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Contents

The cover picture shows a news reporter with
one of the new UHF radiotelephones described
in the article beginning on page 4. One of the
radio taxis can be seen on the left.

Editorial

The Control of Modulation Depth in Domestic Sound Broadcasting

There is a school of thought which holds that the only proper criterion of the correct modulation depth of a transmitter is the appearance of over-modulation no more than once per fortnight. The gain of each item in the programme chain is to be kept constant at all times, of course, and the linearity of every piece of equipment is to be as high as engineering skill can make it.

The adherents of this school tend to live fairly close to high-power transmitters and are blissfully unaware of the interference and noise problems suffered by less fortunate listeners. Those of them who become aware of noise or even suffer it themselves are inclined to regard its appearance as conclusive proof of the perversity of the broadcaster in failing to build more or higher-power transmitting stations. The fallacy in this opinion was clearly exposed in the article 'The Nature of Broadcasting' in BBC Engineering No. 106.

In the very early days of broadcasting, those responsible for the transmissions came to realise that there was an inconvenient limit to the dynamic range of programme that a practical system could deal with. Some types of programme, such as orchestral concerts, could easily exceed that limit and frequently did. The only practicable way to cope with such a problem in the early thirties was to use an operator to make intelligent adjustments of gain, anticipating particularly loud and particularly soft passages by gradual contrary changes of level. The alternative was to accept either that a substantial part of the audience would lose the *pianissimos* in the noise or that all of it would lose the *fortissimos* in a welter of distortion — or both.

In the purist mathematical sense, all variation of gain during a particular programme constitutes non-linearity and thus distortion. Such slow variations as we have just discussed, however, could not possible be heard as distortion products, though the effect could be interpreted as a different style of performance. The system worked well on the whole, though it left a small but significant number of occasions when the level of a *fortissimo* exceeded the expectations of the operator by a sufficiently large margin to result in over-modulation and, therefore, distortion.

Thus it is highly desirable to supplement the manual control of level by some automatic means which could respond sufficiently rapidly to prevent over-modulation. Rapid variation of gain, however, causes the production of spurious components which *can* be heard as distortion, and the problem, in some ways, is one of choosing the least

unpleasant form of non-linearity. The early limiters, while crude by today's standards, did nevertheless succeed in reducing the subjective degradation of the programme.

The story from that time is one of successive improvements in limiters. Some of these improvements are aimed simply at economic benefits — producing the same performance at lower cost; some have been made necessary by changes in the services provided — the pre-emphasis of FM introducing its own problems, and stereo requiring ganging of matched pairs of limiters; and some have refined the performance of limiters to reduce the subjective effect of limiter operation to barely perceptible levels. This story is told in the article by Gleave and Manson in this issue.

Some people have argued recently that one of the various systems for increasing the useful dynamic range of the tape recording process should be applied in broadcasting. These systems work for tape recording where the same limitations apply as in broadcasting — there is a boundary at one end of the range determined by distortion and at the other determined by noise. Why, then, shouldn't the same system be applied to broadcasting? The answer is 'compatibility'. In tape-recording both ends of the system can be under the control of the same person or organisation; the processing introduced prior to recording can therefore, in principle, be precisely compensated during replay. In broadcasting this does not apply, and the broadcaster must take very seriously the millions of existing receivers and avoid making changes which would significantly degrade their standard of reception.

The argument is nothing like so straightforward and clearcut as it appears, however. Manual adjustment of level during a programme and the use of a limiter both constitute compression of the dynamic range and the listener has no means of recovering the original range because he has no way of knowing the criteria on which the adjustments were made. With the commercial systems offered, at least the appropriate expanders can be bought.

The broadcaster must nevertheless be concerned about the fate of the great mass of listeners who are most unlikely to buy the expansion equipment. And even if the necessary expander were easily available and widely used it would be rash to suppose that it would be universally beneficial. The dynamic range of a symphony concert is very desirable in a concert hall but in most domestic circumstances it would be a handicap. Either the very loud passages would infuriate

others who had no wish to listen or the very soft passages would be inaudible against the background of miscellaneous sounds which pervade most of our lives. Very often, neither problem would be avoided. Listening in a car in heavy traffic would be hopeless.

In consequence the broadcaster's decision must be intimately associated with the question of what *form* of compression produces the most acceptable results for the majority of the audience. A subsidiary question — becoming important only when two approaches give very similar results in the first consideration — is that of which system gives the best results for the suitably equipped receiver. And closely related to this is the matter of how much the listener would need to spend in order to obtain these results.

All these questions are very difficult and the answers depend on more than the comparatively simple engineering factors. They are influenced, for example, by such things as the relative importance and the relative frequency of different categories of programme material. In the BBC we take the view that the combination of manual control (by an operator having the skill and training which enable him to make artistic judgements) and high-performance modern limiters provides better overall programme quality for more listeners than would any commercial compression system known to us.

Radio News Reporting in London

UHF Radiotelephones

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Radio Taxis

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Summary: The facilities available to BBC Radio News reporters in London have recently been improved by the replacement of the old radiotelephone systems and radio cars. This article outlines the main features of the new UHF radiotelephones and radio taxis which offer the reporter, among other things, greater mobility.

1   Introduction

Radio news reporters are often required to make their reports in person in the news or current affairs programme concerned. It is usually desirable to make this contribution immediately from the actual location of the event reported or from somewhere nearby. News is a perishable commodity and immediacy is of prime importance.

A VHF radiotelephone system had linked a number of reporters' cars with Broadcasting House for many years, permitting editorial control and direction of the reporters' movements. Direct programme contributions could only be made from a radio car, however.

The old VHF radiotelephone system has recently been replaced by a UHF system which, besides having a variety of operational advantages, has a sufficiently good performance to permit direct contributions to broadcasts. The radio cars have also recently been replaced by two taxis which have a number of particular advantages for this application.

2   UHF radiotelephone system

The UHF transceiver is a modified commercial unit which can be carried by the reporter or mounted in an adapter in his car. (The modification reduces distortion and increases the audio bandwidth to 6 kHz.) Five base stations are used to provide good coverage of most of the Greater London area. Frequency modulation is employed with a deviation of ±5 kHz.

Provided transmission conditions are good, the speech quality is up to broadcast standard. When transmission conditions are not satisfactory, it is often possible to use one of the radio taxis as an intermediate relay point. The base stations radiate an e.r.p. of 5 W and the transceivers operate at half a watt.

2.1   Transceiver Operation

The transceiver is capable of operating on three UHF channels which can be selected by a switch: channel 1 is for communication with the newsroom, channel 2 for broadcast use, and channel 3 for relays via a radio taxi. In each channel, different frequencies are used for transmission and reception but operation is nevertheless simplex because the transceiver is not designed to transmit and receive at the same time. The base stations can and do transmit and receive simultaneously, however, and this allows cue programme to be fed back to the reporter, enabling him to hear the broadcast he is joining, right up to, and including, his cue to transmit.

The unit operates from 15 V nickel-cadmium rechargeable batteries and each reporter has two batteries and a mains-driven charger. In addition, the adapter in the reporter's car allows for a slow charge to be taken from the vehicle battery: this is not adequate to recharge a fully discharged battery but will permit use of the transceiver without loading its own battery.

The mains-driven charger gives a relatively fast charge for five hours, which is sufficient for a fully discharged battery, and then automatically switches to a trickle charge. Charging can be carried out by plugging the complete transceiver into the charger or by fitting the battery into it by means of an adapter. The set can be used, for reception or transmission, while plugged into the charger.

A fully-charged battery will operate the transmitter continuously for about one hour or allow about 48 hours
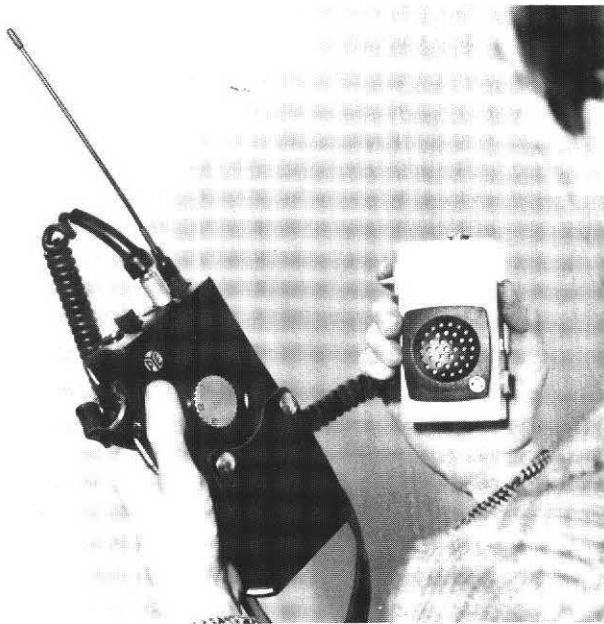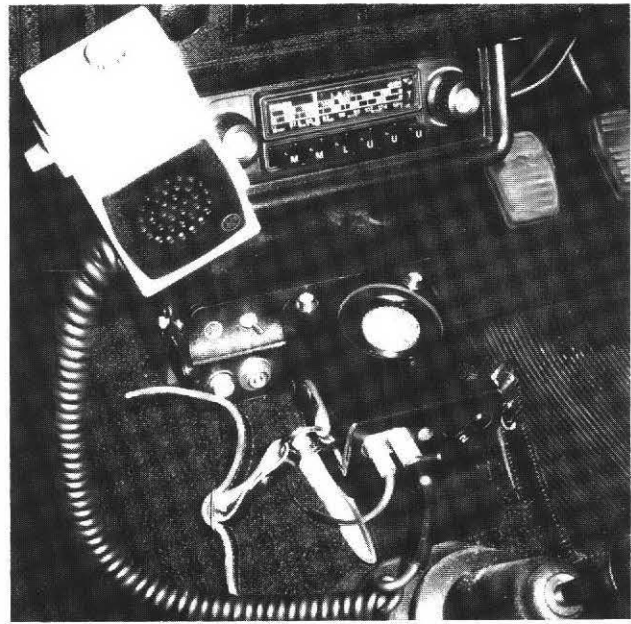
Fig. 1   The new UHF radiotelephone.

Fig. 2   An adapter in the reporter's car provides for use of the vehicle battery and a better aerial.

'standby' operation, i.e. switched on and awaiting calls from base.

Using the transceiver in the adapter in the reporter's car also brings the benefit of a better aerial mounted on the car. This permits satisfactory operation at greater distances from the base stations, giving almost complete coverage of the Greater London area.

The transceiver is supplied with a single transducer which serves as both microphone and loudspeaker but this device, while adequate for communications purposes, is quite unsuitable for broadcasting. For the news reporting application, therefore, a moving-coil microphone and a miniature loudspeaker have been mounted in a small separate unit linked to the transceiver by a coiled flexible cable. Consequently, in situations where the transceiver position is critical (because of transmission conditions) or where the set is mounted in the adapter, the reporter can still retain a limited degree of mobility.

The auxiliary unit also incorporates an earpiece connection (which automatically cuts out the loudspeaker) and a tape recorder input jack as well as a 'remote' transmit/receive switch. This switch has a spring return to the receive position for communications use, but can be set to a continuous transmit position for broadcasting.
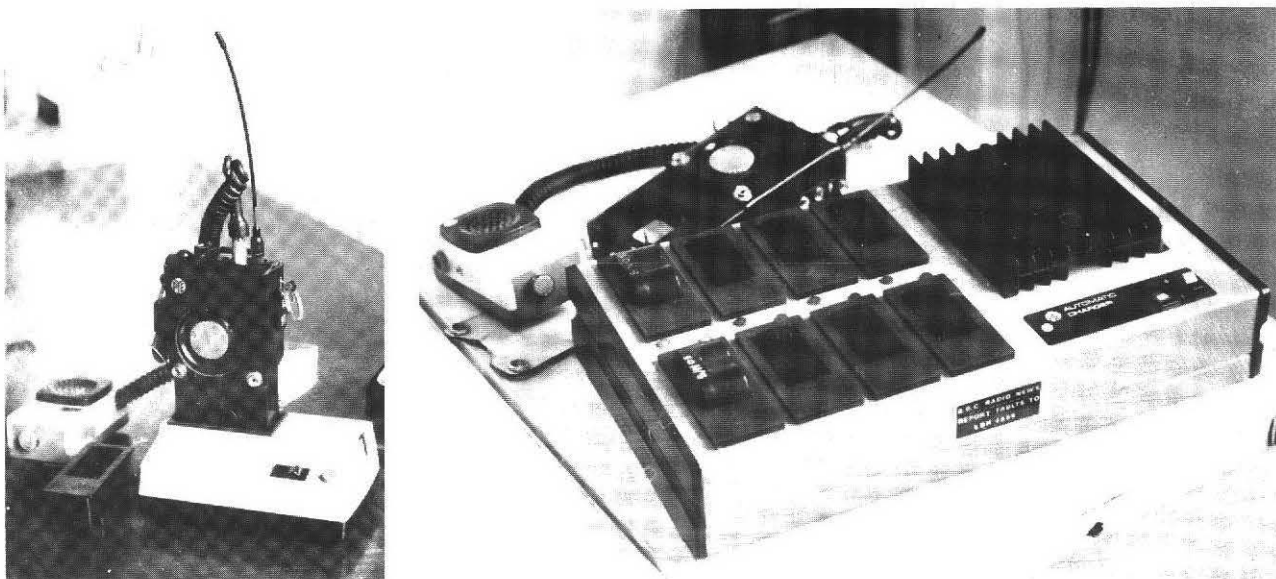

Fig. 3   Mains operated chargers: left, reporter's unit with transceiver plugged in; right, base unit to charge eight batteries.
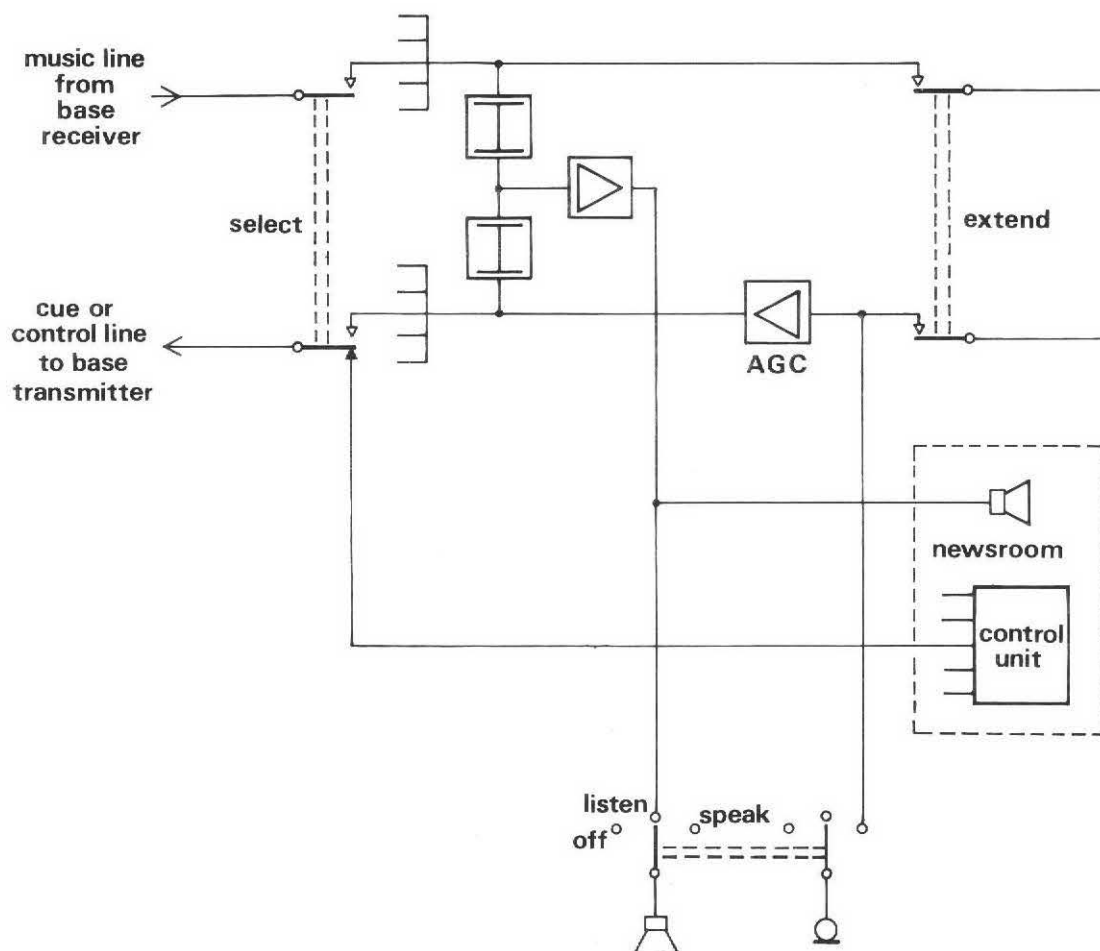
Fig. 4 An outline of the principal connections in Broadcasting House for radio news contributions from the UHF radiotelephones.

2.2 The System at Broadcasting House

A greatly simplified diagram of the switching arrangements at Broadcasting House is shown in figure 4. On the left are the connections to a pair of lines to one of the five base stations: the connections to the other four take the same

Fig. 5 Control unit in Newsroom

form. One of these lines is the incoming 'music' line; i.e. it is equalised to broadcast standards and is used exclusively for broadcasting. The other is the communications line which carries the conversations between the reporter and staff in the Newsroom while the broadcast is being set up in the Control Room. When the broadcast takes place the communications line is used as a cue line, being fed with a mixture of programme and talkback.

The 'Select' relay for each site connects the control or communications line to a control unit in the Newsroom at all times except when that particular site has been selected. This enables reporters to be called from the Newsroom via any chosen base station except when that base station is in use for a broadcast. Such traffic is, of course, carried on channel 1, but when the necessary preliminaries have been completed the reporter is asked to switch to channel 2 and the appropriate base station is also switched to channel 2 by means of phantom voltages (applied by the control unit) on the communications line.

The Outside Broadcasts (OB) operator in the Control Room is informed of the situation and he now takes over, selecting the base station on which contact has been made with the reporter and checking the quality of the incoming

contribution. If the quality is not satisfactory he will ask the reporter to move his position slightly or, if this does not produce the desired effect, to move to a better site if possible. Occasionally it might be beneficial to select a different base station. In very difficult cases a relay via one of the radio taxis might be arranged.

Once the OB operator is satisfied with the incoming quality, he selects the destination required by the Newsroom and operates the 'Extend' relay. The destination might be a recording room or a studio where the incoming signal is treated in the same way as any other outside source.

The 'Select' and 'Extend' relays are interlocked so that it is not possible to operate or release any 'Select' relay while the 'Extend' relay is operated: this prevents accidental interruption of a broadcast. The programme is now automatically routed to the communications line to the selected base station transmitter and thence to the reporter's transceiver. At the same time, the OB operator's ability to speak to the reporter is unaffected (provided the reporter is not transmitting, of course).

3 Radio Taxis

For more comprehensive work in the radio reporting field the use of a versatile and well equipped radio car is required. The BBC has used such vehicles for about ten years and during this time several different standard vehicles have been chosen as the basis for the adaptation. On the last occasion when replacement was required the choice of vehicle was rather unorthodox but was nevertheless rooted in many compelling advantages.

3.1 Choice of vehicle

The vehicle chosen was the type of hire car used by most London taxi drivers, and it compares favourably with nearly all other possible vehicles on most of the factors considered important, namely: reliability, manoeuvrability, long interval between engine overhauls, durability (particularly in the face of much short-distance running), adaptability to broadcasting requirements, comfort for interviewee and interviewer, cost, and ability to be driven and operated by one engineer, who is also available to assist the reporter.

