of this cable the BBC investigated the feasibility of transmitting television pictures over ordinary Post Office telephone circuit pairs. Tests showed that this form of transmission was perfectly practicable over limited distances. Circuits in excess of 1·6 km could be set up in favourable circumstances before a repeater became necessary. The limitations were (and still are) the interference suffered from external sources, e.g. telephone selectors; and the considerable losses at the higher frequencies. Normally up to three equalised repeater sections could be used before the overall performance began to fall below acceptable standards. Typical distortion figures for a short-distance telephony pair circuit were within $\pm 1{\cdot}0$ dB for gain in a 3 MHz bandwidth, 6 per cent non-linearity, and 58 dB peak-to-peak picture to r.m.s. noise ratio. It is worth noting that the techniques developed then are still in use today, suitably modified for the wider bandwidth and other more exacting requirements that must be satisfied for 625-line colour transmission.

To cover television outside broadcasts from more distant sites radio links were required. Two mobile transmitter terminals were therefore built in the late 'thirties. Each terminal

Fig. 4 General view of modern portable link terminal complete with 1·2 m diameter aerial

consisted of a 12-ton vehicle housing a 1 kW vision transmitter which operated in the amplitude-modulation mode in Band I. A second similar-sized vehicle accompanied the transmitter vehicle—this contained a 3-phase generator to supply power. The transmitter terminal was completed by a fire-fighting vehicle-mounted ladder: this ladder could be extended to 33 m and was used to support the transmitting aerial, a simple dipole and reflector.

The receiver terminal consisted of a complete bay of equipment and was a permanent installation. The range of these was some 32 km.

In the late 1940s there was a demand for the extension of Outside Broadcasts, coincident with an expansion of broadcasting. This latter forced the OB equipment out of Band I. Consequently a Band III (190 MHz) system was introduced into service. Again the large power generating equipment and fire-fighting ladder formed part of the transmit terminal.

The transmitter in this case had an output power of 350 watts and the system had a range of about 40 km.

A 25-watt system was also used: this was operated with Yagi aerials, and had a range of 8 km—this was eventually increased to about 19km by the use of a BBC-designed receiver.

Also in the late 1940s frequency modulated microwave (s.h.f.) equipments started to become available: 4GHz equipment was introduced into service in 1952 and 7GHz in 1959.

The equipment used the now familiar parabolic 'dish' aerials, and a standard size of 1·2m was decided upon as being the largest which could be conveniently handled.

The high gain and directivity of these aerials (31–36dB depending on frequency) made path lengths of 65km or more a practical possibility. The low output power (circa 1 watt) meant low prime power requirements; and smaller power generators, capable of being towed by the radio-link vehicle, were introduced into service.

This radio plant was truly portable, and by setting up one or more repeater sites, circuit lengths of over 160km could be established.

As a result of this type of equipment being available, the BBC set up an OB radio link capability on a nation-wide basis, with radio-link teams and equipment at all the major BBC centres. Special vehicles were constructed to house the equipment and also to provide control facilities and shelter for the staff: these repeater units could be rapidly set up at any locality to which a vehicle had access.

An important extension of the OB system was the establishment of a number of permanent OB reception points throughout the country. These were sited at a number of BBC transmitting stations to take advantage of existing masts. Two parabolic dishes were mounted on the mast at a height of about 160m above ground level. These aerials were capable of being rotated about the vertical axis, and thus between the two aerials, complete 360° coverage could be achieved. Permanently installed s.h.f. receivers were provided on the mast, close to the aerials, and the tuning of these receivers and the rotation of the aerials was remotely controlled from ground level. These systems, because of the mast height, made possible single paths of up to 80km in length: because of the geographical locations of broadcast transmitters, most OB points were within 80–160km of a reception point. Many of these installations are still in service with the radio plant replaced by solid state equipment.

In the 1960s a number of events occurred to change the format of the radio links: these were

(i) the loss to the BBC of the 4GHz band for OB use. This meant that the majority of the replacement equipment had to operate in the 7GHz band.

(ii) an allocation of spectrum space at 12GHz: this was a band not previously used by the BBC and a pair of links manufactured in France were acquired to gain operational experience at these frequencies.

(iii) and (fortunately in view of the above) a large proportion of the radio plant was approaching the end of its useful life.

Because of these factors, a programme of replacement of radio links was commenced, the opportunity being taken to bring about other changes: the semi-conductor 'state of the art' was such that OB radio links employing solid-state techniques throughout were possible. Also, with solid-state equipment, power requirements were low enough to introduce battery operation under emergency and short-term operational conditions. The opportunity was also taken to make the equipment fixed-frequency instead of tuneable as had been the previous practice. This simplified the operation of the equipment and increased the performance stability.

The improved and more stable performance made it possible to include a high-quality sound channel with the vision link: this was added to the system by means of a frequency modulated sub-carrier above the upper frequency of the video channel.

The final change in operational technique was concerned with spare equipment. In the past, all equipment was used with 100 per cent spare units at each OB site. In the event of a fault in an operational unit, it was removed and the spare fitted. Such an arrangement could take quite a considerable time—if the working unit was at the top of a 32m tower, for example. It was therefore decided that the main equipment and the spare should be combined into a common aerial. Thus a single-hop link now has two transmitters combined into one aerial, each transmitter operating on a separate frequency; at the receive terminal, two receivers each on the appropriate frequency are fed from a single aerial. The performance of a modern microwave link is summarised in Table 5.

TABLE 5
Typical Performance of 12GHz Solid-state OB Equipment

Parameter	*65km (1 hop)*	*160km (3 hop)*
K rating	1%	2%
Signal/Noise ratio		
unweighted	56dB	52dB
weighted	64dB	61dB
Differential gain	1%	2%
Differential phase	1°	2°
Chrom.–Lum. Inequality		
gain	3%	3%
delay	40ns	40ns

Transmitter Output Power = 400mW
Receiver Noise Figure = 11dB
Gain of 1·2m Aerial = 40dB

Parallel working, combined with battery emergency power, and the reliability of solid-state equipment has opened up the possibility of leaving repeater points unmanned whilst in operation.

Clearly an OB site is chosen for the programme it offers and not for its suitability as a radio link point—so consequently the site may be in a valley, or surrounded by high buildings. To overcome this problem, two other pieces of equipment are often needed.

(i) a suitable structure to enable the transmitter aerial to be raised clear of any local obstruction.

(ii) a radio link system to establish a short 'starter' circuit to the nearest appropriate high point to connect to the 'main line' circuits previously described.

The structure consists of a vehicle-mounted telescopic mast capable of supporting a parabolic aerial and holding this aerial to a stable bearing consistent with the narrow beam-widths involved. The BBC have about twelve such towers with an extended height of 20m and two towers with an extended height of 32m. They are obviously the modern equivalent of the Band I aerial used before the war.

The 'starter' link equipment was more of a problem because despite the use of mobile towers, it was still not always possible to obtain a path which was completely in free space. This usually excluded the use of s.h.f. and so frequencies in the v.h.f. bands were used initially, i.e. in the band 30–300MHz.

Once again, however, the extension of broadcasting after the war made v.h.f. unusable for this purpose. Consideration was therefore given in mid-1950s to the use of u.h.f. (300–3000MHz). No commercial equipment was available and so the BBC's Designs Department undertook the necessary development. Frequency modulation was employed because of its advantages in terms of signal-to-noise performance and its resistance to man-made interference. This system was introduced into service in the late 1950s in Band IV and later in Band V.

Quite apart from the use of this equipment for starter links, it proved very effective for roving conditions where cameras were operated in moving vehicles, in motor boats, and later in helicopters. The BBC continues to use these frequencies with solid-state equipment, again of BBC manufacture. Once again, however, the OB spectrum is being invaded by the broadcast requirement, and with the spread of the u.h.f. transmitter programme the use of Band V is becoming increasingly difficult. Frequency allocations have therefore been obtained in the 2GHz band and solid-state equipment acquired. This band is already proving very effective both as a starter link and in the roving mode.

6 The Future

6.1 Digital Techniques

Already digital transmission has been found to be necessary for the transmission of high-quality sound signals. The same necessity is not yet evident in the case of video transmission, but it will probably arise from a desire for reduced distortion, reduced maintenance and integration of television into the 'digital highways' which will be set up in the UK.

Prototype coders and decoders for digital television signals have been developed by the BBC and brief tests, in co-operation with the Post Office, have already been carried out on a short length of waveguide.

One of these CODECS is shown in Fig. 5.

Units of this type are also being used for R.&D. purposes by COMSAT, the RAI, and the IBA. In 1974 it is hoped to carry out more comprehensive trials on a medium-length 4·4mm diameter co-axial cable digital circuit. The bit rate of the present CODECS is some 120Mbit/s but work is being undertaken to reduce this by, hopefully, a factor of at least two.

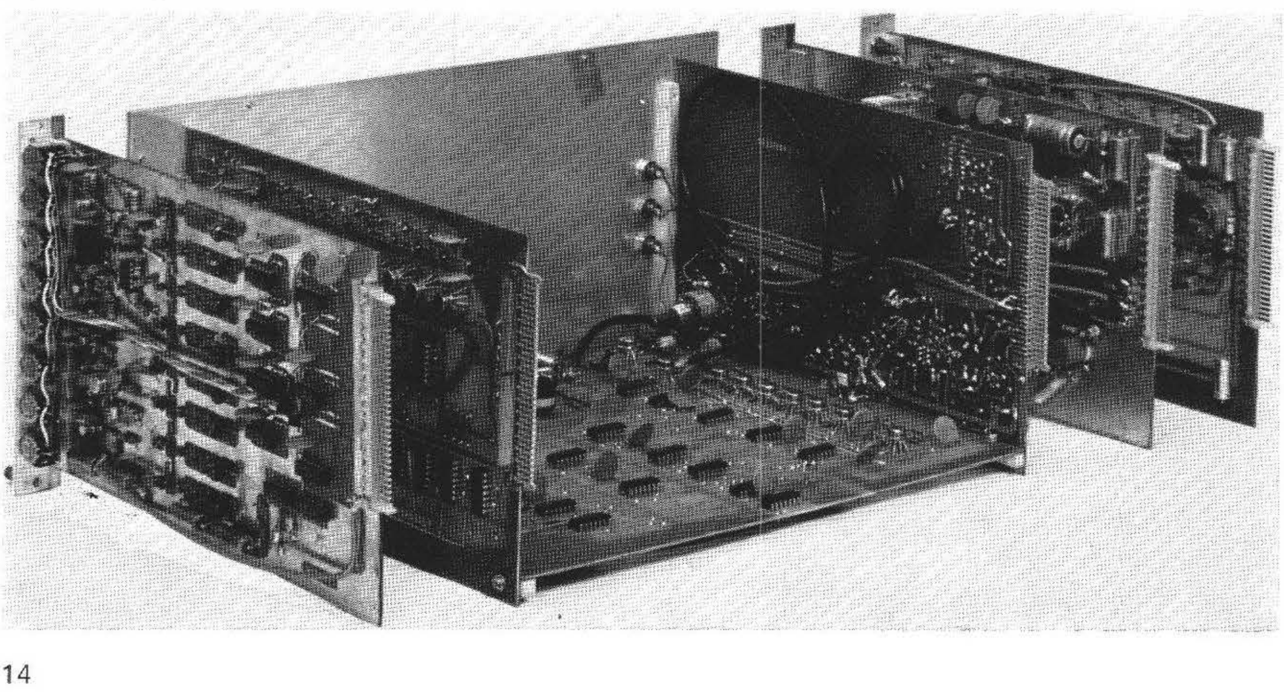
6.2 Guided Systems

The Post Office is currently installing a 60MHz bandwidth analogue coaxial cable system on the London–Birmingham–Manchester route. This will have a capacity of 10800 telephone channels on each tube, but it is proposed that, where required, these could be replaced by up to four television circuits derived by conventional f.d.m. techniques utilising amplitude modulation.

The first digital highways are likely to use conventional coaxial cables repeatered at approximately 2km intervals. It is estimated that a bit rate of 120Mbit/s will be possible on a 4·4mm diameter and that 500Mbit/s will be possible on 9·5mm coaxial.

The next step in guided digital systems is not obvious at this

Fig. 5 Prototype Television PCM CODEC



time. One possibility is waveguide transmission (see Fig. 1e). The Post Office plan to have an experimental waveguide system operating over a distance of some 25 km, by 1976. The bandwidth available on this system will be of the order of 70 GHz. Probably the most significant difficulty with waveguides is that of finding a route which involves a minimum of undulation in both vertical and horizontal planes.

A second possibility is optical fibre transmission. Various forms of fibre are under development and both lasers and light-emitting diodes have been used as transmitters. There are several attractive features to this technique, e.g. the fibres should be cheap to manufacture, they can be bent at will and multiplexing and de-multiplexing can be achieved merely by binding individual fibres into a bundle or cable at one end and unbinding them at the other end of the route. Television signals have already been transmitted experimentally over nearly a mile of experimental fibre, developed at Southampton University (see Fig. 1f), but the service date of a practical system may be some ten years off.

6.3 Free-space Terrestrial Systems

Radio links are already operating in the UK at frequencies up to 12 GHz, and there are plans to go higher still. One of the disadvantages is that the path loss increases with frequency and in particular these s.h.f. channels are vulnerable to rain and fog. Nevertheless the pressure on frequencies increases and so it is planned to use a much closer spacing of repeaters than is at present used in the existing bands. These repeaters would be quite small and it is envisaged that for permanent links they would be mounted on supports not greatly different from telegraph poles.

Looking farther ahead, experimental links have been operated over a distance of a few kilometres at frequencies near to those of visible light. In these the conventional transmitter is replaced by a laser.

6.4 Satellites

The successful launching of the Canadian distribution satellite Anik has demonstrated that it is technically possible to do away with national terrestrial point-to-point programme links of the type now in use. In addition it is probable that in a few years it will be technically possible to broadcast direct to viewers from a satellite, in which case existing broadcasting transmitters could also be dispensed with. However, it seems unlikely that either of these eventualities will occur in the UK in the near future, partly because of the very high percentage of the population satisfactorily served by existing networks and transmitters, into which very considerable sums have been invested, and partly because of the importance of local and regional broadcasting which could not be handled in any practical way by satellite. Three more television services (one, immediately, on u.h.f. and two, in the 1980s, on v.h.f.) could be provided at existing transmitting sites more cheaply than by satellite.

7 Conclusions

In the foreseeable future the existing forms of terrestrial transmitter look like continuing to be the main link with listeners and viewers in the United Kingdom. In the longer term some transmitters may be located 37000 km out in space, i.e. there may be direct broadcasting into the home via satellite but only if the additional programmes make it worth while for the viewer to pay the extra costs. Alternatively, or perhaps additionally, cable television, i.e. the distribution of television direct to the home by cable, will grow, but again only if worthwhile new forms of programme can be forthcoming. For the foreseeable future it seems likely that most parts of the UK will be served by orthodox broadcast means and that there will be a continuing need to provide high quality links to the transmitters.

As has been shown, the distribution to the transmitters of both sound and television programmes began with cable. In the case of sound the distribution is still mainly via cable though the stereo requirements are forcing greater use of s.h.f. wideband circuits. In television the early use of cable soon gave way to radio mainly because the necessarily wide bandwidths could more economically be provided in that form.

It is quite clear that the quality advantage lies with digital rather than analogue distribution and that television as well as sound must eventually be distributed to the transmitters in that form. The various guided-wave systems show considerable promise of making this possible. Thus the indications are that having begun with cable and moved to radio distribution there will eventually be a return to cable—but remarkably different kinds of cable and techniques from those with which broadcasting began.

8 Acknowledgment

An article of this nature must gather information from many sources and I should like to acknowledge with thanks the help of my colleagues David Savage, Frank Rice, and David Grant for their contributions to the writing of this article.

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Quality Criteria, Protection Ratios, and Protected Field Strengths

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1 Introduction

This article reviews the factors limiting the quality of sound broadcasting services in the l.f. and m.f. bands. It gives an introduction to the various measurements and standards involved in the process of planning, with the object of controlling the level of noise and interference to services and thereby ensuring adequate and predictable coverage areas.

2 Limitation due to Noise

In the absence of interference from other stations, a service at m.f. or l.f. will be limited at the range where, as the wanted signal level diminishes, the noise level from natural and man-made electrical interference gives rise to a signal-to-noise ratio lower than that acceptable. We can first consider the levels of noise from different sources, taking 1 MHz as the standard frequency for m.f.

2.1 Atmospheric Noise

CCIR Report 322 indicates noise levels in a series of maps, as a function of time of day and season of the year.[1] The levels are very variable in the m.f. band not only with time but also geographically; for many areas a noise level at 1 MHz of 18 dB (μV/m) in 1 kHz is sometimes reached. It is sometimes advocated that the contributions of noise and interference should be equal at the limit of service, corresponding to the lowest field that can be protected. In view of the variable nature of noise and, as we shall see later, the greater difficulty presented in a plan by night-time interference from other transmitters, it seems more realistic to use sufficient power to ensure that, at night at any rate, the main limitation is set by interference, and that noise hardly contributes at all to the loss or degradation of service. Therefore, if there is a case for controlling the total interference to, say, 30 dB below signal level, as discussed later, we might say that a signal level giving a signal-to-noise ratio of 35 dB in the absence of interference would represent a reasonable lowest value signal strength at the limit of service, assuming that atmospheric noise is the main source of noise. If we take this ratio of 35 dB, a noise level of 18 dB (μV/m) in 1 kHz and a receiver noise bandwidth of 5 kHz (7 dB more noise than in a 1 kHz band) we arrive at $18 + 7 + 35 = 60$ dB (μV/m) as a typical m.f. field for which the signal-to-noise is satisfactory for most of the time. This figure would for example give 35 dB signal-to-noise ratio for about 95 per cent of the time in temperate latitudes (e.g. Europe). As an example of effects at lower latitudes the proposals for minimum field strengths to be considered for an m.f. service at 1 MHz at the 1966 African LF/MF Broadcasting Conference are given in Table 1. In this case the figures were based on achieving 40 dB signal-to-noise ratio for 90 per cent of the time; this leads to a $40 + 7 = 47$ dB difference between the final two columns.

TABLE 1

Zone	*Noise in 1 kHz Exceeded 10% time, 1600–2400 hours*	*Minimum Signal Strength for Protection*
A (Africa N of 20° N, plus E and W areas N of 5° N)	14 dB (μV/m)	61 dB (μV/m)
B (Remaining area between about 20° N and 15° S)	27 dB (μV/m)	74 dB (μV/m)
C (Africa S of about 15° S)	20 dB (μV/m)	67 dB (μV/m)

CCIR Report 322 also indicates that relative to the atmospheric noise level at 1 MHz, that at 1·55 MHz is 3 dB less and that at 500 kHz 6 dB greater.

2.2 Man-made Noise

Man-made electrical interference is even more variable because of possible location of a receiver near electrical machinery. Suppression just reaching typical national and international regulations may still give an electric field strength at 10m from the equipment up to 20dB higher than that from atmospheric noise, giving the field necessary to maintain a 35dB signal-to-noise ratio as high as 80dB (μV/m) at m.f. and even higher at l.f. This, however, gives a false impression with the wide use of battery receivers with ferrite rod antennae, which are less sensitive to locally generated electric fields or mains-borne interference because they respond only to the magnetic field component. They can also be turned to make use of antenna directivity to reduce interference. In the UK levels of wanted signal of 70dB (μV/m) at m.f. and 77 dB (μV/m) at l.f. are considered well protected from electrical interference (including that from television-receiver time-base or video circuits) except in city centres or highly industrialised areas. In less dense urban areas and rural areas most listeners are, of course, much freer from interference, and 60dB at m.f. would be well protected from man-made noise.

2.3 Receiver Noise

In general the sensitivity of receivers is such that 35dB signal-to-receiver noise ratio is achievable at field strengths generally below 60dB (μV/m). At this level internal noise will frequently restrict the signal-to-noise performance to 35dB. Thus, for the most part, receiver noise need not be a serious problem or limitation.

