

**STUDIO ENGINEERING**  
**FOR**  
**SOUND BROADCASTING**

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SOUND BROADCASTING**



*IN THIS SERIES*

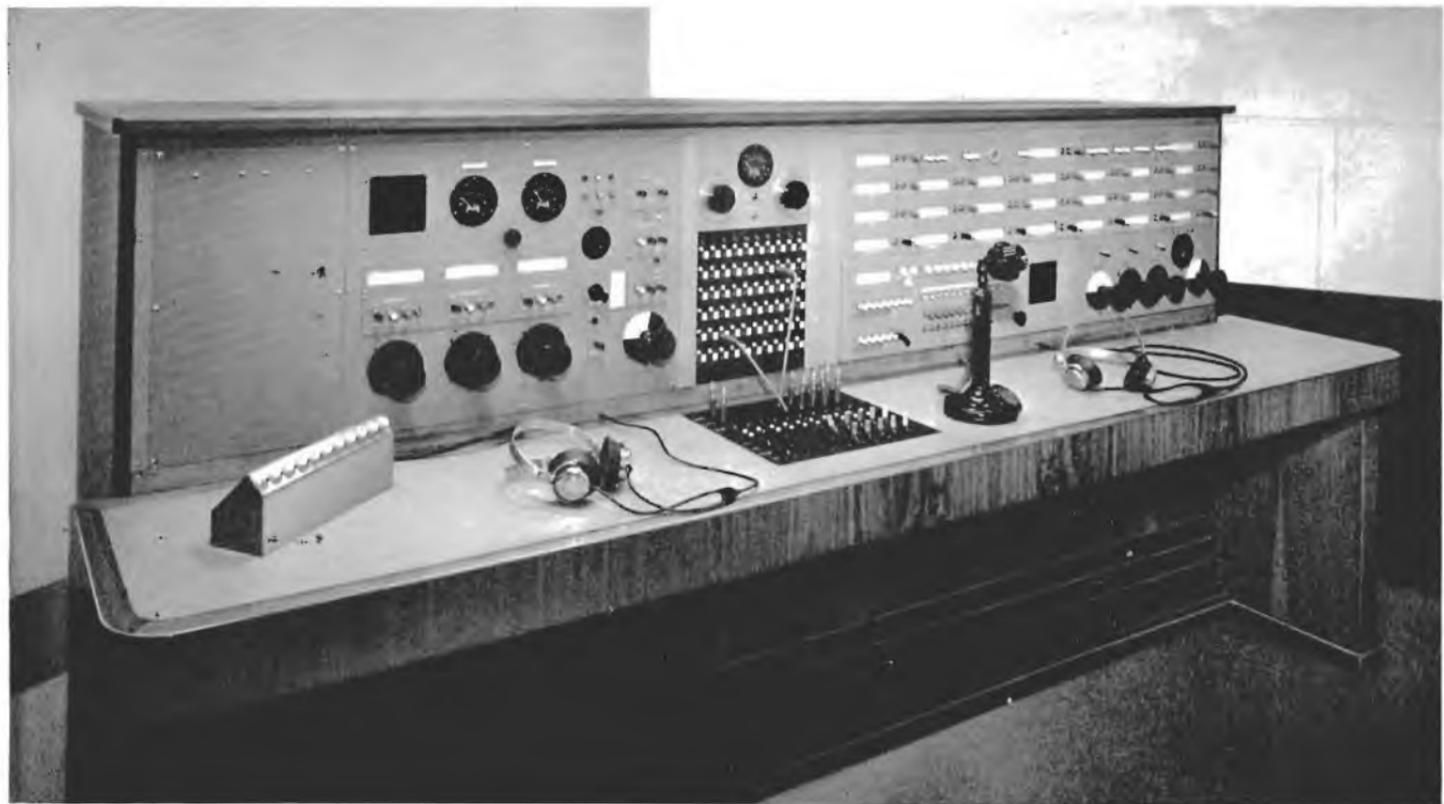
**MICROPHONES** (Staff of the BBC Engineering Training Department)

**SOUND RECORDING AND REPRODUCTION** (J. W. Godfrey and S. W. Amos, B.Sc. (Hons.), A.M.I.E.E.)

**TELEVISION ENGINEERING PRINCIPLES AND PRACTICE** (S. W. Amos, B.Sc. (Hons.), A.M.I.E.E. and D. C. Birkinshaw, M.B.E., M.A., M.I.E.E.)

**VOL. I: FUNDAMENTALS, CAMERA TUBES, TELEVISION OPTICS, ELECTRON OPTICS**

**VOL. II: VISION-FREQUENCY AMPLIFIERS** (In preparation)



CONTROL DESK, BRISTOL (1954)

BBC ENGINEERING TRAINING MANUALS

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STUDIO ENGINEERING  
FOR  
SOUND BROADCASTING

By Members of the BBC Engineering Division

GENERAL EDITOR: J. W. GODFREY

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## PREFACE

THIS manual has been produced for the purpose of training technical staff in the general principles underlying the operational procedure at BBC studio centres. Some of the information is specific in that it relates to equipment designed to meet BBC requirements. In presenting this particular information, it has been necessary to trace the history of studio-equipment design, based on pre-war, war-time and post-war periods. It will be appreciated that this has involved many references to equipment and to procedures which have become or are becoming obsolescent, but which may remain in being for some time to come.

An attempt has been made to supplement the above information by principles of audio-frequency engineering which have a more general application, such as Transmission Quantities, Chapter 2; Acoustics and Microphone Placing, Chapter 3; Programme Circuits on Post Office Lines, Chapter 7; Communication Circuits for telephony and telegraphy, Chapter 9. The BBC is aware that these subjects are of wide interest and for this reason the manual is being published as a contribution to the training of broadcasting engineers in all parts of the world.

The authors of the manual are specialists, each with many years of experience in his respective field, and no effort has been spared to ensure technical accuracy, combined with clear presentation. The technical level is practical rather than academic, and should present no difficulty to the readers for whom it is intended. In this connection, it should be emphasised that the manual is intended for both engineering and non-engineering staff at studio centres.



## CHAPTER 1

# DEVELOPMENT OF THE BROADCASTING CHAIN

By F. C. BROOKER, A.M.I.E.E.

*Operations and Maintenance Department*

### 1.1 INTRODUCTION

IN modern broadcasting, the transmission of information from its source to the listener involves a chain of acoustic and electrical equipment which varies in complexity according to the size of the broadcasting organisation and the character of the information to be broadcast. For example the broadcasting of a talk by a single transmitter can be effectively carried out from a small building, containing a simple studio equipped with a microphone and its associated amplifier, feeding direct into the transmitter. In contrast, a feature programme, involving many sound sources, will require a number of microphones and amplifiers, adequate mixing, control, and switching equipment, and intercommunication between sources; if such a programme is to be broadcast simultaneously by a number of transmitters at different geographical locations, telephone cables or radio links will be needed to convey the programme from its sources to the transmitting stations and to provide inter-station telephone communication.

For convenience, we shall refer to the information to be broadcast as *programme* whether it be a talk on "How to grow Cabbages" or a Brahms symphony played by a full orchestra.

It should be understood, however, that different types of programme present different problems related both to acoustic requirements and to electrical equipment. With all types of programme, it is essential that aural check on quality should be available at various points along the path of transmission. It is also necessary to ensure that the volume of the programme is kept within certain bounds, fixed by the power-handling capacity of the apparatus used on the one hand, and by the necessity to mask unwanted noise on the other. The combined process of aural and visual checking is called *monitoring*.

It will now be seen that the transmission of a broadcast programme from source to listener involves the use of studios, microphones, amplifiers, control units, telephone cables, transmitters and checking facilities, in addition to switching apparatus for selecting programme

sources and destinations. These things combine in a logical sequence to form what is known as the *broadcasting chain*.

## 1.2 DRAWING CONVENTIONS

Before we attempt to describe the broadcasting chain in detail, it is important that the reader should be familiar with certain conventions used in block-schematic and circuit drawings relating to broadcasting apparatus.

### 1.2.1. Block-schematic Drawings

This type of drawing is used to indicate the sequence in which various units of apparatus are linked together; it provides an easy means of tracing the path of a programme from one point to another, when consideration of detail is not essential. Each unit is drawn in geometrical form, for example a rectangle or a circle, inscribed with an identifying legend or symbol, according to specifications recommended by the British Standards Institution and embodied in the BBC Drawing Office Practice handbook.

Connections between the units are shown as single lines, though normally representing two conductors; when it is necessary to show greater detail, the diagram may be opened up into the conventional two-wire circuit, as illustrated in Fig. 1, where 11 and 12 represent

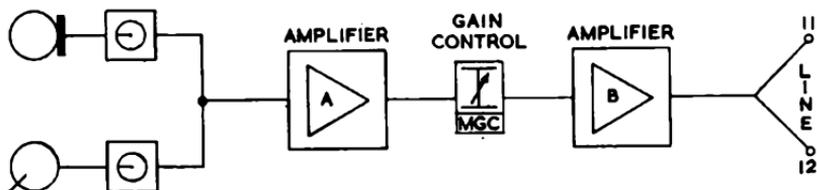


FIG. 1—Block schematic diagram of simple chain

the terminals of a Post Office line. In practice the symbols are self-explanatory and bear no legend; a comprehensive list appears in the BBC Drawing Office Practice handbook.

It is sometimes desirable to include more information on a diagram than that conveyed by block symbols, for example, the *type* of microphone or fader. This can be done by using *circuit* instead of block symbols as in Fig. 2. Comparing the two figures, the first merely shows a microphone and pickup fed into an amplifier through their respective faders, whereas the second indicates a moving-coil microphone and a moving-iron pickup connected to the amplifier through balanced series faders, with a series-shunt gain control between amplifiers. In Figs. 1 and 2, the direction of amplification is indicated by the apex of the triangle. In Fig. 2, the num-

bers represent the programme volume at specific points, referred to a standard called *zero volume* (p. 166) and expressed in decibels; thus amplifier A raises the volume from  $-70$  to  $-10$ , hence its working gain is 60 decibels.

### 1.2.2. Circuit Drawings

A circuit drawing shows the theoretical as distinct from the mechanical construction of an individual unit such as an amplifier, Fig. 3. All the symbols used on this drawing conform to the BBC

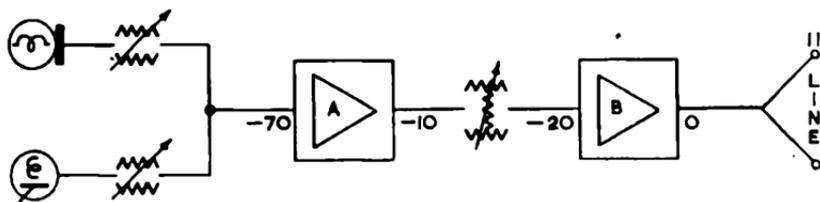


FIG. 2—Block schematic diagram using circuit symbols

Drawing Office Practice handbook, and are based on British Standard 530: 1948, Graphical Symbols for Telecommunications.

The following points are of interest and apply in general to all circuit drawings:

- (a) The sequence of transmission is from left to right, i.e., the input terminals are at the extreme left and output terminals at the extreme right of the diagram.
- (b) No connection is allowed to cross another connection at a junction; the crossing connection is staggered as in the cathode circuit of *V1*. The reason for this is to maintain correct circuitry even if the junction dot is inadvertently omitted.
- (c) Component values are expressed in basic units or their derivatives in a specific manner thus: resistances below 1,000 ohms are given in figures only as in *R11*, *R12*; if 1,000 ohms or above, they are given in kilohms as in *R3*, *R4*. Capacitances are indicated in picofarads or microfarads as follows: up to 999 picofarad, the number followed by *p*, e.g., 50 *p*; above 999 picofarad, the number of microfarads only, e.g., 16; or 0.05. Note that the decimal point must be preceded by 0 which is spaced out from the other figures to prevent misinterpretation if the decimal point is inadvertently omitted. Similarly for inductances up to 999 microhenrys, the number of microhenrys followed by  $\mu$ , e.g., 100  $\mu$ , 1 to 999 millihenrys by *m*, e.g., 85 *m*; 1 henry or above, by the number only.

A B C D E F G H J K L M N O P Q R S

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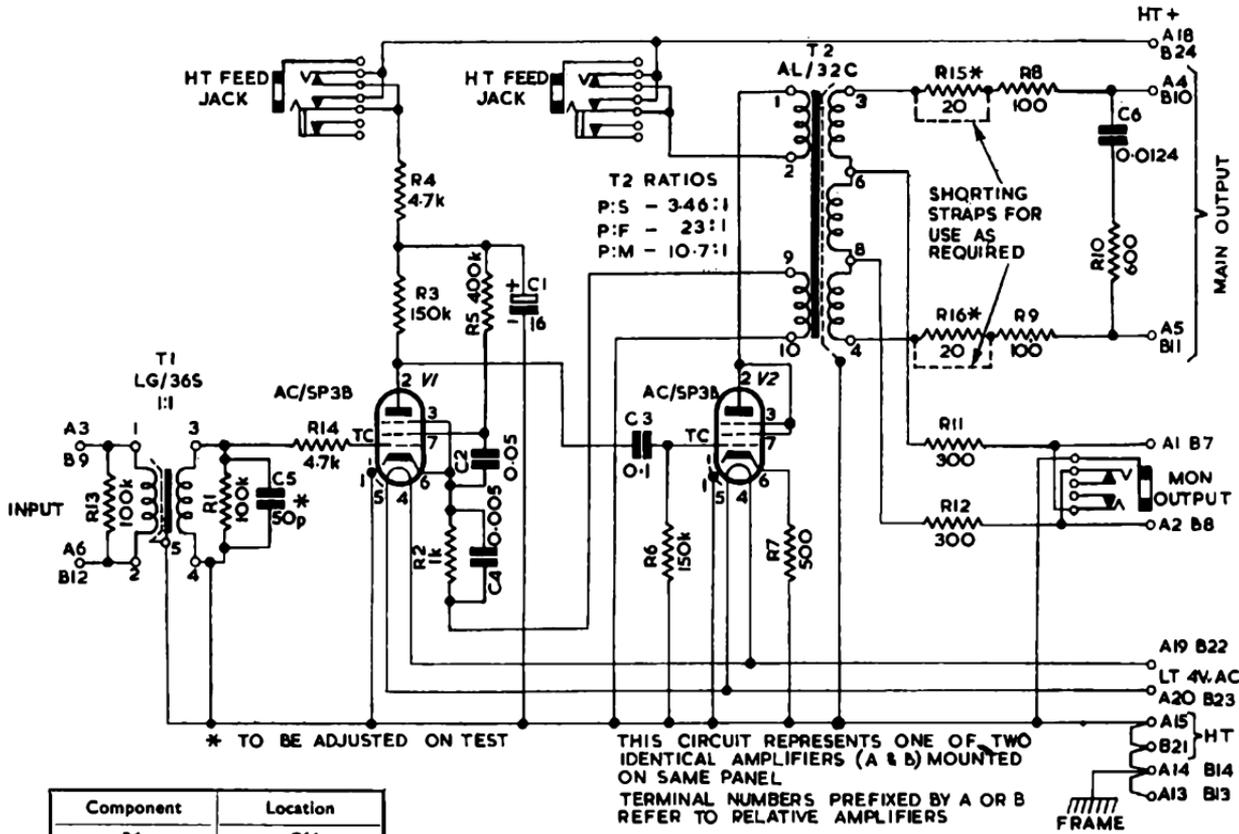


FIG. 3—Typical circuit drawing. The specimen location table indicates use of map reference

## DEVELOPMENT OF THE BROADCASTING CHAIN

- (d) On large circuit diagrams, it is often difficult to locate components referred to in the text of technical instructions. To meet this problem, components are tabulated in numerical order and the diagram surrounded by map references, the use of which is self-explanatory.