The London taxi is outstanding among vehicles of its size and load-carrying capacity in the smallness of its turning circle (7·62 m : 25′) and in the generous headroom it provides — seat to roof measures 0.965 m (3′2″) and floor to head-cloth 1·27 m (4′2″). The estimated interval between engine overhauls is 240,000 km (150,000 miles).

3.2 Modifications and Equipment

The radio cars are required to act as small interview studios and as bases for location reports or interviews. The resulting programme material is to be recorded at the site or inserted live into a broadcast by means of a radio link. Editing of recorded items before transmission to base can also be desirable.

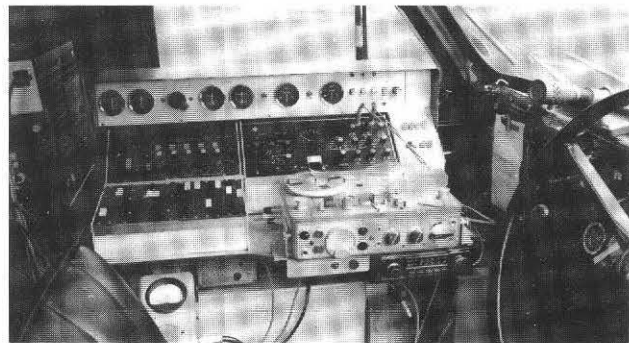Fig. 6 A radio taxi with the mast partially extended.

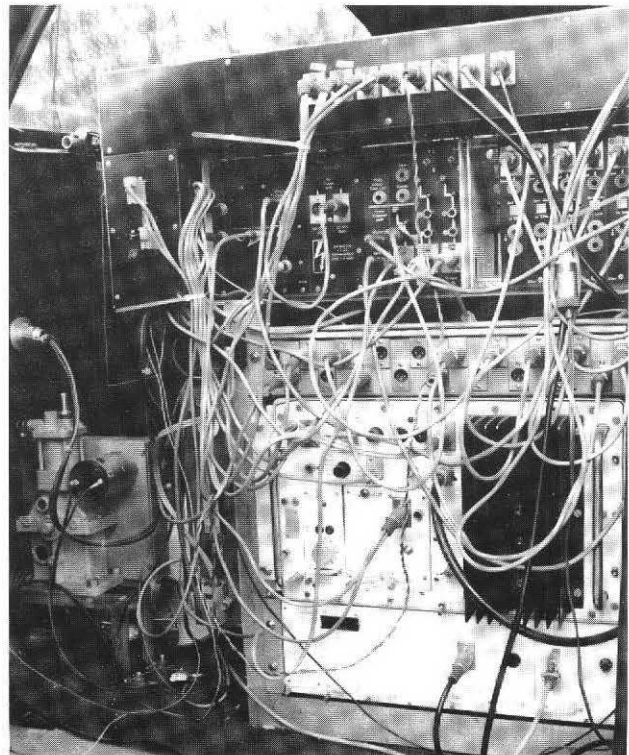Fig. 7 The taxi's equipment as seen from the driving position.

Fig. 8 Access to the rear of the equipment is via the nearside front door of the taxi.

The rear compartment of the vehicle, used for passengers in the normal application as a taxi, has been equipped as a studio, with permanently installed microphones as well as loudspeakers and headphones for cue and talkback purposes. Tinted glass has been fitted to allow privacy to interviewee and interviewer.

The VHF programme link uses an aerial mounted on a 6m (20') pneumatically operated telescopic mast. Frequency modulation is used and the transmitter output power is 5 W for standby working and 40 W for broadcast use. With such a power and such an aerial height coverage of the great majority of the Greater London area is achieved with only three base stations. Editorial control is exercised from Broadcasting House using the UHF radiotelephone system described in section 2. Communication with all the radio reporters, therefore, is by the same system, whether the reporter is using a radio taxi or not.

The taxi can use the UHF system in either of two principal ways:

1) for calls from Newsroom in the same way as it is used by other reporters

2) as a programme link relay using channel 3. In this way the taxi acts as a 'mobile base' for the reporter.

The majority of the technical equipment is fitted on the driver's left, in the space used for luggage in a normal taxi. The controls face the driving position so that the engineer, who also drives the vehicle, can operate it without leaving his seat. Access to the back of the equipment is therefore simply obtained by opening the nearside front door.

Included in the equipment are a reel-to-reel tape recorder and a cassette machine as well as a mixer. Tape editing can also be done on the spot, and a car receiver is provided for checking the broadcast programme.

The mixer permits the vehicle to be used for two purposes simultaneously. For example, an interview being conducted in the studio compartment can be recorded at the same time as another reporter, using a remote microphone, is making a live broadcast via the vehicle's radio link.

A sunshine roof has been fitted to the studio compartment to serve two purposes: it gives access to the aerial when the mast is retracted and provides a reasonable vantage point from which a reporter can observe and report events. This can be done even with the vehicle moving, though not if the mast is raised. As a safety measure the mast controls are interlocked with the handbrake so that as soon as the handbrake is released the mast is lowered. As a further precaution an alarm is sounded if the ignition is switched on before retraction is complete.

The boot houses the four batteries used for the technical equipment, and drums of cable for remote microphone operation and battery charging. Battery charging is normally carried out at base via a special non-standard mains socket (rendering it unsuitable for other purposes and hence always available) mounted on the garage wall adjacent to the taxis' normal parking position. If a taxi spends a long period on site, however, a local mains supply may be used, feeding the cable through a hole cut in the floor of the boot to avoid the need for keeping the boot open while the batteries are being charged. Failing this, the batteries may be charged in pairs from a high-output alternator on the vehicle engine. Switching enables each pair of batteries to be charged in turn at fast idle speed.

Some of the equipment requires 12 V supplies and some 24 V. The four batteries are therefore arranged in two pairs, each of which can be connected in series or in parallel, and battery condition can be checked from meters in the engineer/operator's compartment where the voltage and current drain of each battery can be separately observed. Thus the engineer can decide which battery to use for which purpose and which, if any, should be placed on charge.

3.3 Operation

The radio taxis' contributions can be handled at Broadcasting House in a broadly similar manner to the direct contributions from the UHF hand-held transceivers described in section 2.2. Operation is somewhat easier than with the portable transceiver because the taxi's radio links operate in the duplex mode. Remote indication of the operation of the transmitters and receivers at the base stations is given at Broadcasting House by phantom voltages on the music and control lines.

A microphone mounted in the driving compartment can be connected to the vehicle's transmitter, thus enabling the engineer driving the taxi to keep in touch with the base from the moment he leaves the garage should it be necessary.

The staff duty rotas are arranged to ensure that on Mondays to Fridays at least one taxi can be operated at any time between 0600 and midnight and both can be deployed when required during the busiest hours from 1000 to 2100. Availability at weekends is variable, depending on the other commitments of OB staff at the time.

4 Conclusions

The UHF portable transceivers are a valuable tool for the radio news reporter and offer an important addition to the facilities of the two radio taxis. The taxis themselves are an unorthodox choice of vehicle for radio cars but are proving their worth in a variety of ways.

The Development of Sound-Programme Limiters in the BBC

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Summary: Over-modulation of transmitters produces distortion; under-modulation reduces the service areas. Manual control of audio signal level has been supplemented since the late thirties by automatic limiters to prevent over-modulation. This article describes the development in the BBC of devices of this type from the time of their introduction up to the present.

1 Introduction

Overloading of electronic equipment by audio signals of excessive level produces non-linearity of the transfer characteristic and consequent impairment of the quality of the reproduced programme. When the equipment concerned is a transmitter the overloading can also lead to the radiation of additional components outside the specified channel bandwidth (possibly causing interference to other services) or may even cause automatic switching off ('tripping') of the transmitter to prevent damage. For a variety of reasons, therefore, *significant overloading of audio-signal equipment is to be avoided.*

In the BBC, sound signals are normally controlled manually at source. For programme material of wide dynamic range (such as orchestral music) it is not feasible to preserve the full range in the transmitted signal *without* spoiling sustained quiet passages with obtrusive noise. It is *therefore usual to make manual adjustments of the level* during the course of such programmes, very loud passages being anticipated by gradual reductions of gain and very quiet passages preceded by gradual increases. Though the signals are nominally controlled to a specified maximum level the human operator cannot react *with sufficient speed* to avoid occasional overloading on unexpected programme *peaks.*

Two obvious ways of avoiding accidental overloading are either to use signal levels which are sufficiently low to involve negligible risk of reaching the overload point, or, looking at the subject from the other direction, to use equipment which *is capable of handling signal levels* substantially above the nominal maximum. The former solution inevitably means a degradation of the overall signal-to-noise ratio. The latter solution is generally adopted for low-level signal equipment but is not practicable for an AM transmitter, for example; here the power handling capability can be increased only by raising the carrier power — *a very costly solution even if it were allowable under the* relevant planning agreements.

It is thus evident that there is a need for some form of automatic protection, particularly to prevent over-modulation of transmitters, and quick-acting limiters have been developed which recognise the presence of peaks above the acceptable limit and reduce the gain by an appropriate amount. It should be noted that a limiter in this context is very different from an instantaneous peak clipper; such a device simply removes the peaks of an input waveform which is too large, producing waveform distortion which, as a general rule, is likely to be objectionably audible. Limiters have been used at transmitter sites *since the early nineteen-*forties, and in recent years they have also been used at the

sending terminals of pulse code modulation (PCM) links to permit realisation of the full potential of the digital systems.

In some circumstances reduction of the dynamic range by automatic means is desirable for technical or aesthetic reasons. This can be done by limiters in two ways. Either the input to the limiter is raised above the nominal level so that the limiting action is deliberately invoked for a significant proportion of the time, or the limiter can be modified to give an input/output characteristic with a finite slope less than unity rather than the normal limiter performance with unity slope up to the limiting level and nominally zero slope above it (section 3.2).

2   Early Overload-Protection Practice

In the BBC, signals are, in general, monitored and regulated in level at the studio. In the early days of radio the control of level was based on signal measurement using various metering arrangements, including, for example, a 'slide-back' voltmeter set so that programme at the correct level gave an indication on signal crests only occasionally. Later, in the late nineteen-thirties, the BBC peak-programme meter (PPM)[1] — a pseudo-peak reading instrument — was introduced and provided a better indication of the relevant peak value of the programme signal.

Throughout the early period of broadcasting, continuous monitoring was also carried out at the transmitter. The transmitter engineer was not normally expected to exercise any control, but he could make adjustments when necessary to avoid, say, repeated overloading or a prolonged period of excessively low modulation. To reduce the risk of false 'corrections' he was warned in advance of any passage of programme to be deliberately kept at low level for artistic reasons.

Experienced and skilled operators would be able to predict changes in programme level to a considerable extent but could not be expected to avoid occasional over-modulation and consequent audible distortion. It was thus inevitable, even with the most skilled and conscientious human control, that listeners suffered occasional distortion due to overloading and some also suffered degradation of the service due to low transmitter modulation. The development of an automatic level control device to replace the human operator at the transmitter was thus obviously desirable.

3   Basic Principles of Quick-Acting Limiters

3.1   General

As a means of avoiding over-modulation, manual control falls short of the ideal mainly because of the long reaction time of the human operator. With advancing technology it became possible to develop automatic protection devices which could react to signals of excessive level much more rapidly, and which would allow the gain to rise gradually when the level of the incoming signal had fallen, thus minimising audible interference with the programme.

Figure 1a is a block diagram of one form of simple quick-acting limiter. The circuit consists of two paths; a programme path and a control path or control 'chain'. The programme path is essentially a variable-gain element whose

gain depends on the bias applied to a control input. In operation this bias is derived from the limiter output signal by means of the control chain, acting in the following manner. The output signal is first rectified — full-wave, to ensure that account is taken of the greater peak, of either polarity, of an asymmetrical waveform — then applied to a limiting threshold circuit. This circuit yields no output as long as the signal is below the desired limiting level, but, for higher signals, the crests of the rectified waveform are passed, generally through an amplifier, to a peak-detecting and smoothing circuit. The voltage developed across the smoothing circuit is applied to the control input of the variable-gain element in such a sense that gain is reduced until the control loop is in equilibrium. Peak detection is adopted at the smoothing stage to ensure that the limiter control is related to the crests of the signal waveform, since it is, of course, overloading and waveform distortion on these crests that leads to audible impairment.

The limiter arrangement indicated in figure 1a, known as an output-, or servo-, controlled type, was used in early BBC limiters. The output-controlled configuration is a negative feedback device: it is therefore impossible for it to produce a literally constant output level for a range of input levels though this condition can be closely approached in practice. On the other hand, the use of negative feedback carries with it the advantage that the exact shape of the control characteristics of the variable-gain device is unimportant. The alternative input-controlled configuration can, in principle, compensate completely for input level variations, but demands a precisely-specified control characteristic.

3.2   Static characteristics

Figure 2a shows the gain of a typical output-controlled limiter under steady-state conditions, plotted as a function of input level. Gain is constant for input signals up to the limiting threshold and then decreases by approximately one decibel for each decibel of further increase of input level. The same information is presented in a different form in figure 2b which shows output level plotted as a function of input level; this illustrates more clearly that the limiter output level stays substantially constant as the input level is raised above the limiting threshold. In practice an increase of input level of, say, 20dB beyond the limiting threshold might cause the output level to rise by some $\frac{1}{2}$dB to 1dB.

It must be emphasised that the curve of figure 2b does not imply steady-state waveform distortion (as it would if it related instantaneous input and output voltages) but simply indicates a particular law relating gain and steady-state input level.

3.3   Dynamic characteristics

The dynamic characteristics of a limiter depend on the peak-detection and smoothing circuit, the gain/bias law of the variable-gain stage, and, in the case of the gain-reducing (or 'attack') period, on the gain in the control chain.

Figure 1b shows a typical peak-detection and smoothing circuit for a simple limiter. During the attack period, capacitor C is charged through resistor R_1 (R_2 is normally much larger than R_1) until the bias applied to the variable-
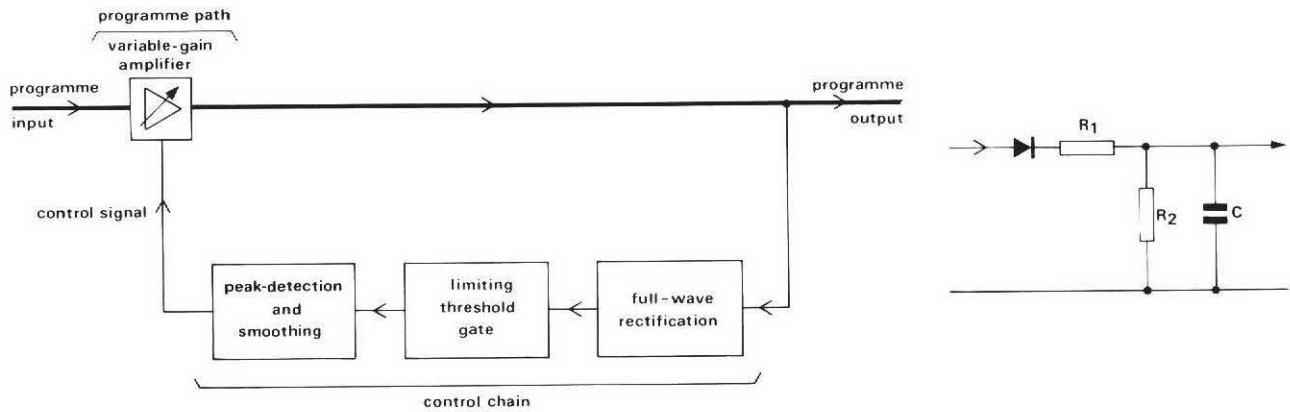
Fig. 1 Simple output-controlled limiter.   a. Block schematic.   b. Peak-detection and smoothing circuit.
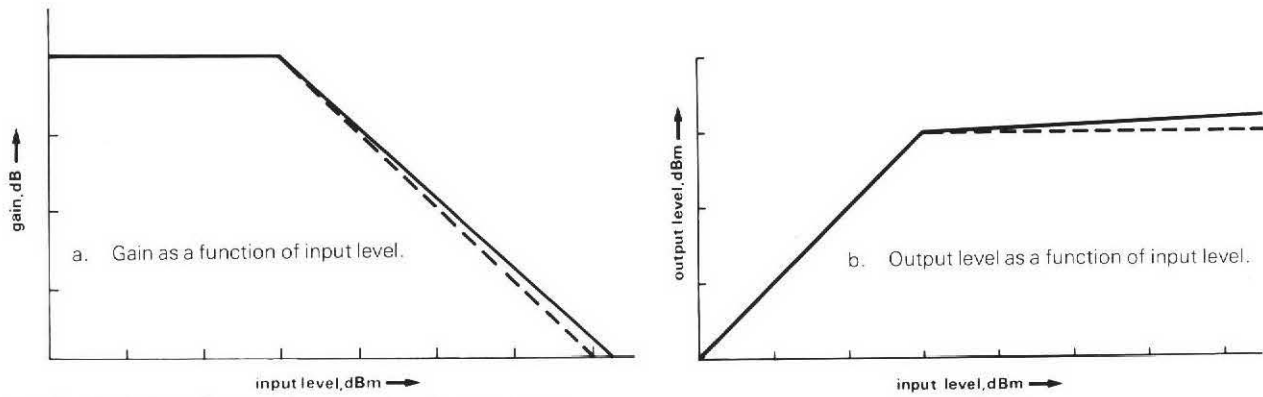

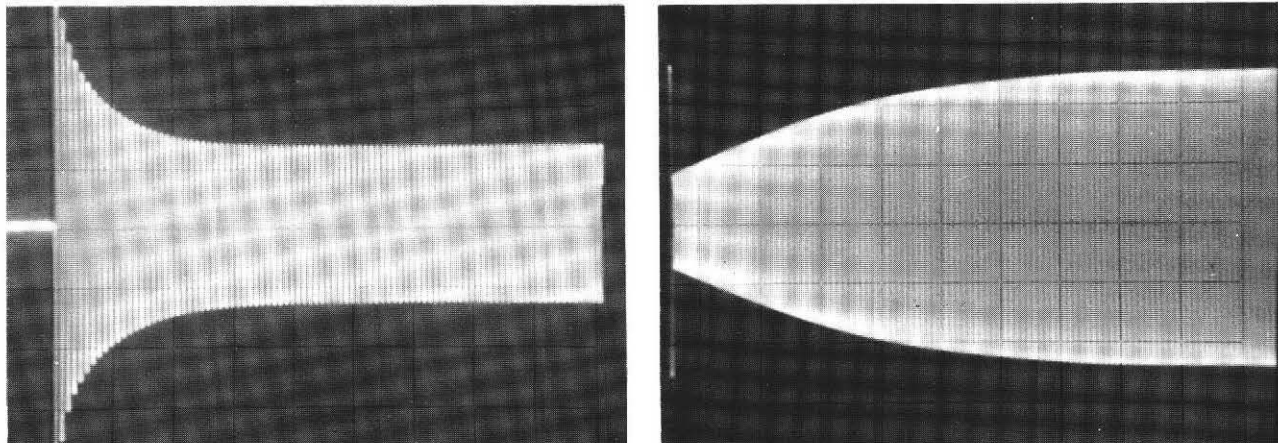**Fig. 2** Typical steady-state characteristics of simple limiter.

Fig. 3 Typical dynamic characteristics of simple limiter.

a. Envelope of output signal during 'attack' period: time scale, 2 ms/major division.

b. Envelope of output signal during recovery period: time scale, 200 ms/major division.

gain stage is sufficient to restore the circuit to equilibrium, the process commonly taking a few milliseconds. Until equilibrium is reached the gain in the limiter is higher than its ultimate steady-state value for the signal applied, so there is a transient overshoot at the limiter output. Figure 3a shows the output signal from a typical output-controlled limiter when a steady signal, well above the input limiting level, is suddenly applied at its input. Equipment further along the programme chain can be overloaded during the transient overshoot period, but before the introduction of

wide-band loudspeakers, impairment due to such overloading was unlikely to be obtrusive.