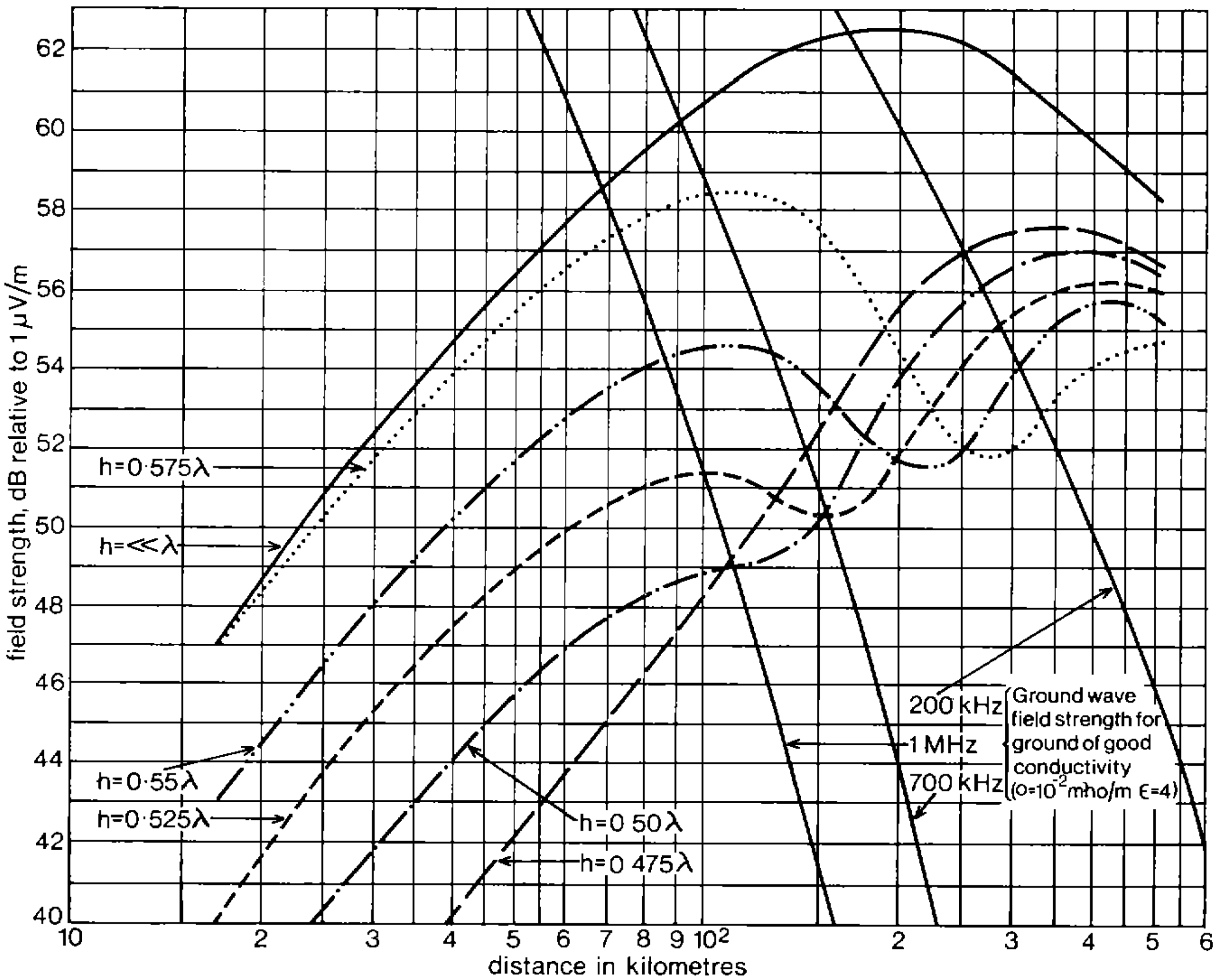
2.4 Conclusions Regarding Noise

In summary it appears that there is not much point, on grounds of noise level, in protecting field strengths, day or night, below about 60dB (μV/m) at 1 MHz, the corresponding values at other frequencies being 68dB at l.f., 66dB at 500kHz and 57dB at 1·55MHz. In low latitudes (less than about 20° latitude) levels might be 14dB higher as proposed in Africa. Also, in urban, city, and industrial areas man-made noise-levels may require that 10dB or more field strength should be provided to give a good service.

3 Ground Wave and Sky Wave

Fig. 1 gives curves showing how the field strength varies with

Fig. 1 Field strength from base-fed vertical aerial on ground of good conductivity
Layer height 100km with unity reflexion coefficient; $E_0 d_0 = 300$V; loop receiving aerial; transmitter-aerial velocity factor 0·9; $Z_0 = 250$ohms; physical height, h; ground reflexion factor, 1·9.



distance for a conventional mast radiator. The case of fairly good ground conductivity is shown. To take account of the influence of the radiation pattern in the vertical plane, the sky-wave curves are labelled with the mast height h in units of free-space wavelength λ. Also, the curves give the field strength that would be obtained for a perfectly reflecting layer at a height of 100 km. The ground-wave curves are shown for four frequencies from 200 kHz to 1,5 MHz. All curves are drawn for a cymomotive force horizontally (which equals $E_0 d_0$, the limiting product of distance and field strength as the distance is reduced), of 300 volts. The radiated power is 1 kW for a short aerial, h ≪ λ, reducing to about 0·5 kW for h = 0·575 λ.

3.1 Ground-wave Service Limitation at Night through Sky Wave

The curves illustrate not only the field strength but also the limit of a satisfactory ground-wave service at night. At the distance where, for a particular mast-height and frequency, in the m.f. band, the ground-wave curve crosses the sky-wave curve, the actual sky wave will (through ionospheric attenuation) have a median value well into the night of about 10 dB below the ground wave. This represents the limit of a first-grade service at night because of interference to the ground wave by the sky wave from the same transmitter. When the signals become nearly equal the reception suffers from fading and frequency distortion effects. In the case of l.f., at distances typical of the limit of service, the ionospheric attenuation is about 5 dB greater than at m.f., and the effective service at l.f. can be considered as extending to the distance where the sky-wave curve in Fig. 1 is 5 dB higher than the ground-wave curve.

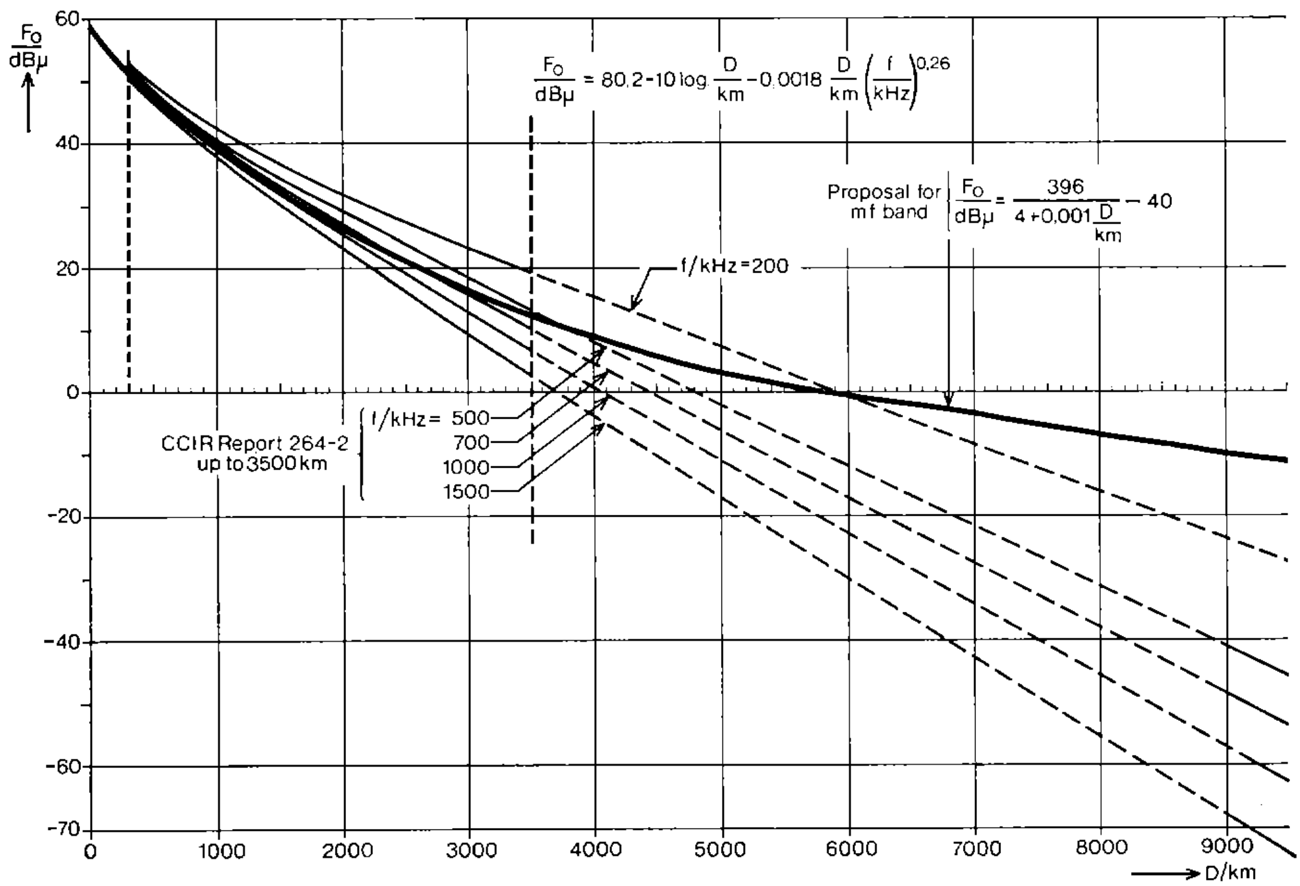
For the conditions applying to Fig. 1 it is possible to see for example that at 700 kHz the limit would be 90 km for a short mast rising to over 150 km for a mast height of 0·5 λ to 0·525 λ. It is also possible to deduce the power to give the desired field strength at the night-time limit. Thus 100 kW from a 0·5 λ mast (2 dB gain) would give 50 + 20 + 2 = 72 dB (μV/m) at about 155 km. Such a field gives reasonable protection against man-made noise. In rural areas in daytime the service might be considered useful to 10 dB lower, 62 dB (μV/m), in which case a greater range (225 km) is achieved.

Finally an example for l.f. may be of interest. Fig. 1, applying to fairly good ground, indicates for h ≪ λ (which is applicable to all l.f. aerials except very special designs such as the ring aerial at Motala) that the night-time range of a good ground-wave service is 250 km at 200 kHz. At this range the sky-wave curve has reached 5 dB above the ground-wave curve, corresponding to the condition given above. The field strength at this limit is 57 dB (μV/m) for 1 kW radiated. For minimum reasonable protection against man-made noise 77 dB (μV/m) or a radiated power of 100 kW is required; higher power is desirable if city or industrial areas are near this limiting range.

3.2 Long-range Sky-wave Propagation

For l.f. and m.f. services at night the interference from co-channel stations, and to a slightly less extent interference from

Fig. 2 Propagation curves.



immediately adjacent channels, are the main limitations controlling a plan. One point of interest in discussing protection against adjacent-channel interference (see later) is the change in sky-wave field strength when the distance from the transmitter changes by a factor $\sqrt{3}$. Fig. 2 (heavy line) gives one example of a recently proposed sky-wave propagation curve applicable fairly accurately to all frequencies in the m.f. broadcast band. Over the distances of interest for planning (co-channel distances of the order of 2000 to 4500km) the ratio of fields at distances $1 : \sqrt{3}$ is about 13dB.

4 Interference from Other Stations

The whole essence of a frequency plan is to enable many services to co-exist with the minimum possible contraction of each service because of interference from other stations. The most important limitation in a plan is co-channel interference.

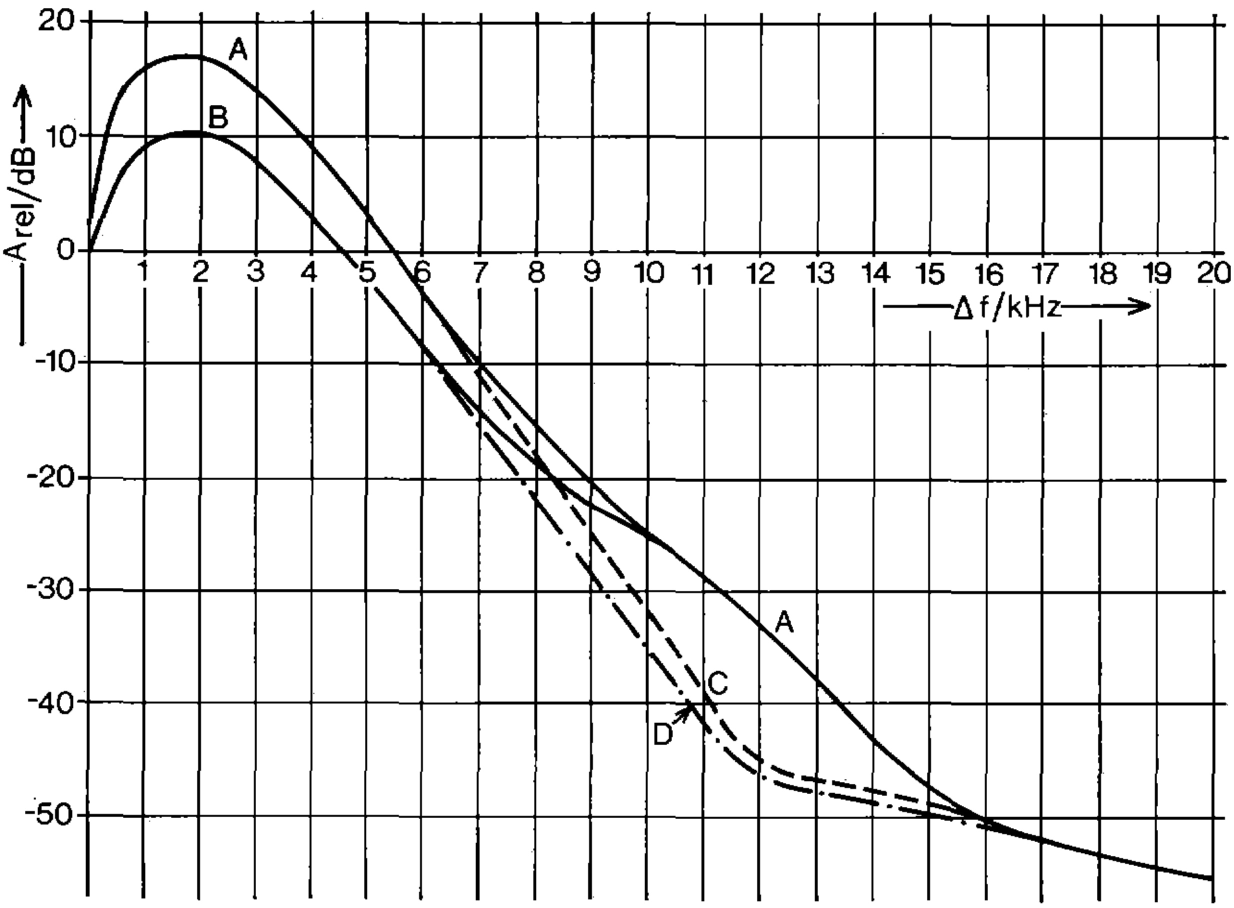
Fig. 3 Relative protection ratio
Curve A, when a limited degree of modulation compression is applied at the transmitter input as in good-quality transmission;
Curve B, when a high degree of compression at the transmitter input (at least 10dB greater than in the preceding case) is applied by means of an automatic device;
Curve C, compression as for Curve A, but with an a.f. bandwidth restricted to about 4·5kHz;
Curve D, compression as for Curve B, but with an a.f. bandwidth restricted to about 4·5kHz.

4.1 Co-channel Protection Ratio

The term co-channel protection ratio is applied to the ratio of wanted to unwanted co-channel signals that exists in a practical situation, usually at the limit of service that can be regarded as satisfactory in the presence of interference. A related concept is protected field strength in dB (μV/m). This equals the sum (in decibels) of the interfering field strength and protection ratio. It is a measure of the level of interfering signal in any situation, and represents the field necessary for the wanted signal in order to give a service which just obtains the basic protection ratio.

On the basis of experiments, and also taking into account studies of the serious limitation of coverage areas that results from attempting to plan for a very high standard of protection, the EBU has recommended that the protection ratio should be 30dB for ground-wave services. This figure applies both for ground-wave interference (mainly daytime limitation) and for the sky-wave interference at night. In the case of night-time, the figure is intended to be applied with respect to the hourly median value at midnight. Well after sunset the interfering field will, through fading, have an instantaneous value alternately above and below the median value. In the case of a single interference, the subjective effect is somewhat worse than for steady interference. For planning purposes, however, it is desirable to combine all interfering signals (on a power basis) and keep the total level within the protection ratio standard. This will to some extent reduce the subjective degradation through fading.

In the case where countries wish to complete their national



services by planned sky-wave coverage, the standard of protection, in terms of median field strengths, should take into account the lower grade of service that this method of coverage provides; the EBU suggests that 27 dB protection ratio is about as high as it would be reasonable to go for protecting national sky-wave services.

Before leaving the subject of co-channel interference it should be noted that a special case arises when a synchronised group of transmitters is used. This is a group of transmitters set up in a country, usually near the centres of different towns, all on the same frequency and carrying the same programme. In this case the co-channel protection ratio required is reduced from 30 dB to 6–10 dB (depending on the type of programme) as a result of the common-programme operation. It may be further reduced if the programme modulation phase and time delay can be equalised in the critical area.

4.2 Interference from Stations in the Adjacent Channel

The amount of interference heard due to a transmission in the adjacent frequency channel depends not only on the ratio of field strengths but also on channel spacing and receiver selectivity. The modulation bandwidth of the transmitter and the amount of programme compression also have some effect on the audible interference.

In the CCIR it has been found convenient to introduce the concept of relative protection ratio. This is generally used when discussing the protection ratio for adjacent-channel interference that is required by receivers. It is equal to the adjacent-channel protection ratio minus the co-channel protection ratio, in decibels. In the context of receiver requirements it can also be regarded as the difference between the dB levels of co- and adjacent-channel signals which give the same degree of audio disturbance. A negative value means that the adjacent-channel signal can be stronger.

Fig. 4 Geometrical relationship of transmitters on adjacent channels ($n-1$), n and ($n+1$)

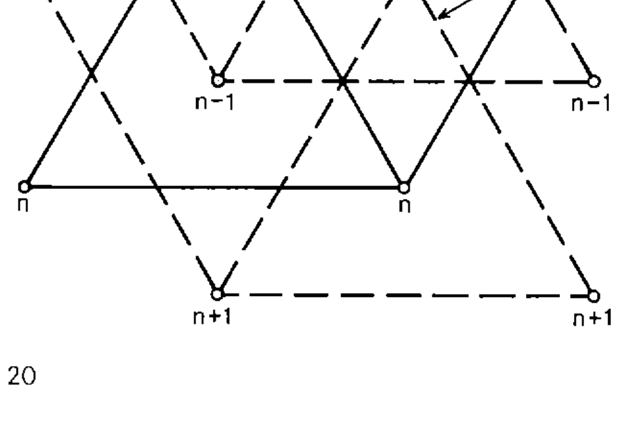


Fig. 3 illustrates curves of relative protection ratio as a function of frequency separation. They are prepared as in the case of the existing CCIR Rec. 449.1 from a consideration of both measurements and calculations based on representative receiver characteristics. The new curves have been extended to show the effect of audio modulation bandwidth restriction as well as compression of the dynamic range of the programme, and have employed a more recent recommendation in regard to noise weighting (Recommendation 468). The curves are valid only when the wanted and unwanted transmissions are compressed to the same extent.

The detailed results of estimates of coverage for idealised or model plans when the channel spacing is varied, which have been undertaken by some member countries of the EBU are described elsewhere. The curves are, however, sufficient to give a general indication of the possibility of controlling adjacent-channel interference in a typical plan. Fig. 4 shows how the positions of adjacent channel stations (channels $n+1$ and $n-1$) in both the higher and lower channels can be arranged to be near the centre of the triangle formed by co-channel stations on channel n. As a result, if the co-channel distance is D, stations on adjacent-channels can be $D/\sqrt{3}$ apart. In Section 3.2 concerning sky-wave propagation it was noted that the relative strengths of signals over path-lengths in this ratio are about 13 dB. This means that, in an ideal plan with equal transmitter powers, the strength of the adjacent-channel signals affecting a service area around a transmitter will be 13 dB greater than signals on the same channel. This statement is valid on the assumption, usually justified, that in a plan there will be about equal numbers of co-channel and adjacent-channel transmitters at the shortest range, up to a maximum of six when at the centre of an area with a fully developed lattice plan. Inspection of Fig. 3 shows that for 8 kHz or greater spacing the adjacent-channel signal may be at least 15 dB stronger before giving impairment equal to co-channel interference. In practice it would be desirable if 8 kHz were to be used to employ the higher degree of compression (permitting 19 dB) and also modulation bandwidth restriction (permitting 21 dB). In the latter case 22−13 = 9 dB margin is provided, that is to say adjacent-channel interference would be 9 dB weaker than co-channel interference in an ideal plan, and this is probably sufficient to allow some departure from a rigidly geometrical plan and still keep adjacent-channel interference a minor component.

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Some Applications of Coding Theory to Broadcasting

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Summary: The paper discusses ways in which certain aspects of message coding theory have been applied to television and sound broadcasting or else are being studied for possible future application. Examples given include work on signal compression, particularly for colour television, digital and hybrid transmission of sound and video signals, television standards conversion, and orthogonal transformation of pictures.

1 Introduction

Early applications of coding theory to broadcasting were chiefly aimed at minimising the required channel capacity. This was particularly important for television where the baseband signal occupied a bandwidth of several megahertz, compared with less than 20kHz employed for sound signals. For example, it was recognised that a better match between picture sources and transmission channels would be achieved if picture areas which were of fine detail, and which therefore required high resolution throughout the source-receiver system, were scanned slowly while plain areas were scanned quickly.[1] To date, however, no means has been found for reducing in this kind of way the capacity required for a television channel while maintaining the high standard of quality that is required for entertainment broadcasting. Nevertheless, an outstanding achievement has been the addition of colour information without demanding any increase in channel capacity.[2] This was done by an ingenious blend of signal coding and interleaving of the luminance and chrominance components in the frequency domain, as illustrated in Fig. 1.

The incentive for applying coding theories to broadcasting has now veered away somewhat from signal compression towards improving the efficiency, stability, and reliability of transmission networks, and also towards reducing operation and maintenance costs. For transmission purposes the general problem is that of matching sources to channels, and here it is important to distinguish between baseband coding and complementary carrier-modulation techniques, since these offer independent means for exchanging bandwidth and signal power.

Straightforward digital baseband coding and logic is being applied to complicated signal processing such as is required for television standards conversion and remote picture source synchronisation. Sophisticated coding, using 'up-dating' techniques, is being developed to achieve high compression of television signals for certain specialised applications, e.g. Picturephone,[3] but severe constraints on the nature of objects that can be satisfactorily accommodated, and on the maximum speed at which those objects can move in a scene without picture break-up, prevent direct application to broadcasting.