### 1.3 SIMPLE BROADCASTING CHAIN

The complete broadcasting chain carrying programme from its source to the listener includes the transmitting station and the listener's receiver; since this manual is intended for engineering personnel employed at studio centres, the chain will be confined to the links between the sound sources and the transmitter. The whole of the apparatus needed for a small broadcasting station serving a limited area could be housed in a single office; the minimum needs would be a microphone and a gramophone with independent fade-units, an audio-frequency amplifier with a volume control (or two

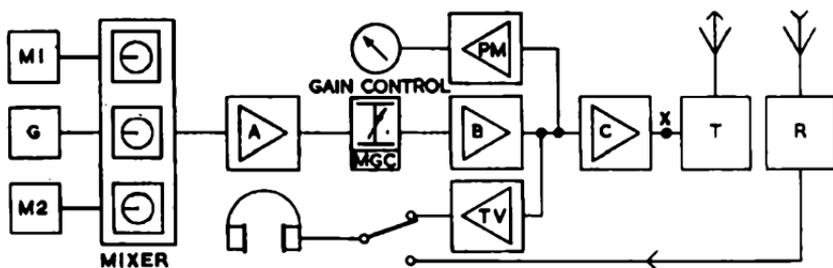


FIG. 4—Chain for self-contained station

such amplifiers separated by a control potentiometer), a low-power transmitter and simple monitoring equipment. The chain for such a station is shown in Fig. 4. It will be observed that the headphones can be switched from the amplifier output to a check receiver; thus the programme radiated by the transmitter can be compared with the programme fed into it. This practice of checking the programme at different points in the chain is widely used and is called *comprehensive checking*.

### 1.4 AMPLIFIER SEQUENCE IN A BROADCASTING CHAIN

In large organisations such as the BBC the broadcasting chain is more complicated than the example shown in the previous section. This is because a large number of programme sources must be available and distributed to numerous transmitters. For this purpose, a number of amplifiers, each performing a specific function is necessary at studio centres. In BBC practice, these are given functional

designations of A, B, C, D, corresponding to their location in the chain.

The A amplifier follows the microphone, pickup, or other electrical source. In certain circumstances, fade-units are interposed between the source and the A amplifier whilst in others the fade-units follow the A amplifier. The output of the A amplifier whether preceded or followed by fade-units is normally connected to a control potentiometer used for adjusting the programme volume; this control potentiometer introduces loss into the chain and is therefore followed by a B or control amplifier.

From the output of the B amplifier it may be necessary to distribute the programme to a number of different destinations via Post Office lines. It will be seen later that it is not convenient to connect a

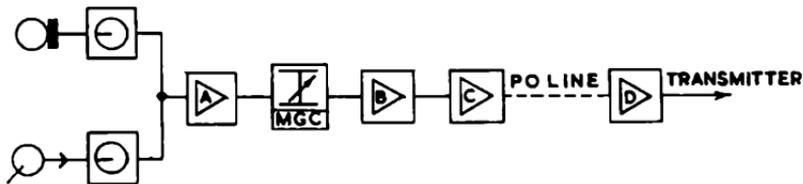


FIG. 5—Chain for single programme

large number of lines direct to the output of the B amplifier; it is therefore necessary to introduce a C or distribution amplifier for each destination.

The distant end of the Post Office line may be another studio centre containing terminations for many such lines; there will usually be some loss in programme volume on the line itself; for this reason it is necessary to terminate each line with a D or receiving amplifier.

Fig. 5 shows the chain for a single programme using amplifiers in the order stated above.

### 1.5 MULTIPLE PROGRAMME SOURCES AND DESTINATIONS

A national broadcasting service has to provide its listeners with many different types of programme every day; this means that a number of sources must be available so that each programme has studio facilities suitable for its composition, and that one programme may be prepared or rehearsed whilst another is being broadcast. For example, a feature programme may have to draw on many sources distributed over the country; to knit these sources into a pattern, the sources must be available some time before the broadcast, and adequate mixing, cueing and communication facilities must be centralised at a control point; there may indeed be more than one control point because each outside broadcast location may use a number of sources. Nevertheless, there will be a final control point,

## DEVELOPMENT OF THE BROADCASTING CHAIN

from which the programme is distributed as a whole to the transmitters from which it is to be radiated.

The main sources can be classified as follows:

- (a) Studios with microphones and/or gramophones
- (b) National Outside Broadcasts, using Post Office lines
- (c) International Outside Broadcasts, using radio links
- (d) Sound Reproduction Rooms, using magnetic or film recordings.

Technical arrangements for each programme are under the control of a studio centre. This centre must be equipped with adequate source-selection apparatus, with equipment for controlling and monitoring, and for distribution to specified destinations.

The main destinations are:

- (i) Transmitters in the country of origin, via Post Office lines
- (ii) Transmitters in other countries, involving cables or radio links
- (iii) Recording rooms or centres
- (iv) Listening rooms or private offices at the studio centre.

### 1.6 BROADCASTING CHAIN FOR A STUDIO CENTRE

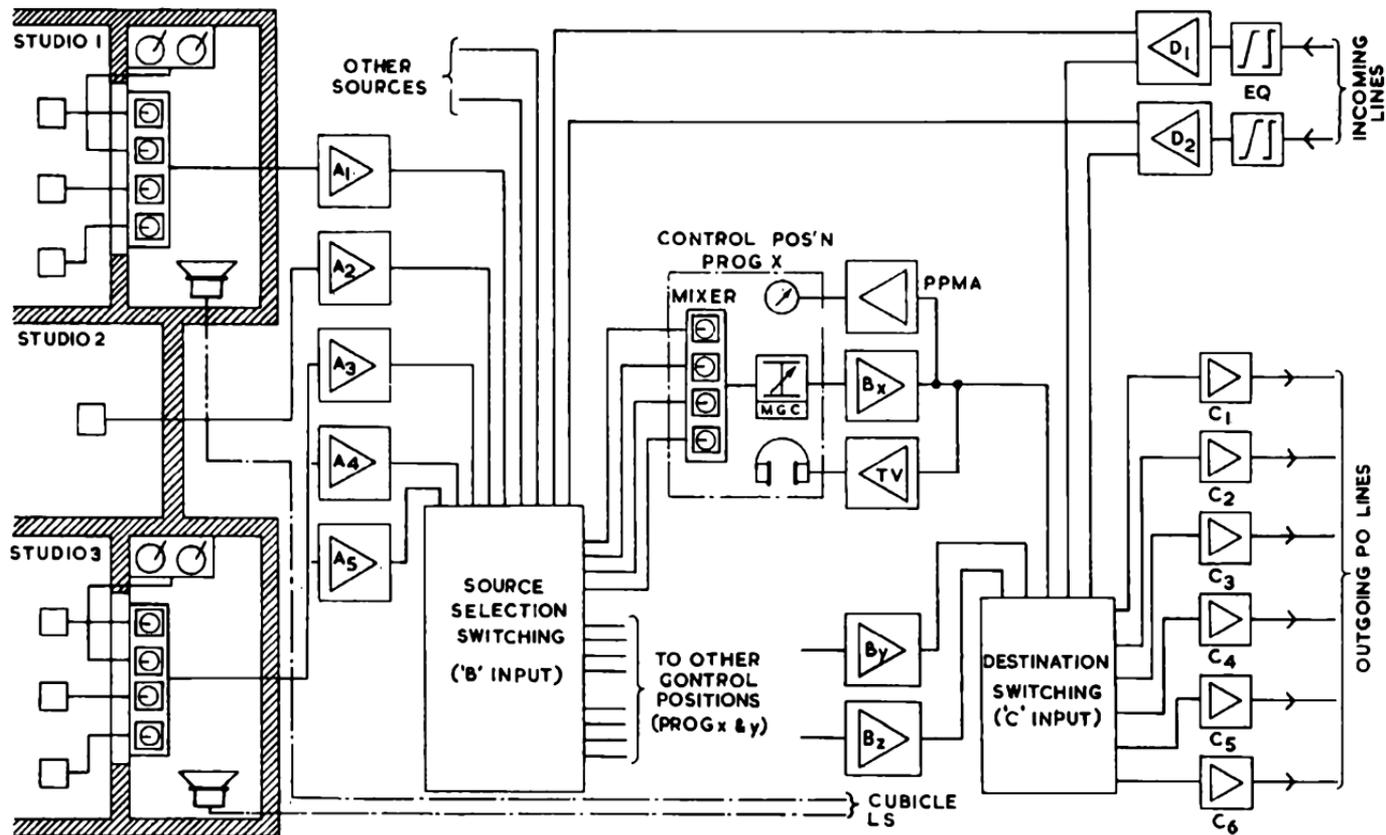
A comprehensive chain incorporating the facilities discussed in the preceding section is shown in Fig. 6, which is a hypothetical arrangement at a main studio centre. A study of this figure in detail will explain the principle of the complex broadcasting chain, though it must be borne in mind that in practice the construction of the chain varies for different requirements and also that modifications take place from time to time with progress in design.

Three studios are shown in Fig. 6, Nos. 1 and 3 being identical multi-purpose studios and No. 2 a talks studio with a single microphone. The mixers in studios 1 and 3 precede the amplifier; this is called *low-level* mixing. Modern practice tends towards *high-level* mixing, a small amplifier preceding each fade-unit. The advantages and disadvantages of the two methods are discussed in Chapter 5.

The outputs of the various A amplifiers are taken to source-selection equipment which can be in the form of plugs and jacks, remotely-operated relays, or uniselectors. The BBC, in the course of time, has used all three methods, in the order stated.

The next point in the chain is the control position, which may be located in a central control room or in a room adjacent to a studio (Chapter 4). A control position is allocated to each programme service (e.g., Home, Light, Third). From this position the source required at a given time is terminated on the mixer panel. The output of the mixer panel is connected to a control potentiometer, used to

FIG. 6—Chain for studio centre



adjust programme volume, thence to a B or control amplifier, the output of which goes to the destination switching panel. The output of this amplifier also goes to the aural and visual monitoring circuits, shown in the figure as headphones and programme meter, each of which has a separate amplifier.

As with source selection, the destination switching panel may use plugs and jacks, relays, or uniselectors (or a combination of these).

In the figure, the programme associated with the control position shown is designated x. At the destination switching panel, programme x is switched to a number of Post Office lines, according to the number of transmitting stations or other studio centres taking this particular programme. To prevent these lines from upsetting the working conditions of the B amplifier (the lines may have varying impedances) and to maintain the correct sending volume to each line, a C or sending amplifier is inserted between the switching panel and the line.

It will be observed that two other programmes, y and z, are assumed to originate at the same studio centre, each of which has its own control and selection equipment. Where this obtains, there is a separate control position for each programme, but the selection equipment for the three (or more) programmes may be centralised. It sometimes happens that sources other than those originating at the studio centre are needed for a given programme. These sources are connected to the centre by Post Office lines; to counteract the *attenuation* or *loss* in the incoming line, a D amplifier is inserted between the line and the selection equipment. If, as is usual, the loss is greater at some audio frequencies than at others, an *equaliser* is inserted between the line and the amplifier. This equaliser is so designed that it introduces small loss at those frequencies which are attenuated by the line, and greater loss at those which are not; thus at the output of the equaliser, the loss is equal at all frequencies. (Chapter 4.)

In Fig. 6 the output of the D amplifier is shown connected to both source and destination switching. The reason for this is that it is sometimes necessary for a programme to be routed through a studio centre when that centre is not using the programme in its own services; in such circumstances, the output of the D amplifier is connected to a C amplifier associated with the line to the appropriate destination.

So far, our attention has been focused mainly on the broadcasting chain at a studio centre. The complete chain must include transmission lines and the transmitter itself. (It can be argued that the chain must also include the listener's receiver but fortunately this is not our concern.)

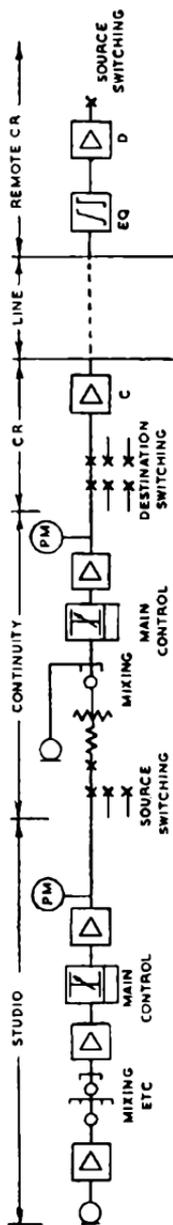


FIG. 7.—Broadcasting chain from source to destination

The skeleton of a broadcasting chain between source and destination is shown in Fig. 7. Communication and monitoring circuits have been omitted deliberately for purposes of simplification.\*

This brief survey of the broadcasting chain serves as an introduction to engineering practice at BBC studio centres. Readers new to the subject will be well advised to master this chapter thoroughly before proceeding to study the detailed information given in subsequent chapters.

\*A detailed explanation of this chain is given in Chapter 6, p. 121.