When the output level falls below the limiting value, capacitor C discharges through resistor R_2, the bias on the variable-gain stage falls to zero, and the limiter gain returns to its normal maximum value. This 'recovery' period may be arranged to take anything from a hundred milliseconds or so, to several seconds, depending on the requirements. Figure 3b shows a typical envelope of the signal at the output of a limiter when a large input signal, above the limiting
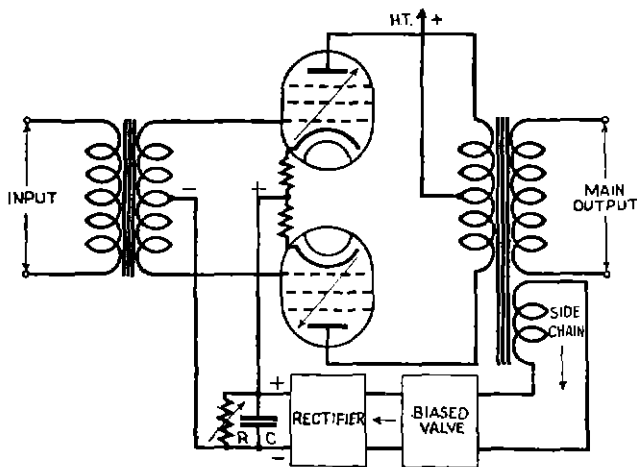
Fig. 4 Simplified circuit diagram of early limiter (from BBC Engineering Division Training Manual, 1942).

threshold, is suddenly reduced to a steady level below that threshold.

Advances in technology have more recently made possible complex attack and recovery characteristics. Delay networks have been used to eliminate the momentary overshoot during the attack period, and the recovery characteristics have been made level- and time-dependent so that the limiting action is still less obtrusive.

4 Valve Limiters

4.1 The first BBC valve limiters

By the thirties the 'variable-mu' valve had been developed in which the grid structure is so arranged that the i_a/v_g characteristic is gently curved, so that valve gain is related to grid bias. The variable-mu valve thus lent itself readily to application as a variable-gain device, the heart of any automatic level control equipment, and was used by the communications engineers of the day in the development of audio limiters. The curved gain/bias characteristic implies a measure of waveform distortion, but the performance was adequate provided the signal level applied was small.

A BBC patent application [2] by Mayo in 1938 describes the principles of a sound-programme limiter using variable-mu valves. The design was developed and went into service in about 1940 as the LIM 2*. Figure 4 shows a schematic diagram of the limiter circuit reproduced from the BBC Engineering Division Training Manual, 1942. This diagram is obviously much simplified and the arrangement used in the LIM 2 was essentially that indicated in figure 1a. It is interesting to note that two variable-mu valves were employed, connected in push-pull. The control bias was applied to the valves in parallel and the push-pull arrangement allowed the control bias voltages, amplified by the two valves, to be cancelled in the anode circuits and thus, ideally,

*The code LIM 1 was allocated to an earlier development version which was not used in service.

excluded from the limiter output. In practice two valves rarely had sufficiently similar characteristics, so a pre-set control was included to allow an adjustment of the balance. It was set for minimum 'plop' at the output when a voltage step was introduced in the control chain.

The attack characteristic of the LIM 2 was of the general form shown in figure 3a, with an effective attack time of a few milliseconds. The recovery characteristic was of the general form shown in figure 3b, the time constant being set by a switch to give any one of five values of effective recovery time in the range from about 100ms to many seconds. For reasons more fully discussed in Section 5.2.4 any single value of recovery time can be no more than a compromise; the performance was improved by a subsequent modification introducing a form of double-time-constant circuit. An input-level control was provided to allow a set amount of programme compression to be introduced if required. The control range was in excess of 20dB and the total harmonic distortion, measured at 1kHz, was less than 0.25% over this range. Valves in the 'British 4-pin and 7-pin' series were used in the LIM 2 which occupied about 23cm (9") on a 56cm (22") bay. The normal output limiting level was −15dBm.

By the early fifties some of the parts used in the LIM 2 were becoming obsolete and the design was therefore revised using currently-available components.

4.2 Introduction of miniature valves

No fundamental changes in design concept were introduced in the replacement, the LIM 5, though there were changes of detail. The use of small components including 'miniature' valves resulted in a reduction in overall size; the bay space occupied by a given number of limiters was approximately halved.

An output amplifier was added to raise the nominal limiting level to the operationally more convenient value of +8dBm, but the limiting characteristics were substantially unchanged.

4.3 A limiter for FM radio

The system specification for the VHF/FM radio service introduced in this country in the mid-fifties included a 50μs high frequency pre-emphasis of the sound signals applied to the transmitter. Thus a conventional limiter, like the LIM 5, operating on the studio output signals, could not protect the transmitter from over-deviation by high-frequency components raised in level by the subsequent high-frequency emphasis. FM transmitters could be protected from over-deviation by a LIM 5 limiter provided it operated on pre-emphasised sound signals, but the output signal would then have to be de-emphasised for monitoring purposes. A modified limiter, the LIM 6, was therefore developed to fit the operational requirements of the time more closely. Here provision was made to include the appropriate pre-emphasis characteristic in the control chain so that the limiting threshold fell with increasing frequency. Thus the output signal was such that when pre-emphasis was applied at the transmitter, the enhanced high-frequency components were not so large as to over-deviate the carrier.

LIM 6 limiters were used for the experimental stereo-phonic FM transmissions in the early sixties, using the pilot-tone system, in which the peak transmitter deviation is determined by the greater of the left- and right-hand audio signals at any time. Transmitter overload protection can thus be conveniently provided by separate limiters in the two channels. For reasons more fully discussed in section 5.4, limiters used in this way must be interconnected and must have carefully matched characteristics. In the case of the LIM 6 arrangement, adequate matching was achieved by careful selection of the pairs of variable-mu valves used. This was a tedious procedure, fortunately made unnecessary by developments in semiconductor technology exploited in the later transistor limiters used for stereophony.

5 Transistor Limiters

5.1 First generation

The first production limiter designed in the BBC to make use of semiconductors went into service in 1964. Known as the AM6/3 it was engineered to meet the same requirements as the LIM 5 and the LIM 6, but with the increased reliability and other advantages that the adoption of transistor circuits offered.

The limiter was feedback controlled, the output being full-wave rectified and threshold-gated to the control terminals of the variable-gain stage. In the AM6/3 the variable-gain element takes the form of a two-stage transistor amplifier shown in schematic form in figure 5.

Negative feedback is applied from the output of this amplifier to the emitter of the input stage via the resistors *a* and *b*. Positive feedback which partially offsets the negative feedback is applied to the base of the input stage via the network *c* and *d*. When the limiter is active the control current reduces the dynamic impedance of the diodes in *d* and thereby reduces the positive feedback, allowing the negative feedback to reduce the gain. This circuit can only function when fed from a non-zero source impedance because a source impedance of zero would prevent the application of any positive feedback, regardless of the dynamic impedance of the diodes. To put it another way, the source impedance from which the amplifier is fed can be regarded as part of *d*.

The operation of the circuit can also be considered by regarding *a*, *b*, *c*, and *d* as a bridge. The output of the amplifier is connected between the junction of *a* and *c* and the junction of *b* and *d*, while the input to the amplifer is taken from the opposite corners of the bridge. Provided the amplifer has a high intrinsic gain, the bridge will be kept approximately in balance and the signal voltage across the diodes will be almost constant when limiting is taking place. The arm *d* is itself arranged as a bridge so that control voltages are well isolated from the programme chain.

The sensitivity of the control loop is such that under steady-state conditions the output is maintained constant to within 0.2 dB over the 20 dB limiting range of the device.

A 50 μs pre-emphasis time-constant can be introduced in the control chain when used for FM transmissions.

The nominal attack time is 1.8 ms and there is a range of fixed recovery time-constants selected by a front panel
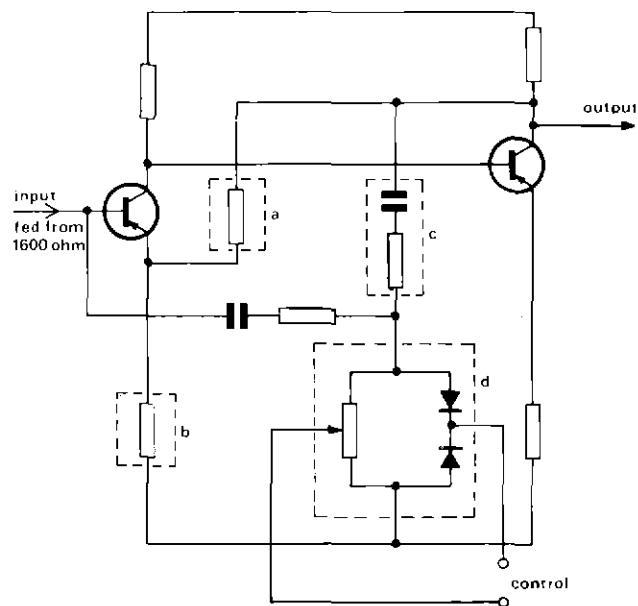

Fig. 5 AM6/3 variable-gain amplifier.

control. An input gain control enables an increasing amount of dynamic range compression to be introduced with a standard input level.

A further development of the device, the AM6/3A, enables the flat-topped limiting characteristic to be changed to a sloping compression law over part of the control range.

5.2 The dynamic characteristics of limiters for sound-programme circuits

5.2.1 General

The dynamic characteristics of the limiters so far described grew out of the conditions obtaining at MF transmitters at the time. In 1964 the subject was re-examined with a view to specifying the dynamic characteristics of the next generation of limiters. This work led to the introduction of the non-overshoot type of limiter which has been described fully elsewhere[3]. For the sake of completeness, however, the main points will be repeated here.

5.2.2 The effects of limiter overshoot

It has already been mentioned that, in the absence of wide-band loudspeakers, overloading on transient limiter overshoot was unlikely to cause significant impairment. However, with the loudspeakers available by the mid-sixties it was noticed that audible distortion could result from the clipping of a momentary overshoot if there was insufficient headroom between the steady-state output limiting level of a limiter and the overload point of following equipment. The headroom necessary to avoid audible impairment depended on the nature of the programme material and on a number of other factors, but values of 3dB or more were typical. In general the headroom required decreased as the limiter attack time was reduced but a condition was reached, as the attack time was still further reduced, where audible distortion was detectable on critical programme material, irrespective of the amount of headroom provided. It was
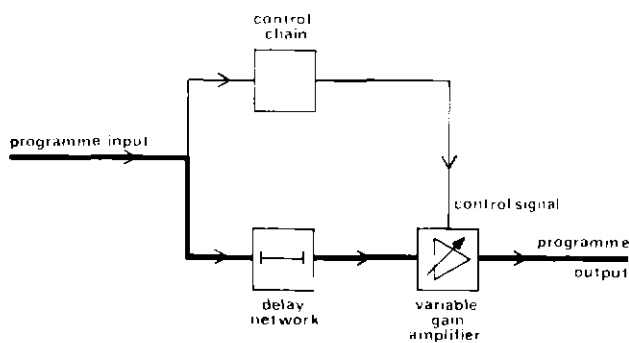
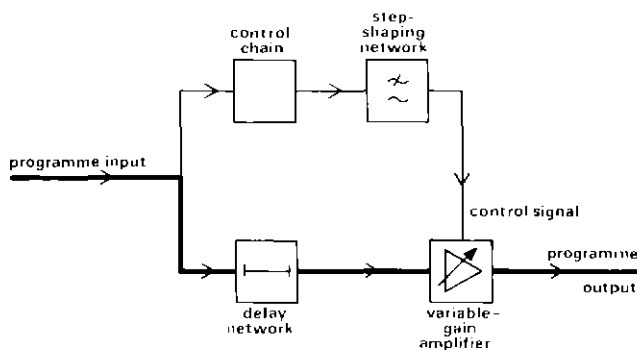**Fig. 6** Input-controlled limiter with delay network.

Fig. 7 Input-controlled limiter with step-shaping and delay networks.

concluded that this distortion — which will be referred to as modulation distortion — was an unavoidable consequence of over-rapid change of gain. Further investigation indicated that, to make impairment from this source inaudible, the gain reduction period had to extend over a few hundred microseconds — long enough for there to be a risk of audible impairment due to overloading of following equipment unless adequate headroom was provided.

5.2.3 Elimination of overshoot

Overshoot in a limiter can, in principle, be reduced to any desired extent by introducing an appropriate delay network in the programme path ahead of the variable-gain element so that, by the time the programme signal reaches this point in the chain, the gain reduction effected by the control circuits is substantially complete. It is clearly essential with this arrangement that the limiter be input controlled, that is that the gain control system is operated by the input signal to the limiter and not by the signal at the output. The resulting circuit is shown in a simplified block schematic form in figure 6.

To eliminate overshoot with all types of signal the peak detection and smoothing system which provides the control signal for the variable-gain element must have a sufficiently short charging time to register fully the crest value of the shortest peak likely to occur — ideally, the shortest peak which the bandwidth of the system will allow to pass. However, in practice, the very rapid change of gain which this implies can itself cause audible distortion in the manner

described in the previous section.

By a slight refinement to the system of figure 6 it was found possible to avoid both overshoot and modulation distortion. In figure 7, which shows the new arrangement[4], the control system is made sufficiently rapid in action to register the crest value of the shortest incoming pulse. The rate of rise of the control signal reaching the variable-gain element is then restricted by a low-pass filter, or 'step-shaping network', so that the resulting rate of change of gain is not so rapid as to cause audible modulation distortion. In practice the attack period adopted was about 300 μs. To avoid overshoot, the arrival of the programme signal at the variable-gain element has to be delayed, as before, for a time sufficient for the control signal to reach its full value, i.e. by about 300 μs. The new arrangement thus avoids both momentary overshoot and over-rapid gain variation, together with the impairments which these effects may cause.

The circuit shown in figure 7 requires a variable-gain element having a particular gain/bias law, arranged so that the desired static characteristics, illustrated in figure 2, are obtained. It may not always be simple to achieve this required law, but the same performance can be produced using the circuit shown in figure 8, which requires two variable-gain stages that have the same gain/bias law but does not demand a particular law. This duplicate variable-gain stage arrangement was used in the experimental non-overshoot limiters and was adopted for the first production version, the AM6/7, described later.

5.2.4 The effects of limiter gain-recovery time

The operation of a limiter may impair the programme quality by the introduction of effects which, while not classifiable as distortion in the ordinary sense of the word, can nevertheless be aesthetically objectionable. These impairments, which increase in severity with the degree of gain reduction, depend largely on the rate of gain recovery.

If the gain-recovery period extends over several seconds it will be observed that occasional exceptionally high signal levels, momentary in duration and in some cases making little direct impact on the ear, produce a sudden and prolonged gain reduction. At the other extreme, if gain recovery is substantially complete within a syllabic period, say less than 0.2 second, other, but equally disturbing, effects may be apparent, particularly if the limiter is deliberately operated above the normal level to reduce the dynamic range of the programme. In speech, breath noises and some initial consonants will be exaggerated by the rise in gain between sentences or even between words. In music, sustained choral or orchestral passages may suffer a fluttering due to the variations of gain produced by the irregular crests of the signal; during sustained piano notes the natural fall in level is opposed — the net result may even include a temporary *rise* in level — and a similar effect occurs during the decay of sound in reverberant surroundings. The rise in gain at every pause in the programme accentuates background noise, such as studio rumble, tape hiss, or 'print-through'. Operation in this manner during a sporting commentary will mean that the level of any crowd noises in the background will follow the

fluctuations in gain, an effect commonly known as gain pumping.

Syllabic compression may be introduced at the studio in the production of some 'pop' music, for artistic reasons, or at AM transmitters to increase the mean modulation depth and thus improve reception under adverse conditions. In these examples the effects of rapid gain recovery may be regarded either as desirable in themselves, or as the price paid to achieve a desirable end. The discussion which follows is, however, concerned with the more difficult case of a protective limiter, the function of which is, like that of a human operator, to apply such overall corrections to the programme level as may be necessary to prevent overload of the transmission system without reducing the mean level of modulation, while making the minimum possible change in the short-term dynamic range.

5.2.5 Double- and triple-time-constant circuits

To avoid the more extreme effects described above, it is usual to arrange, as a compromise, for the recovery time-constant of the control circuit in a simple limiter to be of the order of 0.5 s to 1 s. There is, however, no single relationship between gain recovery and time by which all of the effects described above can be avoided, and various attempts have therefore been made to achieve a better compromise by making the effective recovery time-constant vary automatically with the nature of the signal. One such device introduced in early BBC limiters, gave, in effect, a double time-constant, the gain-recovery period being automatically prolonged according to the amount and duration of the gain reduction which preceded it. While appreciably ameliorating the situation this still failed to bridge the gap between the more extreme requirements, and the 1964 investigations included experiments aimed at extending the range of conditions over which satisfactory operation was possible.

It was found that some of the difficulties associated with a long recovery time could be overcome by making the time-constant dependent on output signal level; a recovery time-constant far in excess of that normally permissible can be allowed, provided that the system is arranged to revert automatically, over a period of about a second, to the usual compromise value of 0.5 s to 1 s during pauses or quiet passages in the programme.

There remained the problem of achieving a sufficiently rapid gain recovery after the limiter had been activated by a signal of less than syllabic duration, while at the same time avoiding flutter effects on sustained passages. These two requirements are in direct conflict; a useful compromise can, however, be achieved by a double-time-constant circuit having fixed parameters, so designed that part of the gain recovery is effected very rapidly, after the limiter has been activated by a signal peak of short duration, leaving the remainder to be restored at a slower rate.

The circuit arrangement finally adopted incorporated both the artifices described in the preceding paragraphs, so that the overall effect might be loosely described as that of a triple time constant[5].

5.2.6 Recovery time-constant parameters selected

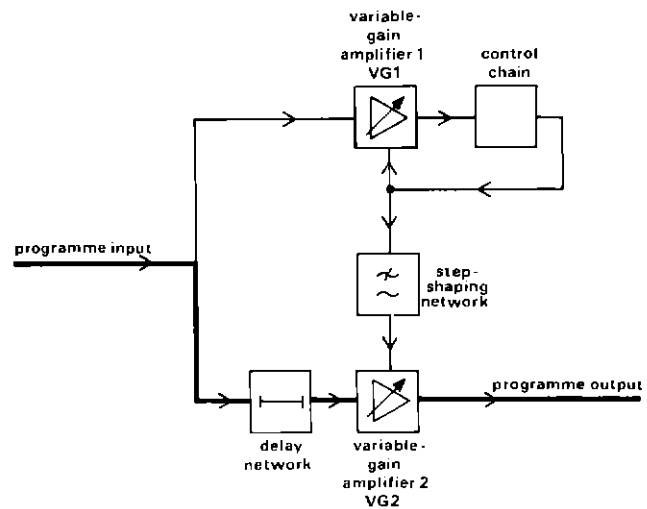Figure 9 shows the essential elements of a peak-detection

Fig. 8 Duplicate variable-gain stage limiter with step-shaping and delay networks.
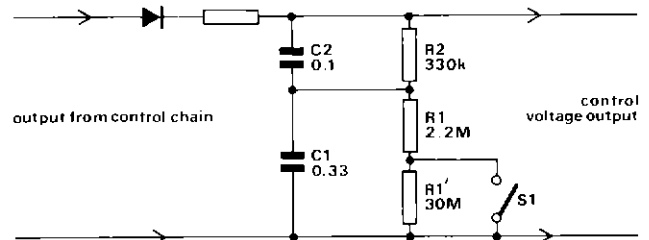

Fig. 9 Peak detection and smoothing system with triple time-constants.

and smoothing circuit giving a triple-time-constant recovery characteristic. The values of the parameters given below are those originally chosen on the basis of subjective assessments by only a few listeners using a variety of selected programme material; however, no significant changes have been found necessary in field trial or service use during the intervening years.