The sections below describe, in roughly chronological order, applications ranging from systems already in service in the United Kingdom to ideas which are currently being developed throughout the world.

2 Recently Established Applications

2.1 'Sound-in-Syncs'

Until recently in the United Kingdom, except where the terrain or economics dictated otherwise, video and sound signals were transmitted separately over networks linking together broadcasting transmitters, network and production centres, and outside broadcast points. The use of separate transmission links arose because of the difficulty of matching a single channel simultaneously to suit sources of both video and sound signals. A solution to the problem, which has recently been applied cost-effectively to operational service on the two BBC television networks, is a system known as 'Sound-in-Syncs'.[4] It depends upon a type of coding in which the analogue sound signal is matched to the video-signal channel by way of a pulse-code modulation (p.c.m.) signal added to each television-line synchronising pulse; the net result is a high-quality sound signal transmitted in time-division multiplex with a video signal.

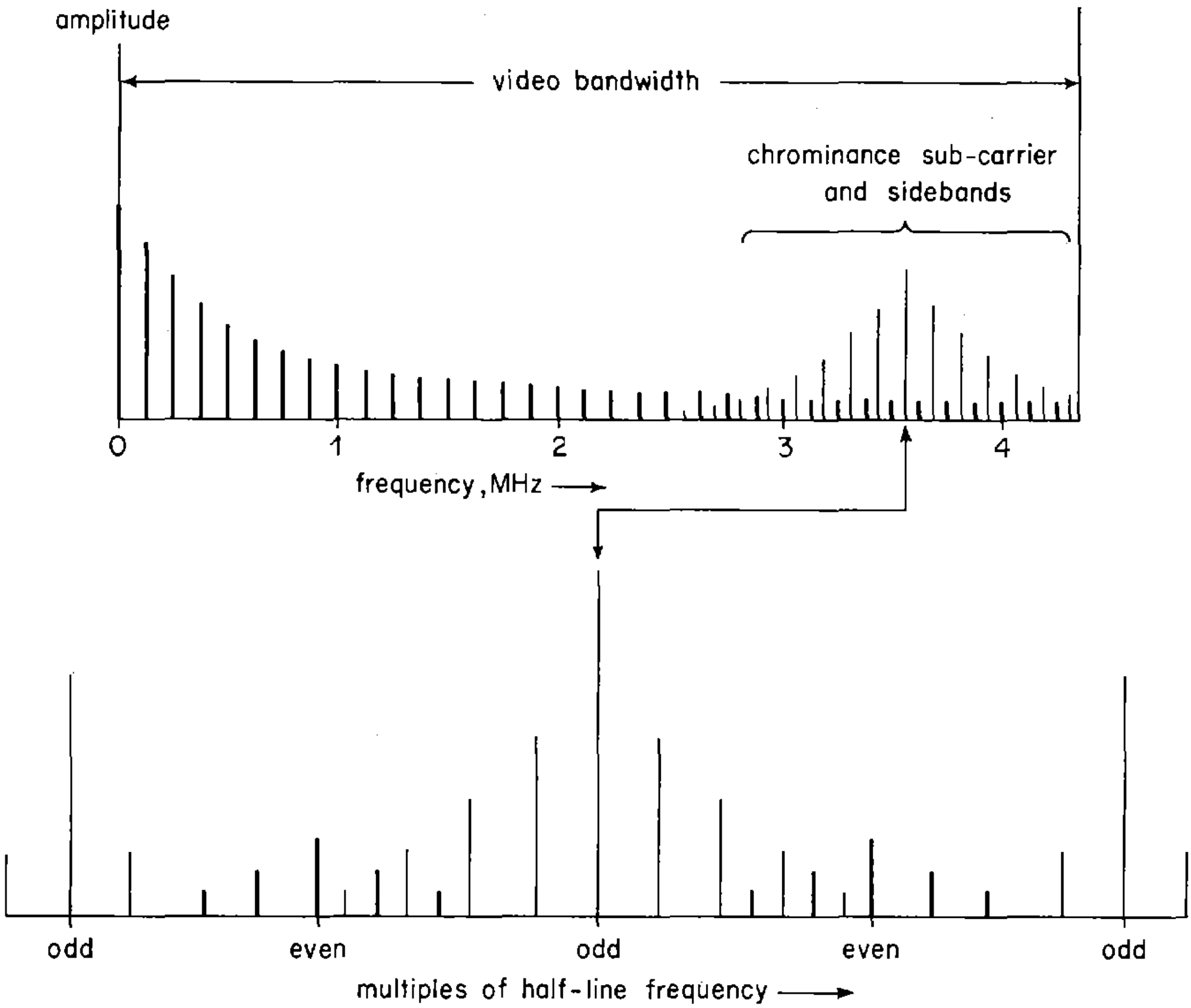


Fig. 1 NTSC colour television amplitude spectrum

2.2 Multiplex Sound-signal Distribution

Stereophonic broadcasting in the UK is currently being extended to more of the existing monophonic programmes and to serve a larger percentage of the population. Together with an increase in the number of national and local radio programmes being broadcast simultaneously, this has led to an increased demand for high-quality transmission circuits. To cope with the increased demand and the special problems presented by stereophony, two principal methods of transmission are being developed.

Firstly, there is transmission of amplitude-compressed sound signals, pre-emphasised at the higher frequencies, in frequency-division multiplex with telephone signals. Although the necessary matching of pairs of circuits for stereophony can be readily achieved with such carrier systems, unfortunately the circuit noise is often not low enough to avoid impairment of the sound programme due to the action of the compressor and complementary expander at the receiving terminal. Also, within a frequency-division multiplex, the spectral positions of the sound signals have to be carefully arranged to avoid crosstalk.

Secondly, p.c.m. is being applied to the distribution of high-quality sound signals.[5] Here the solutions to the problems of matching circuits for stereophony and multiplexing the digital signals are relatively straightforward. Binary p.c.m. by a linearly-quantised sound signal demands about twenty times as much channel bandwidth as that needed for the analogue sound signal, but the p.c.m. signal can withstand about 50 dB more noise power in the channel. A suitable bearer for a p.c.m. sound-signal multiplex is that normally provided for a television signal on s.h.f. radio links; a 13-channel multiplex system using frequency-modulated s.h.f. links is now in service with the BBC.

2.3 Television Line-standards Conversion

For the international exchange of television programmes it is often necessary to convert the signals from one standard to another. Until 1964 such standards conversion was achieved by scanning a television display operating on the incoming standard with a television camera operating on the standard of the receiving country, the requisite redistribution of information in time, and interpolation between adjacent lines and fields, being carried out by virtue of the scanning in the camera tube and the storage properties of the display and camera tubes; the medium-quality pictures obtained by this image-transfer process were acceptable for occasional programmes.

In 1962 it became clear that improved line-standards convertors would be necessary for the plan to change from 405-line to 625-line standards in the United Kingdom, with a long interim period when both standards would be in operation. Consequently the BBC developed what have become known as line-store standards convertors capable of working in both the 625–405 and 405–625 senses, the picture rate being identical for both line standards.[6] Samples of an incoming scanning line are stored on discrete elements, and the samples are read out from the storage elements at the output line-scanning rate. Interpolation inherent in the line-storage action is inadequate and is augmented by convolutional techniques requiring low-pass filters and/or a one-line delay to effect an appropriate vertical-interpolation aperture.

The first line-store convertors were instrumented using analogue techniques and have given many years of operational service throughout the UK. However, the need for dual-standard operation in the UK will continue for some years after the end of the service life of the analogue convertors, and so they will be replaced by digital convertors operating on the same principles. The latter are now nearing the completion of their development, and are proving to be more effective than their analogue counterparts, particularly insofar as they promise to offer better picture quality and greater reliability.

2.4 Television Field-standards Conversion and Synchronisation

The introductory remarks in Section 2.3 also apply, in essence, to the conversion of television signals from one standard to another where the field-rates differ, although the quality of picture obtainable with image-transfer field-standards convertors has proved to be high.

By extending the methods of interpolation and time-redistribution of picture information used in line-store standards convertors, the BBC developed field-store standards convertors, the first of which was put into operational service in 1968 to convert the 525-line, 60-field per second colour pictures from the Olympic Games in Mexico City to the 625-line, 50-field per second standard.[7]

To handle colour signals it is first necessary to transcode the input signal to facilitate interpolation. Compared with line-standards conversion, the chief additional function required of the interpolator in a field-store standards convertor is the reduction of impairment in the portrayal of movement within a scene, particularly that caused by the necessary periodic omission or addition of a field. The requisite control logic is further complicated by the fact that the input and output television signals are fundamentally asynchronous relative to each other.

The last-mentioned feature of a field-store standards convertor suggests the fact that such a convertor can be adapted to the problem of synchronising two television picture sources operating on the same nominal standard but asynchronously. The BBC has modified one of its field-store standards convertors for this purpose.

Digital control techniques were used in the first generation of field-store standards convertors, but it is now practicable also to apply digital techniques to the video-signal processing in digital television synchronisers and convertors.[8] A digital field-store standards convertor has been developed by the IBA.[9]

3 Recent Research and Projected Applications

3.1 Combined Baseband and Carrier Coding

Designers of communication systems have always been concerned with matching sources and receivers to transmission channels, but they have not always been conscious of the benefit which accrues from optimally coding the source signal at baseband frequencies before adapting the resultant signal, usually by carrier-modulation techniques, to match the transmission channel. The latter consideration has been discussed by many authors[10] and two possible applications of this approach in broadcasting are now described.

As outlined in Section 2.2, multiplexed p.c.m. sound signals can be transmitted over frequency-modulated microwave links. A relevant proposal, recently considered by the BBC, optimised the maximum carrier deviation, taking due account of the amplitude spectrum of the digitally coded baseband signals, ensuring that the channel characteristics were such that carrier demodulation failed at the same threshold as that at which the noise power in the demodulated digital signal caused unacceptable digit errors. Such an approach contributes to conservative use of r.f. bandwidth and power.

A second possible application is a scheme recently researched by the BBC in which an 80 Mbit/s (megabit per second) pulse stream, comprising a colour television signal, is matched to a communications satellite channel with a bandwidth of 40 MHz. The 80 M/bit pulse stream is subjected to baseband coding so that it becomes suitable for multi-phase-shift keying of the r.f. carrier.

3.2 Hybrid Pulse-code Modulation

It is known that multi-level p.c.m. is a highly efficient form of coding for optimising the signal-to-noise ratio obtainable in a given channel bandwidth. However, a p.c.m. coder normally causes quantisation noise, but this disadvantage can be removed if the least significant digit of the p.c.m. signal is transmitted as an analogue pulse proportional to the difference between the value of the signal sample and the value of the quantised part of the signal sample represented by the other digits. For most applications such hybrid-pulse-code modulation (h.p.c.m.) is at its most efficient in two-pulse form when the quantised part of the signal is transmitted as a single multi-level pulse preceding the single analogue pulse.

Two-pulse h.p.c.m. is suitable for the transmission of signals over channels in which it is permissible to expand the signal bandwidth by a factor of only two or three to effect an improvement in signal-to-noise ratio; this includes cable circuits, r.f. links, and signal storage systems. It is well suited to television transmission, where, using h.p.c.m., the optimum number of quantising levels has been found to be about five, which is instrumentally relatively simple to provide, giving an effective noise reduction of about 14B while keeping the digital error probability tolerably low at about 10^{-6}.[11] The higher signal-to-noise ratios required for some applications, such as sound-signal transmission, can be attained by increasing the number of quantising levels. It has been found that for a.m. sound broadcasting a 16-level system could offer an improvement in signal-to-noise ratio similar to that obtained with wide-band frequency modulation.[13]

3.3 Video-signal Networks

Video-signals are currently transmitted over networks in analogue form, but economic incentives for digital television transmission are developing in the context of efficiency, stability, reliability, and operating and maintenance costs. P.C.M. has been successfully applied by the BBC to composite PAL colour television signals.[14] Using linear quantisation and sampling at three times the colour subcarrier frequency, seven bits per sample were found to be adequate for one coding and decoding process.

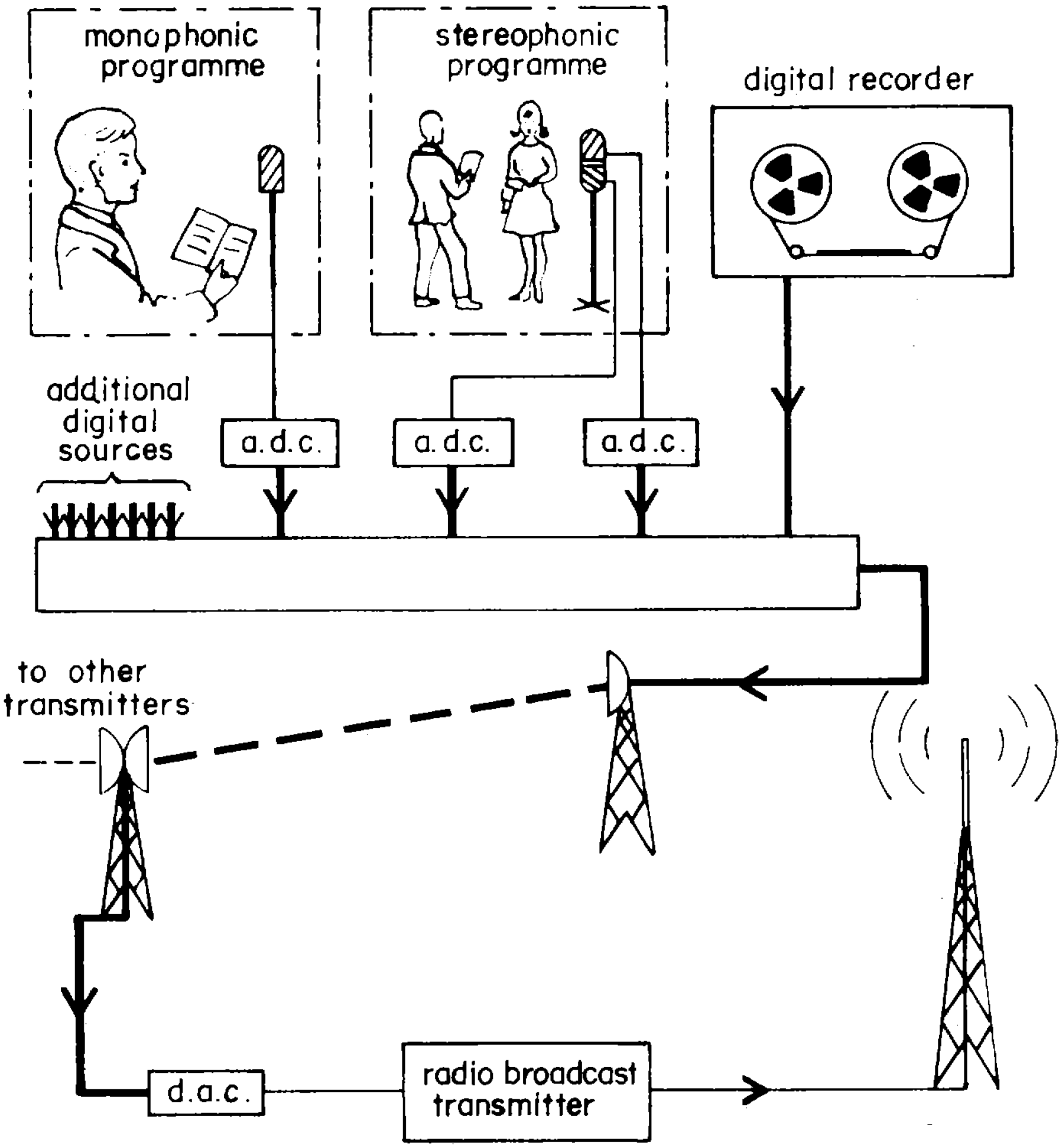
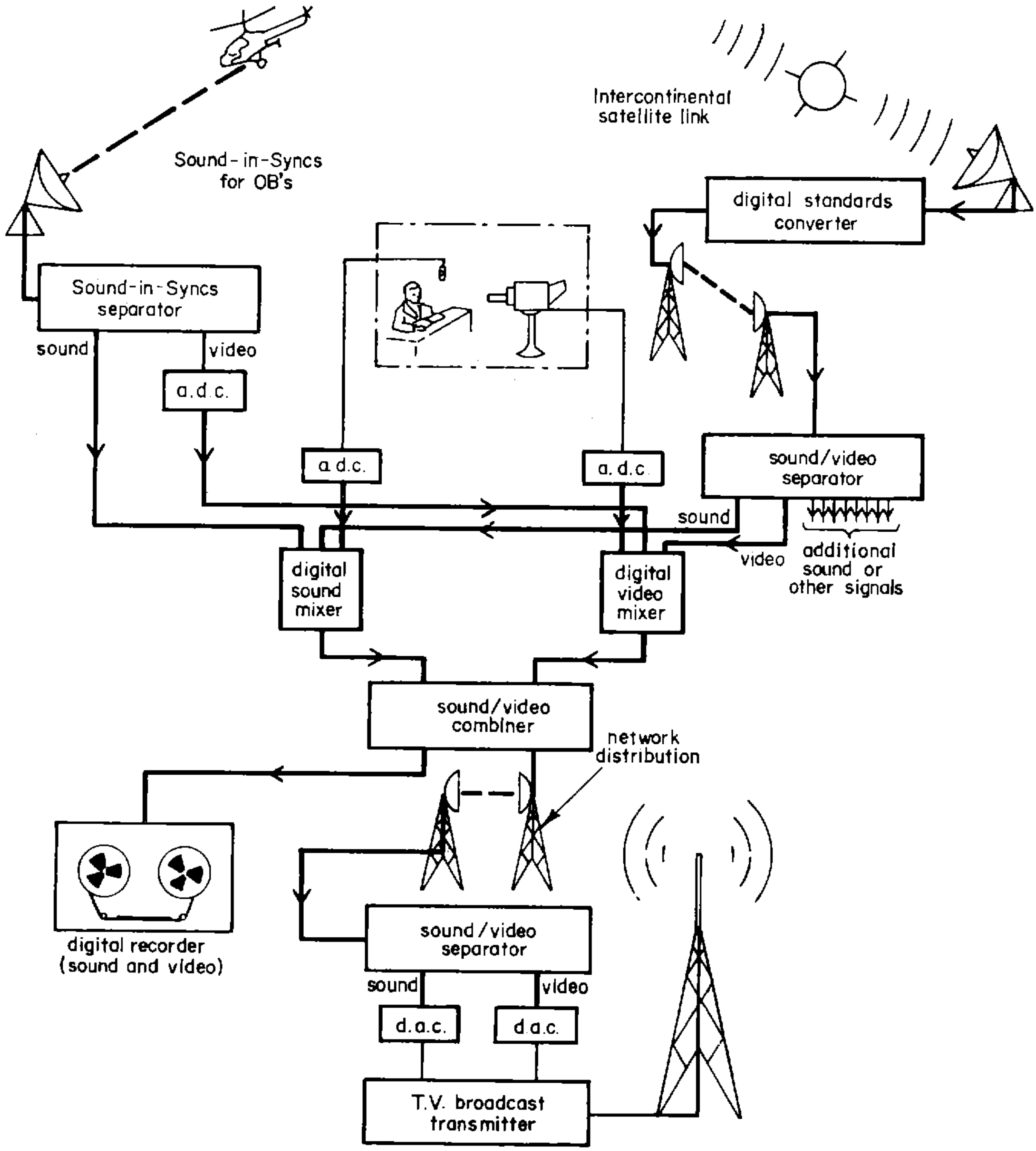


Fig. 2 The application of digital techniques to radio broadcasting

A prime objective in digital television for point-to-point transmission is to minimise the required number of bits per sample. To this end, research effort is being devoted to promising coding methods for a composite colour television signal in which differential p.c.m.[14] is combined with non-linear quantising. Differential p.c.m. (d.p.c.m.) is, at its simplest, the coding of the difference between the current sample and the previous sample; it offers scope for bit-rate reduction because the eye can tolerate relatively coarse (inaccurate) quantising of large differences, complemented by fine (accurate) quantising of small differences. Coarse quantising can clearly give bit-rate reduction, and fine quantising can also do so on differences small enough to be coded into 'words' of fewer bits than those needed for fine quantising of the full range of possible differences.

Bearing in mind the accuracy of fine quantisation, it is advantageous to minimise the differences to be coded. For PAL signals sampled at three times the colour subcarrier frequency, in plain coloured areas of the picture the difference between the current sample and the sample one subcarrier

Fig. 3 The application of digital techniques to television broadcasting



period earlier is virtually zero; such a difference signal is therefore very suitable for d.p.c.m. coding.

The essence of all d.p.c.m. techniques is prediction of the current sample, from either a single earlier sample or a plurality of samples. Such prediction techniques are being investigated, and they offer a reduction of at least two bits per sample for digitised PAL signals.

An alternative approach is to separate the composite signal into its luminance, chrominance, and synchronising components prior to analogue to digital conversion and d.p.c.m. coding.[15] Coding of the chrominance components requires the same or a lower number of bits per sample and a much lower sampling frequency than those needed for the luminance component. Further, to reduce the overall number of bits required, sub-Nyquist sampling, i.e. sampling at a frequency less than twice the highest signal frequency, can be employed for both the chrominance and luminance components in such a way that the unwanted 'alias' spectra are interleaved with the wanted spectra in a manner analogous to that depicted in Fig. 1. This method also offers an overall reduction of at least two bits per sample.