## CHAPTER 2

### TRANSMISSION QUANTITIES

*By R. O. ROWLANDS, M.Sc., A.M.I.E.E.*

*Engineering Training Department*

#### 2.1 MEASUREMENT OF LEVEL

In any programme chain the volume of the signal varies from point to point. In the studio a very weak signal is generated at the microphone and fed through its associated transformer into an amplifier in the control cubicle. It then passes through various units of equipment including a control potentiometer; as the net result is a drop in volume all these are represented in Fig. 8 by an attenuator. It is then

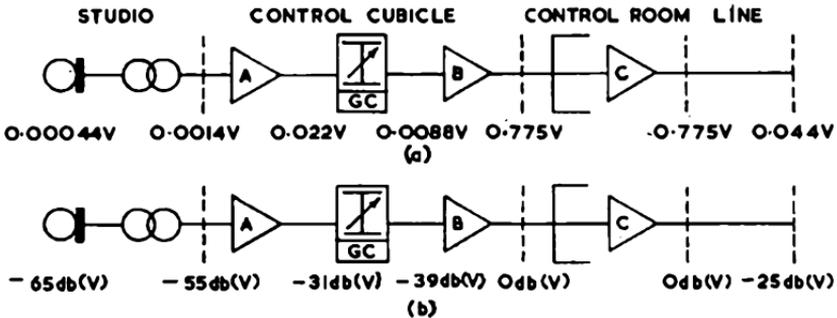


FIG. 8—Simple broadcasting chain

amplified again before being sent on to the control room. Here it is distributed to various destinations, one being via a distribution amplifier into a Post Office line which links the control room to a transmitting station.

Let us now consider in more detail what happens at each piece of equipment. The impedance-matching transformer gives a voltage gain but a small power loss. The main amplifiers give both a power and voltage gain. The C amplifier gives a power gain with usually no voltage gain. The line gives a power and voltage loss. The question therefore arises, "What are we going to measure along the chain, power or voltage?" There are a number of factors in favour of measuring voltage. These are:

- (1) It is easier to measure voltage than power.
- (2) The output of a valve depends upon the voltage applied to the grid, not upon the power dissipated in the grid circuit.

The voltage developed at the microphone will of course depend upon the loudness of the sound reaching it, but let it be assumed that it is going to be used for speech and that the normal speaking voice generates a voltage of about 0.00044 V. If the transformer has a turns ratio of 1 to  $\sqrt{10}$ , at the output of the transformer this voltage is stepped up to  $0.00044 \times \sqrt{10} = 0.0014$  V. Let it also be assumed that the other parts of the chain have the following characteristics. The first amplifier has a gain of 15.8, so the voltage out of the amplifier is  $0.0014 \times 15.8 = 0.022$  V. If the output of the attenuator is 0.4 of the input, its output voltage is  $0.022 \times 0.4 = 0.0088$  V. The gain of the second amplifier is 89 so its output voltage is  $0.0088 \times 89 = 0.775$  V. The third amplifier is a distribution amplifier with a gain of unity, hence the voltage fed to line = 0.775 V. If the output of the line is  $1/17.8$  of the input, the line output voltage will be  $0.775/17.8 = 0.044$  V.

The process of calculating the voltage at various points along the chain is therefore one of multiplication and division. The engineer would do this either with the aid of log tables which change the process of multiplication and division into one of addition and subtraction, or using a slide rule which is just a ruler with logarithmic scales. A further simplification can be made if instead of having a meter with a linear scale to read the voltage we use one which gives a reading directly on a logarithmic scale; we can then dispense with log tables and slide rules. The question then arises, "What shall be the reference voltage on this logarithmic scale?" From purely theoretical considerations one would be inclined to say, "One volt", but practical considerations decree otherwise. The chain considered incorporates a Post Office line and to avoid confusion we use the same *reference level* as the Post Office, which refers to relative power and is taken as 1 milliwatt since the watt would be too large. In the early days most Post Office lines were open wire lines and all these had an impedance of about 600 ohms. The voltage corresponding to 1 milliwatt in 600 ohms is 0.775 V. This is therefore our *reference voltage*.

## 2.2 THE DECIBEL (db)

The logarithmic scale of the meter that we are going to use is calibrated in decibels. Strictly speaking the decibel is a power ratio. If  $P_1$  and  $P_0$  are two amounts of power and  $N$  the number of decibels denoting their ratio, then  $N = 10 \log_{10} \frac{P_1}{P_0}$  and it can only be used in the strict sense as a voltage ratio, when the two voltages are across equal resistances,  $R$ ,

then since

$$P_1 = \frac{V_1^2}{R}$$

and

$$P_0 = \frac{V_0^2}{R}$$

$$N = 10 \log \frac{V_1^2}{R} \cdot \frac{R}{V_0^2} = 10 \log \frac{V_1^2}{V_0^2} = 20 \log \frac{V_1}{V_0}$$

In this case we shall get the same figure in db whether we compare powers or voltages. Because of the usefulness of comparing voltages, many people extend the use of the decibel by making it a measure of voltage ratio irrespective of the impedances across which the voltages are measured. We shall continue with this practice but in order to distinguish between the two applications of the term we shall adopt the following notation: where we are comparing powers or comparing voltages across equal resistances, we shall simply write db, but where the resistances are not necessarily equal we shall write db(v) to show that we are comparing voltages only. Taking the previous example, the generated level at the microphone

$$\begin{aligned} &= 20 \log \frac{0.00044}{0.775} = 20 \log \frac{4.4}{7750} \\ &= 20 \log 4.4 - 20 \log 7750 \\ &= 20(0.643 - 3.889) = -65 \text{ db(v)} \end{aligned}$$

The gain of the transformer is  $20 \log \sqrt{10} = 10 \log 10 = 10 \text{ db(v)}$   
 $\therefore$  the level out of the transformer is  $-65 + 10 = -55 \text{ db(v)}$

The gain of the first amplifier is  $20 \log 15.8 = 24 \text{ db(v)}$   
 $\therefore$  the level out of this amplifier =  $-55 + 24 = -31 \text{ db(v)}$

The gain of the attenuator is  $20 \log 0.4 = 20(-1 + 0.6) = -8 \text{ db(v)}$   
 $\therefore$  the level out of the attenuator is  $-31 - 8 = -39 \text{ db(v)}$

The gain of the second amplifier is  $20 \log 89 = 39 \text{ db(v)}$   
 $\therefore$  the output level is  $-39 + 39 = 0 \text{ db(v)}$

The level out of the distribution amplifier will be the same as it has no voltage gain.

The gain of the line will be  $20 \log \frac{1}{17.8} = -25 \text{ db(v)}$   
 $\therefore$  the level out of line =  $-25 \text{ db(v)}$

### 2.3 MEASUREMENT OF GAIN AND LOSS

As it is necessary to check periodically the gain or loss of the individual units in the chain, the methods adopted for the various units will now be discussed.

2.3.1. *Microphones*

It is not a simple matter to supply a known sound intensity at the diaphragm of a microphone, so a direct measurement of microphone response is not practicable. The procedure adopted is to compare the performance of the microphone under test with that of a calibrated microphone, both microphones being measured under exactly similar conditions.

2.3.2. *Amplifiers*

As we wish to know the amplifier gains under working conditions, it would seem that we must be provided with a tone source of variable output impedance and an amplifier detector of variable input impedance to cater for all possible cases, but since the input and output impedances of an amplifier are generally resistive and independent of frequency it is not difficult to calculate the difference between the gain under working conditions and that as measured in a 600-ohm set. The latter method is therefore adopted. Before proceeding to calculate this difference the concept of *reflection* will first be discussed.

Maximum power is transferred from a source to a load when the resistances of the two are equal. In the case of a load which has a different resistance from that of the source we may regard the source as again supplying this power but due to the mismatch only part of it is absorbed, the other part being reflected back. *Reflection loss* is therefore defined as the ratio, expressed in decibels of the power transmitted into a load of resistance different from that of the source, to the power that would be transmitted if the resistance of the load were equal to that of the source..

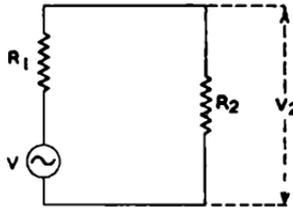


FIG. 9

In Fig. 9

$$V_2 = \frac{R_2 V}{R_1 + R_2}$$

And power in load =  $\frac{V_2^2}{R_2} = \frac{R_2^2 V^2}{(R_1 + R_2)^2 R_2}$

If  $R_2$  were equal to  $R_1$  this expression would become  $\frac{V^2}{4R_1}$

TRANSMISSION QUANTITIES

$$\begin{aligned} \therefore \text{Reflection loss} &= 10 \log \frac{V^2}{4R_1} \cdot \frac{(R_1 + R_2)^2}{R_2} \\ &= 10 \log \frac{(R_1 + R_2)^2}{4R_1R_2} \end{aligned}$$

For positive values of resistance this expression is always  $\geq 0$ . Since we are more concerned with voltage than with power we will use the same definition, with voltage substituted for power, for voltage reflection loss.

So from Fig. 9

$$\begin{aligned} \text{Voltage reflection loss} &= 20 \log \frac{\frac{1}{2}V}{V_2} \\ &= 20 \log \frac{R_1 + R_2}{2R_2} \end{aligned}$$

For  $R_2 > R_1$  this expression is negative which means that there is a voltage reflection gain. The gain of an amplifier under various conditions will now be evaluated.

*Gain of Symmetrical Amplifier Working between Matched Impedances (Fig. 10)*

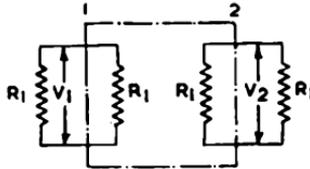


FIG. 10

$$\begin{aligned} \text{Gain (N db)} &= 10 \log \frac{P_2}{P_1} \\ &= 10 \log \left( \frac{V_2}{V_1} \right)^2 \text{ and since } \log X^2 = 2 \log X \\ &= 20 \log \frac{V_2}{V_1} \end{aligned}$$

*Gain of Unsymmetrical Amplifier Working between Matched Impedances (Fig. 11)*

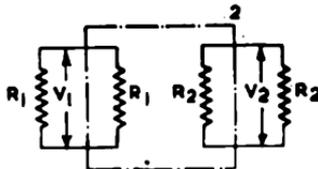


FIG. 11

$$\begin{aligned}
 \text{Gain (N db)} &= 10 \log \frac{P_2}{P_1} \\
 &= 10 \log \frac{V_2^2 R_1}{V_1^2 R_2} \\
 &= 20 \log \frac{V_2}{V_1} + 10 \log \frac{R_1}{R_2} \\
 &\quad \text{Voltage gain + Impedance correction}
 \end{aligned}$$

Gain of Unsymmetrical Amplifier Working into Unmatched Impedance (Fig. 12)

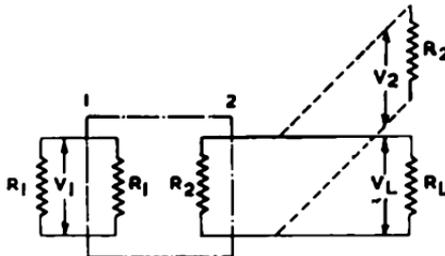


FIG. 12

Gain for working conditions

$$N \text{ db} = \left( 20 \log \frac{V_2}{V_1} + 10 \log \frac{R_1}{R_2} \right) - 10 \log \frac{(R_2 + R_L)^2}{4R_2 R_L}$$

Gain of amplifier for matched impedance condition

Output power reflection loss

$$\text{also} = 20 \log \frac{V_L}{V_1} + 10 \log \frac{R_1}{R_L}$$

Voltage gain for working conditions

+ Load impedance correction

Whence it follows that,

$$20 \log \frac{V_L}{V_1} = 20 \log \frac{V_2}{V_1} - 20 \log \frac{R_2 + R_L}{2R_L}$$

Voltage gain for working conditions

= Voltage gain for matched impedance condition

- Output voltage reflection loss

600-ohm test gain (Fig. 13)

$$N \text{ db} = 20 \log \frac{V_{600}}{V_s}$$

## TRANSMISSION QUANTITIES

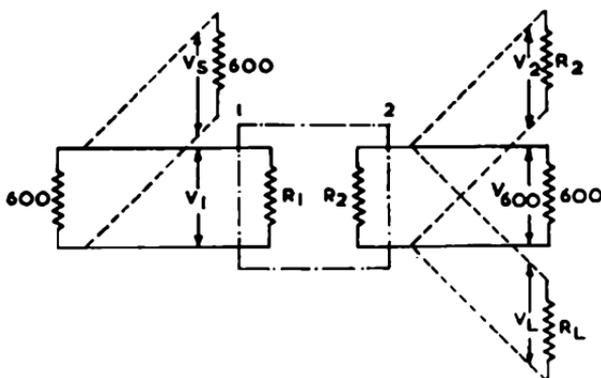


FIG. 13

$$= \left( 20 \log \frac{V_2}{V_1} + 10 \log \frac{R_1}{R_2} \right) - 10 \log \frac{(R_1 + 600)^2}{4 \times 600 R_1} - 10 \log \frac{(R_2 + 600)^2}{4 \times 600 R_2}$$

*Gain of amplifier for matched impedance condition*
*- Sending reflection loss*
*- Output reflection loss for 600-ohm termination*

$$\text{also} = \left( 20 \log \frac{V_L}{V_1} + 10 \log \frac{R_1}{R_L} \right) - 10 \log \frac{(R_1 + 600)^2}{4 + 600 R_1} + 10 \log \frac{(R_2 + R_L)^2}{4 R_2 R_L} - 10 \log \frac{(R_2 + 600)^2}{4 \times 600 R_2}$$

*Gain for working conditions*
*- Sending reflection loss*
*+ Output reflection loss for working load termination*
*- Output reflection loss for 600-ohm termination*

$$\text{and} = 20 \log \frac{V_L}{V_1} - 20 \log \frac{R_L(600 + R_1)(600 + R_2)}{1200 R_1 (R_2 + R_L)}$$

*Voltage gain for working conditions*
*- Combined conversion loss*

Conversely,

*Voltage gain for working conditions*

$$= \text{600-ohm test gain} + 20 \log \frac{R_L(600 + R_1)(600 + R_2)}{1200 R_1 (R_2 + R_L)}$$

### 2.3.3. Attenuators, Lines and Transformers

These must be tested under working conditions but as attenuators and lines are normally used between 600-ohm terminations the procedure is straightforward. In the abnormal cases the correct impedances must be supplied. The gain or loss of a line measured on tone at 1,000 cycles per second between 600-ohm terminations is called the *terminated level*, positive values referring to gain and negative values to loss. Where either of the terminations differ from 600 ohms this must be indicated and correction applied as specified in section 2.4.

## 2.4 PROGRAMME VOLUME

When operating the broadcasting chain at maximum efficiency there is a danger that excessive peak voltages may cause distortion, hence the programme meter has been designed to indicate peak values. It may be used to measure the level of a steady tone as there is a simple relationship between the peak and root mean square voltages of a sinusoidal wave. Although these two voltages are independent in a complex wave it has been established on a statistical basis that compared with a steady tone of equal power, programme peaks at 8 db higher. The volume of a programme is therefore defined as being equal to the level of a steady tone which gives a deflection on the programme meter two divisions (8 db) lower than that of the programme peaks. On this basis, if zero level (tone) gives a programme meter reading of 4, zero volume will cause the meter to read 6 on peaks.