The longest effective time-constant is formed by $C_1(R_1+R_1')$; this product was made about 10 s. For programme levels below a prescribed changeover point, R_1 is short-circuited electronically — a process symbolised in figure 9 by the switch S1 — leaving an intermediate value of time-constant $C_1 R_1$ for which the best compromise was found to be about 700 ms. The shortest time-constant, formed by $C_2 R_2$, was made 33 ms. The values of C_1 and C_2 were chosen so that about three-quarters of the control voltage arising from a momentary overload was allowed to be discharged at the 33 ms rate.

The signal level at which the change of time-constant takes place has to be set sufficiently low to avoid appreciably interfering with the 'tail' of a long reverberation process, yet not so low that gain recovery at the intermediate rate is virtually excluded. For programmes including classical music covering a wide dynamic range, a suitable changeover point was found to be that at which the output signal falls to 20dB below its limiting value.

5.3 A production non-overshoot limiter

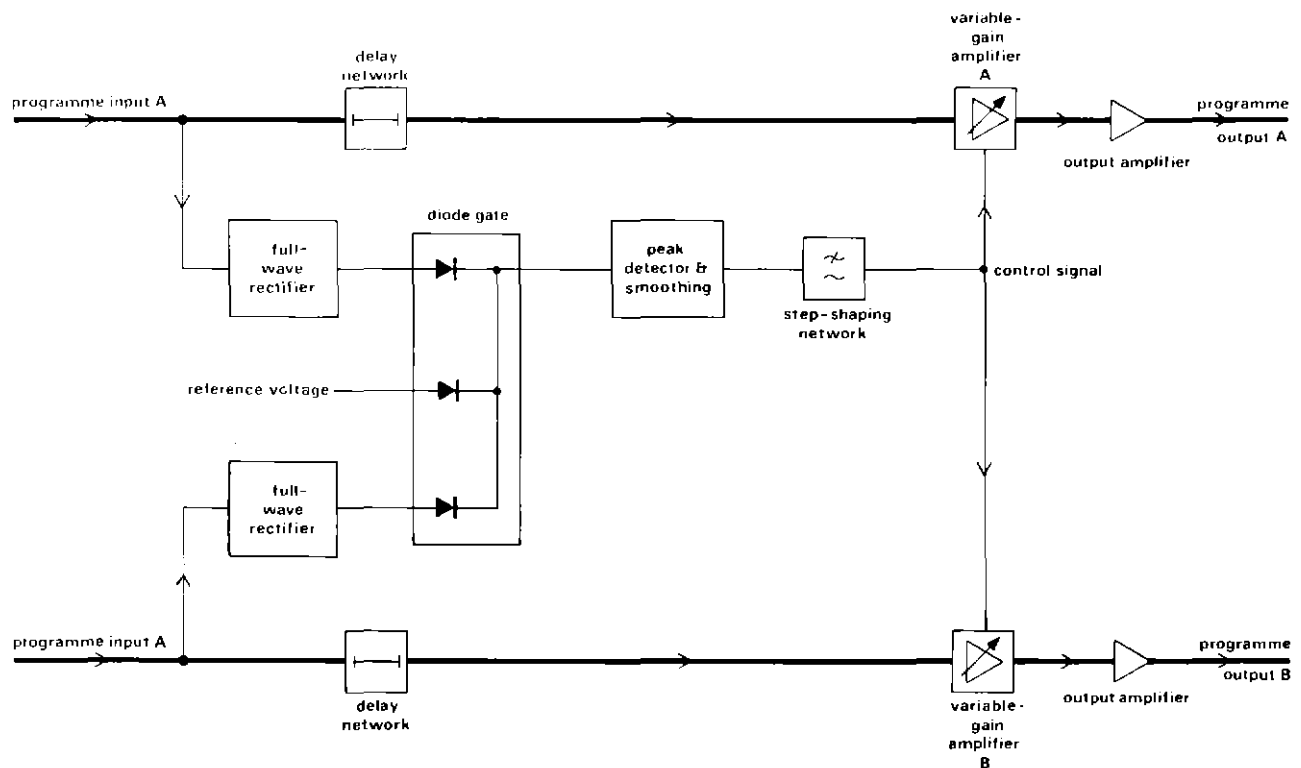The design of the AM6/7 limiter, developed in the late

Fig. 10 Stereophonic limiter arrangement.

sixties, was based on the results of the investigations described above. With its introduction it became possible to eliminate transient overshoot entirely, without risk of introducing impairment in the process.

As indicated earlier the AM6/7 limiter used the method of duplicate variable-gain stages shown in figure 8. The control signal was derived in a fast-acting feed-back limiter circuit, and was transferred at an appropriate rate, controlled by a step-shaping network. to the programme-channel variable-gain element. A lumped-constant delay network having a group delay of about 300µs provided time for the gain in the programme channel to be reduced before the arrival of any programme peak which required control.

The variable-gain elements were selected matched pairs of field-effect transistors (FETs) acting as variable shunt resistors in L-section attenuators.

5.4 Requirements for stereophony

So far only brief mention has been made of stereophony. Stereophony places additional requirements on limiters that were difficult to meet with earlier designs. First, it is essential that, in order to avoid image shift, the two limiters used as a stereophonic pair should be matched and ganged so that they have equal gains at all times: there is experimental evidence that gain matching both statically and dynamically to within about 2dB is required, and clearly matching even more closely than this is desirable within a single link in a transmission circuit. Second, if the output of neither limiter is to exceed its maximum defined level it follows that the common control during gain reduction

should be derived from whichever limiter in the pair has the greater input at the time.

These additional requirements could be met with limiters of the AM6/7 type (see section 5.3), but not without some practical difficulties which were costly to overcome. The most successful ganging technique investigated with this type of limiter was relatively complex: a composite signal, derived from the two programme input signals by time-division multiplex, was used to determine the amount of control effected by the limiters in the two channels.

If, however, the limiters were of the type shown in figure 7, two such devices could conveniently be connected for stereophony in the manner shown in figure 10. Here, each input is full-wave rectified and, by simple diode gating, the greater of the two rectified signals is fed to the remainder of a common control chain. As discussed earlier this approach requires a particular gain-versus-bias characteristic for the variable-gain element in order to achieve the ideal flat-topped static characteristic shown in figure 2b. Where the control signal is linearly related to the limiter input signal, the gain of the variable-gain element must be inversely proportional to the control signal if the desired characteristics are to be achieved. Then, as the input signal exceeds the limiting threshold, the increasing control signal causes the variable-gain element to reduce gain to compensate exactly, and a constant level is maintained at the limiter output. This process may be regarded under steady-state conditions as an arithmetic division of the programme signal by a direct control voltage which, for inputs beyond the limiting threshold, is proportional to the peak of the input waveform.

Such an analogue divider was readily instrumented with the advent of the integrated-circuit multiplier. A multiplier in the negative-feedback path of an amplifier with a high loop gain, as shown in figure 11, results in a variable-gain element of the required type. Any departure from the ideal limiting characteristic depends only on the control circuit and multiplier linearity.

A stereophonic limiter using this technique, the AM6/11, was developed in 1972. The dynamic performance was similar to that of the AM6/7 but the gains could readily be matched within 0.3dB over a 20dB control range with a departure from a flat-topped characteristic of less than 0.5dB.

This form of ganging enables any number of devices to be accurately coupled together and subsequent designs, e.g. the AM6/14, have consisted of monophonic units which, for stereophony, can be ganged to any unit of like design by a *simple wire link*.

5.5 Variable-emphasis limiters
5.5.1 General

Where a flat-spectrum* limiter is used to control pre-emphasised sound signals for FM radio in the manner described in section 4.3, limiting action which is necessary to regulate the emphasised high-frequency components of programme — even though that programme was correctly controlled at source — also depresses the level of low- and medium-frequency components. In some circumstances this action can give rise to an objectionable subjective effect, often referred to as gain ducking. In these circumstances, it has been a common practice for broadcasting organisations to attenuate the signal applied to the limiter until gain ducking is reduced to acceptable proportions — a practice which necessarily reduces the effective service area of the transmitter. In 1971/72 work began on the development of a limiter which, even with full-level signals applied, substantially eliminates impairment due to gain ducking on emphasised sound signals and which prevents overmodulation, even of a momentary nature.

Overloading as a result of high-frequency pre-emphasis can be prevented by a limiter which simply reduces the amount of emphasis as required, without affecting the level of lower-frequency components[6]. Impairment due to gain ducking can thus be made much less obtrusive, and the audible effect of limiter action need be no more than a momentary loss of 'top', generally imperceptible unless unprocessed programme is available for reference.

By its nature a limiter stage of this kind cannot, however, give protection against overloading by low-frequency signal components of excessive level; it must therefore be preceded by a flat-spectrum limiter stage of the non-overshoot delay-line type[4]. The combination of the two limiter stages, plus any necessary emphasis network, will be referred to as a two-stage variable-emphasis limiter.

The arrangement was developed for 50 μs pre-emphasis operation, principally with FM radio and television sound in

*A flat-spectrum limiter can be defined for the present purpose as one which has, at any instant, the same gain for all signal components, irrespective of their frequency.
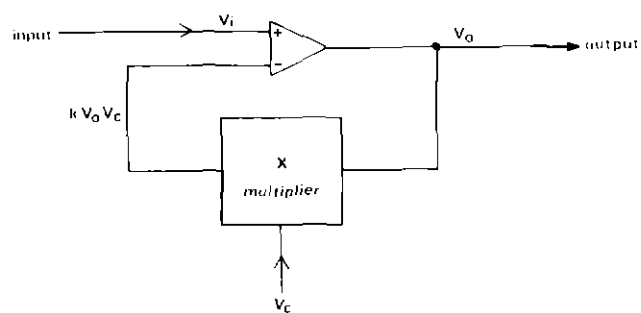

Fig. 11 Voltage-controlled amplifiers. Conventional use of a multiplier in the feedback path of an operational amplifier to produce an analogue divider.
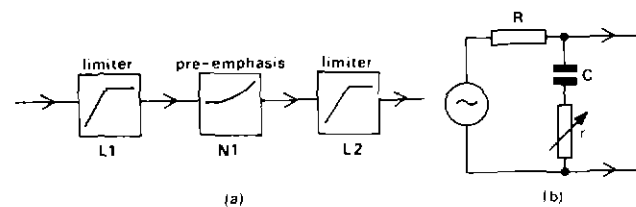

Fig. 12 Experimental two-stage variable-emphasis limiter.
a. Block schematic.
b. Variable-gain element of stage-two limiter.

mind, but it can be applied in other situations where the effective overload level decreases with increasing signal frequency. The principle can be applied to systems having other than 50 μs pre-emphasis, though its performance would have to be reassessed if greater values were being considered. It has found application in association with the BBC's PCM system for sound-signal distribution.

Here the sound signals are pre-emphasised at the sending terminal and variable-emphasis limiters located at that point can give protection to the PCM channels *and*, because of the gain stability of the PCM system, to *all* the FM transmitters they serve.

5.5.2 Experimental circuit arrangement

Figure 12a shows the general arrangement of an experimental two-stage variable-emphasis limiter constructed for evaluation. Incoming signals, when of excessive level, are regulated by a non-overshoot flat-spectrum stage-one limiter L1, then applied to a 50 μs pre-emphasis network N1, and finally to the variable de-emphasis stage-two limiter L2. The flat-spectrum limiter used has a complex gain-recovery characteristic, of the type described in section 5.2.5, and can exercise a considerable amount of control with little impairment of quality.

The circuit of the variable de-emphasis element used initially in stage L2 is shown in figure 12b. It comprises an L section with a series arm of resistance R and a shunt arm having capacitance and variable resistance r connected in series — the latter being in fact the drain/source impedance of a field-effect transistor (FET). The value of r, and hence the amount of top-cut introduced by the circuit, can be controlled by varying the voltage applied to the FET gate. Component values were chosen so that, in the limit (r = 0), a 50 μs de-emphasis characteristic was applied, just offsetting

the effect of the preceding pre-emphasis. The listener thus hears some degree of top cut when programme-signal conditions require L2 to operate.

The subjective effect of this top loss may be minimised by making its duration as short as possible. In flat-spectrum limiters the recovery time-constant is normally made at least several hundred milliseconds in order to avoid audible distortion due to significant gain recovery during the period of one cycle, i.e. 'cycle-following', at low frequencies. The variable de-emphasis stage, however, operates significantly only on high-frequency components, so the rate of gain recovery can be much higher without causing audible distortion; in the experimental apparatus the rate of gain recovery for this stage was set by a circuit having a 25 ms time-constant.

5.5.3 Subjective assessment

Subjective tests were carried out to compare the quality of reproduced programme processed by the experimental arrangement with unprocessed programme and with pre-emphasised programme processed by a conventional flat-spectrum limiter. Three short programme items were used for the investigation — selected because the nature of their high-frequency content made them potentially susceptible to gain-ducking and to degradation due to top loss. Tests were carried out with programme applied to the limiters at normal level and at increased levels such as might occur accidentally in service.

For two programme items having intermittent high-frequency components the subjects showed a marked preference for the experimental arrangement rather than the conventional system. In the item with more continuous high-frequency content, however, the results indicated little difference between the two systems, but in no case was the mean grading of the experimental arrangement worse than that of the conventional arrangement.

The comparisons with unprocessed programme showed that, while the listeners did indeed prefer the unprocessed version of the most stringent test items, the difference was only slight. Programme processed by the variable-emphasis limiter system was, even in these circumstances, less than one grade worse than the unprocessed programme on a seven-point comparison scale running from 'much better' to 'much worse'.

Prototype limiters operating as described above were first subjected to field trial and service use in 1973.

5.5.4 A production design

In the experimental variable-emphasis limiter arrangement non-overshoot delay-line limiters were used for both the stages. Subsequent investigation[7] has indicated that, under the conditions in which the second stage operates, a simplified, non-delay-line arrangement — in which overshoot is prevented by instantaneous clippers — is acceptable provided that the steady-state output limiting level is set about $\frac{1}{4}$ dB below the clipping level. This arrangement has been adopted in the production limiter instrumentation described below.

The nature of the variable-gain element, too, has been

revised during development; the variable-de-emphasis arrangement has been changed to a variable-pre-emphasis arrangement, with the 50 μs fixed pre-emphasis network omitted. As a result the effects on the programme spectrum are somewhat modified but listening tests have indicated that in practice the subjective performance is substantially unchanged.

The experimental two-stage variable-emphasis limiter just described used a field-effect transistor to provide the voltage-controlled resistance, as did the first generation of overshoot-free flat-spectrum limiters, but in the variable-emphasis case the problems of stereophonic ganging are even more acute. Both the flat-spectrum stage-one limiters and the variable-emphasis stage-two device need to be ganged, to an acceptable degree of accuracy, to the corresponding parts of the companion limiter of the stereophonic pair. This requirement makes the use of field-effect transistors in a production design somewhat difficult. Fortunately the needs of the stage-one limiters were met by existing circuits used by the BBC (see section 5.4).

Further development of the stage-two limiter to use multiplier techniques, while still meeting the critical noise requirements of the PCM distribution system, resulted in some modification to the stage-two limiter. Figure 13 shows the basis of the new configuration.

Here the apparent impedance of the network in the emitter circuit of the transistor TR1 depends on the amplitude and phase of the output signal from amplifier A1. This signal is in phase with the input and its magnitude is determined by the control voltage applied to the multiplier. Under normal working conditions this control voltage is zero, and hence there is no signal either at the output of the multiplier or at the output of A1. The system is arranged so that under these conditions the output signal (V) taken from the collector of TR1 has undergone the standard 50 μs pre-emphasis used for FM broadcasting. If, however, the output tends to exceed the defined limiting level, peaks of the full-wave rectified output exceed a reference level and an error amplifier charges capacitor C_2 via a diode and resistor R_4. The resistor and capacitor restrict the attack speed of the control system. The potential difference across the capacitor is fed to the control input of the multiplier and causes a signal in phase with the input signal to appear at the output of A1, increasing the apparent impedance of C_1 and R_3 in the emitter circuit of TR1. This effectively reduces the pre-emphasis time constant until the output assumes a value no greater than the defined maximum. When the rectified output drops below the reference potential, C_2 discharges through R_5 and the standard 50 μs pre-emphasis is restored. The recovery time-constant may be made short for the reason already discussed, and the necessary engineering control is effected with a minimum of subjective disturbance.

The variable-emphasis stage described above is of the non-delay-line type, and clippers are provided, as described earlier, to eliminate momentary overshoot of the pre-emphasised signals. Ganging of two stage-two limiters in a stereophonic arrangement is conveniently carried out by the simple linking arrangement described in section 5.4 for flat-spectrum limiters.

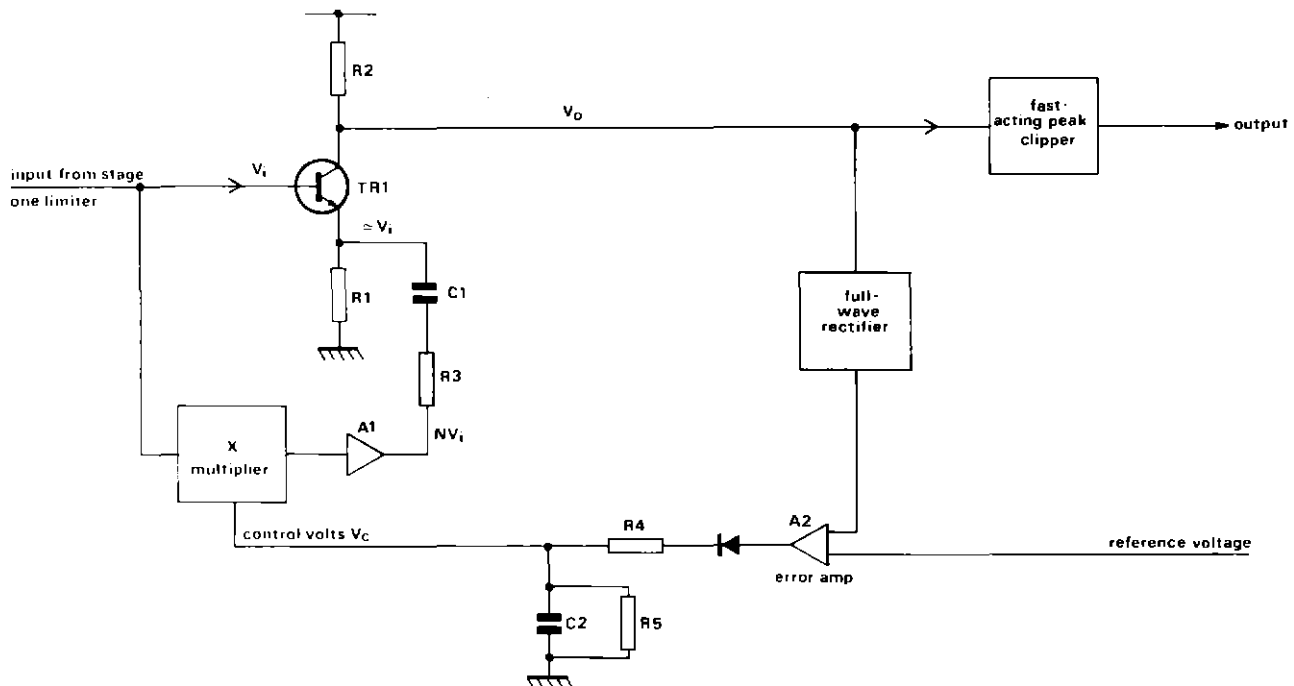Stereophonic variable-emphasis working on the principles

Fig. 13 Stage-two variable-emphasis limiter

described above were used in service in 1974, and the production design, the AM6/16, came into service in 1975.