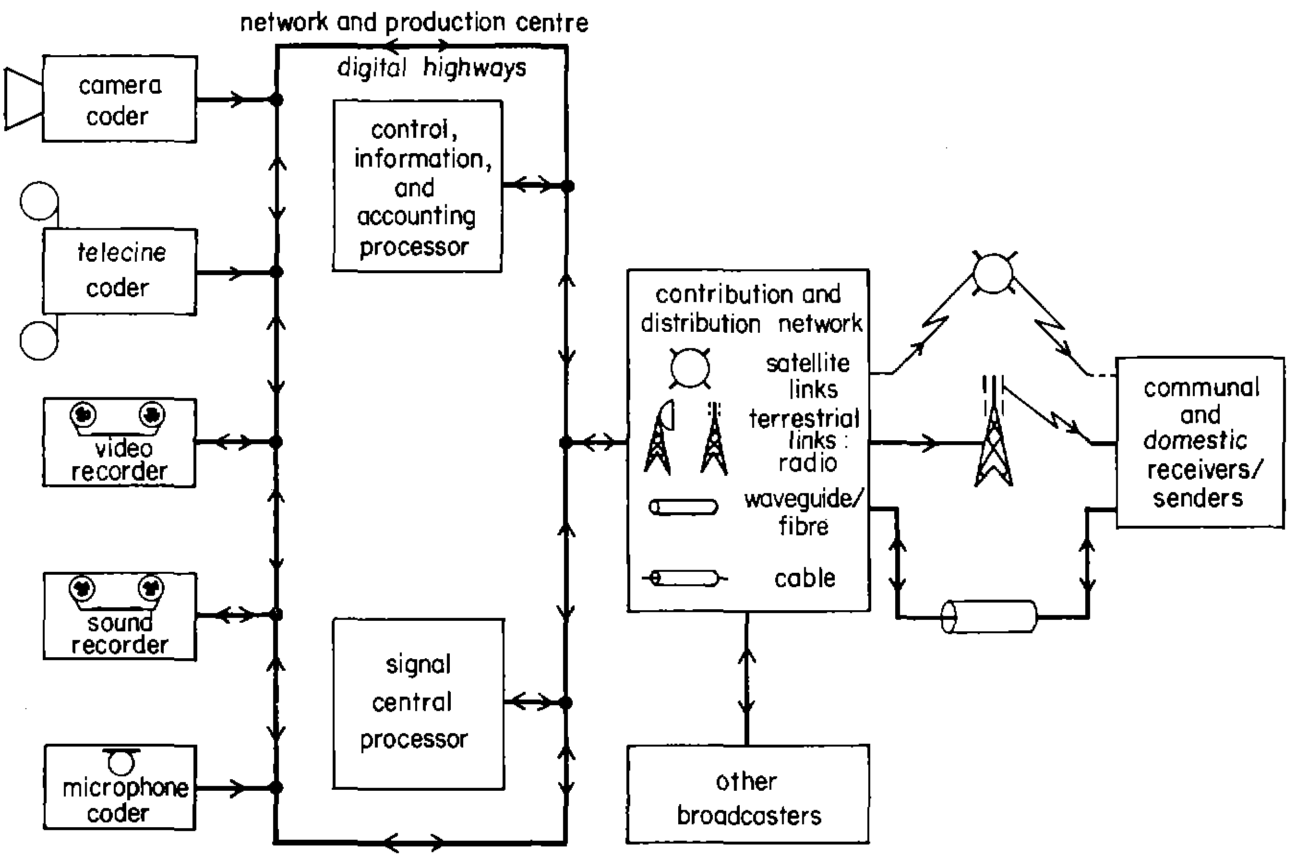
3.4 Recording

With skilled operation of instrumentation, it is now possible to record analogue sound signals on magnetic tape with almost imperceptible impairment upon replay, although non-linear distortion restricts the validity of this statement to only one or two record-replay operations in series. On the other hand, with digital sound-signal recording, again on magnetic tape, it is feasible to pass the signals through a large number of record-replay processes without impairment. Such a 'dubbing' quality is of interest to broadcasters, but the benefit of digital recording will be more valuable in television where it is still somewhat difficult to make high-quality recordings in colour with reliability.

As a preliminary step to making an experimental digital television recorder, the BBC has constructed an experimental digital sound recorder using a conventional tape-transport with normal magnetic tape. Emphasis was placed upon signal coding and processing, and a method was developed for reducing virtually any amount of 'wow' and 'flutter' on the tape transport down to the jitter of a local digital clock. Indeed the work has resulted in digital coding and processing techniques which are likely to be suitable for any recording medium and associated transport which may be forthcoming.[16]

For recording, as for network transmission, there is a strong incentive to reduce the television bit-rate. In addition to the techniques described in Section 3.3, the BBC is investigating the application to this problem of transform coding of pictures. In this regard, transform coding[17] is basically a process in which a group of picture-signal samples is transformed mathematically into an equal number of samples constituting a transformed group. The value of each transformed sample depends linearly upon the values of all the original samples, and the main point of interest is that some transformed samples are usually much more pictorially significant than others. A further point is that the transformed samples are less statistically dependent upon each other than the original samples are. Hence the possibility emerges that insignificant

Fig. 4 A digital broadcasting system



transformed samples could be discarded, or quantised relatively coarsely, before transmission. Thus there is a possibility of bit-rate reduction by producing a truncated transformed signal instead of, say, a p.c.m. signal. The picture signal is recovered by inverse transformation.

Hadamard transforms[17] are of practical interest for the present purpose, because they offer the possibility of bit-rate reduction in the way outlined in the previous paragraph, and because execution of a Hadamard transform is relatively fast as it involves only additions and subtractions.

4 The Future Evolution of Digital Coding

4.1 The Hybrid Phase

It is likely that digital coding techniques will have largely supplanted analogue techniques in broadcasting engineering in twenty or thirty years' time. In the meanwhile there will be a long hybrid phase when great care will be necessary to ensure that analogue techniques are used where they are adequate, and that the application of digital techniques does not degenerate into an uneconomic obsession. It will be necessary to provide analogue to digital and digital to analogue converters, several pairs of which may be connected together in series without impairing sound or picture, and the advantages of systems which closely combine analogue and digital techniques will continue to deserve research and development. Figs. 2 and 3 depict a stage in the hybrid phase when recording and almost all transmission of sound and video signals for broadcasting will be done using digital techniques.

4.2 The Rugged All-digital System

Provided that it attains widespread application, a digital technique can usually be realised in instrumentation which renders cost-effective its technical superiority in noise immunity, electrical stability, and reliability. Therefore as the numbers of broadcasting sources and receivers grow, an economic incentive for developing digital sources and receivers will grow to match technical incentive. If such a development occurs and if the formidable problems of controlling and processing digital television and sound signals on a massive scale can be overcome, then a digital broadcasting system similar to that depicted in Fig. 4 may evolve.

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Artificial Reverberation

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Summary: Existing systems of artificial reverberation are briefly appraised and their defects discussed. Some possible new systems are described and their potentialities assessed, and it is concluded that these would be free from the defects of existing systems, but that they would probably be costly.

1 Introduction

1.1 History

Artificial reverberation systems were reviewed by Axon *et al.*[1] in 1955 and Goodfriend and Beaumont[2] in 1959. Since that time, reverberation plates[3] have come into widespread use and other systems have been proposed. Systems employing torsional oscillations of springs have been developed to the point where they are acceptable to broadcasting organisations for some types of programme.[4] Schroeder and Logan proposed an artificial reverberation system[5] claimed to be free of coloration but there are some practical difficulties to its construction and it does not seem to be in use to any significant extent.

A new class of artificial reverberation systems has appeared in which the sound is picked up by a microphone and replayed over loudspeakers in the room in which the sound originates. These include ambiophony,[6] assisted resonance,[7] and a system devised by Franssen.[8]

An ambiophonic system requires extensive readjustment if the sound source and microphone are moved, so it is inconvenient for broadcasting purposes. Assisted resonance systems which cover the frequency range up to 4kHz have not yet been demonstrated; existing systems are costly to install and maintain. Franssen claims that his system increases reverberation time without the disadvantage of incipient howlback, but it is not clear to the present authors that this necessarily follows from his theoretical analysis. In any event, verbal reports from those who have seen Dr Franssen's installation in Holland appear to establish that the system would require a large number of high-quality microphone-amplifier-loudspeaker channels for music. Understanding of this class of systems is far from complete and there is much scope for further research. Such systems will not be discussed further in this report; even if they were improved considerably, they would have very restricted application to broadcasting. Discussion will be confined to those systems in which *reverberation is added without requiring access to the studio* in which the programme originates.

1.2 Need for New Systems

For applications where an accurate simulation of natural reverberation is unnecessary or unwanted, e.g. pop music, existing devices probably give satisfactory quality and they are not unduly expensive. What is lacking is a system which has all the desirable attributes of natural reverberation from a hall having good acoustics. No system has yet been made which provides artificial reverberation indistinguishable from the reverberation from a large hall of good acoustical quality. In practice, of course, this ideal is unnecessary since most programmes to which artificial reverberation is added already has a considerable content of natural reverberation. However, the nearer this ideal is approached, the more flexible can be the choice of studios and outside halls for broadcasting. If the ideal system were developed it might be possible to broadcast concerts from studios *little bigger than is necessary* to seat the players.

1.3 Defects of Existing Systems

The defects of existing reverberation systems are easier to recognise on listening than to describe in words that are meaningful to someone who has not heard such defects. Until it becomes economic to issue sound recordings with reports, this difficulty will remain.

There are two main defects of existing systems:

1. The presence of colourations. A colouration is the subjective impression that sounds of a particular pitch are being unduly accentuated in comparison with the whole

spectrum. No precise method has yet been developed whereby it is possible from other objective measurement alone, to predict whether or not a system will produce colourations.

2. The presence of flutter echoes. Flutter echoes are most noticeable with impulsive excitation of the system; they appear as periodic repetitions of the input signal. For more continuous excitation, they appear as periodic amplitude modulation of the decaying sound after the excitation has ceased. They are associated with periodicities in the system and are too low in frequency to be heard as colorations.

2 Possible New Systems

2.1 Development of the Reverberation Plate

Van Leeuwen has described his attempt to reduce the size of the reverberation plate without impairing quality.[9] The phase velocity of bending waves in a plate is given by

$$c = (2\pi f)^{\frac{1}{2}} \,.\, \{Yt^2/12\rho(1-\mu^2)\}^{\frac{1}{4}} \qquad (1)$$

where f is the frequency
Y is Young's modulus for the plate material
μ is Poisson's ratio for the plate material
ρ is the density of the plate material
t is the thickness of the plate

If the ratio of wavelength to length and width of the plate is to remain constant, then a reduction in both length and width by a factor r necessitates a reduction in thickness by the factor r^2. Van Leeuwen's plate has the dimensions 170mm $\times$ 130mm $\times$ 5μm corresponding to $r = 1/10$, and is made from nickel foil. The effect of air-damping on so thin a foil is great and in order to achieve a mid-brand reverberation time of 3 seconds the housing for the plate must be evacuated to a pressure of 133 N/m^2 (1 mm Hg).

Dr W. Kuhl of IRT Hamburg, the inventor of the reverberation plate, has for many years been developing an improved version of the device.[10] It is now expected that the new plate will be commercially available in the second half of 1972. A description of the production version of the new plate has been given recently by Rother.[11]

Kuhl's new plate is said to be quite free of colouration and small enough to be carried in a medium-sized saloon car. The area of the plate itself is 0·1 m^2, about one-twentieth of the area of the standard plate. The thickness is about 0·02 mm, about four times as thick as Van Leeuwen's miniature plate.

Unlike Van Leeuwen's plate, Kuhl's plate does not have to be operated in a vacuum to achieve a long reverberation time and the increased thickness accounts at least partly for this. Kuhl indicates that a plate made of nickel is insufficiently heavy to be resistant to unwanted damping, and implies that only alloys of platinum, silver or gold are satisfactory. He points out that problems are to be expected with the transducers used to excite and to pick up the plate vibration and that the bending-wave impedance of a plate 0·02 mm thick is so small that the moving-coil of an electrodynamic transducer attached to the plate must weigh no more than a few milligrammes.

If the new plate lives up to its claims, and is not too expensive, it may well be the answer to the demand for a high-quality artificial reverberation system which is small enough to be easily transportable.

2.2 Proposed Optical Method

In 1966 Gouriet suggested an artificial reverberation device involving optical convolution of the sound with the impulse response of a real or artificial room.

If the impulse response of the path (or paths) between a sound source and a microphone in a room is R(t), then the output voltage of the microphone V(t) is related to the sound pressure waveform generated by the source S(t) by the convolution integral

$$V(t) = \int_0^T R(\tau)\,.\,S(t-\tau)d\tau \qquad (2)$$

where $R(\tau) = 0$, when $\begin{cases}\tau < 0 \\ \tau > T\end{cases}$

In an artificial reverberation machine, R(t) is the impulse response of the machine, S(t) is the input signal and V(t) is the output signal. Gouriet suggested that the convolution could be carried out by making an optical recording of the input signal and passing this over a mask whose optical transmission was a replica of the desired impulse response. The total light transmitted through the mask and recording would be picked up by a photodetector, whose output would be related simply to the required V(t).

The proposal has attractive features. The multiplication in the integrand and the integration itself are fairly easily done optically; the speed with which these operations are carried out clearly presents no difficulties. The mask could represent the impulse response of a known good hall or studio.

However, the method would have serious disadvantages in practice. The expense and delay in producing an optical sound track would be unacceptable operationally. To make the system viable some form of rapidly recordable, erasable and re-usable optical recording medium is required, and no such material is readily available. It could well be a valuable research tool, as an aid to the design parameters of other types of artificial reverberation systems.

2.3 Proposed Scale-model Method

Scale-modelling of a studio or hall is a useful way of evaluating the performance of a room before it is built. In 1968, Gilford suggested that such a model could be used in an 'on-line' artificial reverberation system.

In model work, a recorded tape is played into a $1/n^{th}$ scale model of a room at a speed n times greater than the recording speed, and the sound in the model is re-recorded from a microphone in the model simultaneously. When the new recording is played at the speed of the original tape recording it sounds as if the recording was made in a room similar to the real full-scale room.

As normally operated, such a procedure would be unacceptable for an artificial reverberation machine, because of the delay between the operations of recording and replaying at the different speeds. Gilford's proposed solution was to devise a special tape recorder with two rotating heads. Fig.1

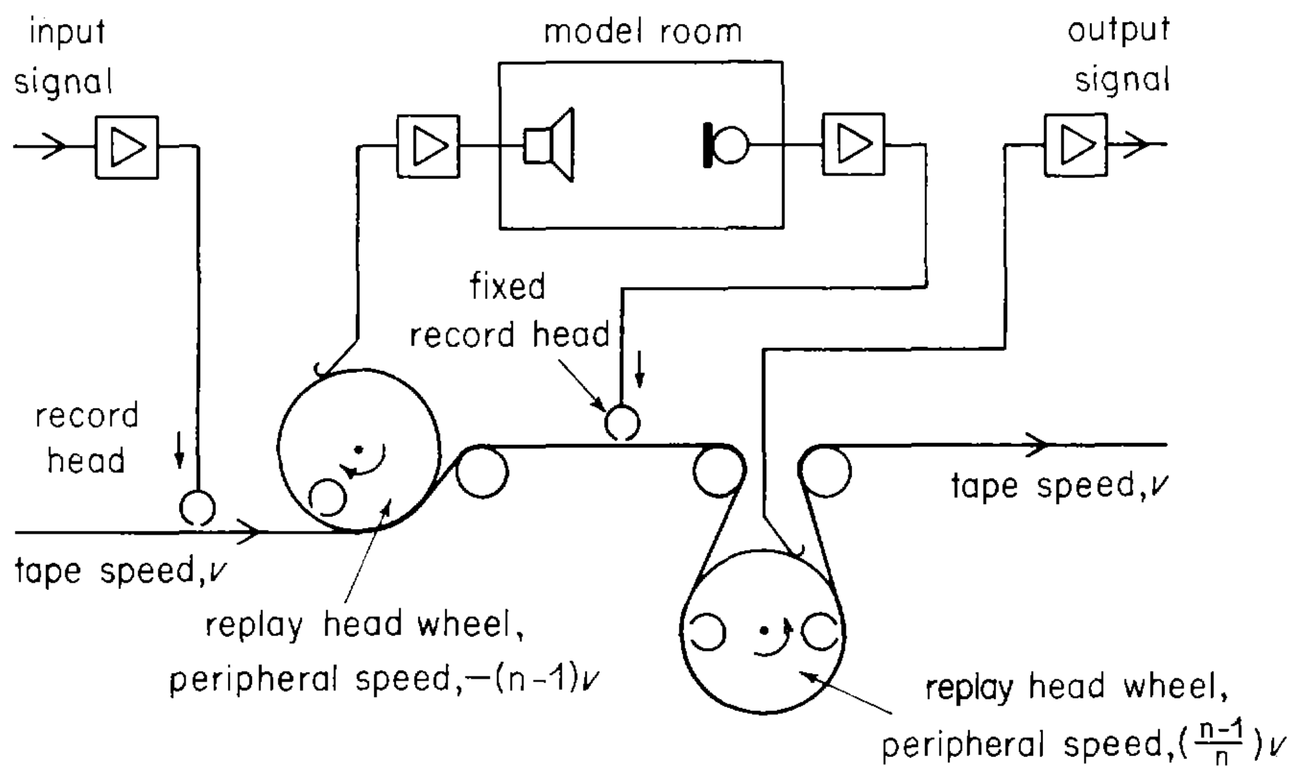
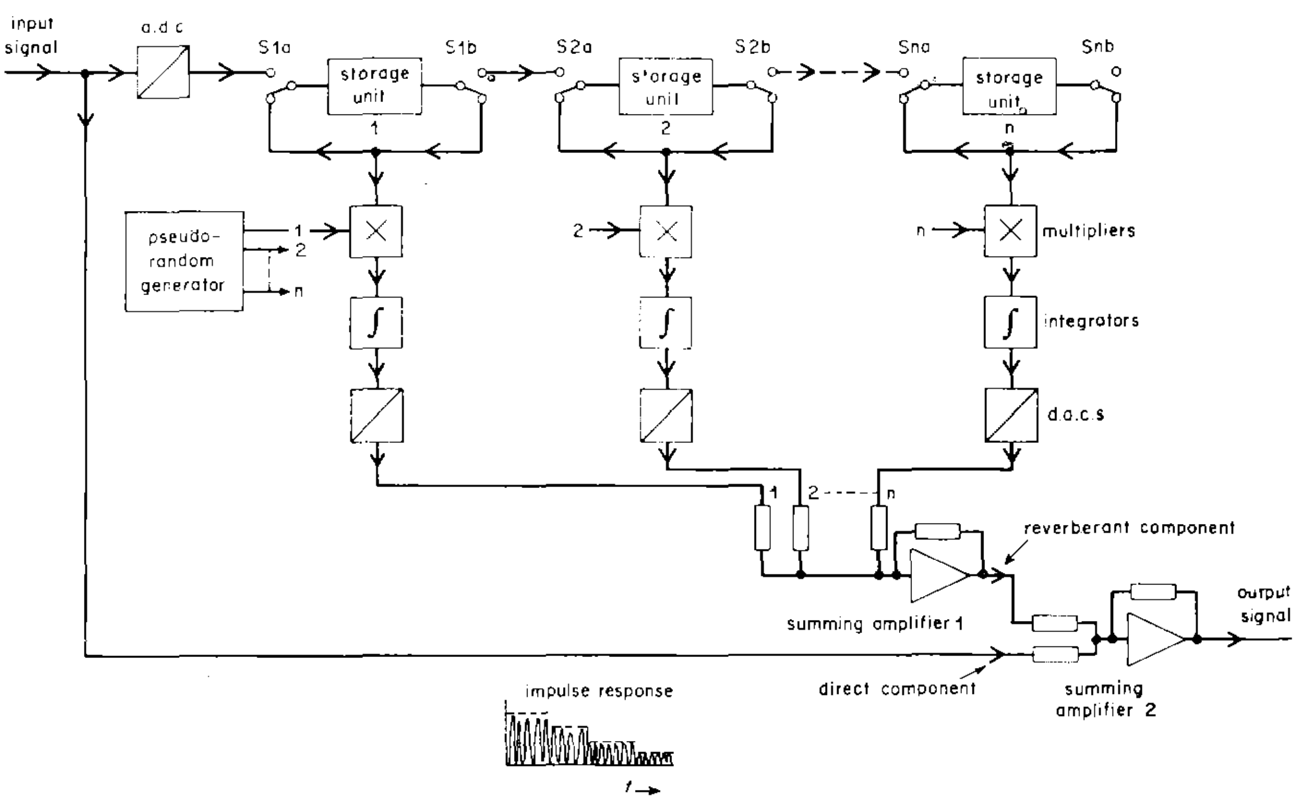


Fig. 1 Proposed scale-model method

Fig. 2 Artificial reverberation using pseudo-random sequences



shows the essential features of the idea. The signal to be reverberated would be recorded on tape in the conventional manner. The recorded signal would then be replayed into the model room from a rotating replay head the peripheral speed of which was $(n - 1)$ times the tape speed and in the opposite direction. The speeded-up sections of programme would then be reverberated in the model, picked up by a microphone and recorded by a fixed recording head, on either a new track or the original track after erasure. The new recording is then replayed by a second rotating head-wheel, rotating at $(n-1)/n$ times the tape speed and in the same direction. Various combinations of number of heads per wheel, head-wheel speed and tape wrap-angle are possible. For the arrangement shown and with $n = 8$, the wrap-angle of the tape around the first headwheel must be $306/8 = 45°$ for the head to scan all of the input tape. The tape must wrap around the second head-wheel for at least 180° with the two heads as shown. The room under consideration could be a scale model of a known good room, or it could be a small reverberation room (echo chamber). The advantage would be that the sound would appear to come from a room much larger than it is necessary to build, and the 'small-room' sound associated with reverberation chambers would be absent.