The programme meter is normally calibrated to read 4 on zero level into 600 ohms. Where the impedance of the circuit is different from 600 ohms and a knowledge of programme power is more important than programme voltage, such as when programme is being sent to line, it is necessary to apply a correction. In such a case the programme power will be greater than that indicated, by  $10 \log 600/Z$  where  $Z$  is the circuit impedance.

*Example:* It is required to send at a programme power of + 4 db into a line having an effective impedance to programme of 100 ohms. At what voltage level should the programme meter be lined up?

The programme power will be greater than that indicated, by

$$10 \log \frac{600}{100} = 10 \log 6 = 8 \text{ db}$$

$\therefore$  Correct lining up voltage level = +4 - 8 = -4 db(v)

## 2.5 DISTORTION

Distortion is defined as the change in waveform which occurs between two points in a transmission system. There are various kinds of distortion, the most important from our point of view being attenuation distortion, phase distortion, harmonic and inter-modulation distortion, and amplitude distortion. Their causes and effects will now be considered.

### 2.5.1. Attenuation Distortion

Attenuation distortion arises from a variation in the loss or gain of a circuit with variation in frequency. Its effect on sound signals would be to alter the balance between the low and high frequencies,

the most common form being loss of high frequencies. The electrical part of the broadcasting chain which introduces the greatest amount of attenuation distortion is the Post Office line, especially one used on an outside broadcast. Attenuation distortion is not regarded as a serious fault provided it is not excessive, because it can be easily corrected by means of an equaliser.

### 2.5.2. Phase Distortion

Phase distortion is closely bound up with attenuation distortion. Where you get the latter you also get the former and in equalising for attenuation distortion, phase distortion is automatically lessened. There is a simple relation connecting the phase difference  $\phi$  of a sinusoidal wave of angular frequency  $\omega$  at two points in a circuit, with the time  $t$  taken for the wave to travel from the first point to the second.

The wave executes  $\omega$  radians in 1 sec

The wave executes  $\omega$  radians in  $\frac{\phi}{\omega}$  secs

$$\text{So } \frac{\phi}{\omega} = t$$

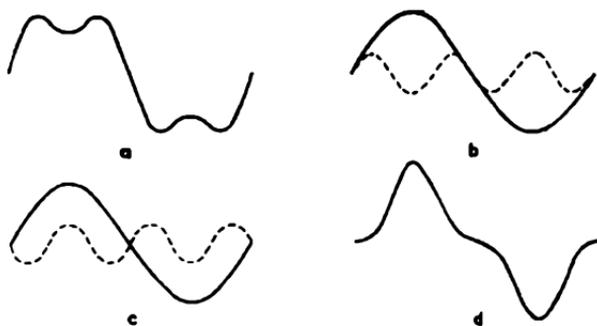


FIG. 14—Phase distortion: half-cycle delay

Phase distortion will therefore arise when the times of transmission of different frequencies through the circuit are unequal. Fig. 14 (a) shows a complex wave consisting of a fundamental frequency and a third harmonic as shown in Fig. 14 (b). If the higher frequency is delayed by half a cycle relative to the lower frequency, the phase relationships take the form shown in Fig. 14 (c), and the complex wave will be as shown in Fig. 14 (d). Now the ear cannot distinguish between sound waves of the form shown in Fig. 14 (a) and those of the form shown in Fig. 14 (d).

Phase distortion cannot be entirely ignored because speech and music are both transient in character. Consider a wave consisting of

a fundamental and third harmonic and persisting for a finite period; if the high frequency were delayed by a large number of cycles relative to the lower frequency, the signal would be changed from that shown in Fig. 15 (a) to 15 (b), and the ear would hear the two components of the signal at different periods. Fortunately such a phenomenon could only occur on a very long transmission line.

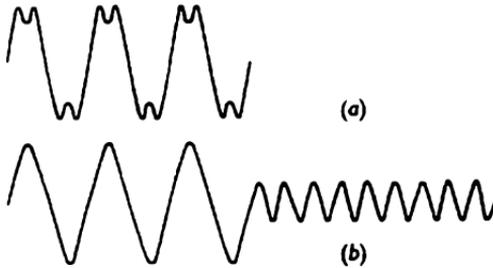


FIG. 15—Phase distortion: nine cycles delay

### 2.5.3. Harmonic and Intermodulation Distortion

Harmonic distortion is the production of harmonic frequencies at the output of a network when a sinusoidal voltage is applied at the input.

Intermodulation distortion is the production of combination frequencies at the output of a network when two or more sinusoidal voltages are applied at the input.

The two forms of distortion are discussed together because they are both due to the same cause, namely the presence in the network of a non-linear component, the characteristics of which vary with the instantaneous value of the signal. Applied to resistance, capacitance or inductance, we mean that its value is not constant but depends upon the current flowing through it. In the case of a non-linear amplifier the curve of anode current plotted against grid-cathode voltage is not a straight line over the working range.

Since the ear analyses a signal into its component frequencies, the output sounds very different from the input; this is the worst form of distortion in sound broadcasting because once it occurs, it cannot be corrected in a manner analogous to the process of equalising for attenuation distortion. It is essential to keep this form of distortion as low as possible in each piece of equipment as the total distortion is cumulative along the chain. Distortion is most likely to occur where the level of the signal is high in equipment using valves or iron-cored inductors.

Harmonic distortion is discussed more fully in BBC Engineering Training Supplement No. 3.

### 2.5.4. *Amplitude Distortion*

This again is a form of non-linear distortion but whereas harmonic and intermodulation distortion arise in a system which is non-linear to the instantaneous value of the signal, amplitude distortion is the product of a system which is completely linear to a steady signal but is non-linear to changes in the r.m.s. value of the signal. It is deliberately introduced in the sound chain in order to compress the dynamic range of the programme into one that the chain can accommodate. This is dealt with in greater detail in a later chapter.

## 2.6 NOISE

We will consider briefly the main sources of noise and the steps that are taken in each case to minimise it.

First of all we have acoustic noise which may be external or internal to the studio. One of the chief problems of studio design is to provide sufficient insulation against external noise (Chapter 3). This is achieved in the horizontal plane by using cavity walls. A more complicated construction must be used, however, in the vertical plane. The volume of internal noise will depend upon the type of programme, e.g., whether or not there is a studio audience. It is always high in a television studio, hence it is desirable here to use a microphone with a cardioid or unidirectional characteristic.

Secondly we have amplifier noise. This again may be of two kinds, namely hum, due to the power supply being obtained from the a.c. mains, and hiss from thermal agitation, which is random in character. The former may be taken care of by adequate smoothing in the h.t. supply, by ensuring that the valves used in the initial stages of the amplifier have minimum coupling between heater and the grid-cathode circuit, and by careful design and screening of transformers. The initial stage is the most important as any noise produced here is amplified by the subsequent stages. It is not practicable to reduce thermal noise and the only thing that can be done is to make certain that the strength of the signal at the input of an amplifier does not fall to a very low value relative to the noise.

The third major source of noise is the telephone line. The majority of modern telephone lines are carried in underground cables, and the wires being in close proximity must be accurately balanced as the signal in one circuit may transfer into an adjacent circuit by either inductive or capacitive coupling, producing what is known as *cross-talk*. In a well-balanced circuit most of this cross-talk will be longitudinal, i.e., the noise voltages in each wire will be equal and of the same sign and so may be prevented from travelling farther than the end of the line by terminating the line in a transformer with balanced primary. Another form of cross-talk which is more

common to communication circuits than to music circuits, and therefore of less importance, is that which occurs on a line used for carrying more than one channel, such as a teleprinter, as well as a speech channel or multi-channel carrier telephony. The cross-talk here may result from a number of causes such as inadequate filtering or from harmonic and intermodulation distortion. Noise on lines is discussed more fully on p. 133.

The most disturbing of the various types of noise we have discussed are intelligible cross-talk and continuous or interrupted tone in the middle-frequency range. The least disturbing are low-frequency hum and random noise.

The volume of noise generated in each unit of equipment must be kept as low as possible. The actual figure will depend upon the type of apparatus and the way it is used, for example, in a low-gain amplifier the noise may be 60 db or more below peak signal but in a high-gain amplifier it may only be about 45 db on maximum setting. The tolerance for signal-to-noise ratio on lines used for outside broadcasts is 46 db.

## CHAPTER 3

# ACOUSTICS AND MICROPHONE PLACING

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### 3.1 INTRODUCTION

FROM the earliest days of broadcasting it has been recognised that the acoustical requirements for good sound transmission are more exacting than for direct listening in the actual hall or studio. For orchestral performances in the average concert-hall, most of the seats are usually considered satisfactory, although fastidious listeners may have their preferences. But for the best quality of reproduction on the loudspeaker, the position of the microphone in the hall is critical, unsatisfactory results being usually obtained by placing the microphone near even the most favoured seats. In small rooms, in common with medium-sized halls, a listener at the far end can hear a speaker with almost 100 per cent intelligibility; but when broadcasting it is essential for the microphone to be placed within a few feet of the speaker; in small rooms it may be even closer, say 18 inches.

Clearly, the microphone and the transmission channel to the loudspeaker are not equivalent to the two ears of the listener in the studio or hall. Just as with two eyes *stereoscopic vision* makes it possible to see spatially and estimate distances, so with two ears *stereophonic hearing* implies the ability to estimate direction and to concentrate on sound coming from one direction only. Hence, the listener in the hall can concentrate, largely unconsciously, on the sound coming direct to him from the orchestra or speaker, ignoring the sound reflected from walls, ceiling, or other reflecting surfaces. The microphone, however, has no such ability to discriminate between the direct and reflected sound. When the microphone is too far away from the sound source the reproduction becomes confused and lacks clarity because of the pickup of a high proportion of reflected to direct sound. While a listener with good binaural hearing can enjoy a performance even if seated at the back of an auditorium, the microphone must be comparatively near the orchestra or speaker and as far as possible from any object that may reflect the sound.

For satisfactory broadcasting or recording, the microphone or microphones have to be skilfully placed in relation to orchestra,

artists and speakers. Many concert-halls, although adequate for direct listening, are quite unsuited for broadcasting, which requires much more exacting acoustic conditions.

### 3.2 ACOUSTICS FOR BROADCASTING

Until the pioneer work of W. C. Sabine in the U.S.A., at the beginning of this century, there had been no serious attempt to apply systematically the already well-established science of sound to the building of auditoria and concert-halls. Even the new Sabine approach was not at all well known, and its principles seldom put into practice until the advent of broadcasting in the 1920's, with its more stringent acoustical demands. Also in this same period the replacement of the traditional wood and brick by modern building materials raised new problems, for floors and ceilings and walls of reinforced concrete absorb much less reflected sound than those of the older materials. Moreover, the modern architectural trend of simple construction, avoiding ornamentation, did not tend to produce good acoustics; as will be seen later elaborate ornamentation scatters the sound and leads to good diffusion. Hence, in the last 30 years the study of practical acoustics has increased greatly, and application of its principles has become universal in building practice.

#### 3.2.1. *Direct and Indirect Sound*

In broadcasting or recording from any enclosed space, the sound waves which reach the microphone from musical instruments, singers or speakers can be divided into two classes:

- (i) those which arrive by the shortest possible path, i.e., the direct sound;
- (ii) those which arrive after one or more reflections from the surfaces of the hall, i.e., the indirect or reverberant sound.

To the radio listener direct sound is more faithful and more characteristic of the original than indirect sound which, to a greater or lesser extent, is acoustically distorted as a result of the one or many reflections, depending on the reflecting properties of the hall's surfaces (walls, ceiling, and floor), the type of seating used and the size of the audience present. *The science of room acoustics is primarily the study of the characteristics of the reflected sound, and the art of microphone placing is largely the adjustment of the ratio of the direct to the indirect sound.*

#### 3.2.2. *Reflected Sound*

Sabine was the first to make a study of reflected sound. When a steady note from an organ-pipe or from an electrical tone source is

sounded in an enclosed space, the sound energy increases with each of the numerous reflections and quickly builds up to a maximum value which remains constant until the steady note is cut off; when this happens, the reverberant sound energy is gradually absorbed and the sound dies away to inaudibility, the time taken depending upon the absorption properties of the surfaces of the enclosed space. The interval of time for the reverberant sound to decay to inaudibility is called the *reverberation time*: it is more precisely defined as the time taken for an intensity to fall to one-millionth of its initial value, i.e., a decrease of 60 decibels from the original intensity. Sabine showed that the reverberation energy is directly proportional to the reverberation time, and utilised a stop-watch for measuring the time

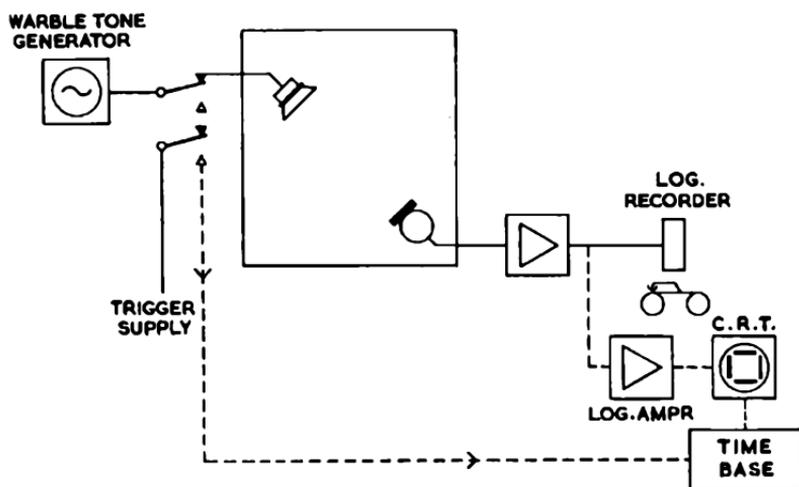


FIG. 16—Apparatus for measuring reverberation.  
The recorder incorporates a logarithmic amplifier

taken for the reverberant sound to die away. In modern practice, the stop-watch has been replaced by high-speed electrical recorders or cathode-ray oscillographs, which show and record the manner in which the reverberant sound decays (Fig. 16). Sabine proved also that the sound decay is approximately exponential, analogous to the current discharge of a capacitance through a resistance. If a logarithmic amplifier is incorporated in the measuring chain, it is only necessary to measure the slope of the decay pattern in order to find out the reverberation time (Fig. 17).