6 Conclusions

This review of the development of sound-programme limiters within the BBC mirrors advancing technology and changing requirements.

The physical size of limiters has been reduced, first by the introduction of miniature valves and later by the introduction of transistors and integrated circuits. Limiter size is likely to continue to decrease with the increasing application of integrated circuits.

Limiter performance has been improved by successive refinements of a basic technique, to meet the needs of the day. The special requirements for FM radio and for stereophony have been met and new designs have been introduced to reduce the impairment of programme quality by eliminating momentary overshoot and by minimising the effect of gain fluctuations.

Future developments are less easy to predict, but it seems inevitable that analogue limiters, essentially similar to those described, will continue to be required for the foreseeable future, at least until an economic digital equivalent is developed.

7 References

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Acknowledgements

Some Considerations on the Use of Pre-Amplifiers at Low-Power UHF Relay Stations

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Research Department

Summary: Several inexpensive, commercially available pre-amplifiers are examined and their performances in respect of gain, noise factor, linearity, and impedance match are compared. Some theoretical planning curves are presented for considering low-field conditions and formulae are derived for dealing with high-signal conditions.

A re-broadcast link front end, made up from the most suitable of the amplifiers examined, is assessed in respect of its overall performance. An indication is given of the amount of isolation which is required between transmitting and receiving aerials and the desirable response of the mast-head filter.

List of symbols

f	frequency, MHz, subscripts for identification
s	signal power level, dB(mW)
	subscripts: i = input, o = output 1, 2, 3 identification
s'	relative power level, dB, subscripts as for s
v	general voltage amplitude
	subscripts: i = input, o = output
x	slope of filter band-edge, dB/MHz
B	bandwidth
C	output intercept point, dB(mW)
D	feeder loss (ratio of power at input to power at output)
E	field strength, μV/m
F	noise factor, subscripts for indentification
G	power gain, subscripts for identification
G_A	aerial gain (relative to a half-wave dipole)
G_a	amplifier power gain
I_A	aerial isolation, dB
i_E	emitter current
M	noise measure
N	noise power
R	resistance, subscripts for specific values
T	temperature, K
V	voltage amplitude, subscripts for identification
Y_s	source admittance
Z	constant
α, β	constants
λ	wavelength, m
ω	angular frequency, subscripts for identification

Several of the above symbols represent voltage or power ratios which, in parts of the article, are more conveniently considered in terms of decibels. Where this is done the symbol is adapted thus: G represents power gain, expressed as the ratio of output power to input power; $_{dB}G$ represents the gain in decibels — thus $_{dB}G = 10 \log G$, but $_{dB}E = 20 \log E$.
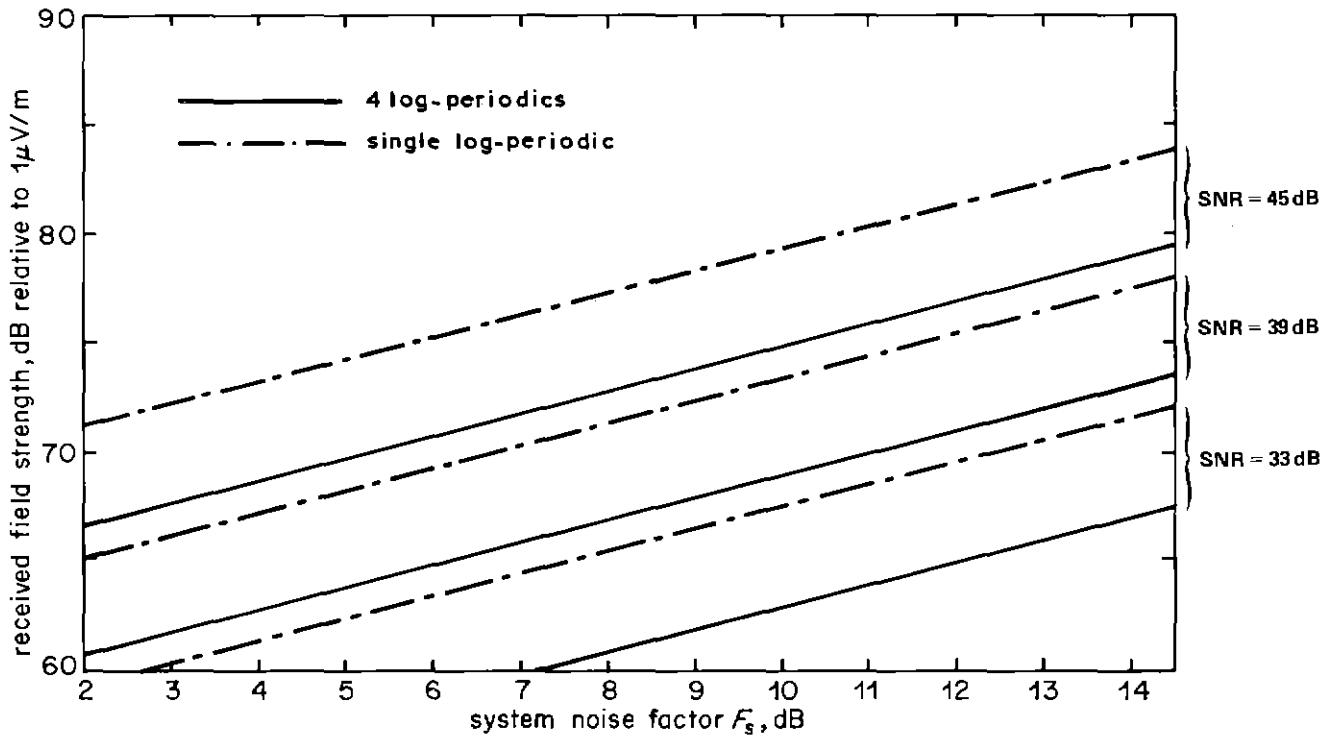
Fig. 1 The dependence of the minimum required field strength on system noise factor at 650 MHz

1 Introduction

As the total number of UHF television relay stations in the UK is likely to be several hundred, ways of reducing costs are continually being considered. A preliminary theoretical survey has indicated that a reasonable number of stations will require only one, or maybe two, log-periodic aerials for the receiving array. This is is permissible when co-channel interference conditions allow a broader horizontal radiation pattern to be used, e.g. when the terrain provides some protection. By using pre-amplifiers to compensate for reduced aerial gain, the number of sites at which the simpler receiving aerial is suitable can be increased.

A second benefit of using pre-amplifiers is the improvement in the signal-to-noise performance in low field conditions. The benefit is greatest when the amplifier is used at the top of the mast immediately after the aerial. The attenuation of the feeder then contributes little to the overall noise performance and a cheaper, higher-loss cable can be used.

As an alternative, an amplifier at the foot of the mast, immediately in front of the channel-splitting equipment and transposers, would also be beneficial provided its noise factor is better than that of the transposer. In this case a low-loss feeder is desirable.

The report first considers some of the theoretical aspects of using pre-amplifiers at UHF relay stations. An assessment is then given of a number of commercially available inexpensive amplifiers. In addition a series of measurements has been made to establish the aerial isolation templet which is appropriate to low-power relay stations.

2 Theoretical considerations

2.1 Low signals

The sensitivity of a receiver, which determines the lowest acceptable signal levels, is related to its noise factor, F_s (or to its noise temperature).[1]

The video signal-to-noise ratio, in decibels, at the output of a system is numerically given by (see Appendices 1 and 2)

$$SNR = {}_{dP}E + 20 \log \lambda + {}_{dB}G_A - {}_{dB}F_s - 25 \cdot 7$$

where ${}_{dB}E$ = received field strength, $dB(\mu V/m)$
λ = signal wavelength, m
${}_{dB}G_A$ = net aerial gain, dB
${}_{dB}F_s$ = receiving system noise factor, dB

If the required signal-to-noise ratio at the output of the receiver is specified, the lowest acceptable field strength can be found from this expression.

The required transmission standards for UHF low-power relay stations have already been established and depend on the type of station. Details of the two standards, which are referred to as P and Q, are given in Appendix 3. The minimum acceptable received field strength at a frequency of 650 MHz in relation to the system noise factor, is shown in figure 1. Two types of receiving aerial are assumed: one comprising a single log-periodic aerial and the other an array of four such aerials.

The system noise factor depends on the arrangement adopted, i.e. whether the pre-amplifier is placed at the mast-
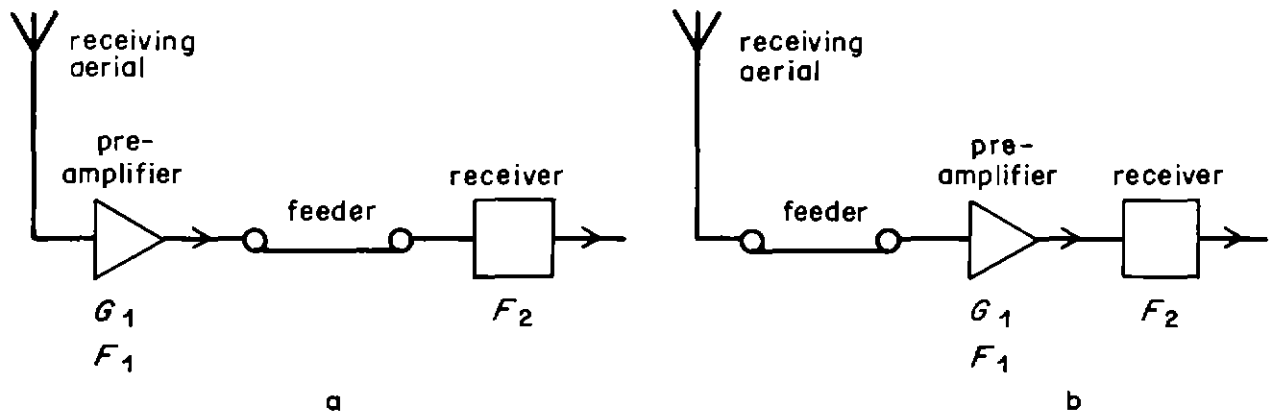
Fig. 2   Receiving systems (a) Mast-head pre-amplifier (b) Mast-base pre-amplifier
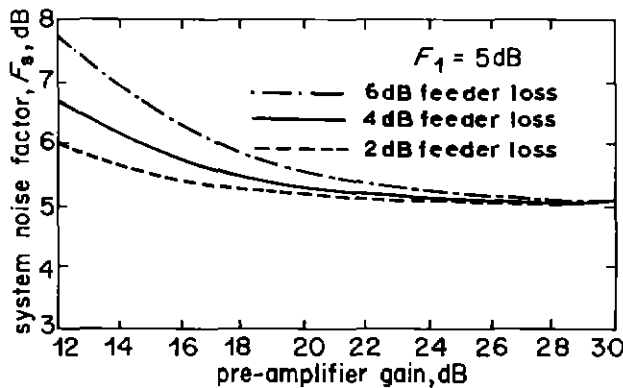

Fig. 3   The noise factor of system of figure 2a

head immediately after the aerial, or at the foot of the mast following the main feeder.

Referring to figure 2a and 2b and assuming a matched system, the respective noise factors are given by:

$$F_{s(a)} = F_1 + \frac{F_2 D - 1}{G_1}$$

$$\text{and } F_{s(b)} = D\left(F_1 + \frac{F_2 - 1}{G_1}\right)$$

where

F_1, G_1 are the pre-amplifier noise factor and gain respectively,

F_2 is the receiver noise factor, and

D is the feeder loss.

The performance of these arrangements is compared with that of a conventional arrangement in table 1. A receiver noise factor of 10 dB, a feeder loss of 4 dB, a pre-amplifier noise factor of 4 dB and gain of 15 dB, are assumed.

The advantage of a pre-amplifier in low-signal conditions is readily seen; moreover, the advantage is greatest when a mast-head pre-amplifier is used. For this case, figure 3 shows how the system noise factor depends on the feeder loss and pre-amplifier gain. A gain of 20 dB is seen to be adequate for this case. Only if the feeder loss were higher or

the receiver more noisy, would a greater gain be required.

2.2 High signals

At high signal levels an amplifier's non-linearity causes it to generate intermodulation products (IPs) and eventually to overload. Let the amplifier characteristic be represented by a third-order polynomial such that the output response, v_o, to an input signal v_i, is

$$v_o = \alpha v_i + \beta v_i^3$$

where α and β are constants. For the signal excursions being considered, this representation gives a good approximation for calculations involving solid-state amplifiers.

Consider the effect of applying two tones to the amplifier:

$$v_i = V_r \cos \omega_r t + V_s \cos \omega_s t$$

The output spectrum in the neighbourhood of the input frequencies is of the form shown in figure 4, where the amplitudes of the IPs are $0.75\beta V_r^2 V_s$ and $0.75\beta V_r V_s^2$. In normal television applications this type of IP is the most troublesome.

The principal third-order IPs mentioned above can be used to define the linearity and hence the dynamic range of an amplifier. A 'two-tone' test is applied in which two equal sinusoidal signals of different frequencies at the input are progressively increased in amplitude. The input/output characteristics showing the power levels in dB(mW) associated with the fundamental and IPs are plotted in figure 5.

TABLE 1

Minimum received field strength in dB (μV/m) required for some RBL systems using 4 log-periodic aerials at 650MHz

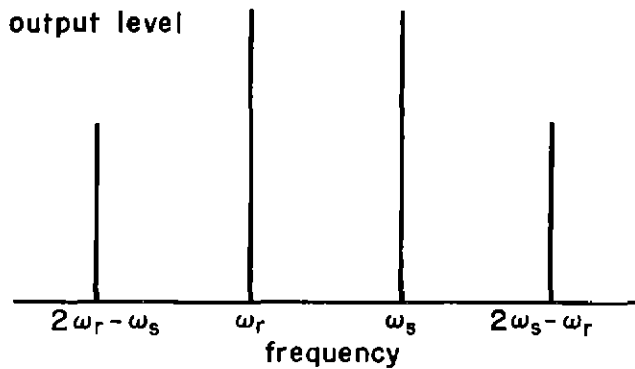| SNR	45 dB	39 dB(P)	33 dB(Q)
No pre-amp $F_s = 14$ dB	78·9	72·9	66·9
Mast-base amp $F_s = 8$ dB	72·9	66·9	60·9
Mast-head amp $F_s = 5$ dB	69·9	63·9	57·9

Fig. 4 Part of the output spectrum of a non-linear amplifier with two input tones

The curves are linear for low input levels and their linear portions, when extrapolated, cut at the intercept point, C. The output level in dB(mW) corresponding to this point is called 'the output intercept point' and is a convenient indication of the linearity of the amplifier. From the intercept point, the maximum input an amplifier can handle before the IPs become intolerable can be predicted. Sometimes the 'input intercept point' is referred to, which is less than the output intercept point by the amplifier gain.

An alternative test of linearity is the three-tone test.[2] In this test the vision, colour, and sound carriers are simulated by separate tones set at particular levels. These are referred to as f_v, f_c, and f_s respectively. The vision tone is set at 8 dB below the reference level of peak carrier power to correspond to an average level of luminance.

In the three-tone test the most visible IP appears at f_v+1·57 MHz. If the fundamental output represents the peak sync level, the three-tone test IPs at f_v+1·57 MHz are shown in figure 5. They are 26 dB lower than the two-tone input IPs for the assumed characteristic (see Appendix 4).

For RBL applications an amplifier must be capable of handling four television channels. The maximum amplifier inputs per channel to meet the requirements of standards P and Q (see Appendices 3 and 4) can be expressed in terms of the intercept point:

Standard P: $s_{i(max)}=(C - _{dB}Ga)-24$ dB(mW)
Standard Q: $s_{i(max)}=(C - _{dB}Ga)-20$ dB(mW).

Incident field strengths are unlikely to exceed 90 dB(μV/m) at sites where pre-amplifiers are considered. The most stringent requirement corresponds to the lowest frequency, where the output from four log-periodic aerials is −26 dB(mW) which leads to minimum input intercept points of

−2 dB(mW) for Standard P
−6 dB(mW) for Standard Q:

2.3 Return loss

The impedance match required at the input and output terminals of a pre-amplifier depends on the level of delayed images which can be tolerated at the final output. These levels are related to the delay times associated with the
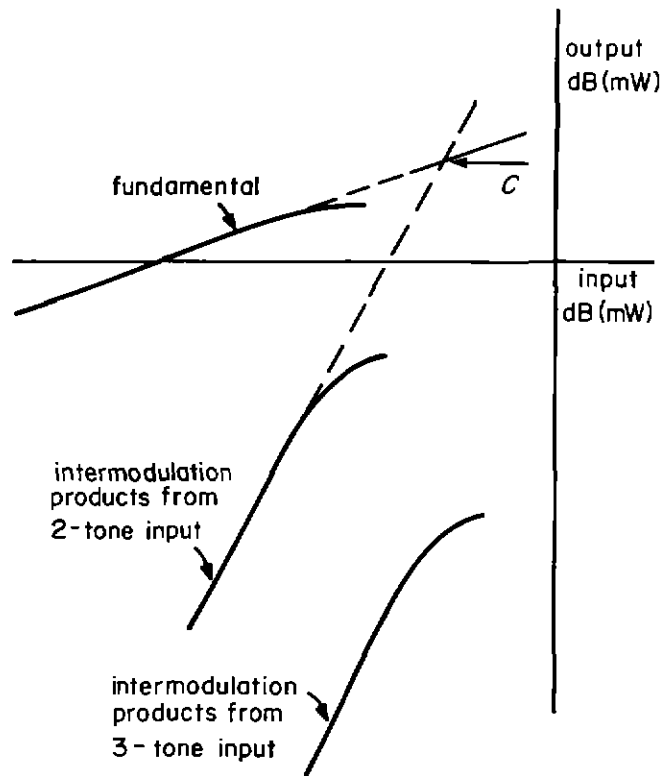

Fig. 5 The performance of a non-linear amplifier

appropriate signal path. When considering the amplifier input impedance the delayed signal is produced by reflection at the amplifier input and re-reflection at the aerial. Similarly, when considering the amplifier output impedance the delayed signal results from reflection at the receiver input and re-reflection at the amplifier output.

At low-power relay stations the feeder lengths are short enough to keep the delay time well below 1 μs. The delayed signal attenuation required is given in Appendix 3. For a Standard P transmission it is 28 dB and for Standard Q it is 24·5 dB.

If the feeder loss is 3 dB and the return loss at the aerial terminal is 16 dB, the minimum return loss at the amplifier input is given in table 2 for both mast-head and mast-base situations. Similar figures are given for the minimum return

TABLE 2

Amplifier minimum return loss (dB)
VSWR figures are given in brackets

	Standard P		Standard Q	
	Input Impedance	Output Impedance	Input Impedance	Output Impedance
Mast-head amplifier	12 (1·7)	12 (1·7)	9 (2·1)	9 (2·1)
Mast-base amplifier	7 (2·6)	18 (1·3)	4·5 (4)	14·5 (1·5)

amplifier No.1
50 Ω
input →
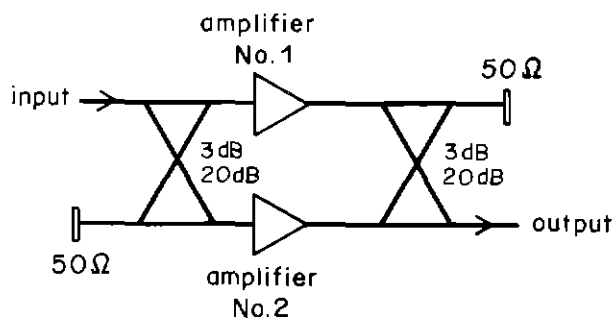3 dB 20 dB   3 dB 20 dB
→ output
50 Ω
amplifier No.2

Fig. 6 An Engelbrecht arrangement of two amplifiers

loss at the amplifier output assuming that the receiver input return loss is 10 dB.