The foreseeable difficulty in the proposal is that of joining up accurately the segments of programme which have been time-compressed and then time-expanded. The problem lacks an economical rather than technical solution, as timing correction systems of adequate performance are available, although they are costly.

A similar proposal, involving a modified transverse-scan videotape recorder, has been suggested by Boutros-Attia.[12] The technological difficulties in this proposal appear to be more formidable than the problems of Gilford's method.

2.4 Proposed Digital Method

It should be possible, in principle at least, to construct a special-purpose digital computer to carry out the required convolution given in Section 2.2 Equation 2.

A sampled version of the room impulse response $R(\tau)$ would have to be stored in the computer for multiplication with digitised samples of the input signal $S(t)$. Samples of the input signal ranging from the 'present' sample $S(t)$ and all the earlier samples as far back as $S(t-T)$ would need to be stored at any one time; the integration would be merely replaced by addition. For real-time operation, the multiplications and addition would need to be carried out within one sample period of the input signal.

A tentative example will show the scale of the problem:

Bandwidth 5kHz, therefore sample rate 12000 samples/second.

Programme samples coded into 10-bit words.

Simulated reverberation time 2 seconds, therefore $T \approx 2$ seconds.

The number of samples required to simulate $R(\tau)$ adequately is unknown. It would be hoped that 100 would be sufficient. In one sample period the machine would have to carry out the following operations.

1. Read and multiply 100 10-bit words with another 100 10-bit words and sum the products.
2. Shift the programme samples one place, discarding the oldest sample and reading in the newest.

Consideration of these requirements indicates that such a machine could be made to work in real time, but that the cost would be high. It is estimated that the cost of the 'hardware' alone for such a machine would not be less than £8000 and would perhaps be as high as £20000. In any case it would be unwise to embark upon such a device without a clearer idea of the number of samples required adequately to approximate $R(\tau)$. Simulation of a proposed machine by programming a general-purpose digital computer would be a safer (but still expensive) first step.

2.5 Proposed Pseudo-random Artificial Reverberator

A device employing a pseudo-random sequence generator to eliminate the colourations inherent in recirculatory delay systems has recently been proposed by Jones.[13] A schematic drawing to illustrate the principle is shown in Fig. 2.

The signal to be reverberated is converted to digital form as for the method of Section 2.4. Consecutive sections of the signal are then, in effect, convolved with a pseudo-random sequence of numbers, thus producing sections of signal having no colouration. The resultant sections of signal are then weighted progressively so that the envelope of the impulse response of the system approximates the decaying form characteristic of natural reverberation.

Referring to Fig. 2, the input signal is sampled, digitised and passed to an assembly of N storage units formed from shift registers. Each storage unit has a capacity of n signal samples. At the beginning of a sampling period the switches S1a, S1b, S2a, . . . etc., are in their upper positions and one signal sample is moved from the analogue-to-digital converter to the first storage unit, one from the first storage unit to the second, one from the second to the third and so on throughout the chain of storage units. The switches are then set to the positions shown and the samples in each storage unit are rapidly circulated once before the signal sampling period is complete. During circulation, each sample is multiplied by a different 'word' (i.e. a number) from the pseudo-random generator and the sum of the resultant products is accumulated in each integrator. The pseudo-random generator has a number of different outputs so that the samples in each storage unit are multiplied by a different set of numbers.

At the end of the summation process, the totals from the integrators are fed to digital-to-analogue converters* and the resulting analogue signals combined in a summing amplifier, after having been attenuated progressively to give the required decaying envelope for the impulse response. At the end of each sampling period, the signal samples are moved one place to the right along the storage unit chain and the pseudo-random generator and integrators are reset in readiness for the next multiplication/summation cycle.

The impulse response of such a reverberator would be a pseudo-random sequence, amplitude-modulated by a staircase function. It seems very unlikely that such a response

* The function of multiplication, integration and digital-to-analogue conversion may be combined within a single unit.

would give rise to colorations. The likelihood of flutter from the periodicity in the envelope would also appear to be small if there were something of the order of 100 steps in a reverberation time of 2 seconds, i.e. about 1 dB average fall in amplitude per step.

The storage units could be made from currently-available MOSFET integrated circuits. It is estimated that a reverberator having the same basic design standards as those of the non-random device described in Section 2.4 above would cost about £4000 (hardware only).

3 Conclusions

Existing artificial reverberation devices are all unsatisfactory in one respect or another, but apart from projected improvements to the reverberation plate, there seems little prospect of making a high-quality device at an economic price at the present time.

Of the new methods of producing artificial reverberation discussed, the optical method would be an attractive and flexible device if it could be operated in real-time with a re-usable recording medium, the scale-model would be attractive were it not for the expense envisaged in solving the timing-correction problem and the digital method would be prohibitively costly even in its simplest form. The pseudo-random method would appear practicable, although it would be too expensive for operational use in the precise form described. The construction of a prototype would permit possible simplifications (and cost reductions) to be assessed subjectively.

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Subjective Study of Two Large Music Studios

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Summary: Users of two BBC large music studios (Maida Vale 1 and Glasgow 1) were asked to give their opinions on various acoustic characteristics of these studios using a specially devised questionnaire. Three statistical methods of analysing the raw data were used and the results obtained are outlined in this report. One aspect of the work concerns users' views on what would constitute an 'ideal' music studio and these are compared with their assessments of a real studio. Some tentative conclusions are drawn about which individual characteristics are most in need of improvement. The report also attempts to answer the question as to which characteristics determine the overall opinion on a 'like/dislike' axis, and simple equations are deduced which correlate well with the stated opinions of the observers. Factor analysis was used to determine whether a smaller number of variables (derived from the characteristics used for the raw data) could adequately account for the results but little success was achieved using the principal-components method.

1 Introduction

Over a period of some years there has been a mild but growing dissatisfaction with the acoustic quality of the large music studio in Maida Vale (MV1). This dissatisfaction applies particularly when a full symphony orchestra or a large ensemble of players are performing in the studio; when the studio is used for recitals involving only one or two instruments or soloists it is regarded with much more favour, even to the extent of being regarded as completely satisfactory by some people. The dissatisfaction felt when large orchestras are using it, and in particular the BBC Symphony Orchestra which is based there, has been fairly clearly expressed by some individuals although a concensus of opinion is more difficult to define. Some ten or so years ago this particular studio was regarded as a good one by some experts[1] and it would be interesting and instructive (if this were possible) to know why there appears to have been a change of opinion. Be that as it may, there is a feeling of less than complete satisfaction with this orchestral studio and, in view of the possibility of using acoustic modelling[2] to effect improvements, it was considered highly desirable to determine the nature of these complaints.

The aforementioned acoustic modelling work made use of a model of the Maida Vale Studio 1 for two purposes: the first was to establish whether technology had reached the point where modelling was a useful and helpful technique, and whether the quality of sound produced using the model could be considered to be a sufficiently close replica of sound from the real studio. If and when this point had been established, it was then proposed to use the model to determine whether any practicable variations of the acoustic treatment of the studio of layout of the orchestra would give rise to better sound quality.

Before starting these experiments, it was felt desirable to ask the various users (sound engineers, musicians, audience) what was wrong with Maida Vale Studio 1. To this purpose a questionnaire was designed and distributed to the appropriate people and this report consists principally of an account of an attempt to analyse the results obtained. The ultimate aim of all work of this kind is to establish significant correlation between subjective opinions about concert halls and objective measurements of relevant acoustical parameters: in this way one hopes eventually to be able to state in objective terms what is responsible for producing good acoustic quality.

Although the original intention of the present work had been restricted to an investigation of Maida Vale Studio 1, it was subsequently found possible to visit Glasgow and to secure the co-operation of the BBC Scottish Symphony Orchestra. Thus the work includes a study of Glasgow Studio 1 and also some interesting comparisons between the two studios.

2 The Questionnaire

A draft questionnaire was first distributed to some senior sound engineers intimately concerned with sound recording in Maida Vale, and in view of their favourable comments the main experiment used the questionnaire in a virtually unmodified condition (See Appendix 1). Nineteen individual characteristics were included in addition to a general overall response on a 'Like/Dislike' axis. It is not considered that as many as nineteen characteristics are required to specify sound quality, but at this stage in the investigation it was felt desirable to include all the terms which might usefully help to convey a full picture. Hopefully the total number of completely independent variables would not exceed four or five, but there was no guarantee that a set of truly independent variables (orthogonal in multi-dimensional space) would correspond precisely to any of the terms normally used to describe acoustic quality, e.g. the terms used in this questionnaire. It will be noted that in fact all the terms do relate directly to acoustic quality and that other matters which might have had a bearing on the subject, but far less directly, were not included, e.g. conditions of lighting (this affects the musicians with respect to ease of reading their scores), air conditioning, décor, etc.

In compiling the list of terms to be used, attention was paid to the lists used by other workers for similar purposes, e.g. Beranek,[3] Hawkes[4] and hopefully the terms selected include, explicitly or implicitly, all those previously used.

3 Analysis

The number of completed questionnaires relating to the acoustics of the Maida Vale studio was disappointingly small, and in fact numbered only twenty-five. The 'catchment area' did not include as wide a range of personnel as had been hoped, and mainly embraced engineering staff engaged in sound broadcasting. Due to various difficulties it was not possible to obtain replies from all members of the BBC Symphony Orchestra, but the Orchestral Committee agreed to take part. In addition to inquiring about the observers' reactions to the real Maida Vale studio, fourteen engineers and musicians also filled in questionnaires giving their views on the rating of an ideal music studio, and some of the analysis which follows will refer to these questionnaires.

In purely numerical terms, greater success was achieved with the study of Glasgow Studio 1, where fifty-six completed questionnaires were obtained relating to the real Glasgow studio and another fifty-six relating to the musicians' opinions about an ideal music studio. In due course certain interesting similarities will be pointed out between the assessments in London and Glasgow.

Three methods of analysis will be reported upon:

1. Simple statistical analysis making use of means and standard deviations. This will include comparison of the real with the ideal studio.
2. Correlation analysis which, *inter alia*, attempts to establish a simple mathematical relationship between the overall assessment and those individual characteristics which contribute significantly to the overall assessment;
3. Factor analysis. This is a more sophisticated version of correlation analysis and gives as its end product the relationship between the overall assessment and combinations of the individual variables which are, in the mathematical sense, orthogonal. These combinations of the individual variables are termed factors.

3.1 Simple Statistical Analysis

3.1.1 Real Maida Vale Studio 1

The prefix 'real' is used here to distinguish the studio from the 'ideal' studio which will be reported on in Section 3.1.2, or the 'model' studio used in the acoustic scaling experiment.

The means and standard deviations for the twenty qualities reported upon by the twenty-five observers are given in Table 1 (for compactness and ease of comparison Table 1 also gives data on the 'ideal' studio). The polarity of the numbering is given in Appendix 1 and usually proceeds from the 'undesirable' condition, rated at 0, to the 'desirable', rated at 10. This

TABLE 1

Summary of Assessments of Observers on Maida Vale 1 Studio

Quality	*Real Studio*		*'Ideal' Studio*	
	mean	*standard deviation*	*mean*	*standard deviation*
1. Coloration	4·96	2·45	7·65	2·78
2. Diffusion	5·90	2·64	8·00	2·21
3. Echoes	8·34	1·27	7·92	2·62
4. Pitch Changes	8·01	2·03	9·03	1·05
5. Singing Tone	4·87	2·54	9·62	1·41
6. Ensemble	4·23	2·67	9·19	0·68
7. Fullness of Tone	5·17	2·49	8·94	1·39
8. Hardness	3·88	2·17	6·44	2·40
9. Loudness	6·47	1·72	5·77	1·62
10. Intimacy	3·80	2·42	5·73	1·97
11. Liveness	5·06	2·55	7·57	1·68
12. Tonal Warmth	4·56	2·42	8·73	2·00
13. Definition	4·77	2·78	8·19	1·36
14. Brilliance	5·10	2·37	7·23	1·70
15. Balance	4·26	2·52	9·33	0·68
16. Blend	5·17	2·68	9·06	9·74
17. Attack	6·00	2·00	8·98	1·39
18. Dynamic Range	5·06	2·54	8·88	1·62
19. Timbre	4·08	2·14	9·50	0·92
20. Overall	4·10	2·39	9·79	0·38

simple system does not obviously apply to some qualities, since either extreme may be undesirable and in that case Appendix 1 must be consulted. For example, the polarity is obvious for characteristic No. 20 'overall', where 0 = disliked, 10 = liked, or characteristic No. 13 'definition', where 0 = muddy, 10 = clear. A more difficult case, however, is characteristic No. 10 'intimacy', where it has been decided to label 'distant' as 0 and 'intimate' at 10, although one may not wish for either extreme in practice.

Examination of Table 1 shows that certain charactersitics secure high numerical gradings. Thus 'echoes' (No. 3) are graded at 8·34 ± 1·27, 'pitch changes' (No. 4) at 8·01 ± 2·03 and 'loudness' (No. 9) at 6·47 ± 1·72.

The first two of these three characteristics certainly imply a favourable grade in the sense that the studio is substantially free from echoes and from pitch changes; for the third (loudness) this is not necessarily so because a studio can sound too loud, and this may well be the case for Maida Vale 1 (see Section 3.2.2, for correlation analysis tending to substantiate this).

Turning now to the characteristics securing the lowest gradings, 'intimacy' (No. 10) is rated at 3·80 ± 2·42, 'hardness' (No. 8) at 3·88 ± 2·17, 'timbre' (No. 19) at 4·08 ± 2·14 and 'overall' (No. 20) at 4·10 ± 2·39. It does not automatically follow that a low numerical grading proves that the studio is deficient in a particular quality and characteristic; No. 10 'intimacy' has already been discussed in order to illustrate this point. Discussion of this is usefully deferred until the characteristics of the 'ideal' studio have been considered (Section 3.1.2). Nevertheless, little doubt attaches to the significance of low ratings of characteristics 19 and 20 and it would appear that the studio is to some extent lacking in timbre and is below the half-way mark on the 'dislike' to 'like' axis used for assessing the 'overall' response. Of course, it may be argued that if a criticism of a studio is sought, it is almost certain that one will be forthcoming, and therefore the fact that the overall grading is less than five should not be taken as too significant. In fact, the distribution of opinions on 'overall' show a bimodal distribution: fifteen observers gave ratings below 4, and ten gave ratings above 5·7 with a gap in the middle range from 4 to 5·7. This suggests two schools of thought about Maida Vale 1, those who view it favourably, and those with a moderate to strong dislike (two observers graded the studio at 0!).

Numerical data of the kind listed in Table 1 is not easy to interpret and Fig. 1 is offered as a 'profile' of the studio. No particular meaning attaches to the shape given in Fig. 1, but it is useful in comparing other studios when plotted in the same way; similarities and differences then become immediately apparent. Fig. 1 looks rather like the shape that might be expected if the peaks of random noise pulses were joined by straight lines, except for the peaks at characteristics Nos. 3 and 4 which have already been commented on. Most of the mean gradings lie in the range 4–6 and the mean of all the mean gradings is 5·19.

3.1.2 'Ideal' Studio (Maida Vale Observers)

Fourteen of the twenty-five observers who filled in questionnaires on the real Maida Vale studio gave their opinions on what would (within the terms of reference) constitute an ideal

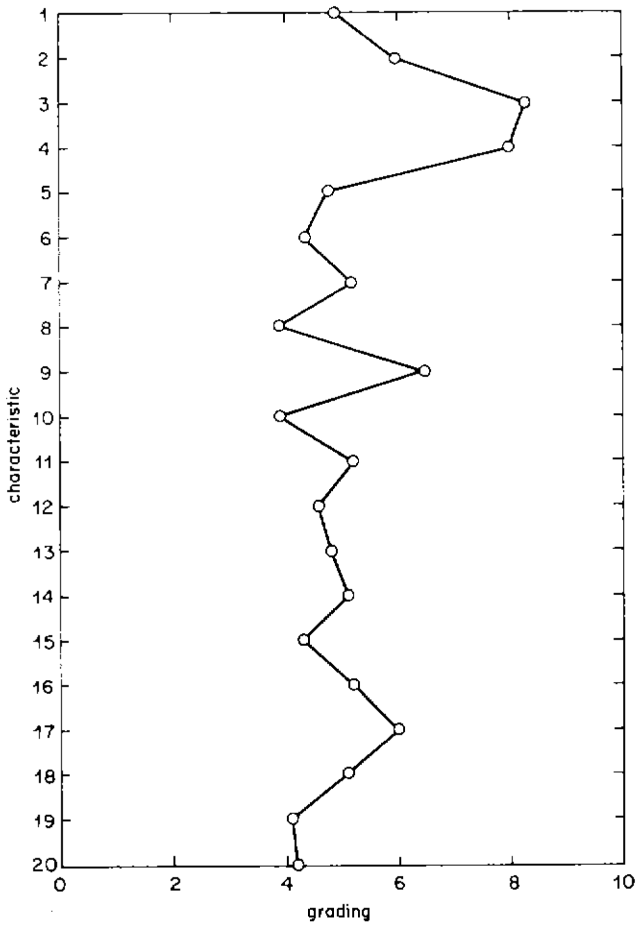

Fig. 1 Profile of real Maida Vale 1 Studio

studio. The mean gradings and standard deviations are given in Table 1 and a profile is shown in Fig. 2. One immediate observation on Fig. 2 is that some 'Ideal' characteristics are well removed from a rating of 10 and these are 'hardness' (No. 8), 'loudness' (No. 9), and 'intimacy' (No. 10) where a rating of about 6 is chosen. Although values close to 10 might be expected for most of the characteristics, in fact the average rating is only about 8, and only in the case of characteristics Nos. 5, 6, 15, 19, and 20 are the mean ratings above 9. It will also be noted that the profiles shown in Figs. 1 and 2 are different in both shape and level. The 'ideal' rating for 'loudness' is 5·77 as against 6·47 for the 'real' studio, and this supports the view expressed that Maida Vale Studio 1 is too loud (although not by too large an amount if the numerical difference in grading has quantitative significance).

Some of the answers are a little surprising and imply that there may be a linguistic or semantic difficulty. In the view of acousticians, a good studio should be free from colourations and thus characteristic No. 1 should be rated 10. In fact it was rated at 7·65 with a relatively large standard deviation (2·8). It is possible that, for some observers, absence of colouration is wrongly thought to mean a colourless studio in the sense of lifeless or dead. Coloration was one of the few terms defined in the questionnaire (Appendix 1) but this may not have been read or understood by every observer.

A similar difficulty arises with characteristic No. 3 (echoes),

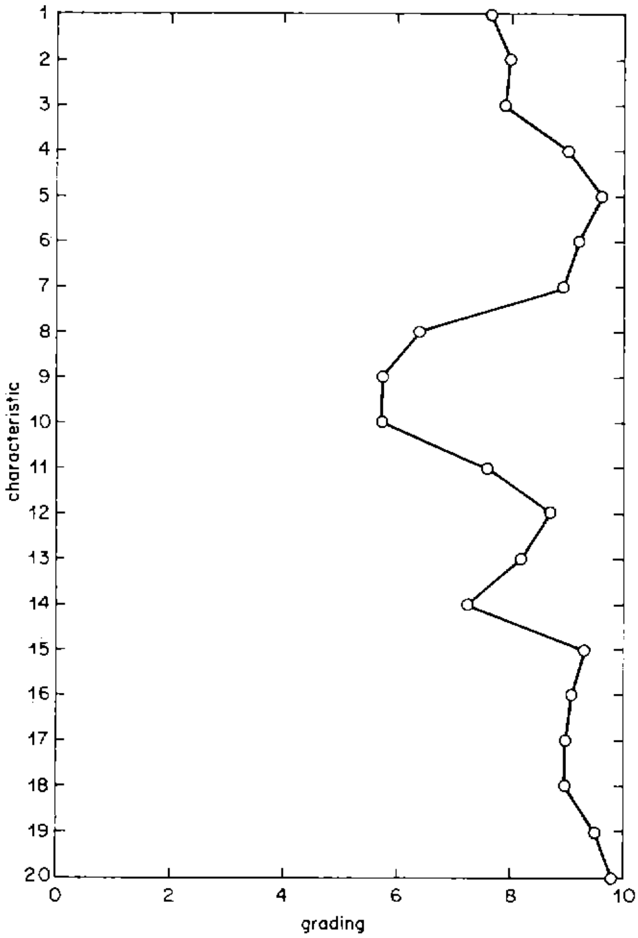

Fig. 2 Profile of ideal studio (Maida Vale observers)

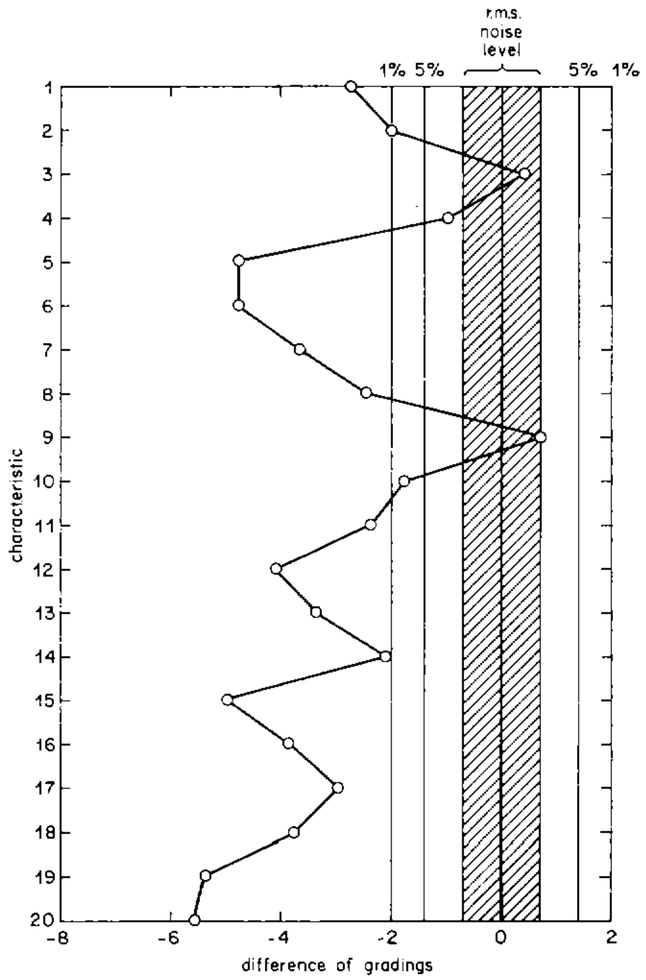

Fig. 3 Real studio compared with the ideal (Maida Vale observers)

and here the confusion may be between discrete echoes, which are usually considered harmful, and reverberation which, in the right amount, is a very desirable characteristic. Echoes were rated at 7·9 (standard deviation 2·6).