Sabine in his original experiments used as sound sources a series of organ-pipes of different lengths, and showed that the reverberation time usually varied greatly with the sound frequency. Hence,

it is essential to carry out reverberation time measurements for a large number of frequencies over the audible frequency range, usually from 60 c/s per sec to 8,000 c/s. Electric tone generators have replaced the Sabine organ-pipes, but a sine-wave note sounding in a reverberant room sets up a standing-wave pattern, and when the sound source is cut off there are violent changes in the standing-wave

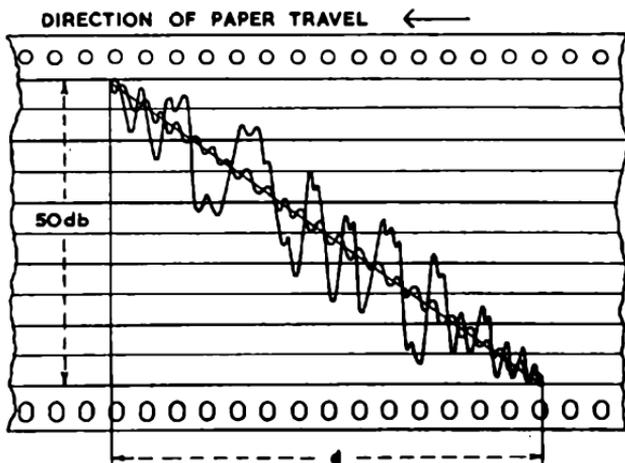


FIG. 17—Decay pattern as measured by high-speed recorder

Speed of paper = 30 mm/sec or 100 mm/sec  
 Time =  $\frac{60}{50} \times 30x$  or  $\frac{60}{50} \times 100x$   
 where  $x$  is in millimetres

pattern resulting in a very irregular decay. Hence, in order to minimise the standing-wave pattern it is customary to use a warble note, i.e., a note varying usually plus or minus 10 per cent in frequency rather than a note of constant frequency. This is illustrated in Fig. 17, the smaller curve indicating the application of a warble note.

### 3.2.3. Reverberation Time

Sabine, in addition to showing that the reverberant energy is directly proportional to the reverberation time, gave a simple formula and demonstrated that the decay time  $T$  was proportional to the volume  $V$  of the hall, and inversely proportional to the sum of the hall's surface areas  $S$  and their respective absorption co-efficients  $a$ .

$$T = \frac{0.05 V}{(a_1 S_1 + a_2 S_2 + \dots)}$$

The Sabine absorption co-efficients vary from 0 to 1, i.e., from a perfect reflector to a total absorber of the impinging sound. He took an open window as the standard of a total absorber with co-efficient 1. If we imagine a room having no absorption other than that due

to an open window, the reverberation time of the room will depend on the area of the window. The sound is absorbed by virtue of this space, and by taking one square foot of open window space as a standard, we have a unit for comparing the absorption of other materials. For example, if the reverberation time for a room having an open window space of 0.25 sq. ft is similar to that for the same room when the open window is replaced by 1 sq. ft of absorbent material, then the absorption co-efficient of that material is 0.25.

By applying the Sabine formula and using materials with the necessary absorption characteristics, it is theoretically possible to design a hall or studio of any desired reverberation time for any given frequency. Usually, however, the aim is to have the reverberation time relatively independent of frequency, at least for broadcasting, for reasons which will become more apparent later on in this chapter. Hence, it would appear to be necessary to find materials which absorb the whole frequency gamut equally, i.e., a

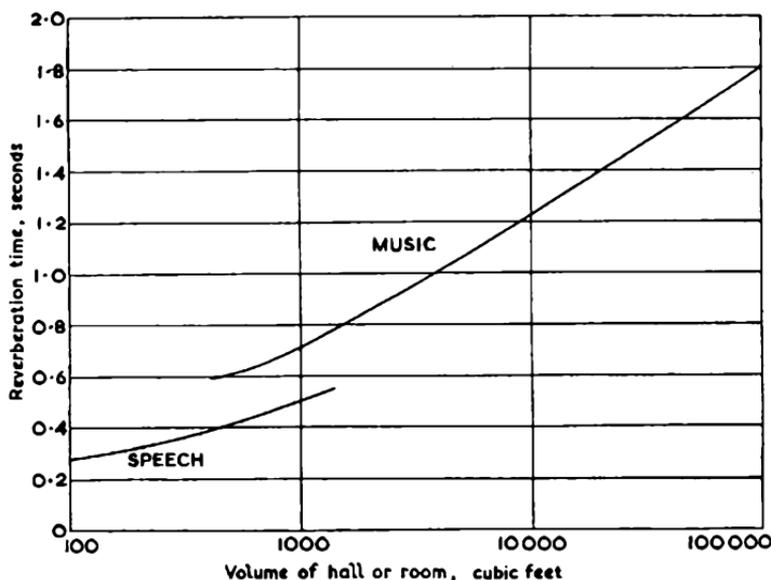


FIG. 18—Reverberation time curves, after Beranek

co-efficient of absorption constant with frequency. Such materials have yet to be developed, so that it is usual and more convenient to use various types of materials with varying co-efficients over the frequency range.

Practical experience has shown that large concert-halls with audiences of normal size present should have reverberation times to

the order of 1.5 to 2 seconds. Smaller halls or studios for medium-sized orchestras should have reverberation times round about 1.2 to 1.5 seconds; studios for variety purposes, 0.8 to 1 second; drama studios, anything between complete deadness, i.e., reverberation time of the order of 0.1 of a second, up to 0.6. Studios for speech, talks or discussions should be of the order of 0.3. It must be pointed out, however, that the above reverberation times are subject to changes of taste, for instance at the time of writing, variety and light music producers like to work in studios of longer reverberation than was considered desirable a few years ago. Various authorities have prepared diagrams showing how the reverberation time should vary with the volume of the room or hall (Fig. 22).

Simple Sabine theory as outlined above can only be regarded as a first approximation to a more general theory; it assumes complete diffusion of the sound, which seldom occurs in practice. In a room possessing simple geometry, e.g., a rectangular room, there is a complicated system of natural resonances similar to the natural resonances of an organ-pipe, but whereas the organ-pipe has a one-dimensional series, the rectangular room has a three-dimensional system.

The natural frequencies of such a rectangular room with hard plane surfaces for walls, ceiling and floor are determined by the Rayleigh formula,

$$f = c/2\sqrt{\left[\left(\frac{p}{L}\right)^2 + \left(\frac{q}{W}\right)^2 + \left(\frac{r}{H}\right)^2\right]} \text{ cycles per second}$$

where  $p$ ,  $q$ ,  $r$  are independent integral numbers (0, 1, 2, 3, etc.) and  $L$ ,  $W$ ,  $H$  the length, width and height, respectively, of the room, and  $c$  the velocity of sound propagation.

In a very large hall the fundamental resonances are so low in frequency, e.g., 10 c/s, that over the audible frequency range the natural resonances are very close together, so that one is not conscious of individual resonances, but only of the general reverberant energy. Hence the Sabine theory is most successful in predicting the acoustic behaviour of large halls.

But for small rooms or studios with dimensions between 10 and 20 feet, the fundamental resonances are in the bottom end of the audible range; the natural resonances therefore are no longer closely spaced, and one is very conscious of individual resonances, so that certain notes of the speaking voice may be greatly exaggerated compared with other notes. Hence it is far more difficult to design small speech studios than much larger studios.

A more evenly spaced distribution of the natural resonant frequencies can be obtained by several expedients:

- (i) irregular surfaces and ornamentation much favoured in the past;
- (ii) non-parallel surfaces;
- (iii) the avoidance of certain dimensional ratios, such as the cube;
- (iv) irregular distribution of the absorbing materials in small patches.

### 3.3 ACOUSTIC MATERIALS

#### 3.3.1. Porous Materials

When sound waves impinge on a porous material, such as thick curtains, the air in the interstices in the material is set into vibration. Some of the sound energy is converted into heat due to the friction within the pores of the material and is consequently absorbed; the remainder of the sound is partly transmitted and partly reflected. For porous materials to be effective absorbers they must have a thickness approximately proportional to the wavelength. The most effective absorber of this class is glass-wool, or rock-wool, which for a thickness of 6 in will give an absorption co-efficient of 0.8 from 250 c/s to 8,000 c/s, with a value of 0.6 at 100 c/s and 0.3 at 60 c/s. Porous absorbers are more effective at the lower frequencies if they are spaced at some distance from the reflecting wall surface. It is usual to cover the highly absorbent materials of the glass-wool type

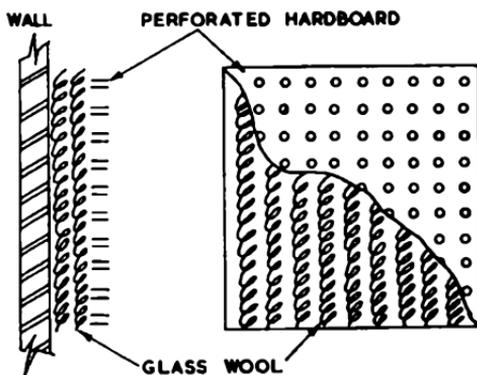


FIG. 19—Use of hardboard panelling to restrict absorption at high frequencies

with fabric. Unfortunately, such fabrics tend to become electrostatically charged and attract dust particles which in the average industrial centre results in begrimed material within a short time; it is, therefore, becoming more usual to use plastic fabrics which can be easily washed *in situ*.

In practice high absorption co-efficients at the high frequencies are often undesirable. The furnishings, the occupants of a room or hall, and the air itself tend to absorb the higher frequencies. This absorption can be restricted very effectively by covering the glass-wool with perforated panelling (Fig. 19). The size and number of holes per unit area determine the absorption co-efficient for the higher frequencies. A panelling of 5 per cent perforation causes the absorption to tail off above 1,000 c/s. When more absorption at the higher frequencies is required, a higher proportion of perforation or slotted panelling is used.

### 3.3.2. Membrane Absorbers

Membrane absorbers consist of non-porous materials, such as plywood, building board, hardboard, roofing felt, linoleum, etc., placed at some distance from a surface. The space between the material and the surface is often partially filled with porous absorbing materials and sealed to prevent air escaping when the membrane vibrates (Fig. 20). Such absorbers have pronounced resonant fre-

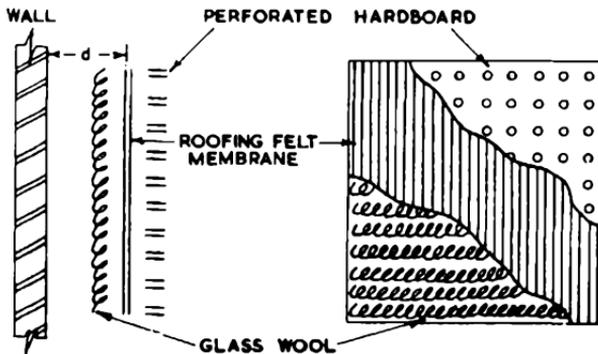


FIG. 20—Application of membrane absorber

quencies in the lower range, and have quite high absorption co-efficients in that range. Wood panelling and lath-and-plaster were the principal materials used in traditional building practice, which accounts for the good bass absorption characterising older concert-halls and theatres. As was noted above, porous materials are not at all effective absorbers at the lower frequencies; membrane absorbers are therefore very useful for supplementing the porous materials. When absorption over a comparatively narrow band of low frequencies is required, for example in a room with a pronounced standing wave, the roofing felt or linoleum type of absorber is very useful. The frequency of resonance depends on the thickness of roofing felt and its spacing from the walls. Often in modern studio

construction the low frequencies from 60 to 250 c/s are effectively absorbed to the required amount by using a number of such absorbers tuned to staggered frequencies in this range.

### 3.3.3. Resonant Absorbers (Helmholtz)

A resonant absorber, often called a Helmholtz resonator (Fig. 21), consists essentially of a vessel with rigid walls connected to the surrounding air by an opening called the neck. Sound energy is absorbed round about the resonant frequency of the resonator,

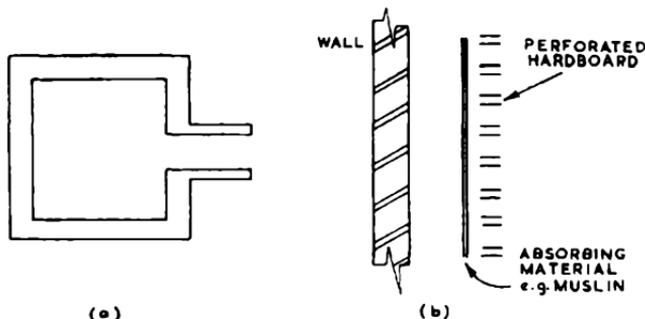


FIG. 21—Resonant absorber

which therefore acts as a fairly sharply tuned device—more so than the membrane roofing felt absorber. In practice, resonant absorbers are mounted on the wall or built into the wall.

For higher-frequency absorption, a common method is to use perforated or slotted panelling spaced an inch or so from the wall with the intervening space lightly packed with absorbing material; this behaves as a Helmholtz resonator with a sharp peak, before the tail-off referred to in section 3.3.1.

### 3.3.4. Measuring Absorption Co-efficient of Materials

There are various methods for measuring the all-important absorption co-efficients. The more usual and straightforward method is to use a room with very hard walls, ceiling and floor which have absorption co-efficients approaching zero, and to measure the reverberation time in the empty state. Samples of the material are then arranged on the walls, ceiling and floor, and the reverberation time again taken (Fig. 22). The absorption co-efficient of the sample material can then be deduced by simple substitution of the Sabine formula. Measurements, of course, are carried out over the wanted frequency range.

Acoustic absorbing materials can be divided into three types, porous materials, membrane absorbers, resonant absorbers.

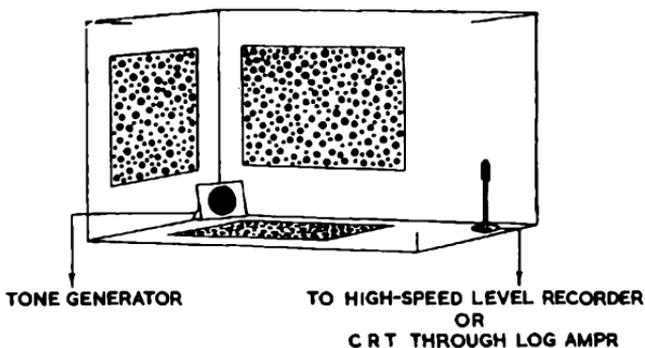


FIG. 22—Method of measuring absorption co-efficient of materials

### 3.4 STRUCTURAL RESONANCES

Attention must be paid to the influence the structure itself has on the general absorption, usually confined to the low-frequency end of the aural spectrum.

It is possible to erect structures, such as brickwork several feet thick, which are quite rigid. But economical considerations usually dictate the building of simple structures which have their own resonances, for example wood-joint floors, suspended ceilings, walls of varied compositions. It is essential that these resonances should be reasonably well damped. In designing new buildings, or where the existing structure is completely determined, allowance can be made for the absorption due to resonances in the structure. But it is often preferable to measure the reverberation times of the bare room over the frequency range and then apply the necessary absorption materials to reduce the reverberation time at any given frequency to the required value.