2.4 Engelbrecht arrangements

A parallel arrangement of two amplifiers can be used to improve performance.[3] The amplifiers are interconnected by hybrids as shown in figure 6: the input signal is divided equally between two paths and fed through separate amplifiers. The output signals are recombined in a second hybrid. The hybrids are designated by two figures which are the output coupling and directivity respectively. Because the reflections can be diverted to a 50 Ω termination, the input and output return loss of the system is greater than that of either amplifier alone. The signal path is shared so that each amplifier handles half the input power. This gives the system a 3 dB advantage as far as linearity is concerned. The performance in respect of both noise figure and gain is

unchanged. If one amplifier should fail for any reason the output signal is maintained at a 6 dB lower level — hence reliability is increased.

3 Assessment of some commercial amplifiers

Altogether twelve amplifiers or arrangements of amplifiers were examined. Table 3 gives details of the measurements made and also the manufacturers' quoted performance. Amplifiers were chosen either for low noise figure or high linearity, together with low cost. A BBC-designed amplifier (reference J) was used as a comparison. The amplifiers' nominal impedances are generally 75 Ω but in an Engelbrecht arrangement designed for 50 Ω the mismatch is effectively suppressed.

3.1 Gain

Measurements of gain were made throughout the UHF Bands IV and V* at input levels of about 10mV, which is well below the overload for these amplifiers. The figures quoted in table 3 are averages for Bands IV and V with the measured variation over this range.

3.2 Noise

The measured noise factors are the averages taken from a series of spot noise figures measured at 50 MHz intervals in Bands IV and V using a receiver with an IF bandwidth of

*Band IV   470-582 MHz: Band V   614-854 MHz

TABLE 3
Results of amplifier tests

Model	Power supply	Gain (dB) in Bands IV and V		Noise			Linearity				Impedance match	
				Noise Factor (dB)		Noise § measure (dB)	Output intercept dB(mW)	Input intercept dB(mW)	Max. input dB(mW)#		Input	Output
		Quoted	Measured	Quoted	Measured				Quoted	Measured	VSWR	VSWR
A	24V 34mA	18	16±1	5	4·5±0·5	3·49	21·5	+4·5	−15·5	−18·5	2·1	2·45
A ENGELBRECHT	24V 68mA	18	16±2	5	5±0·5	4·24	24·6	+6·6		−16·4	1·95	1·77
B	24V 24mA	22	20±2	5	5±0·5	4·13	27·8	+3·8	−27·5	−19·2	2·60	2·25
C	16V 3mA	16	12±2	3·5	5±0·5	4·52	14·3	+0·3	−24·5	−22·7	−	−
D	24V 34mA	22	20±1	3·5	6±1	6·09	24·3	+3·3	−16·5	−19·7	7·2	2·60
D ENGELBRECHT	24V 68mA	22	20±1·5	3·5	6·5±0·5	6·10	28·3	+6·8		−16·2	1·69	1·74
E*	INTERNAL	28	27±2	4·5	6±0·5	5·41	32·0	+3·0	−17·9	−20·0	3·1	2·0
F†	24V	15	14±2	7	7·5±0·5	7·53	23·7	+7·7	−13·8	−15·3	1·18	1·21
G†	24V	25	26±4	5·5	7·5±1	7·87	20·6	−9·4	−28·8	−32·4	1·55	1·70
H	INTERNAL	20±1	22±1·5	6·5-10	8·5±1	9·02	31·0	+7·5	−22·5	−15·5	1·77	2·33
I	24V 20mA	16±2	14±1	7	5·5±1	5·62	25·0	+10·0	−25·5	−13·0	3·55	4·15
J	−12V 80mA	21±3	22±2	7	9·0±0·5	9·03	23·0	−1·0	−27·0	−24·0	2·20	1·86

†Engelbrecht arrangement   *VHF also   §See section 3.2   #for 4 UHF channels, 46 dB IPs
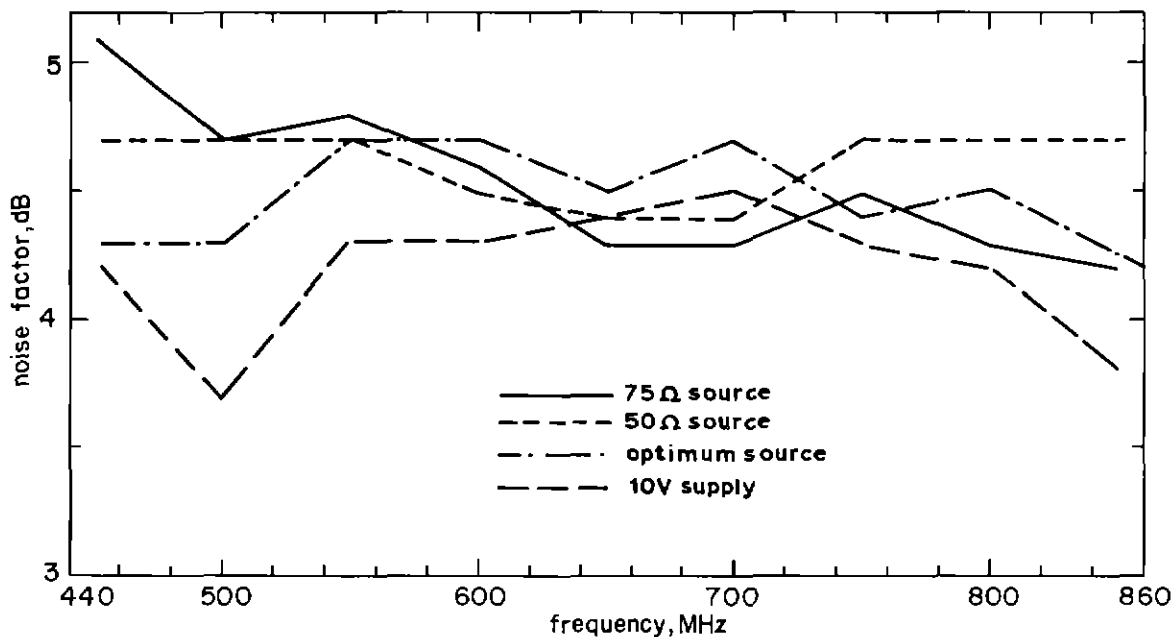
Fig. 7 The noise factor of amplifier A under different source impedance conditions

2 MHz and driven from a source matched to the input impedance of the amplifier. The level of the inherent noise from the pre-amplifier was measured on the receiver. A calibrated noise source was used to double this level.[4] Also tabulated is the comparative noise measure, M, which is given by[5]

$$M= \frac{F_a-1}{1-(1/G_a)}$$

where F_a is the noise factor of the amplifier and G_a is its gain. This is a useful noise performance figure of merit because it takes account of the gain. The lower noise measure indicates the better amplifier.

3.3 Linearity

The output and input intercept points are determined from two-tone test measurements at 650 MHz. From these the maximum level for a four-channel input, assuming a worst IP of 46 dB relative to the output, can be calculated (see Appendix 4). Manufacturers normally state the linearity performance of amplifiers in terms of the German specification DIN45004.[6] The quoted figures in table 3 have been derived from this.

3.4 Impedance match

The input and output VSWRs are given referred to the nominal characteristic impedance. This is 75Ω in all cases except for the Engelbrecht arrangements and the BBC-designed amplifier (J) for which the value is 50 Ω.

4 Improvement of amplifier noise performance

The noise factor is defined as the ratio of

(i) the total noise power per unit bandwidth available at the output port when the noise temperature of the input termination is standard (290 K), to

(ii) that portion of (i) engendered by the input termination.

The spot noise factor is that measured in a narrow frequency band and is useful for specifying the noise factor at individual frequencies. As defined the noise factor depends on the internal structure of the amplifier and on its input termination — but not on its output termination, i.e.

$$F=f(F_0,Y_s,i_E)$$

where F_0 represents the contribution from the amplifier itself, Y_s is the source admittance and i_E is the emitter current of the first transistor stage. There exists, therefore, at any given frequency, an 'optimum' noise factor [1,7,8] which is the lowest noise factor which can be obtained by adjustment of the source admittance and transistor operating point.

For amplifier A an attempt was made to improve its noise performance by adjusting the source admittance. This could be done for two amplifiers in an Engelbrecht configuration without making the overall impedance match unacceptable. Figure 7 shows how the noise factor of a single amplifier varies with frequency under matched conditions (7a) and driven from a 50 Ω source (7b). Further, a microstrip matching network was constructed consisting of a 50 Ω line with three shunt variable capacitors at eighth wavelength
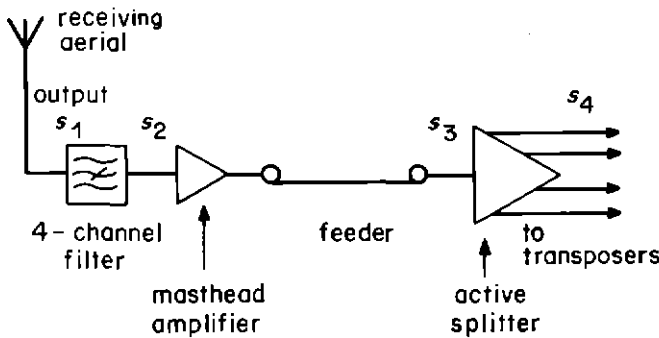
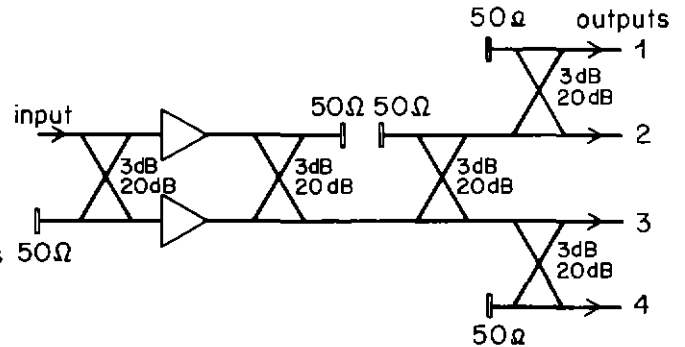**Fig. 8** An RBL system using inexpensive amplifiers

Fig. 9 An active 4-way splitting unit

intervals. The best noise performance which was found for a particular source impedance maintained across the frequency band is shown in figure 7c. It is only marginally better than the matched case and that only over a limited frequency range.

Also, for the same amplifier A, the operating point was effectively changed by reducing the supply voltage — but this also reduces the gain. The result of a 10 V supply voltage is shown in figure 7d.

To achieve a good all-round performance it is important to have the correct balance of noise factor, gain, and linearity.

5 An RBL system using inexpensive amplifiers

The proposed RBL front end shown in figure 8 comprises a four-channel bandpass mast-head filter, a mast-head amplifier, a main feeder, and an active splitter. The filter is required to protect the pre-amplifier from out-of-band signals and from the local transmission. Both the filter and the pre-amplifier require a weatherproof box for mounting and are expected to work with a high degree of reliability. The active splitter is a wideband device which provides four separate outputs with an adequate degree of isolation between them.

The filter, with a bandwidth of about 100 MHz to cover a standard four-channel grouping, must be of low loss, say less than 1 dB, to avoid significantly degrading the overall low-signal performance. Suitable filters are commercially

available: the band-edge slope is dependent on the number of sections chosen for the filter.

From the amplifiers examined, a suitable choice has been made to fit into the general arrangement of figure 8. The mast-head amplifier must be of the low-noise type — an Engelbrecht pair using amplifier A units would be appropriate. For the wideband splitter a suitable choice would be amplifier F units in Engelbrecht form. The arrangement is described in more detail in the next section. Typical operating levels for this arrangement are indicated in table 4 for 650 MHz.

The assumptions are that

Mast-head filter loss	= 1 dB
Pre-amplifier gain	= 16 dB
RBL feeder loss	= 4 dB
Active splitter, insertion gain to each port	= 8 dB

Three different receiving aerials are considered, comprising one, two, or four log-periodic arrays (LPs). Transposers of 2 W and 10 W are considered separately and the minimum and maximum levels given are based on typical transposer operating levels. The minima are set by signal-to-noise ratio limits and the maxima by intermodulation, assuming a standard P transmission: 10 W transposers introduce more intermodulation than 2 W installations and thus demand lower IPs at the input.

6 An active 4-way splitting unit

A four-way splitting unit was constructed from an Engel-

TABLE 4

|		Receiving aerial and gain (dB)			Aerial output S_1 dB(mW)	Pre-amp input S_2 dB(mW)	Active splitter input S_3 dB(mW)	Transposer input S_4 dB(mW)
	Transposer power	1LP (8dB)	2LPs (11dB)	4LPs (12·5dB)				
		Received field strength (dB(μV/m))						
Minima (required by noise considerations)		69·9	66·9	65·4	−53·4	−54·4	−42·4	−34·4
Maxima (determined by inter-modulation performance)	2W	92·5	89·5	88·0	−30·8	−31·8	−19·8	−11·8
	10W	89·3	86·3	84·8	−34	−35	−23	−15
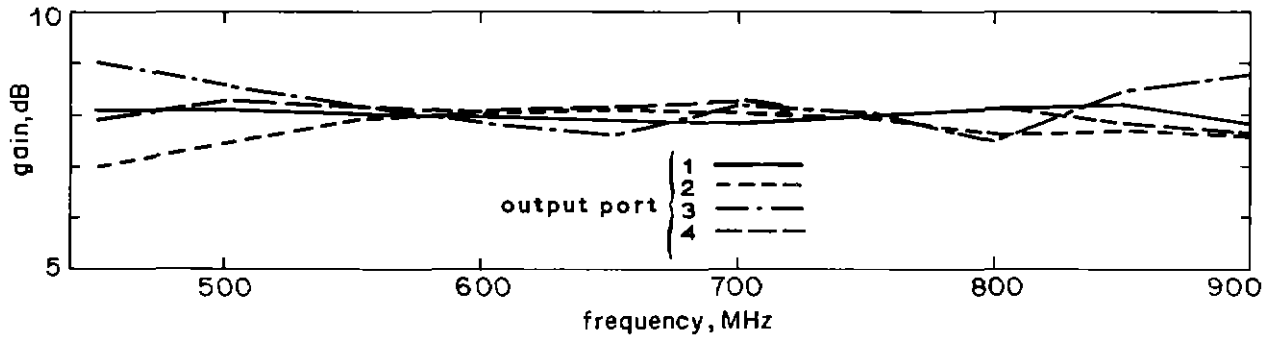
Fig. 10   Gain of the active splitter using wireline hybrids

brecht pair of amplifier F units and five hybrids made from wireline and interconnected by microstrip on double-sided printed laminate. The arrangement is shown in figure 9.

For the splitter the overload performance of the amplifier is more important than its noise performance. The gain need only be sufficient to overcome the losses introduced by the hybrids in the splitting process. As the diagram shows, two of the hybrids are used in the Engelbrecht pair and the other three perform the split in two stages.

Measurements on the splitting unit gave the results shown in figure 10 for insertion gain and in figure 11 for isolation between output ports. The worst return losses at each output port are tabulated below:

Port No.	Worst return loss (dB)
1	22
2	20
3	20
4	24·5

7   Aerial isolation and input filtering

The isolation required between the terminals of the transmitting and receiving aerials at a low-power relay station is generally determined by the generation of unwanted intermodulation products. With up to four television channels amplified in a common amplifier, the worst of these take the form of spectral components near the vision carrier frequency. Subjectively this appears as a patterning similar to that produced by some forms of co-channel interference. In general, closed-loop effects such as system stability problems and the visibility of delayed images created in the loop, will be insignificant if the IP criteria are met.

In principle, any desired loop transmission loss may be achieved by inserting a channel-group filter before the masthead amplifier. The specification for this filter is derived. Maximum acceptable IP levels are derived from the limits for a standard P transmission (see Appendix 3).

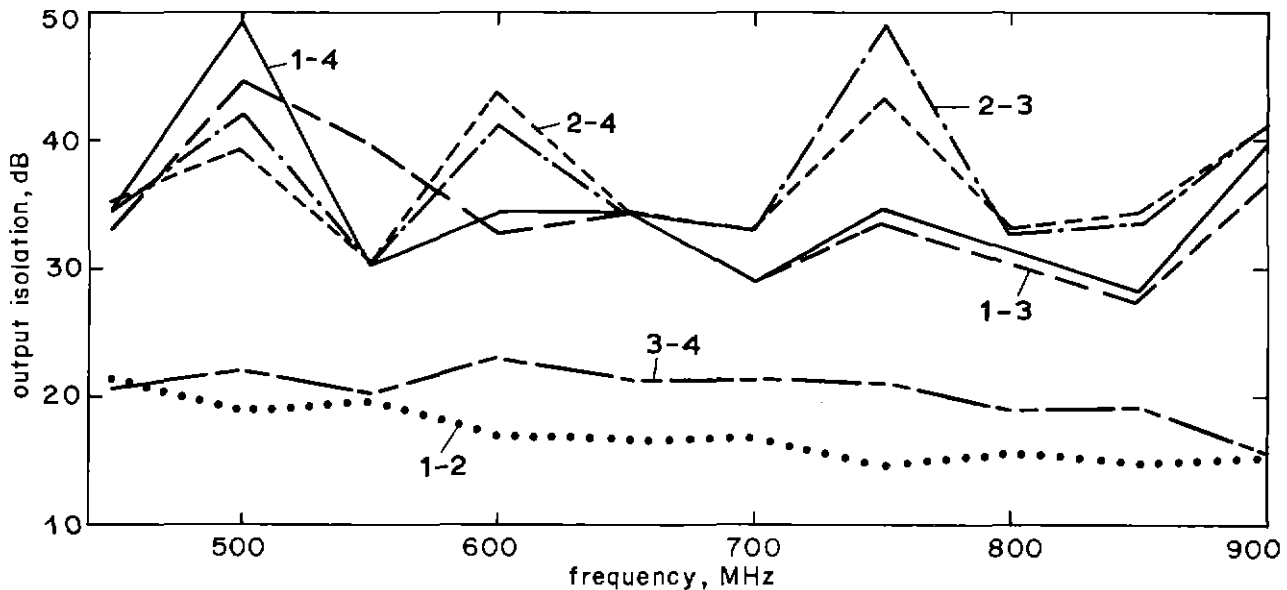It is convenient to consider the example of the RBL system

Fig. 11   Isolation between output ports of the active splitter using wireline hybrids
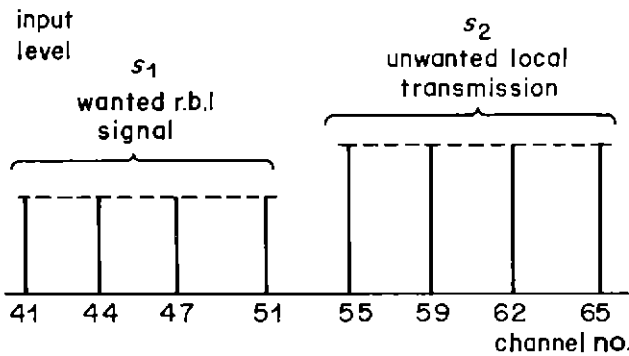
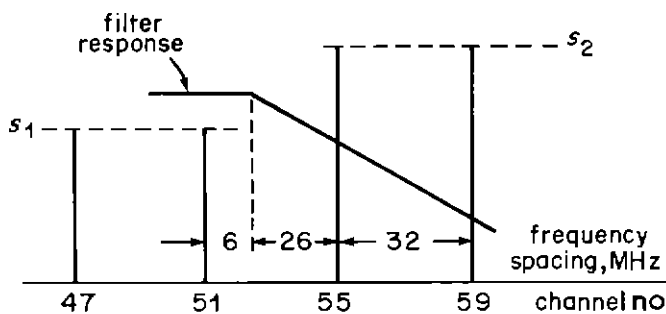**Fig. 12** The signal spectrum at the input of an RBL system

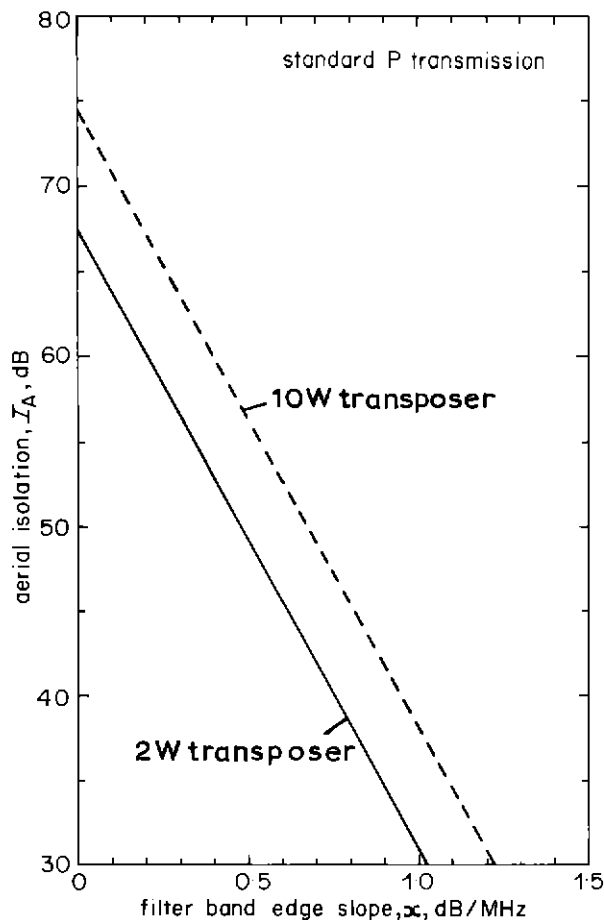**Fig. 13** Filter response at band-edge

Fig. 14 The interrelation of aerial isolation and mast-head filter response

described in section 5. Assuming that the loss associated with the main feeder and distribution transformers etc. is 4 dB, the transmitting aerial input levels are

for 2 W transposer: +29 dB(mW)
for 10 W transposer: +36 dB(mW).