In spite of these misgivings, the difference between the profiles shown in Fig. 1 and Fig. 2 is thought to be significant and indicative of those characteristics which should be changed. Fig. 3 plots the differences together with 5 per cent and 1 per cent levels at which the differences are significant (as derived from the standard deviations taken as a whole, not for each individual characteristic). Taking the points outside the 1 per cent level (which means the probability of the two means belonging to the same statistical population is less than 1 in 100) the following changes to MV1 would appear to be desirable, and are listed in Table 2 opposite.

This seems a formidable list and it does not follow that all these changes are possible simultaneously. For instance, Nos. 8 and 13 may be incompatible in the sense that a more mellow studio is unlikely to have better definition (or for that matter, better attack (17)).

It is to be noted that loudness (9) does not appear in the list because the difference in grading is not sufficiently large.

3.1.3 Real Glasgow Studio 1

As stated in the introduction, the original intention of this investigation was to restrict attention to the Maida Vale No. 1 studio, but it was found possible to carry out a similar investigation on the studio which houses the Scottish Symphony Orchestra, namely Glasgow Studio 1, and this section reports the results obtained. Fifty-six observers took part and in contradistinction to the London results, these were obtained

TABLE 2

Characteristic	*Description*	*Direction of change*
1	Coloration	Less coloured
5	Singing tone	More singing tone
6	Ensemble	Better ensemble
7	Fullness of tone	More fullness of tone
8	Hardness	More mellow
11	Liveness	More live
12	Tonal warmth	More warmth
13	Definition	More clarity
15	Balance	Better balance
16	Blend	Better blend
17	Attack	Better attack
18	Dynamic range	More dynamic range
19	Timbre	Better timbre

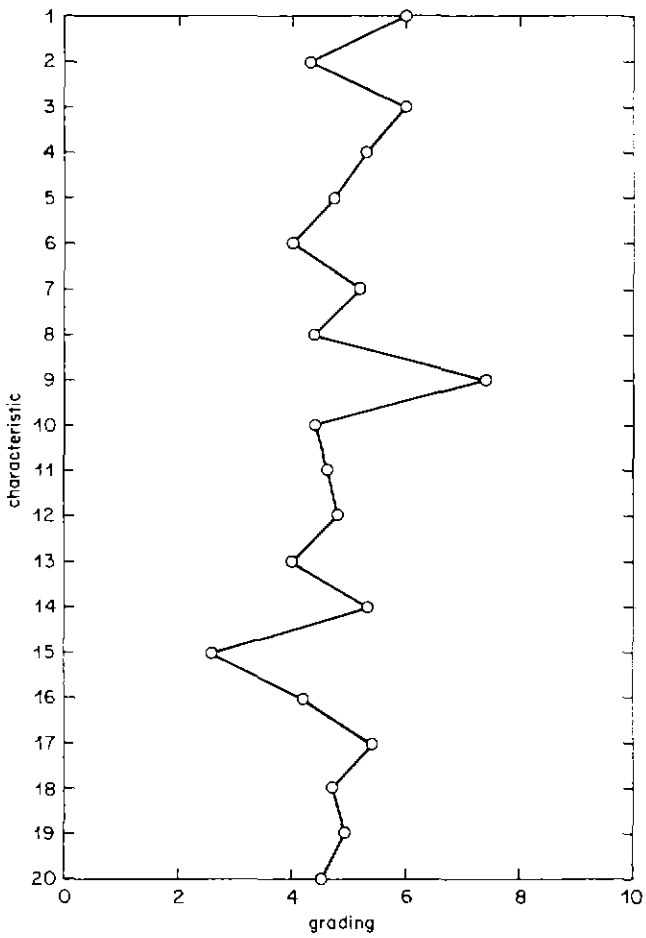

Fig. 4 Profile of real Glasgow Studio 1

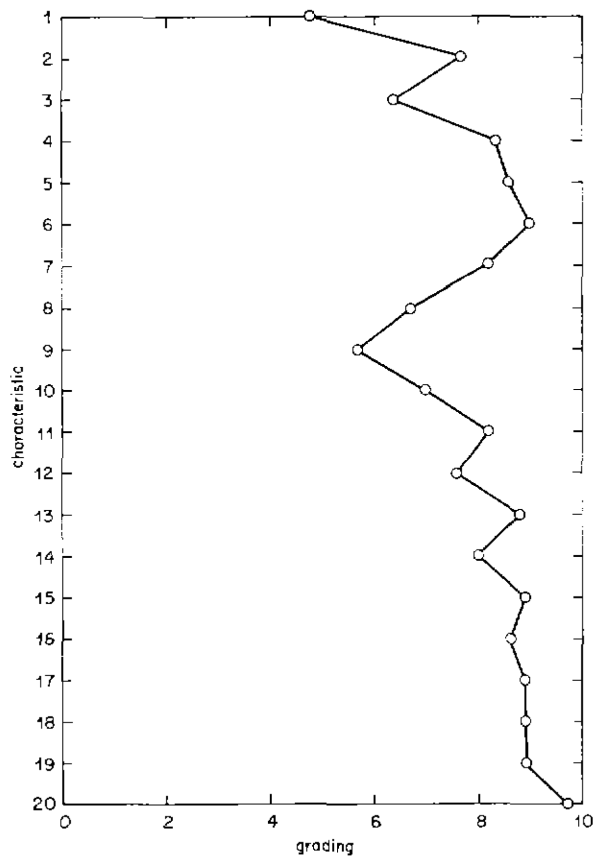

Fig. 5 Profile of ideal studio (Glasgow 1 observers)

exclusively from musicians. The results are given in Table 3, on page 38 (for the real and the 'ideal' studio). The 'profile' for this studio is given in Fig. 4. Most of the mean gradings lie in the range 4–6 as was the case with Maida Vale Studio 1; only two characteristics lie outside this range, namely No. 9 (loudness) rated at 7·4, and No. 15 (balance) rated at 2·6. The overall assessment is $4{\cdot}5 \pm 2{\cdot}5$ and like Maida Vale 1 is just below the 'half-way' mark, i.e. 5. Whether the comparatively luke-warm comments on this studio imply some measure of dissatisfaction is uncertain, although there have been comments by the conductor and members of the orchestra which imply some criticism. However, as commented for the London studio, it may well be that if one asks for criticism, one is almost certain to receive it.

3.1.4 'Ideal' Studio (Glasgow Observers)

The ratings of the fifty-six musicians of the Scottish Symphony Orchestra on what, in their opinion, constitutes an 'ideal' studio are also given in Table 3 and in Fig. 5. Once again it will be noted that the 'ideal' studio is not defined by a set of characteristics all of very high grading, i.e. between 9 and 10. Some characteristics are graded around the half-way mark, and in particular, Coloration (No. 1), Echoes (No. 3), and Loudness (No. 9), are graded at 4·8, 6·4, and 5·7 respectively. The linguistic difficulty again appears to apply to characteristic No. 1 (Coloration) in that the mean opinion for an 'ideal' studio suggests a grading of approximately 5, whereas Coloration, in the acoustician's definition of the term, is something that should be absent and, when absent, produce a grading of 10 or near to 10. The grading of Echoes at 6·4 is likewise not fully understood and maybe the same comment applies as with Maida Vale (Section 3.1.2). The two characteristics which rate between 9 and 10 are No. 6 (Ensemble) at 9·0, and No. 20 (Overall) at 9·7.

The differences between the real and 'ideal' ratings are plotted in Fig. 6, and as with Maida Vale Studio 1, a large number of characteristics give significant differences at the 1 per cent confidence level; in fact this particular studio would appear to be marginally satisfactory in respect of only two characteristics, namely Coloration (No. 1) and Echoes (No. 3): it is significantly too loud (characteristic No. 9) and in respect of all the other characteristics it is apparently deficient. This adverse judgement is possibly a little extreme, but nevertheless it is what emerges from the analysis.

3.1.5 Comparison of the 'Ideal' Studios

Sections 3.1.2 and 3.1.4 report on the views of two very different groups of observers as to what constitutes an 'ideal' studio. If the results are compared, i.e. Tables 1 and 2 or Figs.

TABLE 3
Summary of Assessments of Observers on Glasgow Studio 1

Characteristic	*Real Studio*		*Ideal Studio*	
	mean	*standard deviation*	*mean*	*standard deviation*
1. Coloration	6·0	2·4	4·8	2·7
2. Diffusion	4·3	2·3	7·7	1·9
3. Echoes	6·0	2·8	6·4	2·0
4. Pitch Changes	5·3	2·7	8·4	2·4
5. Singing Tone	4·7	2·7	8·6	2·0
6. Ensemble	4·0	2·9	9·0	1·4
7. Fullness of Tone	5·2	2·7	8·2	1·7
8. Hardness	4·4	2·4	6·7	1·8
9. Loudness	7·4	2·2	5·7	1·9
10. Intimacy	4·4	2·6	7·0	2·6
11. Liveness	4·6	2·8	8·2	2·0
12. Tonal Warmth	4·8	2·4	7·6	2·4
13. Definition	4·0	2·9	8·8	1·3
14. Brilliance	5·3	2·5	8·0	1·5
15. Balance	2·6	2·1	8·9	2·2
16. Blend	4·2	2·5	8·6	1·6
17. Attack	5·4	2·7	8·9	1·4
18. Dynamic Range	4·7	2·6	8·9	1·4
19. Timbre	4·9	2·3	8·9	1·4
20. Overall	4·5	2·5	9·7	0·6

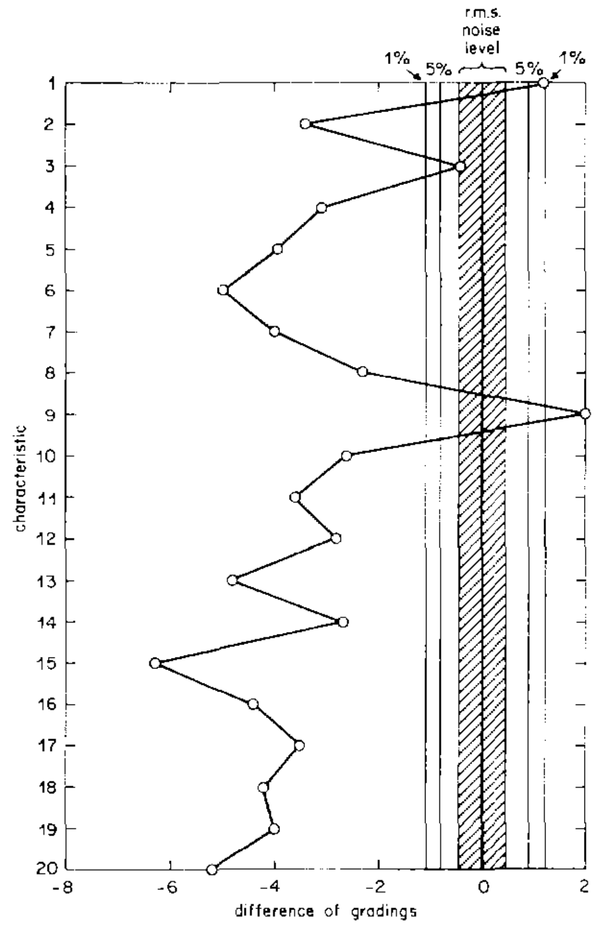

Fig. 6 Real studio compared with the ideal (Glasgow observers)

2 and 5, it will be observed that there is a significant degree of agreement on almost all the characteristics. The two about which there is some disagreement are Coloration and Echoes, and it has been pointed out that some, if not the majority, of observers appear to have difficulty in understanding these two characteristics, in the sense in which they are defined by acousticians. Not only are the backgrounds of the two groups of observers different, but also the studio which they are most accustomed to use. Thus Maida Vale 1 is relatively squat and has a height to width ratio of 0·44 at the centre falling to 0·28 at the sides, whereas Glasgow Studio 1 is much taller and has a height to width ratio of 0·74. The volumes of the studios are roughly similar; Maida Vale = 6250 cubic metres, and Glasgow = 5,150 cubic metres.

Although the 'ideal' studios envisaged by the two groups of observers appear to have strong similarities, it is difficult to be sure whether or not this is pure coincidence. Statistical tests suggest that it is highly significant.

3.1.6 Comparison of the Real Studios

The similarity between the 'ideal' studio just discussed would also appear to apply to a comparison of the real studios, although not quite to the same extent. If Figs. 1 and 4 are superposed, both graphs show a peak at characteristic No 9. (i.e. both studios sound too loud); the general trend of ratings between the gradings of 4–6 appears to be roughly similar and the only points of possibly significant difference are characteristics Nos. 3 and 4 (Echoes and Pitch Changes), where Maida Vale 1 scores high ratings, and characteristic No. 15 (Balance) where Glasgow has a lower rating. It may or may not be relevant that both studios are the subjects of criticism at the present time.*

3.2 Correlation Analysis

3.2.1 General

Correlation analysis is a powerful mathematical tool whereby it is possible to determine from a mass of data whether certain characteristics are closely (or not so closely) related to others and to determine, say, by linear regression whether a simple linear equation can serve as an adequate statistical substitute for the data. It is not usually possible to claim that the equation explains the phenomena involved because correlation analysis does not deal with cause and effect. In the present study, it may be possible to determine a linear equation which predicts with reasonable accuracy the overall assessment from a selected list of individual characteristics; in such circumstances a mathematical model has been devised which more

* Maida Vale 1 has just had its orchestral layout modified in an attempt, *inter alia*, to improve the acoustics, and Glasgow 1 is being refurbished for stereophonic operation.

or less accounts for the data and which may give some insight into how different characteristics are (subconsciously or otherwise) weighted to give the overall results.

Correlation deals with variations from mean values. It therefore has nothing to say about the importance or significance of quantities that, in a given experiment, are invariant or nearly so. To take an example, 'Echoes' (characteristic No. 3) are consistently rated highly in the real Maida Vale Studio 1 and this means that the studio is agreed to be substantially free from echoes. Therefore it is unlikely that correlation analysis will include 'echoes' as a relevant characteristic. It would be quite erroneous, however, to assume that freedom from discrete echoes is unimportant. The analysis about to be discussed concerns characteristics about which there is a variation of opinion and how this variation affects individual judgements of overall assessment.

3.2.2 Real Maida Vale Studio

The symmetrical matrix of 20 × 20 correlation coefficients relating to observations on the real studio was evaluated by computer. An examination of this matric indicated that eight independent variables were significantly related (at the 1 per cent level or lower) to the overall assessment and, on this basis, linear regression for nine variables was carried out resulting in Equation (1).

$$R_{20} = -0{\cdot}002R_5 + 0{\cdot}201R_6 - 0{\cdot}387R_9 - 0{\cdot}114R_{13} + 0{\cdot}136R_{15} + 0{\cdot}204R_{16} + 0{\cdot}148R_{18} + 0{\cdot}361R_{19} + 2{\cdot}455 \quad (1)$$

where R_{20} = rating of characteristic 20 (overall)

R_5 = rating of characteristic 5 (singing tone)

R_6 = rating of characteristic 6 (ensemble)

R_9 = rating of characteristic 9 (loudness)

R_{13} = rating of characteristic 13 (definition)

R_{15} = rating of characteristic 15 (balance)

R_{16} = rating of characteristic 16 (blend)

R_{18} = rating of characteristic 18 (dynamic range)

R_{19} = rating of characteristic 19 (timbre)

The multiple correlation coefficient was 0·893, which is a measure of the success of this particular mathematical approach. Fig. 7 shows the overall assessment derived from Equation (1) versus that directly stated by each of the twenty-five observers. Clearly there is a trend in the results shown in Fig. 7, but in terms of accuracy of estimation, Fig. 7 reveals errors of up to 2·4. The standard error of estimate is given by the formula.

$$\text{S.E.} = \sigma(1 - r^2)^{\frac{1}{2}} \quad (2)$$

where S.E. = standard error of estimate

σ = standard deviation (of R_{20})

r = multiple correlation coefficient.

In the present case $\sigma = 2{\cdot}09$, $r = 0{\cdot}893$ whence S.E. $= 0{\cdot}94$.

Thus the standard error of estimate has a value of approximately one unit, or one-tenth of the total scale used in this study. The weightings of the individual characteristics reveal some interesting features:

(*a*) The weighting of R_9 (loudness) is negative, implying that an increase produces, on the average, a lower value of R_{20} (overall).

(*b*) The weighting of R_5 (singing tone) is very small, implying that it has little effect on the overall assessment R_{20} (overall).

(*c*) The largest positive weighting coefficient, 0·361, applies to R_{19} (timbre) which thus appears to exert most influence on the overall assessment R_{20}.

(*d*) The weighting of R_{13} (definition) is negative which would appear to imply that the 'definition' of Maida Vale Studio 1 is too clear on a scale which goes from 'muddy' to 'clear', and possibly agrees with a comment that the studio is too 'hard'.

(*e*) The weighting of R_{16} 'blend' has the second largest positive coefficient and this possibly supports the negative weighting of R_{13}, 'definition'; against this, however, R_2, 'diffusion', does not appear in the correlation analysis.

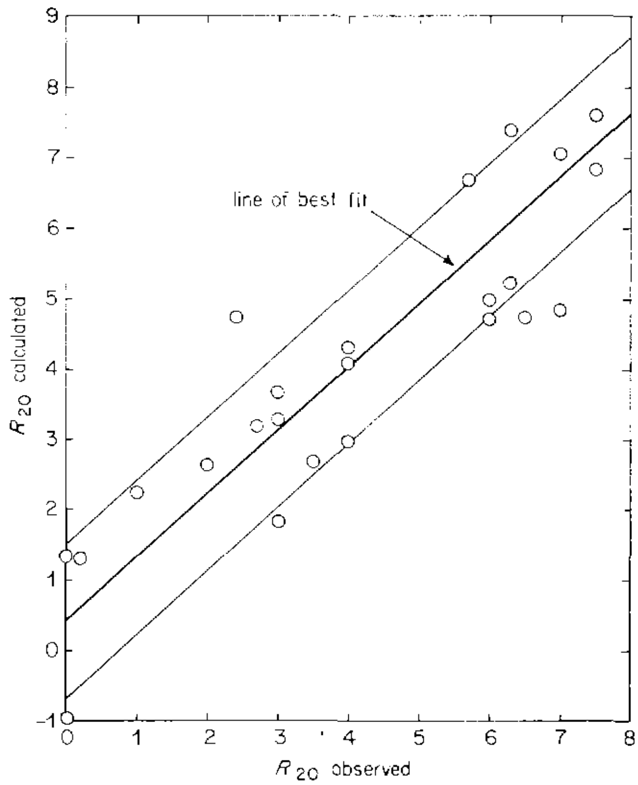

Fig. 7 Linear regression applied to the real Maida Vale Studio 1

Fig. 3 and Table 2 (Section 3.1.2) showed the extent to which the 'real' studio fell short of the 'ideal'. It is pertinent to enquire whether correlation analysis shows the same characteristics. Table 4 shows (in order of importance*) the first seven characteristics.

* It is here assumed that importance is directly related to either weighting coefficient or difference of mean gradings (see Appendix 3).

TABLE 4
First Seven Characteristics Requiring Attention

Order	*Difference of mean gradings (Fig. 3)*	*Weighting coefficient (Equ. 1)*
1	Timbre (No. 19)	Loudness No. 9) (Negative)
2	Balance (No. 15)	Timbre (No. 19)
3	Singing tone (No. 5)	Blend (No. 16)
4	Ensemble (No. 6)	Ensemble (No. 6)
5	Tonal warmth (No. 12)	Dynamic Range (No. 18)
6	Blend (No. 16)	Balance (No. 15)
7	Dynamic Range (No. 18)	Definition (No. 13) (Negative)

One immediate difficulty in lining up the two sets shown in Table 4 is the negative weighting coefficients for 'loudness' and 'definition'. Loudness (No. 9) is on the opposite side of the zero line in Fig. 3 to almost all the other characteristics although not to a significant extent. With 'definition' (No. 13) there is a complete contradiction because Fig. 3 and Table 2 imply that more clarity is required: Equation (1) says that there is too much clarity. All the remaining five characteristics, however, are in both lists, namely timbre, balance, ensemble, blend, and dynamic range. Thus it can be said that agreement exists on five characteristics when two different approaches to the observers' data are used.