By making use of acoustical absorbing materials of the above three categories, and taking into consideration the acoustical characteristics of the actual structure, it is possible to design halls, studios and rooms to have reverberation times of any desired values and with reverberation frequency curves of any desired shape, although usually an approximation to a straight line is desirable, for broadcasting at least.

### 3.5 SOUND INSULATION

Broadcasting and recording studios require to be adequately insulated from external sounds to a somewhat greater extent than for direct listening in the studios. Outside windows, if provided, are sealed and are of double and even triple plate glass; consequently conditioned air supply has to be installed with baffles or acoustic

filters in the trunking carrying the air supply, to cut out all outside extraneous noise.

Sound insulation theory indicates that the reduction factor, i.e., the ratio of the sound pressure at the face of one side of the wall to the corresponding pressure on the other side, is proportional to the product of the mass per unit area of the wall and the frequency. Hence for any given wall the sound reduction factor or the insulation decreases 6 decibels per octave as the frequency falls. In practice a lower value is obtained, 4 to 5 decibels, so that for adequate reduction of outside air-borne noise, massive walls, ceilings and floors are necessary. It is, however, possible to attain higher sound reduction factors by using double walls of lighter construction, separated by an air space. A sound absorbing blanket suspended in the cavity gives a further improvement in insulation.

Structural-borne noise can be reduced by floating floors and even walls on glass-wool blankets, and by suspending ceilings on resilient hangers.

Special attention has to be paid to studio doors, which may be either strong massive doors of special design or two or more doors of more orthodox construction separated by vestibules.

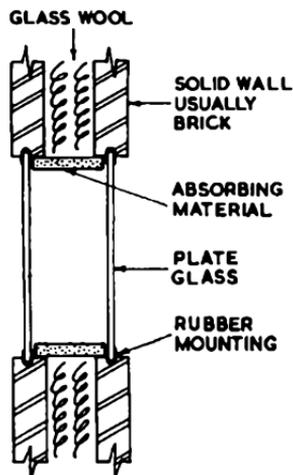


FIG. 23—Method of insulating a double window between two rooms

### 3.5.1. *Sound Insulation between Studio and Control Cubicle*

A modern studio has an adjacent control cubicle with an observation window looking into the studio so positioned that the producer and technical operators have a full view of at least the working end of the studio. For good sound insulation between studio and control cubicle, double panes of plate glass are necessary; in a double wall

of cavity construction, one pane is fixed to each wall. The plate glass is mounted in rubber and the edges of the air space between the panes are treated with sound absorbing material (Fig. 23).

### 3.6 MICROPHONE PLACING

Now that the theory of acoustics of enclosed spaces has been outlined, it is necessary to see what bearing it has on the practical problems of arranging microphones in relation to instruments, or speakers, to maintain artistic balance. In the BBC, the art of microphone placing is referred to as *balancing*.

#### 3.6.1. *Orchestral Halls and Studios*

The pressure on the diaphragm of the microphone is inversely proportional to its distance from the orchestra; however, the reverberant energy in a concert-hall is to a rough approximation reasonably constant in the open parts of the hall, i.e., away from seating, and from areas under balconies, etc. Therefore, the ratio of direct sound to indirect sound picked up by the microphone becomes less as the microphone is moved away from the orchestra. Even where the acoustics are very good and the reverberation time reasonably independent of frequency, the indirect sound is quite characteristic, lacking the clarity of the original, and sounding somewhat confused. The effect is more noticeable with some instruments than others; it is much more apparent on violins than woodwind instruments. Hence when the microphone is far away from the orchestra the reproduction on the loudspeaker becomes confused, lacking in clarity and definition.

*The art of microphone placing is fundamentally the art of adjusting the ratio of direct and indirect sound in order to obtain a pleasing artistic reproduction on the loudspeaker.* When the microphone is very near to the orchestra, there is a large ratio of direct to indirect sound and the reproduction, although a faithful reproduction of the instruments, is harsh and lacking in musical quality; this results in a poor orchestral blend or ensemble. With the microphone farther back, so that there is a smaller ratio of direct to indirect sound, the greater amount of reflected sound softens the reproduction and also gives liveliness with a feeling of spaciousness.

The softening effect has often been compared to what happens in photographic portraiture. Special portrait lenses are used, which introduce various types of lens distortion, so that the needle-point sharpness obtained when using high-quality lenses is removed, thus blurring over harsh, sharp lines.

The general theory of acoustics outlined above has shown that the indirect sound consists of waves reaching the microphone after one

or many reflections from the hall's surfaces, some of which may be comparatively near the microphone, others far off, such as at the back of the hall. Experiment and general experience tends to show that all these reflections are not of equal value in softening the reproduction. The first reflections from the surfaces near the orchestra seem to be the most important for obtaining an ensemble, and the general over-all softening of tone. It is well known that when broadcasting in studios which are acoustically divided into a dead part and a live part, the orchestral reproduction is very much better with the orchestra in the live part than in the dead part, even though the microphone is so positioned that the ratio of direct to indirect sound is the same in each case.

Nevertheless, although the general ensemble effect depends upon the first reflections from the surfaces near the orchestra, this is not enough for a satisfactory transmission; there must also be a general spacious effect conveyed by the over-all reverberations, i.e., the countless reflections from every part of the hall.

In this discussion so far an omnidirectional microphone, i.e., one capable of responding to sounds coming from every direction, has been assumed. There are numerous types of microphone: electro-dynamic (moving-coil, the ribbon), electro-static (condenser), piezo-electric (crystal), having characteristics of various types. However, for all practical purposes, microphones can be divided into three classes, irrespective of their modes of operation:

- (i) omnidirectional
- (ii) bi-directional, with a figure-of-eight polar diagram
- (iii) cardioid, so-called single-sided.

For all three types, the directional properties should be independent of frequency.\*

The omnidirectional type of microphone, which accepts reflected sound from every direction, is most useful for halls of short reverberation times. On the other hand, the bi-directional and cardioid types do not respond to anything like the same extent to the reflected sound; in fact, such microphones pick up only one-third of the reflected sound energy. Hence, these two types of microphone can be used in very lively halls of longer reverberation times. The cardioid microphone is more suitable than the bi-directional when it is desired to pick up more reflected sound from the front, ignoring reflections coming from the back, e.g., when broadcasting an orchestra from an empty concert-hall, or when the orchestra is stretched out over a wide platform. In such cases the wider acceptance

\*See BBC Engineering Training Manual *Microphones*, by the Staff of the Engineering Training Department (Iliffe & Sons, Ltd., 1951).

angle of the cardioid is especially useful. But, in most concert-halls with a normal-sized audience present, the bi-directional microphone, which ignores to a large extent reflections from the sides, and responds to those from the front and back, usually gives the better over-all reproduction.

In balancing an orchestra for broadcasting, the optimum position is found empirically, and is usually determined by the balance of the strings in relation to the woodwind and brass. It so happens that the string instruments, perhaps because of their characteristic transient tone, require a higher ratio of direct to indirect sound; consequently, as the microphone is taken farther and farther away from the orchestra, the strings tend to be blotted out by the other instruments for which a lower ratio of direct to indirect sound is acceptable, because of their more substantial tonal quality. Fortunately, the string instruments are usually quite a few feet in front of the others.

Most concert-halls have their acoustical peculiarities and the orchestra is often placed on platforms which are not entirely suitable for broadcasting, so that much experimenting has to be carried out to find the best positions, which are often a compromise between various conflicting factors. In most halls it is usually possible to find at least one position where a single microphone can be so placed that the balance of instruments is satisfactory for broadcasting purposes. Nevertheless, there are halls which present special difficulties, such as those having a very wide, shallow platform, necessitating the use of two microphones; other halls may be lacking in reverberation and for these a microphone well to the back of the hall has to be used in order to pick up additional reverberation. But even in these halls presenting special difficulties no attempt is made to alter the relative sensitivities of the two microphones during the actual performance of a particular composition. By adopting such principles, the technicians are not interfering with the balance of instruments as determined by the conductor for actual listening in the hall.

Microphone placing for orchestral transmission from studios is very similar to that for concert-halls but the problems are simplified because the studio is especially designed acoustically for the purpose, and care is taken that the acoustical properties are the same at rehearsal as at transmission. The audience is usually small, and the seating provided is heavily upholstered, so that the acoustics are very little affected by the presence of an audience. In the older halls, the difference in acoustics between the hall empty and full is very great, and quite often changes the reverberation time from 4 or 5 seconds to 1.5 seconds between the two conditions. This makes it very difficult to judge what the actual performance will be like by listening to the rehearsal.

The floor area occupied by the orchestra has an important bearing on microphone placing. It must not, however, be assumed that it is possible to design a studio for orchestral playing in which practically the whole floor is needed to accommodate all the players. In practice it has been found that no more than a third, ideally much less, of the floor area should be occupied by the players.

The height of a studio for music purposes should be in keeping with the length and breadth. The main orchestral studio at the BBC premises in Glasgow has dimensions of 80 feet in length, 60 feet in breadth, and 40 feet in height. This studio is considered very satisfactory for orchestras of up to 70 or 80 players. Just as curves have been prepared from empirical data showing optimum reverberation time for studios and halls of varying volume, similarly curves have been prepared giving the maximum number of players for any given volume (Fig. 24).

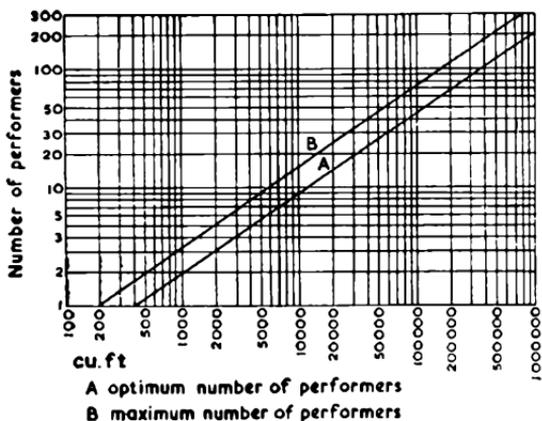
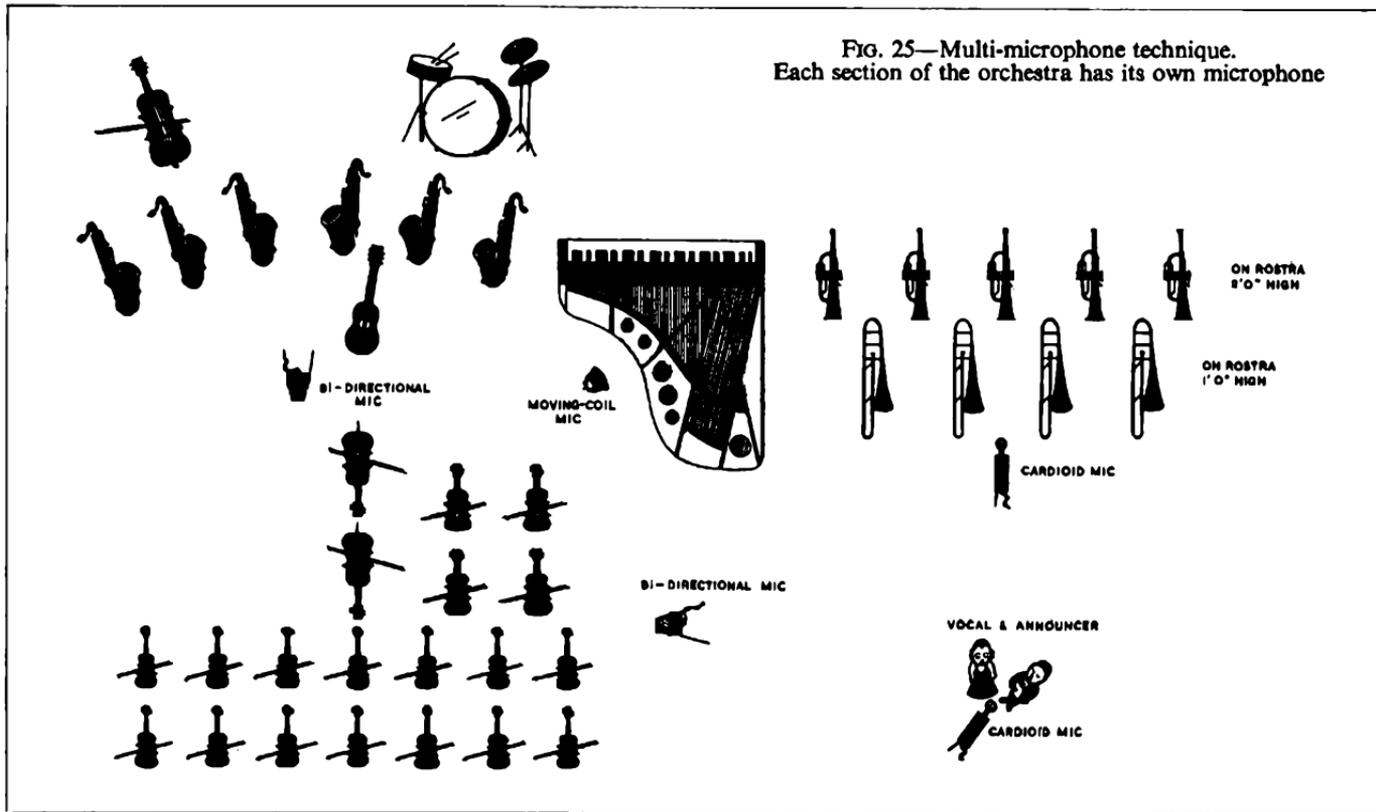


FIG. 24—Relationship between volume of hall and number of performers

In the broadcasting of symphony orchestras from studios the microphone is placed a considerable distance from the orchestra, determined in the same manner as in the concert-hall, there being considerably more latitude in the studio. However, many of the orchestras regularly broadcasting from studios are much smaller although playing music intended for full-sized orchestras. They usually have more or less the full complement of brass and woodwind instruments, but are relatively weak in string instruments, e.g., six first violins, four seconds, two violas, two cellos and one double-bass, compared with a full orchestra of 16 firsts, 16 seconds, 12 violas, ten cellos, eight double-bass. Such orchestras are internally unbalanced, and for any playing louder than mezzo-piano the strings are much too weak for woodwind and brass; hence, in order