If I_A is the isolation between the terminals of the transmitting and receiving aerials, the local transmission appears at the RBL aerial terminals at a level of

for 2 W transposer: $(29 - I_A)$ dB(mW) } (1)
for 10 W transposer: $(36 - I_A)$ dB(mW) }

The maximum permissible level of this unwanted input is set by the linearity considerations. One of the worst situations for a UHF standard channel group transposition is shown in figure 12. This shows the spectrum of vision carriers present at the receiving aerial terminals. The peak sync level of the wanted signal in a lower Band V channel group is represented by s_1. The peak sync level of the unwanted signal in an upper Band V channel group is represented by s_2. It is assumed that without a mast-head filter the local transmission may be a stronger signal than the wanted.

The worst channel IP in this situation is that in channel 51 at the vision carrier frequency which is produced by the vision carriers of channels 55 and 59 when they reach peak sync level. The level of this IP at the transposer input is given by s_{02} (see Equation (9) Appendix 4).

$$s_{02} = 3(s_2 + {}_{dB}G_a) - 2C \qquad \text{dB(mW)}$$

where ${}_{dB}G_a$ is the gain between the receiving aerial and the transposer input and C is the output intercept point.

The wanted signal level at the transposer input is

$(s_1 + {}_{dB}G_a)$. The relative IP level is therefore given by

$$s'_{02} = s_{02} - s_1 - {}_{dB}G_a \qquad \text{dB}$$
$$= 3s_2 - s_1 - 2(C - {}_{dB}G_a) \qquad \text{dB} \qquad (2)$$

and $(C - {}_{dB}G_a)$ is the input intercept point which, in this example, is −4·3 dB(mW).

The IP performance applicable to a standard P transmission is given as −48 dB in a three-tone test: this implies a single channel only. When more channels are amplified simultaneously in a common amplifier, the cross-modulation of the vision carriers is significant and the protection figure of −55 dB for co-channel interference is more appropriate.

From equation (2) the maximum unwanted input level is found by assuming that the lowest wanted input level is −54·4 dB(mW)

$$s_2(\text{max}) = [s_1 + 2(C - {}_{dB}G_a) + s'_{02}]/3$$

$$= -39 \cdot 3 \text{dB(mW)}. \qquad (3)$$

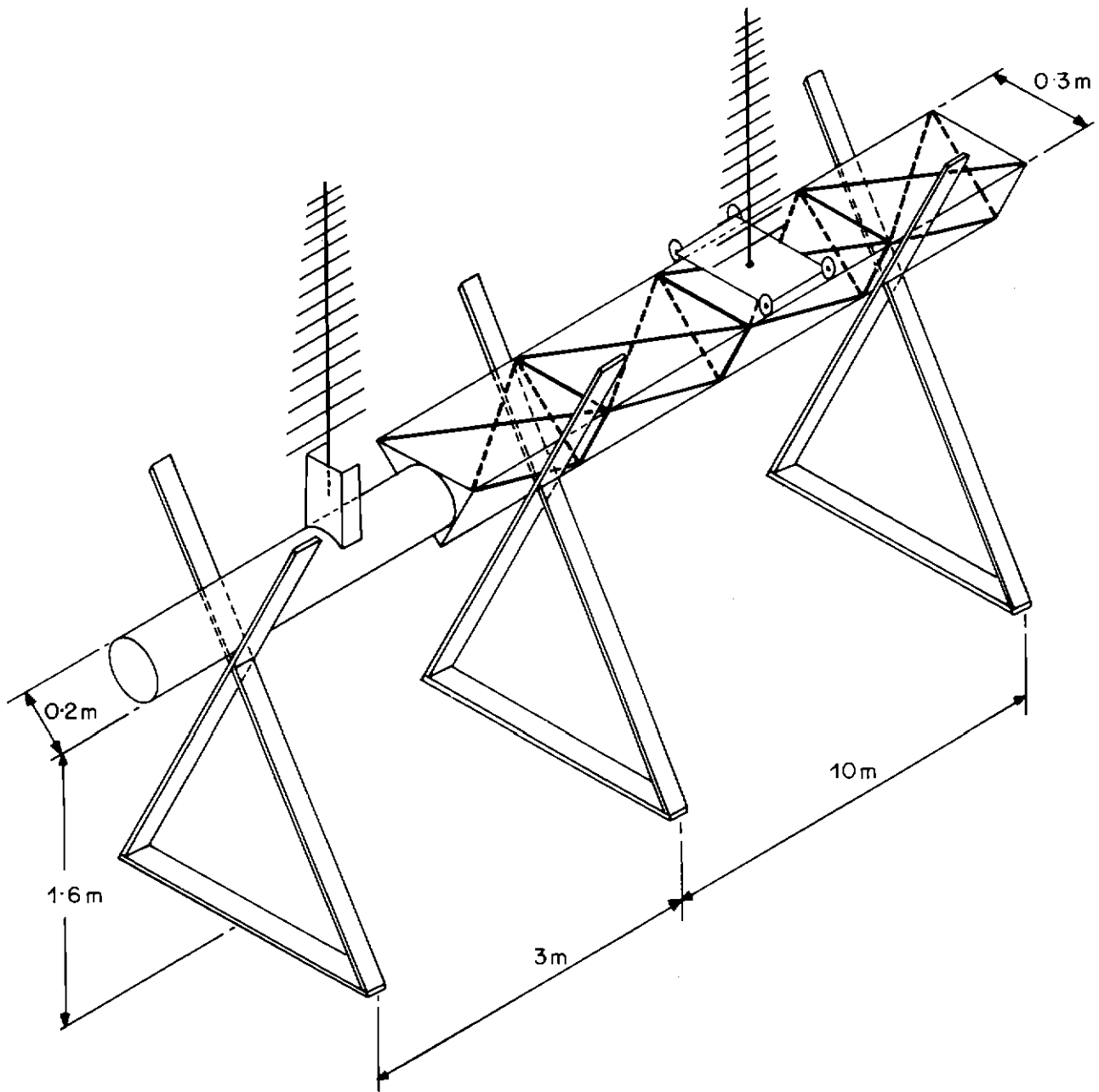Therefore, from (1) and (3), the minimum aerial isolation required is

Fig. 15 Measurement of aerial isolation

for 2W transposer: $29 + 39 \cdot 3 = 68 \cdot 3 \, \text{dB}$,

for 10W transposer: $36 + 39 \cdot 3 = 75 \cdot 3 \, \text{dB}$.

The addition of a mast-head filter allows the aerial isolation requirement to be relaxed. Consider the effect of a channel group filter such that the response at the band edge is as shown in figure 13.

The level of the unwanted signal at the RBL aerial terminals is reduced by the filter. If the slope of the filter response is x dB/MHz, the IP amplitude is reduced by a total of

$$(2 \times 26)x + 58x = 110x \qquad \text{dB}.$$

Assuming that the filter insertion loss is 1 dB, the unwanted signal level (i.e. the value of s_2 which would yield the same IP level) is given by:

for 2W transposer: $(28 - I_A) - \dfrac{110}{3} x \qquad \text{dB}$

for 10W transposer: $(35 - I_A) - \dfrac{110}{3} x \qquad \text{dB}.$

These can be equated to the maximum unwanted signal level given by (3). We have, therefore, a numerical relationship between the aerial isolation and the filter response.
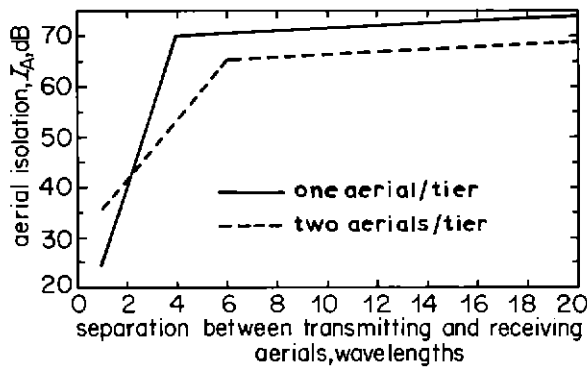
Fig. 16 The isolation between the receiving and transmitting aerials for relay stations

For 2W transposer:

$$(28-I_A) - \frac{110}{3}x = -39 \cdot 3 \text{ dB}$$

$$\therefore 3I_A + 110x = 202 \text{ dB}.$$

For 10W transposer:

$$(35-I_A) - \frac{110}{3}x = -39 \cdot 3 \text{ dB}$$

$$\therefore 3I_A + 110x = 223 \text{ dB}.$$

These relations are shown graphically in figure 14.

8 Measurements of aerial isolation

The aerials at low-power UHF stations will usually be log-periodics, although up to two panels or a simple cardioid aerial may be used instead. A series of experiments is described in this section to evaluate the aerial isolation which it is likely to be possible to achieve at these stations.

A simulated mast, consisting of a main triangular section of braced steel extended at one end by a circular section, of diameter similar to a cantilever pole, is supported horizontally about 1·6 m above the ground. The arrangement is illustrated in figure 15. The receiving aerials are mounted on a trolley which can be moved along the top face of the triangular section to vary the physical separation between the aerials; the separation is measured to the first transmitting aerial element. The isolation is determined by substituting a calibrated attenuator for the aerial path and maintaining the signal level constant.

Measurements were made initially at several frequencies in Bands IV and V and a periodic variation with frequency was noticed at a given separation. A similar variation is apparent as the separation is varied at a fixed frequency. Since it is the worst case which is of interest, all subsequent measurements were made at 700 MHz, a frequency which was affected little by local off-air transmissions.

Two templets indicating the minimum aerial isolation are

produced in figure 16. One is for simple arrays of log-periodic aerials, i.e. one, two, or four tiers of one aerial per tier in the transmitting case and one or two aerials in the receiving case. The second templet covers more complicated transmitting arrays with two aerials per tier. Measurements were made with the angle between transmitting and receiving aerials set at 0°, 45°, and 90° and with the receiving aerial set for either polarisation. A greater isolation would be expected if four aerials were used in the receiving array.

The results show that, for a large number of cases, the minimum aerial separation will be 3·8 m at channel 21 and 2·2 m at channel 68. These separations assume that there is no significant contribution to the isolation from the channel filter.

9 Conclusions and recommendations

There are two main advantages of using pre-amplifiers at low-power UHF relay stations where co-channel interference conditions allow.

a) the number of stations where only a single or double log-periodic receiving aerial is required may be increased;

b) a useful increase in the sensitivity of the receiving system may be achieved; thus, by using a mast-head pre-amplifier, the received field strength may be 9 dB lower than when a pre-amplifier is not used. The corresponding figure for a mast-base amplifier is 6 dB.

Some inexpensive amplifiers have a performance in respect of noise figure and linearity comparable with that of more expensive professional units. Their impedances are usually specified to be 75 Ω but an acceptable match to 50 Ω can be achieved in an Engelbrecht configuration. It is therefore recommended that the use of amplifiers of this type should be considered in the design of receiving arrangements at UHF relay stations.

Critical matching at the input terminals of the pre-amplifiers was found to achieve a marginal improvement only and is not considered to be a useful technique without amplifier re-design.

An active four-way splitter has been described which can replace the splitting filters which have been used to date. It may be possible to produce a cheaper version using micro-strip techniques.

A mast-head bandpass filter is desirable for protection from strong out-of-band signals. The protection required from the local transmissions is less stringent. The band-edge slope of the filter requires only a few sections to meet the specification.

Two templets of aerial isolation have been produced — one for simple transmitting arrays comprising a single aerial per tier and a second for more complicated arrays with two aerials per tier.

10 References

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Appendix 1

The power available from a receiving aerial

The equivalent circuit of a receiving aerial and load is shown in figure 17. Under matched conditions, the voltage developed across the load is given by:

$$_{dB} v_L = 20 \log \frac{\lambda}{2\pi} + _{dB}E + _{dB}G_A \qquad dB(\mu V)$$

where $_{dB}G_A$ is the aerial gain relative to $\frac{\lambda}{2}$ dipole, dB

$_{dB}E$ is the received field strength, $dB(\mu V/m)$

λ is the wavelength in metres.

The power in the matched load is numerically related to $_{dB}v_L$ by

$$_{dB}W_L = _{dB}v_L - 108 \cdot 6 \qquad dB(mW)$$

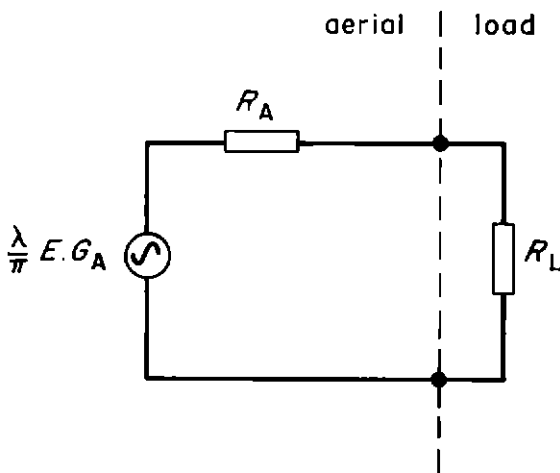**Fig. 17** Equivalent circuit of a receiving aerial and load

$$\therefore {}_{dB}W_L = {}_{dB}E + 20 \log \lambda + {}_{dB}G_A - 124 \cdot 6 \qquad dB(mW) \quad (4)$$

For the UHF range:

Freq. (MHz)	$20 \log \lambda$
470	-3·9
600	-6·0
650	-6·7
720	-7·6
860	-9·1

The net aerial gain including feeder loss, $_{dB}G_A$ may be assumed to take the following values for log-periodic (LP) arrays

1 LP	:	8 dB
2 LPs	:	11 dB
4 LPs	:	12·5 dB

Appendix 2

The signal-to-noise ratio of a receiving system

The unweighted signal-to-noise ratio of a base-band television signal is defined as

$$SNR = 20 \log_{10} \left(\frac{\text{peak-to-peak picture voltage}}{\text{RMS noise voltage}} \right) dB.$$

Assuming that the peak-to-peak picture voltage is 56% of the peak sync. carrier voltage (-5 dB) at the receiver input, the peak-to-peak picture voltage is given by

RMS level of carrier at syncs. $-5 + 3$ dB

$\therefore$ SNR at receiver input

$$= 20 \log_{10} \left(\frac{\text{RMS sync. carrier voltage}}{\text{RMS noise voltage}} \right) - 2 \, dB$$

$$= 10 \log_{10} \left(\frac{\text{sync. carrier power}}{\text{noise power}} \right) - 2 \, dB.$$

Also,

SNR at receiver output $=$ SNR at input $- _{dB}F_s - 6$

where $_{dB}F_s$ is the noise factor of the receiver and the reduction of 6 dB accounts for the receiver response.[9]

$\therefore$ SNR at receiver output

$$= 10 \log_{10}$$

$$\left(\frac{\text{sync. carrier power at input}}{\text{noise power at input}} \right) - _{dB}F_s - 8.$$

The sync. carrier power available at the input of a receiver matched to the aerial is given by (4).

The noise power associated with an aerial, assumed to be at a temperature, T, of 290 K in an effective bandwidth, B, of 5·08 MHz is given by

$$N = kTB = -106\cdot 9 \qquad \text{dB(mW)}$$

where k is Boltzmann's constant.

∴ SNR at receiver output =
$$_{dB}E + 20 \log \lambda + {}_{dB}G_A - {}_{dB}F_s - 25\cdot7 \text{ dB}.$$

Appendix 3

Relevant minimum transmission standards at the output of UHF relay stations

Impairment	Standard P	Standard Q
Signal-to-noise ratio	39 dB	33 dB
Attenuation of delayed images:		
$\leqslant 1\mu$s delay	28 dB	24· 5 dB
$>1\mu$s delay	30· 5 dB	28 dB
Intermodulation products (level of 1· 57 MHz IP in a 3-tone test)	−48 dB	−40 dB

Appendix 4

Amplifier linearity

Let the amplifier's transfer characteristic be represented by a third-order polynomial such that the output, v_o, is related to the input, v_i, by

$$v_o = \alpha v_i + \beta v_i^3$$

where α and β are constants.

When the input takes the form of two components of equal amplitude,

$$v_o = V_i(\cos\omega_r t + \cos\omega_s t),$$

the amplitude of the IPs, v_{o2} is $0\cdot 75\beta V_i^3$. The power level corresponding to this amplitude is

$$s_{o2} = -2\cdot 5 + {}_{dB}\beta + 3s_i \quad \text{dB(mW)} \qquad (5)$$

where s_i is the input power level.