3.2.3 'Ideal' Studio (Maida Vale Observers)

It is not possible to perform correlation analysis with reference to twenty variables when only fourteen completed questionnaires are available. Further, since the overall assessments are in the restricted range 9–10, any form of correlation analysis is not likely to yield very useful results (because the dependent variable is nearly constant). Regression analysis, using the variables found to be significant in the study of the real studio, was carried out, however, and the resulting 'best fit' equation is as follows:

$$R_{20} = 0{\cdot}084R_5 + 0{\cdot}664R_6 + 0{\cdot}066R_9 + 0{\cdot}086R_{13} + 0{\cdot}346R_{15} - 0{\cdot}063R_{16} + 0{\cdot}027R_{18} - 0{\cdot}399R_{19} + 2{\cdot}262 \quad (3)$$

using the same symbols as in Equation (1).

This equation has the surprisingly high multiple correlation coefficient of 0·955: the restricted range of R_{20} makes it doubtful how much significance in the psycho-acoustic sense should be inferred, although statistical significance is apparently very high.

The characteristics which appear to have most control over the final assessment are, in order of weighting coefficient:

ensemble (No. 9)
timbre (No. 19)
balance (No. 15)

(other characteristics have a weighting coefficient of less than 0·10).

The negative coefficient for 'timbre' is not understood, since good timbre is an undisputed requirement of a studio and it is not one of the characteristics where an extreme is obviously undesirable. Perhaps this result should be taken as a warning about the limitations of input data of restricted range whatever the statistical tests of significance may imply. In view of this, it was not considered profitable to analyse the consequences of Equation 3 in greater detail.

3.2.4 Real Glasgow Studio 1

Linear regression analysis applied to the 56 questionnaires relating to the real Glasgow Studio 1 was carried out with reference to the nine independent variables which were significantly correlated (at the 5 per cent or lower level*) with overall assessment. Equation (4) gives the resulting linear formula.

$$R_{20} = 0{\cdot}203R_5 + 0{\cdot}095R_6 + 0{\cdot}206R_{10} - 0{\cdot}047R_{12} - 0{\cdot}091R_{14} + 0{\cdot}380R_{15} + 0{\cdot}253R_{16} + 0{\cdot}106R_{17} - 0{\cdot}620 \quad (4)$$

The ninth variable R_{19} has been omitted because the coefficient (−0·0003) was so small a value as to have no perceptible influence on the result.

The multiple correlation coefficient is 0·747 which is highly significant (the change of this occurring by random probability is much less than 0·1 per cent), but the standard error of estimate is (as Equation (2))

$$\text{S.E.} = \sigma(1 - r^2)^{\frac{1}{2}} = 0{\cdot}4415 \qquad \sigma = 1{\cdot}10$$

The negative sign describing R_{12} (tonal warmth) is not understood, but at least the magnitude of the coefficient is small compared to the other coefficients.

It is instructive to compare (as in Section 3.2.2) those characteristics which were shown to be most deficient when the mean gradings of the ideal studio were subtracted from those of the real studio (Fig. 6) with the characteristics in Equation (4) taken in order of weighting coefficient. Table 5 collates the data.

TABLE 5
The First Eight Characteristics Requiring Attention

Order	*Difference in mean grading (Fig. 6)*	*Weighting coefficient (Equ. 4)*
1	Balance (No. 15)	Balance (No. 15)
2	Ensemble (No. 6)	Blend (No. 16)
3	Definition (No. 13)	Intimacy (No. 10)
4	Blend (No. 16)	Singing Tone (No. 5)
5	Dynamic Range (No. 18)	Attack (No. 17)
6	Fullness of Tone (No. 7)	Ensemble (No. 6)
7	Timbre (No. 19)	Brilliance (No. 14)
8	Singing Tone (No. 5)	Tonal Warmth (No. 12) (Negative)

* In the corresponding analysis for Maida Vale 1, the 1 per cent level was chosen because a higher general level of correlation held in that studio and eight variables 'qualified'. At the 1 per cent level, five variables 'qualified' for Glasgow 1 and it was decided to include four additional variables in the 5 per cent to 1 per cent category. The analysis effectively eliminates variables which are not important in terms of the final 'best-fit' equation.

In the case of Maida Vale Studio 1, five of the seven characteristics listed in Table 4 were common to both lists: here the agreement is rather less in that only four of the eight characteristics are common, namely balance, blend, ensemble, and singing tone. Balance is the most important characteristic in both lists; the others vary in their respective placings. Since two dissimilar methods give these four characteristics it would seem highly probable that suitable improvements relating to balance, blend, ensemble, and singing tone would produce an appreciably better studio, at least in the view of the members of the BBC Scottish Symphony Orchestra.

3.2.5 Ideal Studio (Glasgow Observers)

Correlation/regression analysis was not applied to the individual ratings of the ideal studio because it is not thought to be a likely source of useful information and the anomalous results reported in Section 3.2.3 tended to support this view.

3.3 Factor Analysis

3.3.1 General

Several approaches to factor analysis are described in the literature.[5] The one which has been used in the present work is the 'principal components' method, which is capable of producing 'm' orthogonal factors from 'n' original variables contained in N sets of data ($m \leq n < N$); the factors are selected in decreasing order of their variances. One undesirable feature of this method is that each factor uses all the original 'n' variables, although, hopefully, certain of the variables might dominate and others be of much less importance. Whether the 'm' orthogonal factors correspond to qualities which have any physical or psycho-physical significance is something that cannot be predicted. The mathematical procedure amounts to finding the latent roots in an $m \times n$ matrix equation: orthogonality and decreasing order of variances can be guaranteed but are not necessarily of physical psycho-physical significance.

The purpose of factor analysis is to reduce the number of variables required to describe a situation to the minimum compatible with no significant loss of information. Nineteen independent variables have been used in the questionnaire, but as stated in Section 2, it is thought unlikely that nineteen variables (when optimally defined) will be required to describe the acoustic quality of a studio. In more recent studies, the number of factors has been arbitrarily set to seven in the hope that seven or less factors (i.e. qualities or combinations of qualities) should be sufficient.

3.3.2 Real Maida Vale Studio 1

Factor analysis was applied to the nineteen independent variables contained in the twenty-five questionnaires and the first seven factors were extracted by the Elliott 803 computer using a factor analysis programme. The first seven factors extracted 83·5 per cent of the total variance. A detailed and careful examination of these factors failed to reveal any physical significance attached to the individual factors: one disappointing feature was the absence of any clear differentiation between qualities which dominated and those which were much smaller in importance. For example, the first factor

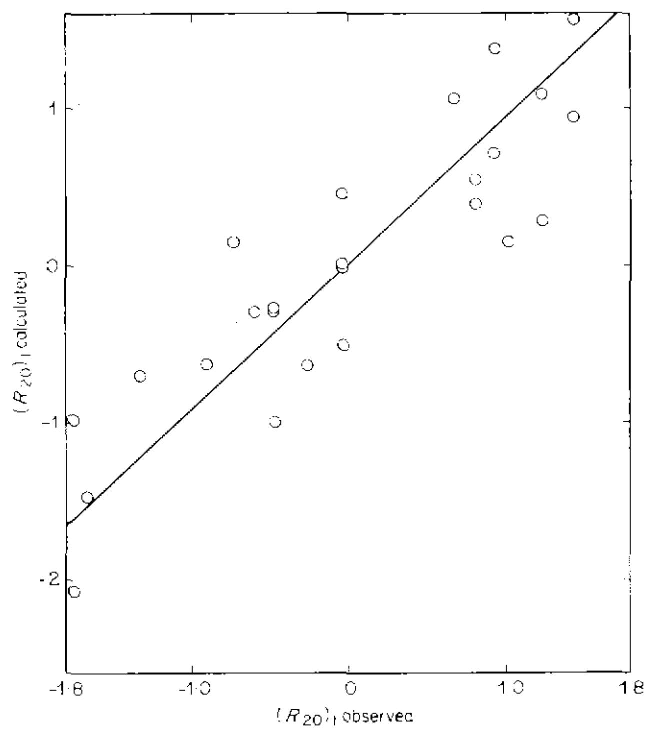

Fig. 8 Factor analysis applied to the real Maida Vale Studio 1

(F_1) (standard deviation, $\sigma = 5{\cdot}7098$) included fifteen of the nineteen variables with gradually decreasing weighting coefficients (starting with No. 5 (singing tone) coefficient $= 0{\cdot}8279$ and finishing with No. 10 (intimacy) coefficient $= 0{\cdot}2082$) before there was any appreciable change in ratios of successive coefficients. This factor (F_1) correlates well with the overall assessment No. 20 (correlation coefficient $r = 0{\cdot}809$), but a factor which includes almost all the variables is not capable of any easy interpretation. The second factor, F_2, included seventeen variables before any 'break-point'. The only factors where there was a clear tendency for a single (or just a few) variables to separate out were F_4 and F_5, where echoes and intimacy, respectively, were outstanding from the remaining variables by ratios of nearly two to one. Neither of these factors, however, correlates significantly with the overall assessment. The complete results are given in Appendix 2.

During the course of the work, it became apparent that the original data from a rather limited number of questionnaires ($N = 25$) did not much exceed the number of independent variables ($n = 19$), and therefore some degree of correlation may well have existed even if the original data has been sets of random numbers. Indeed, this thought prompted a run of the programme with two sets of data consisting entirely of random numbers. The variances extracted by the first seven factors were 72 per cent and 70 per cent respectively, which is well above the 37 per cent (7/19ths) which would be expected from random numbers if N greatly exceeded n. In other words, short sequences of random numbers are bound to show correlations and the notion of a short sequence of random numbers is untenable. The conclusion from this part of the study was that the number of questionnaires was very inadequate for this kind of powerful mathematical analysis.

An attempt to circumvent the above difficulty was made by applying factor analysis using only the eight most significant

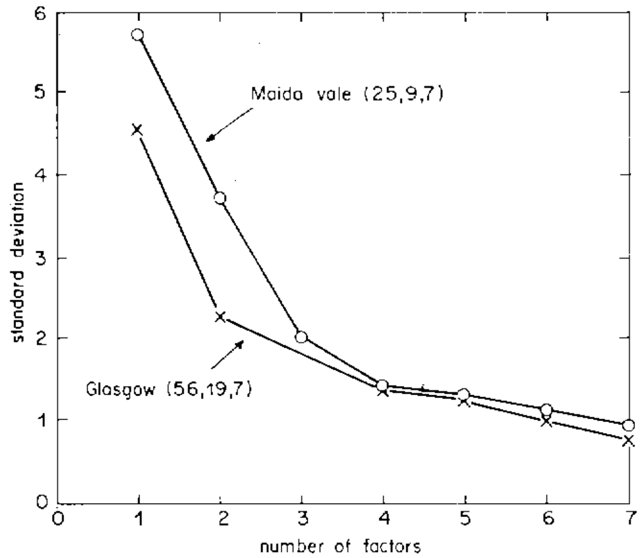

Fig. 9 Extraction of the factors and their corresponding standard deviations for the real Glasgow Studio 1 and real Maida Vale Studio 1 results. The three numbers in the bracket are numbers of sets of data, number of independent variables and number of factors respectively

variables (Section 3.2.2, the variables used in Equation (1)) and extracting the first seven factors ($N = 25$, $n = 8$, $m = 7$). In one sense, this approach was much more successful in that the first factor F_1 extracted 55 per cent of the total variance. None of the factors were capable of any obvious psycho-physical interpretation, however, and from this point of view, the exercise was no more successful than the previous one using nineteen variables. Only two of the factors, F_1 and F_6, correlated well with characteristic No. 20 (overall assessment) and Equation (5) gives a multiple-correlation coefficient of 0·883. Fig. 8 shows the 'goodness of fit' for the twenty-five

$$R_{20} = -0{\cdot}195F_1 + 0{\cdot}590F_6 \qquad (5)$$

observers. Note that the results are here plotted in statistical 't' units, i.e. with zero mean value and unit standard deviation, but this involves only a linear transformation of the original data, i.e. shift of zero and scale change. If it were possible to find names or concepts relating to F_1 and F_6 it would be possible to claim, substantially, to have 'explained' overall assessment in terms of two factors, at least as far as Maida Vale 1 is concerned.

3.3.3 Ideal Maida Vale Studio 1

No attempt was made to apply factor analysis to these results on account of the relatively small number of questionnaires ($N = 14$), and the fact that the less sophisticated regression analysis yielded results which were statistically significant but physically meaningless, or, to say the least, difficult to interpret.

3.3.4 Real Glasgow Studio 1

The problem of N exceeding n by a relatively small number does not arise with the Glasgow results where $N = 56$; $n = 19$, and $m = 7$ as before. Hence the probability of relatively high correlations from chance coincidence of sequences of numbers is low. The way in which the variance is extracted by the first seven factors is shown in Fig. 9, which also gives the Maida Vale result for comparison. The levels are well above those expected from a truly random set of numbers. Examination of the weightings of the original variables to form the factors once again reveals no obvious psycho-physical significance in that no 'break-points' occur after the first few variables, thus separating them out from the remainder. The first factor, F_1, correlates very significantly with the overall assessment ($r = -0{\cdot}57$)* and the second is probably significant ($r = -0{\cdot}27$).* The remaining five factors do not correlate significantly. Arising out of this factor analysis, Fig. 10 shows the measure of agreement between the overall assessment calculated from Equation (6) and the directly observed overall assessment.

$$R_{20} = -0{\cdot}198F_1 - 0{\cdot}193F_2 \qquad (6)$$

The units in Fig. 10 are statistical 't' units as in Fig. 8. Although a general trend is clearly discernable, the accuracy of estimates is relatively poor, so that Equation (6) cannot be regarded as 'explaining' the way in which judgements were made. The first five characteristics comprising F_1 and F_2 are listed in Table 6, but it must be acknowledged that the choice of five is entirely arbitrary as there is no 'break-point' between variables 5 and 6.

A comparison of Factor F_1 in Table 6 with the correspond-

* No significance attaches to the negative sign in this context because the polarity of the factors is arbitrary and a new F_1 and F_2 could easily be defined with opposite signs which would have all the requisite mathematical properties.

TABLE 6

Order	*Factor 1*		*Factor 2*	
	Characteristic	*Weighting Coefficient*	*Characteristic*	*Weighting Coefficient*
1	17 Attack	−0·735	3 Echoes	−0·588
2	5 Singing tone	−0·641	11 Liveness	0·552
3	19 Timbre	−0·633	6 Ensemble	−0·473
4	7 Fullness of tone	−0·608	7 Fullness of tone	0·450
5	18 Dynamic range	−0·607	13 Definition	−0·440

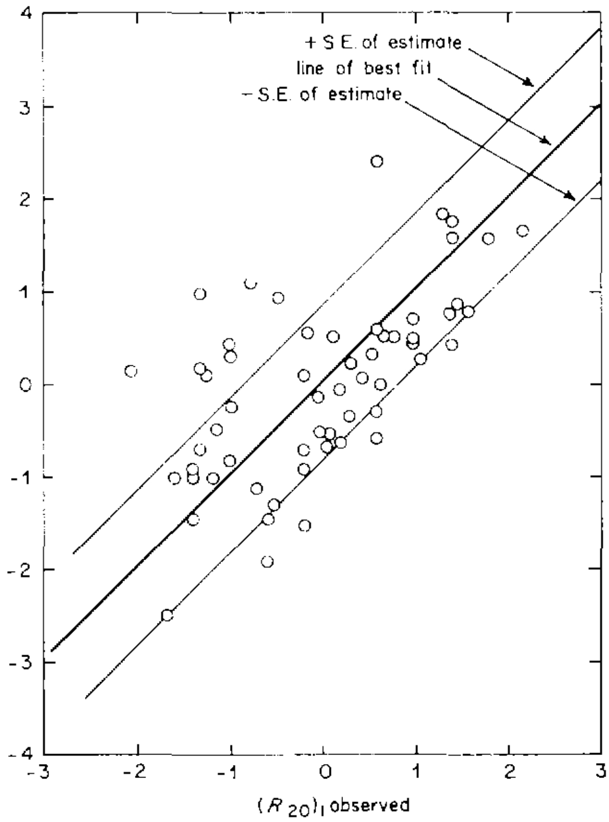

Fig. 10 Factor analysis applied to the real Glasgow Studio 1

ing result for Maida Vale 1 (Appendix 2) shows that three of the five characteristics are common, namely singing tone (5), dynamic range (18), and timbre (19). This commonality is possibly significant but too much should not be inferred because, as stated above, there are no logical grounds for selecting the first five characteristics.

Factor analysis was not carried out on the questionnaires relating to the Ideal Glasgow Studio, not because there was an inadequate number of questionnaires, but rather because of the almost complete lack of success in the preceding applications.

4 Conclusions

Various types of analysis have been applied to the raw data and the results have not always yielded significance in the psycho-acoustical sense or in the sense of correlating well with general experience. However, some of the results are probably significant and are certainly interesting.

(*a*) The considerable similarity in the characteristics of an ideal music studio as given by two dissimilar groups of observers instils confidence that the answers given are a true (or very near to a true) representation of a suitable combination of individual characteristics to define this type of studio.

(*b*) The real Maida Vale Studio 1 is shown to fall short of the ideal in respect of a considerable number of characteristics. Likewise Glasgow Studio 1 also falls short of the ideal and in many ways the deficiencies appear to be similar to those of Maida Vale 1. Both studios were designed with similar objectives and there is little doubt that the objectives have been largely achieved. This leads one to question whether these objectives are consistent with present-day ideals: perhaps some rethinking of BBC methods of acoustic design is called for. Both studios are of similar volume and it is just possible that they are both not really large enough for a full symphony orchestra. The present analysis is incapable of determining whether this is true or false.

(*c*) Linear regression analysis applied to the two real studios has produced formulae each of which has a high multiple correlation coefficient. More confidence might be attached to this approach if the formulae had been very similar, but this was not the case in that only four of the eight variables used were common (see Equations (1) and (4)).

(*d*) The failure to find a few significant factors is disappointing, and poses the query whether the raw data is unsuitable (in the sense that each set of data relates to only one studio), or whether alternative methods of factor analysis should be investigated. In view of the importance of finding answers to the questions, it is felt that further work is required on this problem, which is not capable of an easy solution. One hopeful new approach is offered by the use of a scale model and work using this is now proceeding. A scale model enables changes to be made to a single feature (or a small group of features) and the consequent effects on the acoustics of the hall can then be estimated by subjective assessment techniques coupled with appropriate objective measurements of relevant parameters in the model.

5 Acknowledgments

I am grateful to A. N. Burd and H. D. Harwood for much helpful discussion and assistance during the progress of this work.

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FORM 1A

1. Colouration
0 Highly Coloured — 10 Not Coloured

2. Diffusion
0 Poorly Diffused — 10 Well Diffused

3. Echoes
0 Prominent Echoes — 10 No Echoes

4. Pitch Changes
10 No Pitch Changes — 0 Large Pitch Changes

5. Singing Tone
10 Good Singing Tone — 0 Poor Singing Tone

6. Ensemble
0 Poor Ensemble — 10 Good Ensemble

7. Fullness of Tone
10 Full Tone — 0 Thin Tone

8. Hardness
0 Hard — 10 Mellow

9. Loudness
0 Quiet — 10 Loud

10. Intimacy
10 Intimate — 0 Distant

11. Liveness
0 Dead — 10 Live

12. Tonal Warmth
10 Warm — 0 Cold

13. Definition
0 Muddy — 10 Clear

14. Brilliance
10 Brilliant — 0 Dull

15. Balance
0 Unbalanced — 10 Balanced

16. Blend
10 Well Blended — 0 Not Blended

17. Attack
0 Poor Attack — 10 Good Attack

18. Dynamic Range
10 Large — 0 Small

19. Timbre
0 Poor — 10 Good

20. Overall
10 Liked — 0 Disliked

21. Remarks:

Signature .... Date ....

Appendix 1

Assessment of Acoustic Quality of Maida Vale Studio 1

In order to help us arrive at an agreed assessment of the acoustic qualities of Maida Vale 1 Studio as used for symphonic or choral works, we are asking a selected number of people to fill up the attached questionnaires. You will see that under each heading is a scale running between two extreme descriptions and we would like you to place a cross at a point which indicates your estimate of the extent to which this parameter is possessed by the studio. The cross can be placed anywhere on the scale, not necessarily at the markers.

Example:

Temperature

Warm — Cold

It is important that the forms be returned by 15 August to the following address:

Mr H. D. Harwood,
BBC Research Department,
Kingswood Warren,
Tadworth,
Surrey.

In order to avoid affecting the investigation we would ask you not to discuss it with anyone else. Do not spend more than five minutes filling up the form.

If you think that there are any important features of Maida Vale 1 Studio which we have not covered would you please mention them in the final remarks column.

To clarify matters we define below a few unusual terms we have used:

Coloration
A characteristic timbre possibly having a locateable pitch.

Diffusion
A sense that the reverberant sound comes from all directions at once.

Dynamic Range
The difference in loudness between the quietest and loudest sound in the studio.

In addition to help us assess the meaning of your judgement we would ask you to fill in Form 2 indicating your idea of an ideal studio.

Appendix 2

Maida Vale 1 Factor Analysis

Factor F_1 is highly correlated with R_{20} ($r = 0{\cdot}809$); it contains all nineteen variables (as do factors F_2 to F_7), but range of weighting coefficients vary from 0·8279 (No. 5) to 0·0446 (No. 3).