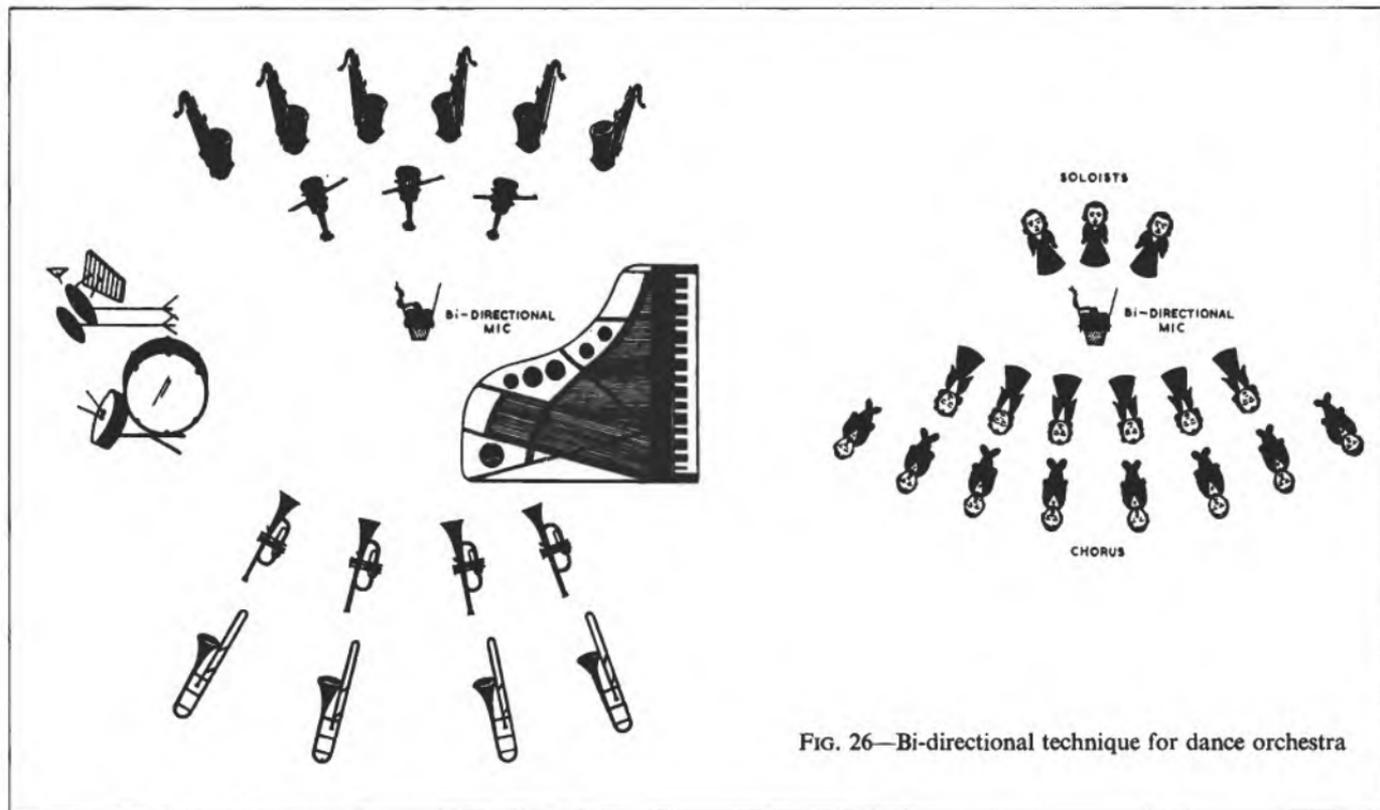


FIG. 26—Bi-directional technique for dance orchestra

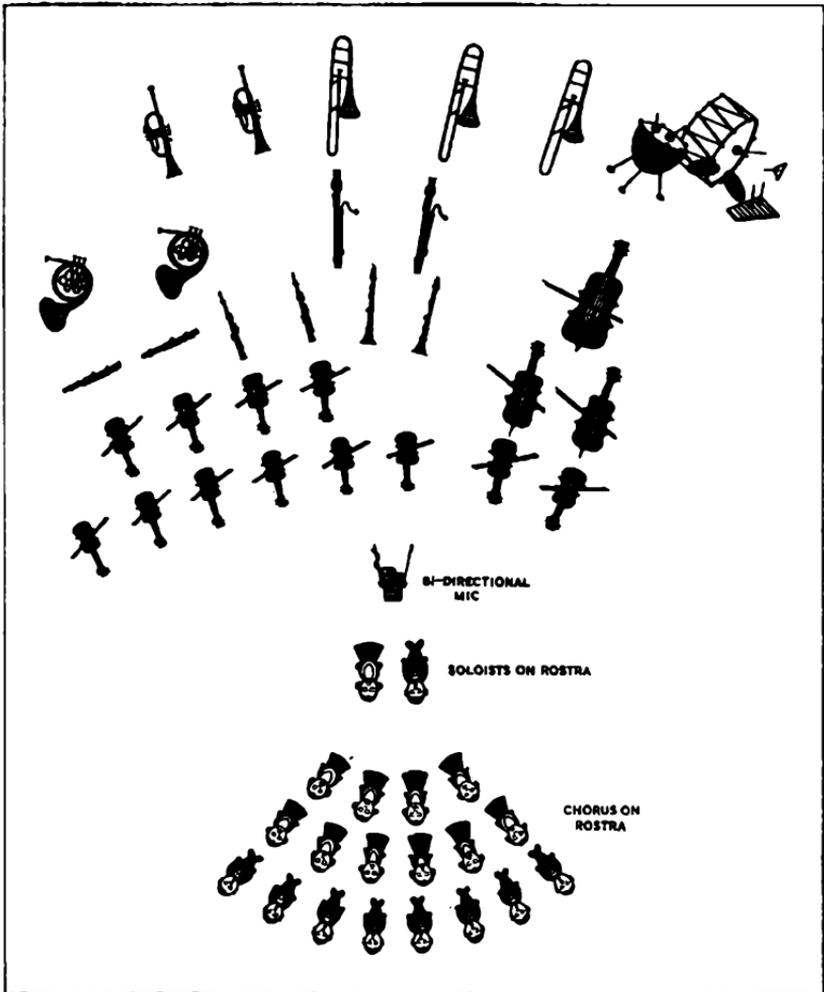


FIG. 27—Single microphone technique

to maintain a good balance for the radio listener, the microphone has to be placed relatively near the strings to make up for their numerical weakness.

At the present time, composers and arrangers are orchestrating more and more with a radio audience in mind, and are exploiting many unusual tone colours by placing microphones very close to the instruments such as muted brass and softly-played clarinets, which are practically inaudible to the audience in the hall. Nevertheless, in balancing these orchestras playing modern arrangements, the principle of using one microphone is often adopted, plus additional microphones close to the muted brass and other instruments

## ACOUSTICS AND MICROPHONE PLACING

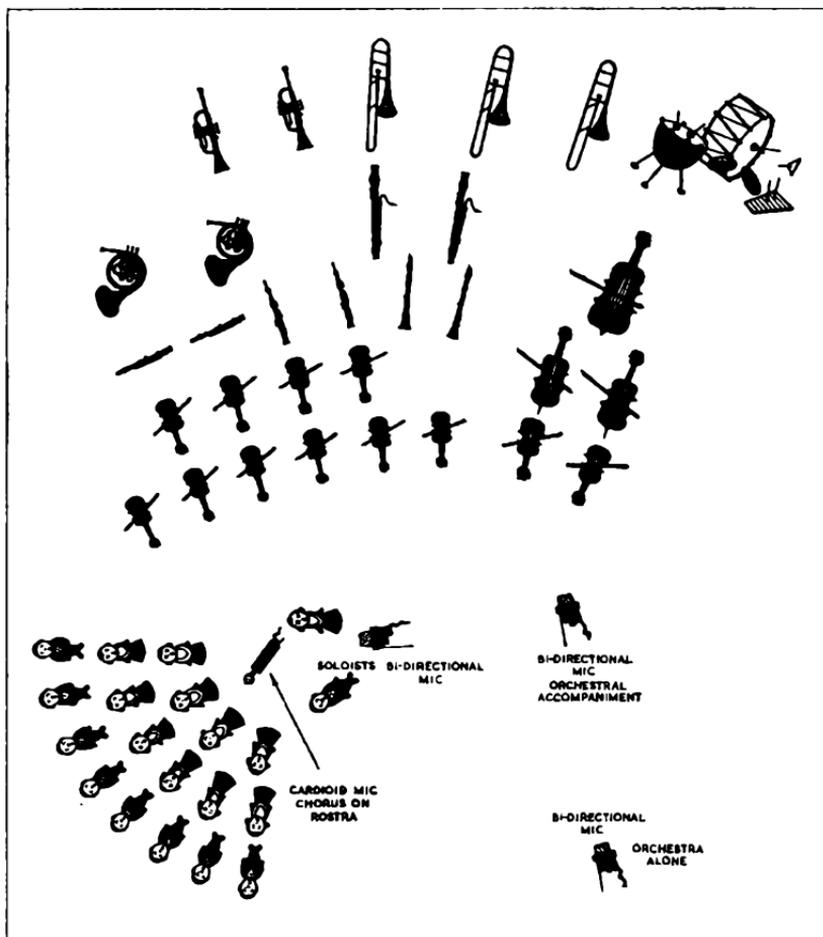


FIG. 28—Multi-microphone technique

producing the special tone colours. Other orchestral arrangements necessitate a multi-microphone technique, originally evolved for the broadcasting of dance orchestras; each section of the orchestra has its own microphone or even microphones. Fig. 25 is an example of this multi-microphone technique, *but it should be understood that there are almost unlimited possibilities in the field of microphone placing and instrumental layout.*

For orchestral transmission it is seldom that both sides of a bi-directional microphone are used for direct orchestral pickup; but, in dance orchestra broadcasts, where often a close-up reproduction with the minimum amount of reflected sound is desired, the players are divided on each side of the bi-directional microphone, Fig. 26 is an example with an additional microphone for soloists and singers.

It is sometimes possible to broadcast an orchestra, with solo singers and even a choir, using only one microphone, either double-sided or single-sided, as shown in Fig. 27, but the balance of sound between the singer and orchestra or chorus and orchestra is much more critical for the radio listener than for the listener in the studio. Hence, it is usually desirable to use separate microphones so that the critical balance between singers, chorus and orchestra may be obtained by adjusting the relative sensitivity of the microphones. Fig. 28 shows a layout using several microphones.

Orchestras are often broadcast from halls or studios which are acoustically dead, the studio probably having been designed for other purposes. This defect can be corrected to a considerable extent by using small echo rooms (see Chapter 4). A proportion of the energy picked up by the microphone is routed to a loudspeaker

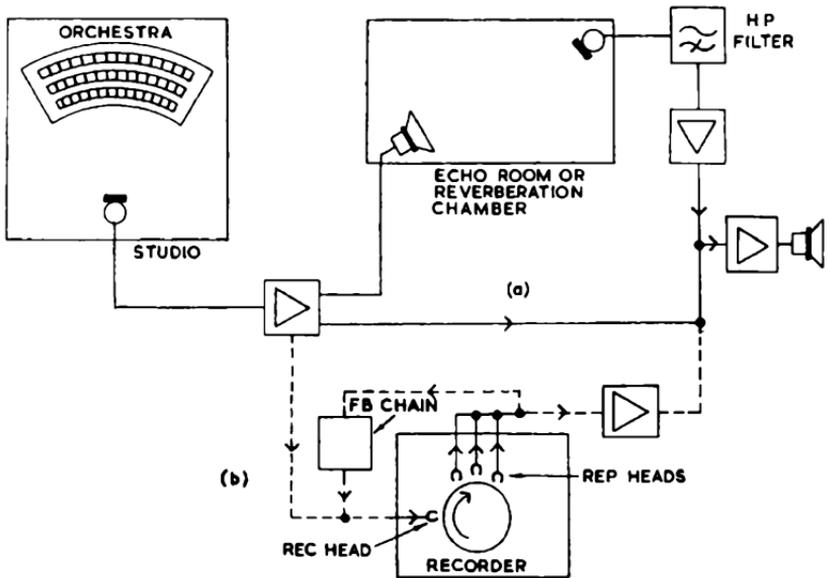


FIG. 29—Artificial reverberation methods.

The lower part of the diagram illustrates the use of variable delay by means of staggered reproducing heads on a magnetic recorder

placed in a small, very reverberant room having highly reflecting walls, ceiling and floor. A microphone picks up the many reflections from the echo room's surface, and its output after amplification is superimposed on the direct output from the microphone in the studio (Fig. 29). The improvement in creating an illusion of space by these means depends considerably upon the size of the echo room available and the quality of the loudspeaker and ancillary electrical apparatus. Although improved results can be obtained from a small

echo room by filtering out the low and middle frequencies coming from it (especially when the studio in question has adequate low-frequency and middle-frequency reverberation but is lacking in high-frequency), the best results are only obtained when the acoustics of the echo room are comparable with the acoustics which it is desired to simulate on transmission and which are lacking in the hall where the orchestra is playing. The superimposition of echo or reverberation is being used more and more in the broadcasting of dance bands and light orchestras with vocal items. In a multi-microphone set-up, the producer, if he so desires, can obtain special artistic effects by applying echo to individual microphones at will.

Many attempts have been made to produce artificial reverberation without a reverberation chamber; for example, coiled springs, long tubes, artificial transmission or delay lines. The most encouraging results have been obtained by taking advantage of the delay between recording and reproducing heads in magnetic tape recording apparatus; with this method, a number of reproducing heads are suitably spaced along the tape with feedback from each to the recording head. By careful adjustment of the amplitudes from each reproducing head and controlling the amount of feedback the apparent reverberation can be varied over quite wide limits and an approximation to an exponential decay obtained (Chapter 6).

### 3.7 DRAMA PRODUCTION TECHNIQUE

In drama productions the question of acoustical perspective is all-important. At one moment the producer wants his artists to sound quite close up as heard on the listeners' loudspeakers, the next moment he may desire an artist to be quite distant. The placing of the artists at varying distances from the microphone, i.e., adjusting the ratio of direct to indirect sound, is the basis of drama technique. Except for plays of very simple construction, drama producers usually require two or more quite different acoustical conditions. The simple adjustment of the acoustic ratio is not sufficient. In the past it was common practice to use several studios, each with different reverberation times and different acoustical properties. Each studio was linked with a production panel, often situated some distance from the studios. It is now more common to use one large studio in which the local acoustics can be varied at will; for example, one end of a production studio may have walls and ceiling treated with glass-wool with carpet on the floor, whilst the other end has its walls and ceiling reflecting, and its floor of bare wood. The live and dead ends can be effectively separated by two canvas curtains; also, localised acoustics can be built up by the use of special acoustic screens, which have one side of polished or painted wood, and the

other side treated with absorbing material, such as felt or glass-wool. In addition, it is now usual for such production studios to have a small adjacent so-called *narrator's* studio with observation window between it and the main studio. Built-in sound effects are provided, consisting of a stairway, the steps partially wood and partially stone, effects doors, window, water-tank, gravel-pit, etc.