The constant α represents the gain of the amplifier and is

hereafter written G_α or, in decibels, $_{dB}G_\alpha$. The following relationships can be deduced:

$$s_o = s_i + {}_{dB}G\alpha \quad \text{dB(mW)} \qquad (6)$$

while, at the intercept point,

$$s_o = s_{o2} = C \quad \text{dB(mW)}.$$

Substituting in (5)

$$C = -2\cdot 5 + {}_{dB}\beta + 3s_i, \quad \text{dB(mW)} \qquad (7)$$

and in (6)

$$C = s_i + {}_{dB}G_\alpha \qquad \text{dB(mW)}.$$

Substituting for s_i in (7) we get

$$C = {}_{dB}\beta - 2\cdot 5 + 3C - 3{}_{dB}G_\alpha \qquad \text{dB(mW)}$$

which, on rearranging, gives

$$_{dB}\beta = 3{}_{dB}G_\alpha + 2\cdot 5 - 2C \qquad \text{dB} \qquad (8)$$

When this value is substituted for $_{dB}\beta$ in (5) we get

$$
\begin{aligned}
s_{o2} &= -2\cdot 5 + 3{}_{dB}G_\alpha + 2\cdot 5 - 2C + 3s_i && \text{dB(mW)} \\
&= 3(s_i + {}_{dB}G_\alpha) - 2C && \text{dB(mW)} \\
&= 3s_o - 2C && \text{dB(mW).} \quad (9)
\end{aligned}
$$

If the input signal has three separate components of different amplitudes we may write

$$v_i = V_v\cos\omega_v t + V_c\cos\omega_c t + V_s\cos\omega_s t$$

and in a normal test procedure ω_v, ω_c, and ω_s are appropriately spaced to represent the vision, colour, and sound carriers and V_v, V_c, and V_s represent their amplitudes. The amplitude of the IP at $f_v + 1\cdot 57$ MHz is $1\cdot 5\beta V_c V_c V_s$
The power level of this IP is therefore

$$s_{o3} = 3\cdot 5 + {}_{dB}\beta + s_v + s_c + s_s \quad \text{dB(mW)}$$

where s_v, s_c, and s_s are the power levels of the vision, colour, and sound components respectively.

In the normal test the levels of the three components are given by

$$
\begin{aligned}
s_v &= s_i - 8 && \text{dB(mW)} \\
s_s &= s_i - 7 && \text{dB(mW)} \\
s_c &= s_i - 17 && \text{dB(mW)}
\end{aligned}
$$

where s_i is the input level corresponding to the tips of the synchronising pulses. Hence

$$s_{o3} = {}_{dB}\beta + 3s_i - 28\cdot 5 \qquad \text{dB(mW)}.$$

Substituting from (8) we get

$$s_{o3} = 3_{dB}G_\alpha + 2 \cdot 5 - 2C + 3s_i - 28 \cdot 5 \qquad dB(mW)$$
$$= 3(s_i + {}_{dB}G_\alpha) - 2C - 26 \qquad dB(mW)$$
$$= 3s_o - 2C - 26 \qquad dB(mW) \qquad (10)$$

Comparing with (9)

$$s_{o3} = s_{o2} - 26 \qquad dB(mW)$$

Only the relative level of the IP is important, defined as s'_{o3} where

$$s'_{o3} = s_{o3} - s_o \qquad dB.$$

Substituting from (10) gives

$$s'_{o3} = 2s_o - 2C - 26 \qquad dB$$
$$\therefore s_o = \tfrac{1}{2}s'_{o3} + C + 13 \qquad dB(mW).$$

This equation will give the maximum usable output level if the permissible level of IP is subsituted for s'_{o3}. The effect of additional channels has been assessed by computer and leads to the more general equation

$$s_{o(max)} = \tfrac{1}{2}s'_{o3} + C + Z \quad dB(mW) \qquad (11)$$

where Z is a constant depending on the number of channels. For one channel, $Z = 13$; for two channels, $Z = 4$; for four channels, assuming unequal spacing such as the common $3 - 3 - 4$ arrangement, $Z = 0$. The same equation gives $s_{i(max)}$ if C is replaced by the input intercept.

Notebook

Television Noise-Reduction System

The BBC has developed a digital signal-processing system for reducing the amount of transmitted video noise; it is the first such system to be successfully used with PAL colour. The prototype equipment was used recently over a period of ten days when about fourteen hours of live and recorded programmes were processed. They included a Silver Jubilee Concert from the Royal Albert Hall where the difficult lighting conditions led to rather noisy pictures which were greatly improved by the new technique.

Some formidable problems had to be overcome by BBC Research Department in developing the new system, which uses a television picture store in a recirculating mode, so that many successive television pictures are added together. The effect is to reduce noise by integration. The wanted picture detail, being present on every picture, is reinforced relative to the noise, which is random. Unfortunately, this technique cannot be applied to areas of the picture which contain rapid movement because integration of successive pictures would result in smeared images of moving objects. An additional problem is that the colour subcarrier would be reduced along with the noise because it is transmitted in a sequence of eight television fields.

Both of these problems have been successfully overcome and are the subject of patent applications.

The photograph shows the reduction in visible video noise that can be achieved; the top half of the picture has been processed by the noise-reduction equipment and is obviously improved relative to the bottom half. (A synchronised-shutter technique was used to take the photograph so that only two television fields were exposed.)

No serious problems have been encountered in the use of the prototype equipment, which requires manual adjustment to suit the noise-level on the pictures to be processed. Work is now in hand with the object of building an automatic system for full operational use. The system can easily be adapted to work with NTSC colour on 525 lines and it is likely that a version of the equipment will be designed that can be used with both PAL and NTSC.

## BBC's 1000th Domestic Transmitter Service
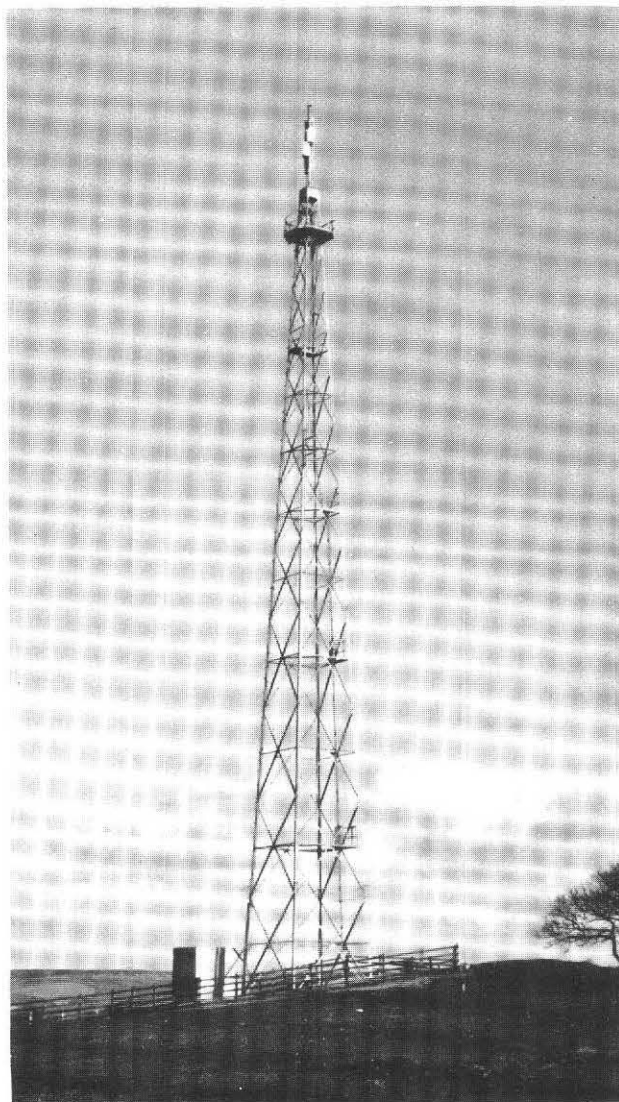
By the time of publication of this issue the BBC will have well over one thousand transmitters in operation for its domestic radio and television services. The 999th and 1000th started operating with the opening of the BBC1 and BBC2 colour television services from the Limber Hill (North Yorkshire) relay station on 15 April. Limber Hill, shown in the photograph, is one of 26 uhf relay stations opened in the seven months to the end of July, from Bressay in the Shetland Islands to Alderney (Channel Islands) some 1200km to the south.

One of the relay stations, at Grinton Lodge in Yorkshire, is similar to one of the types proposed for serving small communities, as described in BBC Engineering 106, April 1977, pages 24-29. A 17-metre wooden pole carries the receiving and transmitting aerials and the transmitter equipment is housed in a small metal box fixed to the pole.

625-Line Programme Link to Channel Islands

The Alderney relay station mentioned in the previous item serves the island of that name and is also the receiving point for the 625-line programme feed from the British mainland to all the uhf transmitters in the Channel Islands.
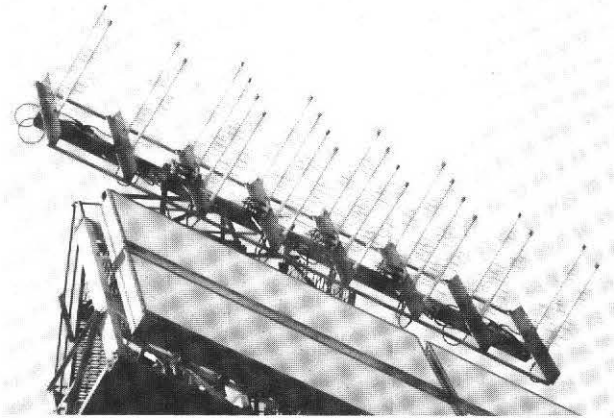
BBC engineers have designed and built a special diversity receiving system which is used at the relay station to pick up the BBC1 and BBC2 signals from the high-power uhf transmitter at Stockland Hill in Devon, 137km away. The system includes the array of 24 log-periodic aerials shown in the photograph, mounted at a height of 26 metres on the aerial tower and used in diversity with an IBA dish aerial near ground level. A diversity switch operates very rapidly during field blanking so that the minimum disturbance is caused to the television waveform.

Records made during the first few months of operation have shown that the log-periodic array is in use for much the greater part of the time and that signals of good quality are available to the Channel Islands uhf transmitters.

Quadraphony

As announced in the last issue, a series of quadraphonic broadcasts using BBC Matrix H encoding is now in progress. Among the broadcasts are all the Promenade Concerts from the Royal Albert Hall.

In order to provide an opportunity of quadraphonic listening (there being no suitable encoding equipment available commercially) a number of public listening sessions were arranged during the first week of the broadcasts. The public reaction to these was generally very favourable. Also listeners at home using stereo or mono welcomed the broadcasts and there were a large number of enquiries about the availability of Matrix H decoders. It seems likely that many listeners would buy such equipment if it were available.

Early in June it was announced that the BBC and the National Research Development Council (NRDC) had agreed to share their knowledge and experience of quadraphonic, or surround-sound, systems. The BBC's Matrix H and the NRDC's Ambisonic 45J, although developed independently, are similar in several important respects. The objective of the co-operation will be the derivation of the optimum unified coding specification for a system which can be used with tape and disc recording as well as broadcasting.

WARC 1979

'In September 1979 a World Administrative Radio Conference (WARC) will convene in Geneva under the auspices of the International Telecommunication Union for the purpose of revising, where necessary, the international Radio Regulations, a process which was last carried out by a similar conference almost twenty years ago. Without doubt, the most important task of the Conference will be to amend the international Frequency Allocation Table (Article 5 of the Radio Regulations) to take account of the changing frequency requirements of the various radiocommunication services and to meet any additional frequency requirements likely to arise in those services up to the end of this century.'

The above paragraph begins the Introduction to a booklet in which are set out the proposals which the BBC has submitted to the Home Office in respect of broadcasting frequency allocations. In the United Kingdom the Home Office is the Government department responsible for radio regulatory matters and so for preparing the UK proposals to WARC.

Before preparing its submission the BBC carried out an extensive appraisal of its frequency requirements up to the end of this century, not only for its domestic radio and television and external broadcasting, but also for the associated back-up services.
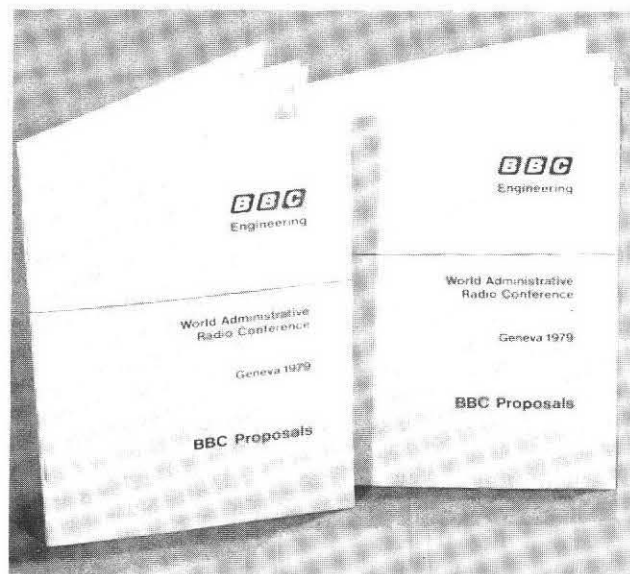
The booklet is available free of charge from BBC Engineering Information Department, Broadcasting House, London W1A 1AA, England.

ENG

At a press conference on 13 July, the BBC's Editor, Television News, Andrew Todd, gave some details of the BBC's plans to be the first organisation in Britain to use ENG — Electronic News Gathering. A one-year experiment will be carried out using a specially-equipped Range Rover with a two-man crew drawn from Television News film and engineering staff.

The equipment will include a shoulder-mounted lightweight camera, portable cassette video recorder, radio links and radio-telephone communications. Camera pictures can be recorded on the spot by the cassette recorder or can be transmitted live via the radio link(s) back to Television Centre. One of the radio links is a portable short-range unit which can be carried to the news site with the camera, which it effectively converts into a radio camera, the signals being passed back to the vehicle for relaying via the second link and a base station to Television Centre. Initially one base station on top of a skyscraper block near central London will be used but other sites have been successfully tested and may be used subsequently.

At Television Centre a special ENG video tape recording and editing suite is being set up.

It is hoped that a fuller account of the ENG experiment will be published in a later issue of *BBC Engineering*.

Research Department Reports

The following BBC Research Department Reports have been published since our last issue.

1977/13	Two colour-television tubes with precision in-line-gun assemblies.
1977/14	Permissible timing jitter in broadcast PAL colour-television signals.
1977/15	CEEFAX: optimum transmitted pulse-shape.
1977/16	M.F. lattice planning on a spherical earth.
1977/17	Correction of primary chromaticities of display tubes by matrixing of gamma-corrected RGB signals.
1977/18	CEEFAX: field trials at v.h.f. using System B — Bavaria — April 1975.
1977/19	Film lighting using metal-halide lamps: some operating conditions giving freedom from picture luminance fluctuations.
1977/20	BBC Test Card No. 61 (flesh tone reference): colorimetric and other optical considerations.
1977/21	Digital video: multiple sub-Nyquist coding.
1977/22	An aerial diversity switch for the Channel Islands television re-broadcast link.

A subscription to BBC Research Reports, of which about 35 are published each year, costs £25.00. Further information and subscripton forms are available from:

Research Executive
BBC Research Department
Kingswood Warren
TADWORTH
Surrey, England.

New Low-Noise 15dB Video Amplifier
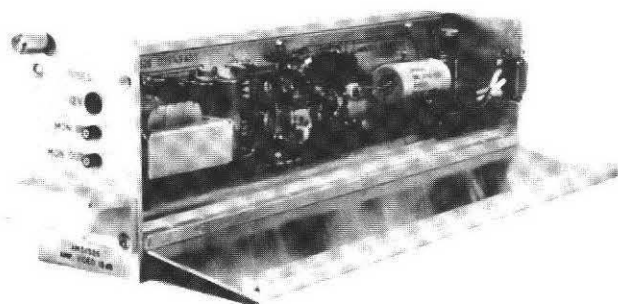
Among recent BBC designs is this video amplifier (AM5/526) which is intended for use at the receiving end of coaxial cables. Its low noise makes it particularly suitable for use as the first amplifier in a system. The amplifier has a gain of 15dB and will provide a composite 1V signal output. Brief details of its specification are as follows.

Maximum input level	−15dB composite video
Input impedance	75Ω (return loss >30dB)
Output level for input level of −15dB 1)	0dB ±0.1dB
2)	−15dB ±0.3dB
Output impedances	75Ω (return loss >30dB)
Gain	15dB ±0.1dB
Frequency response	50Hz to 6MHz ±0.1dB
Noise factor	<7dB (typically 6dB)
50Hz bar tilt	<0.5%K
Differential gain	<0.1%
Differential phase	<0.05°
Chrominance/Luminance gain inequality	<1%
Chrominance/Luminance delay inequality	<2ns
Chrominance/Luminance crosstalk	<1%
Power requirement	200 to 270V 10mA a.c. or ±15V 160mA d.c.

The above equipment is typical of many items of BBC design which are available commercially from licensed manufacturers or are available for such licence arrangements to approved UK manufacturers. Further information is available from: Liaison Engineer, BBC Designs Department, Broadcasting House, London W1A 1AA.

Contributors to this issue

Stan Davies joined the BBC as a Youth-in-Training at Shrewsbury Transmitting Station in 1942 and moved to Tatsfield Receiving Station the following year. He was called up into the Royal Navy in 1944 and rejoined the BBC in 1948 at Tatsfield.

In 1952 he transferred to Studios at 200 Oxford Street and then joined Studio Capital Projects Department in 1957. In 1963 he went to Malaysia on secondment to advise Radio Malaysia. He was responsible for setting up five regional studio centres and a receiving station.

He returned to the BBC in 1967 to SCPD and moved into Radio Projects in 1971 where he is now Senior Projects Engineer.

M. M. Gleave graduated in Electrical Engineering at the University of Wales in 1967 and joined the BBC Designs Department that year. Since then his work has mainly been involved with the design of sound studio equipment for both domestic and external broadcasting and with the design of limiter/compressors for studio use and transmitter protection.

John Levett joined the BBC in 1969 direct from University as a graduate trainee in Transmitter Capital Projects Department, where he was involved with the commissioning of UHF relay stations. He moved to his present post of Systems Engineer in Links Unit of Transmitter Capital Projects Department in 1970 where he is primarily concerned with the planning, installation, and commissioning of radio links and associated equipment. He has recently been responsible for the supply of a Radio Outside Broadcast Communication Vehicle and a number of radio telephone systems.
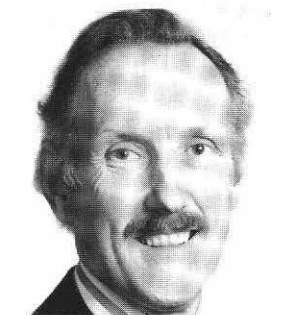
Duncan MacEwan has been Chief Engineer, Radio Broadcasting, since April 1971. He was educated at Ayr Academy and Glasgow University, where he graduated in mathematics and radio-physics in 1944. After commissioned service in the Royal Signals which included two years in the Far East, he joined the BBC in 1947 and worked on radio in Glasgow and Edinburgh until early 1953, when he transferred to television. He continued spare-time work for the Army Emergency Reserve and the Territorial Army, working on special communication systems, until 1966. After leaving Scotland in 1956, he worked in both radio and television, first as a senior lecturer in the Engineering Training Department and afterwards in senior posts in the Regions, being latterly Head of Engineering, Northern Ireland. In 1968 he moved to London to become the engineering member of the BBC Policy Study Group, whose work resulted in the paper 'Broadcasting in the Seventies', the plans for which have since been implemented. From August 1969 until he took up his present appointment, he was Chief Engineer, Regions.

W. I. Manson graduated at the University of Aberdeen in 1947 and joined the BBC in the same year. After a short time at studio centres he joined Research Department in 1948, working on the development and testing of microphones and loudspeakers.

In the late fifties and early sixties he was concerned with investigations of studio techniques and sound-signal system tolerances for stereophonic radio broadcasting.

More recently he has been concerned with analogue sound-signal processing, particularly limiters.

John Riley graduated in Electrical Engineering at Leeds University and a year later took an MSc. in microelectronics at Southampton University. After a two-year teaching appointment overseas, he joined Research Department in 1970 and has spent most of his time with Transmitters and Propagation Section of Radio Frequency Group. He has been chiefly concerned with UHF aerial design and the development of the UHF relay station programme.