Complete list is as follows:

Order	Coefficient	Quality	Description
1	0·8279	5	Singing tone
2	0·7851	6	Ensemble
3	0·7637	18	Dynamic range
4	0·7631	13	Definition
5	0·7051	19	Timbre
6	0·6756	17	Attack
7	0·6585	7	Fullness of tone
8	0·6061	15	Balance
9	0·5947	12	Tonal warmth
10	0·5940	16	Blend
11	0·5013	14	Brilliance
12	−0·4555	9	Loudness
13	0·3695	8	Hardness
14	0·3600	11	Liveness
15	0·2082	10	Intimacy
16	0·0678	1	Coloration
17	0·0654	2	Diffusion
18	0·0468	4	Pitch changes
19	0·0446	3	Echoes

Factor F_2 is much less well correlated with R_{20} ($r = 0{\cdot}260$); it contains all nineteen variables with weighting factors ranging from 0·7993 to 0·0305.

Complete list:

Order	Coefficient	Quality	Description
1	0·7993	1	Coloration
2	−0·7514	11	Liveness
3	−0·6609	9	Loudness
4	−0·6075	2	Diffusion
5	−0·5213	8	Hardness
6	−0·5011	12	Tonal warmth
7	−0·4862	7	Fullness of tone
8	−0·4511	4	Pitch changes
9	0·4378	10	Intimacy
10	0·4190	13	Definition
11	−0·3075	17	Attack
12	0·2925	18	Dynamic range
13	0·2713	6	Ensemble
14	0·2173	14	Brilliance
15	−0·1905	3	Echoes
16	0·1747	15	Balance
17	−0·1559	5	Singing tone
18	0·0822	16	Blend
19	0·0305	19	Timbre

Factor F_3 has correlation coefficient of 0·0581 with R_{20}; No. 4 (Pitch Changes) is highest on the list with a coefficient of −0·6841; No. 6 (Ensemble) is lowest.

Complete list is as follows:

Order	Coefficient	Quality	Description
1	−0·6841	4	Pitch changes
2	0·6282	8	Hardness
3	−0·4856	14	Brilliance
4	0·3754	12	Tonal warmth
5	0·3404	7	Fullness of tone
6	−0·3338	17	Attack
7	−0·3203	2	Diffusion
8	−0·3120	10	Intimacy
9	−0·2990	11	Liveness
10	−0·2816	9	Loudness
11	−0·2139	13	Definition
12	−0·1711	15	Balance
13	−0·1694	16	Blend
14	0·1448	5	Singing tone
15	0·1336	19	Timbre
16	0·1124	18	Dynamic range
17	0·0875	3	Echoes
18	0·0766	1	Coloration
19	−0·0698	6	Ensemble

Factor F_4 correlates with R_{20} with coefficient 0·0400; Echoes (No. 3) is the predominant characteristic; seven of the characteristics have coefficients less than 1/10 that of 'Echoes'.

Order	Coefficient	Quality	Description
1	−0·8251	3	Echoes
2	0·4501	15	Balance
3	−0·3975	14	Brilliance
4	0·2615	19	Timbre
5	0·2492	2	Diffusion
6	−0·2428	17	Attack
7	0·2337	4	Pitch changes
8	−0·1858	6	Ensemble
9	0·1720	1	Coloration
10	0·1302	5	Singing tone
11	0·1247	16	Blend
12	−0·1188	13	Definition
13	0·0709	7	Fullness of tone
14	0·0689	10	Intimacy
15	0·0473	18	Dynamic range
16	−0·0310	8	Hardness
17	0·0255	11	Liveness
18	−0·0207	9	Loudness
19	0·0043	12	Tonal warmth

Factor F_5 correlates with R_{20} with a coefficient of 0·0169; Intimacy (No. 10) appears to the one predominant characteristic; $r = 0{\cdot}0169$, highest coefficient $= 0{\cdot}6361$, lowest coefficient $= 0{\cdot}0220$.

Order	Coefficient	Quality	Description
1	−0·6361	10	Intimacy
2	0·3767	19	Timbre
3	−0·3158	4	Pitch changes
4	0·3110	15	Balance
5	0·2896	2	Diffusion
6	−0·2807	7	Fullness of tone
7	−0·2320	12	Tonal warmth
8	0·2265	13	Definition
9	0·2250	9	Loudness
10	−0·2214	17	Attack
11	0·1916	14	Brilliance
12	0·1795	18	Dynamic range
13	−0·1743	5	Singing tone
14	0·1302	3	Echoes
15	−0·0890	6	Ensemble
16	0·0664	11	Liveness
17	0·0449	16	Blend
18	−0·0445	1	Coloration
19	0·0220	8	Hardness

Factor F_6 correlates with R_{20} with a coefficient of −0·1536; there is no outstanding characteristic and the weighting coefficients vary from −0·4516 to −0·0171.

Order	Coefficient	Quality	Description
1	−0·4516	2	Diffusion
2	0·3582	14	Brilliance
3	−0·3555	10	Intimacy
4	−0·3477	18	Dynamic range
5	−0·3344	3	Echoes
6	−0·2571	8	Hardness
7	−0·1840	16	Blend
8	0·1758	12	Tonal warmth
9	0·1657	13	Definition
10	−0·1528	6	Ensemble
11	−0·1474	1	Coloration
12	0·1309	5	Singing tone
13	0·1261	7	Fullness of tone
14	0·1082	19	Timbre
15	−0·0814	15	Balance
16	−0·0741	9	Loudness
17	0·0647	11	Liveness
18	0·0489	17	Attack
19	−0·0171	4	Pitch changes

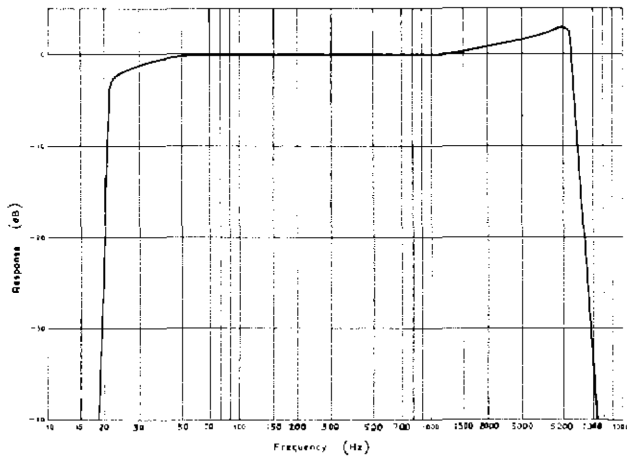
Factor F_7 correlates with R_{20} with a coefficient of −0·1066. Thirteen qualities are included before any 'break-point' appears. Highest coefficient $= 0{\cdot}4072$, lowest coefficient $= 0{\cdot}0129$.

Order	Coefficient	Quality	Description
1	−0·4072	19	Timbre
2	0·3637	1	Coloration
3	−0·3304	6	Ensemble
4	0·2626	2	Diffusion
5	0·2293	7	Fullness of tone
6	0·1985	12	Tonal warmth
7	0·1850	13	Definition
8	−0·1772	4	Pitch changes
9	0·1530	15	Balance
10	−0·1527	18	Dynamic range
11	0·1500	16	Blend
12	0·1468	14	Brilliance
13	−0·1450	5	Singing tone
14	−0·0729	9	Loudness
15	0·0514	17	Attack
16	0·0431	11	Liveness
17	−0·0348	10	Intimacy
18	0·0225	8	Hardness
19	−0·0129	3	Echoes

Appendix 3
Relevance of Weighting Coefficients

It is not necessarily true that weighting coefficients in a linear regression equation describing acoustic quality are related to deficiencies in the acoustics of a studio. In fact, as a general statement it is probably untrue. In the present investigation, however, the situation is that fifteen of the twenty-five observers gave adverse judgements of Maida Vale 1 (i.e. overall rating on a like/dislike axis of less than five) and it is probable that those qualities which correlate highly with the overall rating are also those which caused the unfavourable judgements. Hence these are the qualities the ratings of which, in the view of more than half the observers, ought to be changed so that the acoustics of Maida Vale 1 should be improved. This is clearly not an argument that can be strictly proven on logical grounds and the whole approach of using correlation analysis on only one studio makes assumptions which are doubtful, to say the least. However, the apparent success of the exercise (in the sense of producing multiple correlation coefficient of about 0·9) tends to strengthen the view that this approach may well give some useful and significant information.

Low-pass Audio Filter



Low-pass Audio Filter

The low-pass audio filter FL4/55 has been designed for incorporation in the programme-chain feeding a medium-frequency transmitter, for the purpose of attenuating frequencies above 5 kHz, and hence the corresponding radiated sideband-frequencies, in order to reduce adjacent-channel interference between m.f. transmissions.

The filter is a three-section pi-type, fed from an emitter-follower and driving an output amplifier via an attenuator-pad and an equaliser giving 'top lift' at 5 kHz of 3 dB relative to the level at 1 kHz. Above 5 kHz and below 20 Hz the response of the system cuts off sharply.

The input and output of the unit are transformer-coupled and earth-free. The unit has unity gain, and is designed to operate with a zero-level programme input. The input impedance is high, and the output load should be 600 ohms or higher.

The filter and associated circuits, with a power-supplier, are constructed in a cast aluminium box measuring 270 mm × 170 mm × 55 mm, bolted to a standard 483 mm × 90 mm panel for bay-mounting. Connections are via a seven-pin connection and a three-way mains connector.

New Audio Limiting Amplifier

The audio limiting amplifier AM6/14 has been designed primarily for transmitter protection, but has facilities which also make it suitable for studio use. Two of the amplifiers can be ganged for stereophonic use by means of simple wired connections between tags of their mating connectors. This facility is not available in earlier limiting amplifier designs now in use in the BBC, and the new amplifier offers another important practical advantage in so far as the earlier designs needed specially-selected field-effect transistors, while the AM6/14 uses an integrated circuit which makes this unnecessary.

The limiting action is free from overshoot. A meter on the front panel indicates gain reduction, and the gain-recovery time is switchable to any of seven fixed values between 160 ms and 3 s or to 'automatic' recovery, i.e. a condition in which the recovery-time is normally 10 s but changes to approximately 0·5 s if gaps of 1 s or more occur in the programme. Another facility activated by gaps in the programme (in this instance gaps of 0·6 s or more) is a noise gate which can be switched into circuit to reduce the gain of the amplifier by 12 dB when such gaps occur.

A 'voice-over' input circuit is provided to enable a second incoming signal to be used, in addition to the signal in the side-chain of the amplifier, to reduce the gain of the main chain.

An input attenuator is provided to enable the amplification of an incoming 0 dB-volume signal to be varied, in 2 dB steps, so as to produce up to 20 dB of gain-reduction by the limiting-circuit.

The gain-recovery, noise-gate and attenuator switches are all front-panel controls.

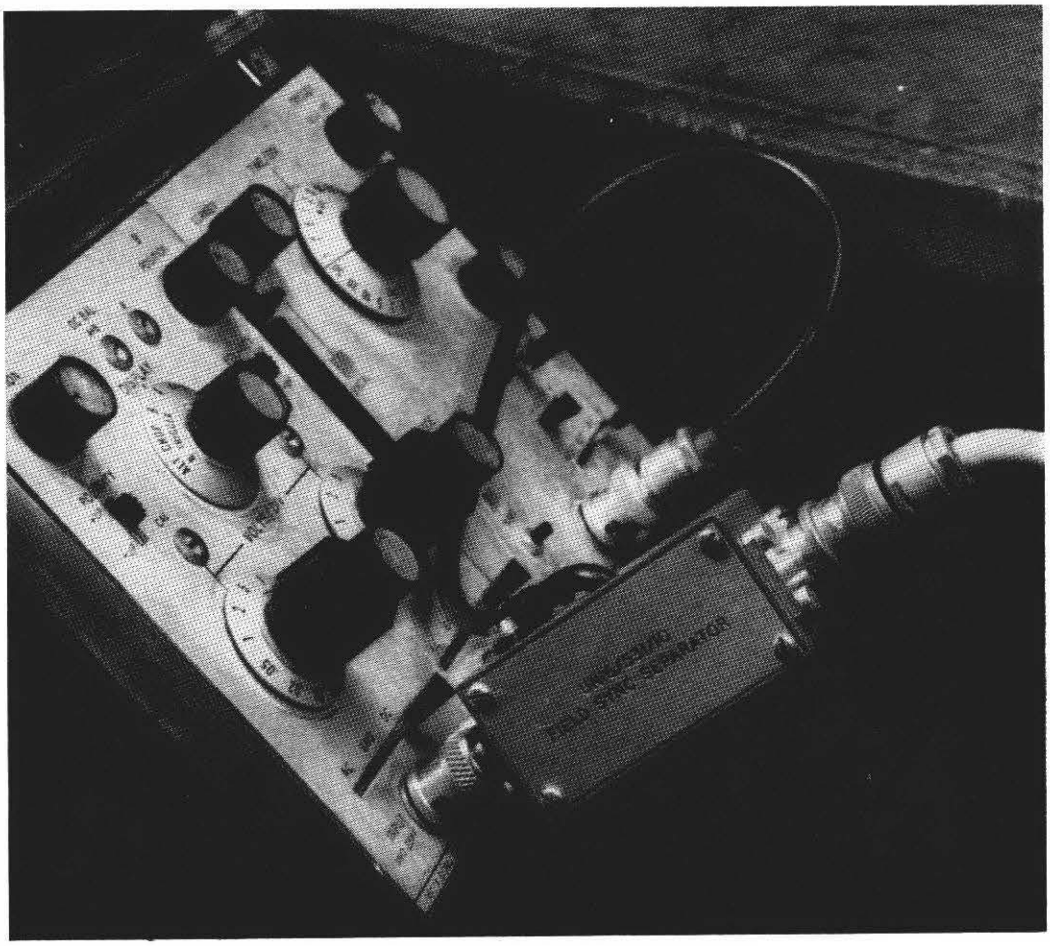
A pre-emphasis network with a time-constant of 50 μs can be introduced into the control-chain feed.

The onset of limiting causes a relay to operate, the contacts of which are available via pins of the unit connector to mute automatic monitors.

The unit is constructed on a chassis type CH1/26B.

Performance Data:

Power requirement	135 mA d.c. at 24 V
Design source impedance	600
Input impedance	10 k
Output impedance	60 (maximum)
Design load impedance	600 (minimum)
Input signal level	+ 8 dB peak
Output signal level	+ 8 dB peak (− 12 dB peak, by internal adjustment at f.j. transmitters)
Limiting range	20 dB



Field Sync-pulse Separator for Triggering Oscilloscopes

The field sync-pulse separator UN 16/531 will enable general-purpose oscilloscopes to be triggered at field-repetition rate. It is designed to operate from an input of either composite video signal or mixed synchronising-pulses from a $75\,\Omega$ source in the level-range -20dB to $+14$dB with respect to 1 V, from which it produces field-synchronising-pulses with an amplitude of 1 V p–p. In use, the unit is interposed in the main input circuit to the oscilloscope (where it introduces a simple d.c. restorer) and the separated synchronising-pulses are fed to the triggered input of the instrument.

The unit is constructed in a diecast metal box, measuring 57 mm $\times$ 28·5 mm $\times$ 22 mm, fitted with BNC connectors which enable it to be fitted at the input connector to the oscilloscope. The synchronising-pulses are fed to the trigger input of the instrument via a flying lead, connection of which closes the circuit of the internal dry cell which powers the unit. This cell (Mallory type RM265) is claimed to have a life of six months' continuous use at the rate of discharge due to the unit.

Pilot-tone Synchronisation System for B62 Audio Recorders

This system has been designed to enable a B62 twin-track audio recorder to be operated in synchronisation with a television waveform. This is achieved by recording, on one of the tracks, a pilot-tone derived from the television waveform.

The system consists of an electronics unit, comprising two printed-circuit boards, which plugs into the spare module position in the programme-electronics nest of the B62. The first board accepts the mixed synchronising pulses of the television waveform and produces a 50 Hz waveform suitable for recording on track-2 of the B62. The other printed-circuit board accepts the recorded 50 Hz from the tape and produces a control voltage which is fed into the capstan-servo amplifier of the recorder.

Mounted on the rear top horizontal member of the trolley and located mid-way between the spools, are an indicator lamp which, when illuminated, indicates that the recorder is running in synchronism, and a key which enables the tape speed to be advanced or retarded by 5 per cent. The indicator lamp, which is energised by the local 50 V supply, can be remoted by means of a switched-earth available on pin B11 of the 50-way Hypertac connector.

The power required for the system is obtained from the $\pm$12-volt supplies within the B62.

Minor modifications are required to the B62 capstan-servo board to enable the $\pm$5 per cent speed change to be achieved, together with some wiring changes to the existing socket which receives the electronics unit. The modified B62 with trolley and electronics unit is coded RD4/10.

Contributors to this issue

James Redmond, a Scot, was educated at Falkirk Technical School and Caledonian Wireless College, Edinburgh. After serving as a Marine Radio Officer he joined the BBC in 1937 as a Sound Engineer at Edinburgh, and during the war years he served once again in the Merchant Navy. When he rejoined the BBC in 1946 he became a Television Engineer at Alexandra Palace, and in 1949 he joined the Planning and Installation Department. From 1956 to 1963 he was successively Assistant Superintendent Engineer, Television Films; Superintendent Engineer, Television Recording; and Superintendent Engineer, Television Regions and Outside Broadcasts. He became Senior Superintendent Engineer, Television, in 1963 and in 1967 he assumed responsibility for the whole field of BBC engineering as Assistant Director of Engineering. In 1968 he succeeded Sir Francis McLean as Director of Engineering.

Mr Redmond is President of the Society of Electronic and Radio Technicians and Vice-President of the Institution of Electrical Engineers.

Geoffrey Phillips read Natural Sciences at Cambridge University from 1942 to 1944, and again in 1947 after three years with the Admiralty. Following post-graduate work at Cambridge on movements in the ionosphere, he joined the BBC in 1951. He has worked most of the time in the Research Department on aerials, receivers, and transmission systems. Since 1972 he has been Head of Transmission Group of Research Department, and has become involved in work on digital systems and in studies on the use of satellites for broadcasting.

Bruce Moffat graduated in engineering science at Oxford University in 1959. He remained at Oxford to carry out research in microwave electronics, for which he received a D.Phil. degree.

He joined the BBC in 1962 and worked in the Acoustics Section of Research Department where he was chiefly concerned with the application of computer techniques to studio acoustics.

In 1966, Dr Moffat transferred to Television Section in Research Department and investigated the problem of head-clogging in video tape recorders.

After a two-year spell working in Industry, Dr Moffat returned to the BBC in 1970 to Electronics Group of Research Department and was engaged in the development of the PCM system now used for the distribution of high-quality sound signals. In 1971 he was appointed to his present position as Head of the Baseband Systems Section of Transmission Group.

Neil Spring is a physics graduate of Imperial College, London. He joined the BBC Research Department in 1964 after spending two years with B.I.C.C. Ltd and five years with EMI working on problems in room acoustics, artificial reverberation, and stereophony.

His work with the Physics Section of Studio Group in Research Department has followed similar lines to his work with EMI; he had been concerned particularly with automatic measurement of reverberation time and, more recently, the construction of models to predict the acoustical performance of studios.

William Sproson is a graduate of Christ's College, Cambridge, and a Fellow of the Institute of Physics. He joined the BBC in 1950 to work on colour television after spending three years with Dufay Chromex (manufacturers of the last of the additive photographic colour processes).

In his present position as Head of Physics Section, Research Department, he is responsible for optics, colorimetry, and acoustics. He has also been involved in several external committees, serving on the committee of the Colour Group* on two occasions; as secretary of the Optical Group (Institute of Physics and Physical Society) for three years; and at present on two National Illumination Committee panels on colorimetry and colour rendering and also the panel organised by the Scientific Instrument Research Association on assessment and specification of image quality.

* Originally the Colour Group of the Physical Society. Now the Colour Group (Great Britain) Limited.

At the time of publication, biographical notes for Dr Gilford were not available.

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Publications available from Engineering Information Department

Information Sheets on the following subjects can be obtained from Head of Engineering Information Department, Broadcasting House, London W1A 1AA, and are available free of charge, except where otherwise indicated.

General

9002 Wavebands and Frequencies Allocated to Broadcasting in the United Kingdom

Television

4006 UHF Television Reception

9003 Television Channels and Nominal Carrier Frequencies

2701 Television Interference from Distant Transmitting Stations

4101 Television Receiving Aerials

4306 Test Card F

2001 Transmitting Stations, 405-line Services (BBC-1 and BBC Wales): Channels, Polarisation, and Powers

2901 Transmitting Stations, 405-line Services (BBC-1 and BBC Wales): Map of Locations

4003 Transmitting Stations, 625-line Services: Channels, Polarisation, and Powers

4919 Main Transmitting Stations, 625-line Services: Map of Locations

2020 405-line Television: Nominal Specification of Transmitted Waveform

4202 625-line Television (Colour and Monochrome): Brief Specification of Transmitted Waveform
How to receive BBC TV – 625 lines and colour

Radio

1042 BBC Local Radio Transmitting Stations (MF2 VHF): Frequencies and Powers

1701 Medium-wave Radio Services: Interference

1603 Stereophonic Broadcasting: Brief Description

1604 Stereophonic Broadcasting: Technical Details of Pilot-tone System

1605 Stereophonic Broadcasting: Test Tone Transmissions

1034 VHF Radio Transmitting Stations: Frequencies and Powers

1919 VHF Radio Transmitting Stations: Map of Locations

Service Area Maps

Individual maps showing the service areas for many radio and television transmitters are also available.

Specification of Television Standards for 625-Line System I Transmissions

A detailed specification of the 625-line PAL colour-television signal transmitted in the United Kingdom is published jointly by the British Broadcasting Corporation and the Independent Broadcasting Authority, and can be obtained for 50p post free from Head of Engineering Information Department, Broadcasting House, London W1A 1AA.