In general, directional microphones are preferred except in dead studios of very short reverberation time. The omnidirectional type picks up too much ambient sound, so the bi-directional ribbon type is mostly used, and the fact that two speakers can face one another using the same microphone is of great use in dramatic production.

Most effects and much of the music incidental to a play are obtained from disk recordings. The output of the pickups is controlled and mixed electrically on the control panel. An alternative method is for the output of the pickups to be fed to a loudspeaker in the studio. The effects and music reproduced on this loudspeaker are picked up by the artists' microphone or by a special microphone. The advantage of this method is that the recorded effects are transmitted with the acoustical characteristics of the studio superimposed, and are audible to the cast in the studio.

### 3.8 TALKS

Most speech, talks and discussions originate from comparatively small studios. It was explained earlier on in this chapter that such small spaces usually have pronounced discrete natural resonances, chiefly in the lower frequency range, coinciding with the fundamental frequencies of normal male voices. Hence the bi-directional ribbon type microphone which does not respond to two-thirds of the natural resonances, i.e., those parallel to the dead plane of the microphone, is usually preferred to an omnidirectional microphone for which the speaker has to be much closer than with the ribbon type in order to minimise the effect of studio acoustics. It must be remembered that if a speaker is closer than approximately 2 feet from a ribbon microphone, the bass frequencies of his voice become greatly exaggerated (see Engineering Training Manual *Microphones*, p. 74).

### 3.9 DYNAMIC RANGE

It is well known that the volume range of all radio transmissions has to be restricted between upper and lower limits. This is frequently referred to as the *dynamic range*. The upper limit is fixed by the power handling capacities of the transmitters at 100 per cent modulation. A lower limit must also be imposed so that the quietest passages of the music or dramatic speech are not masked by receiver

noise and general interference. At present the BBC transmits a nominal dynamic range of 20 db, although in practice this range is somewhat extended. The symphony orchestra has potentially the greatest range of all types of transmissions, and can be of the order of 80 to 90 db. Its actual range, however, is seldom anything approaching this value, and even in compositions fully exploiting the resources of the orchestra the range seldom exceeds 50 db. Much of the classical music of Haydn, Mozart and early Beethoven symphonies seldom exceeds 30 to 35 db in the dynamic range. Although a 20 db requirement is laid down, the fact that a peak and not a mean reading instrument is used for measuring the programme volume helps considerably. The 20 db requirement is satisfied provided that the soft passages succeed in making the programme meter (p. 166) reach the lower permitted figure at least once every 10 or at most 20 seconds.

For modern compositions a certain amount of compression is necessary, and if it is to be carried out successfully so that the radio listener is not conscious of it, considerable skill and artistic ingenuity have to be exercised by the person at the control desk. A good knowledge of the music being played and ability to follow the full orchestral score are necessary.

The process of compression usually termed "control" is carried out according to the following broad general principles:

- (i) control potentiometer is adjusted only when absolutely necessary.
- (ii) dynamic contrast between soft and loud playing is preserved as far as possible; for example, consider a prolonged pianissimo passage followed by a sudden fortissimo; at the beginning of the pianissimo, the electrical output has to be increased by adjusting the main control but in order to preserve the contrast between the pianissimo passage and the fortissimo which follows, it is necessary to reduce the setting of the main control gradually towards the end of the pianissimo (see also Chapter 8).
- (iii) necessary alterations in control settings are less noticeable if they are made during a passage where there is a change of orchestral tone colour, such as violin chords contrasted with woodwind or brass chords.

## CHAPTER 4

### AMPLIFIER EQUIPMENT

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#### 4.1 HISTORY

BEFORE proceeding to describe the various types of amplifiers used in the BBC it is necessary to outline a little of the historical background relating to the general equipment and its use.

Until about 1938 all BBC studio and control room equipment was battery operated and to keep feeders as short as possible all apparatus was centralised in the control room (p. 82). The amplifier units themselves were large and had only moderate gains. They were designated in accordance with their functional types; thus there were A amplifiers for raising the microphone level, B amplifiers for restoring the loss in the main gain control, C amplifiers for feeding Post Office lines, and D amplifiers to raise the level of incoming lines after equalisation.

It was about 1938/9 that amplifier equipment was first designed to be supplied from the public electricity supply mains via a "mains unit". The major factor which determined the retention of batteries for so long was the extreme reliability of this form of supply, and it was not until improvements in valves, smoothing equipment, etc., justified the change that batteries were replaced by mains-operated supply units. At first the mains units took the form of large centralised assemblies but these were soon abandoned in favour of small individual units which are so much more flexible and no more costly. Moreover this arrangement allowed equipment to be decentralised and that associated with studios to be installed as part of the studio suite. With the adoption of this method it became possible to provide a separate mains unit for each amplifier, and in certain instances to build the mains unit into the amplifier itself; experience has shown, however, that it is often more economic to make each mains unit capable of supplying a number of amplifiers. Such a unit is flexible in function because it can be used for many different purposes.

The amplifiers themselves became more compact with the more efficient valves available, and fewer and more versatile amplifier types were introduced to take the place of the multiplicity of designs

which had grown up. Such amplifier types could, therefore, be used for more than one of the purposes outlined in the previous paragraph; the nomenclature has been retained, however, to designate the function of an amplifier in any given set-up.

## 4.2 BASIC FUNCTIONS

Fundamentally there are three main purposes for which an amplifier may be needed (excluding some miscellaneous requirements which will be dealt with separately). These purposes are:

- (a) To raise the volume of the programme signals.
- (b) To isolate one circuit from another or to feed one programme to a number of different destinations.
- (c) To provide an output of considerable power.

Each of these requirements will be considered separately.

### 4.2.1. *Level-raising Amplifiers*

Under heading (a) may be classed all A, B and D amplifiers, the only essential difference between them being the maximum gain wanted and the incidental requirements which arise from this difference. In general, they all work between 600-ohm terminations and have input and output impedances of this order, although there are variations. This value has been chosen as the nominal sending and load impedance for long Post Office lines and has consequently been adopted for interconnecting circuits between apparatus units on BBC stations. The major exception has always been microphone circuits, which have grown up as 300-ohm nominal-impedance circuits because this was roughly the impedance of the Reisz carbon microphone originally used. A 300-ohm input impedance has therefore been used for A amplifiers to provide a matched load to the circuit; this usually necessitates interposing a matching transformer between the microphone and the amplifier input terminals.

Before negative feedback was used the output impedance was mainly determined by the output valve; some control was available by the use of suitable transformers, but generally the desired impedances could only be obtained by this method at the expense of reduction in power output. Because of this the actual output impedance was a compromise and usually of the order of 200 ohms.

Since the introduction of feedback the tendency has been towards standardising input and output impedances at 600 ohms. The advantage of this is that a single design of amplifier can be used for many purposes and amplifiers can be interchanged without upsetting the sending levels in the circuit. The figure of 600 ohms has been chosen since (as explained above) this has already been accepted as the standard sending and terminating impedance of Post Office lines,

where the value is critical on account of its effect on the frequency characteristic of the line and its equaliser.

#### 4.2.2. *Distribution Amplifiers*

These amplifiers fall under heading (b) above, and include C amplifiers for feeding branch circuits, and monitoring or programme meter amplifiers. Such amplifiers differ from level raisers in that, first, their gain is generally small, and secondly, their input impedance is high, so that one or more can be bridged across a circuit to provide a branch without appreciably affecting the voltage level in that circuit. For this reason this class of amplifier is often referred to abroad as a *bridging* amplifier. The higher the input impedance of such amplifiers, the greater the number which can be connected in parallel across the circuit, and a figure at least twenty times the nominal impedance of the circuit is aimed at; on this basis the bridging of one such amplifier across a common point would cause a loss of 0.2 db at that point.

#### 4.2.3. *Power Amplifiers*

Amplifiers in category (c) are designed to provide considerable output power either to feed a loudspeaker unit, a public address system, a disk recording head, or a large number of headphones. Such amplifiers are generally arranged to have a high input impedance for bridging purposes (except recording amplifiers) and have a very low output impedance of the order of a few ohms. The maximum undistorted power output is very dependent upon the correct value of load impedance, partly on account of reflection due to mismatch as discussed in Chapter 2, but principally because of the characteristic of output valves which give minimum distortion when feeding a particular value of load.

### 4.3 AMPLIFIER DESIGN CONSIDERATIONS

Amplifier design is inevitably a matter of compromise and it is usually impossible to obtain the best performance in one respect without sacrificing something in another. There are however certain parameters that call for special attention, for example, gain, power output, noise, distortion, and operating impedances.

#### 4.3.1. *Gain and Power Output*

The reader should not be confused between high gain and high power output in an amplifier. High gain is required when the input is very small, e.g. the output of a microphone or a sound reproducing head, and such an amplifier is only required to deliver moderate power; the standard volume adopted for programme feed circuits

between various points in BBC premises is zero (see Chapter 2) which demands a peak output of less than 10 milliwatts. To raise the output of a microphone which may be 100 microvolts to this level may require a voltage gain of the order of 90 db. On the other hand, an amplifier required to give considerable power, say up to 10 watts to feed a loudspeaker, may only need to have a moderate voltage gain of something like 10 db.

#### 4.3.2. *Noise in Amplifiers*

The major difficulty in the design of a high-gain amplifier is the elimination of unwanted background noises such as mains hum, microphony (or "pong"), and hiss, all of which, if amplified, will cause serious interference with the programme. To begin with, there is a fundamental background hiss which it is impossible to reduce, and which arises in the electrical circuit of the microphone or other voltage source. It is due to the random movement of electrons within matter itself and amounts to approximately  $1/4 \mu\text{V}$  on a 300-ohm audio-frequency circuit. Then there are the less fundamental sources of noise such as hum and microphony. As a general rule most additional noise arises in the first valve and its circuits, so that by stepping-up the microphone output voltage which is applied to the grid of this valve by means of a high-ratio transformer, such noise becomes relatively less important. But such a high step-up can only be achieved at the expense of the frequency band which can be passed and of the constancy of the input impedance. Here, then, is one of the points where compromise is necessary in design.

#### 4.3.3. *Distortion*

No valve amplifier gives a completely faithful copy at the output of the applied input voltage, and the departure from similarity is called distortion. One form of distortion, for example, is due to the variation of gain with frequency, while another is the variation of gain with amplitude. All such forms of distortion can be reduced materially by the technique of negative feedback. Briefly, this consists of feeding a portion of the output back to the input of the amplifier in a negative sense and through a linear network, such as a resistance attenuator. The effect of this is that the linear network swamps the non-linear characteristic of the amplifier and tends to make the output a more faithful copy of the input in every respect. The improvement is obtained at the expense of the maximum possible gain but, as mentioned earlier, high gain is not difficult to provide. On the other hand, the improvement extends in other directions, for not only is the distortion within the amplifier reduced, but so also is any noise generated within it, and variations in performance due to

alterations in supply voltages or valve parameters are also much lessened.

It should be mentioned here that negative feedback so applied can only reduce distortion but never eliminate it. For all practical purposes variations of gain with frequency can be virtually eliminated by using a feedback network which is frequency dependent; this is not the same thing as a non-linear network, and such a network is very rarely used in studio amplifiers. Further information on this subject will be found in Engineering Training Supplement No. 3.<sup>1</sup>

#### 4.3.4. *Over-all Performance Requirements*

The performance of the broadcasting chain, so far as passing the whole audio-frequency band is concerned, should be just good enough for the ear to be unable to detect any discrepancy. To aim at a performance better than this would not be sound economically, and in general it may be said that a flat characteristic over-all within  $\pm 1$  db in the band 100 c/s to 5 kc/s and  $\pm 6$  db down to 30 c/s and up to 15 kc/s, represents a reasonably good amplifier performance. However, many amplifiers may be included in such a chain, so the performance of each must be much better than this, particularly if their discrepancies are all likely to be in the same direction. Figures of  $\pm 0.2$  db and  $\pm 1$  db respectively are therefore adopted as design limits for individual amplifiers, not because this is always good enough but because it is a good economic compromise.

The amount of harmonic and intermodulation distortion present in the signal is also governed by economic factors. It is difficult to quote precise figures, for the ear may be prepared to accept a relatively large amount of distortion at one frequency and yet be quite intolerant of a small amount at another. For this reason, therefore, no figures will be quoted.

#### 4.3.5. *Input and Output Impedance*

It has been mentioned above that nominal input and output impedances of 600 ohms are now regarded as standard except for the input of bridging amplifiers and the output of power amplifiers. By suitable choice of circuit in the feedback path, however, it is now possible to obtain almost any desired value at will.

The accuracy of input and output impedances is not of vital importance except for amplifiers associated with equalised lines; to meet this special requirement for the output impedance of C amplifiers and the input impedance of D amplifiers a value of 600 ohms  $\pm 5$  per cent is specified; the reactive component of these impedances is not greater than 10 per cent at any frequency between 50 c/s and 10 kc/s.

## 4.4 SPECIFIC AMPLIFIER TYPES

Detailed information regarding various particular amplifiers will be found in Technical Instructions.<sup>2</sup> It is intended here to refer only to major trends in design or to specially interesting details.

4.4.1. *Early Feedback Amplifiers*

The first BBC amplifiers to employ negative feedback (1938/9) incorporated an output stage with feedback over that stage only. The amplifier types concerned are: A/11, C/4, D/9, D/10, TV/17, TV/18, TV/19; all these amplifiers have a similar output stage in which feedback taken from a winding on the output transformer is fed to the cathode circuit of the output valve (see Fig. 30).

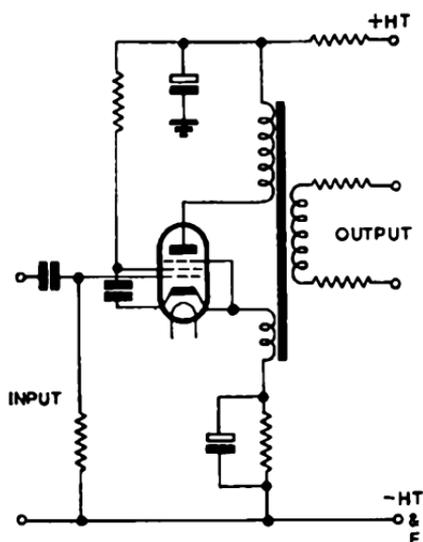


FIG. 30—Cathode-injection feedback

Another notable amplifier employing negative feedback is the Outside Broadcast Amplifier OBA/8 (and its rack-mounted companion APM/1).

With these amplifiers variable feedback is used in the first stage and forms the main gain control (see Fig. 31). The reasons for this are a little involved; first, to economise in space and components it was desirable to limit the number of stages to two; second, to raise microphone output volume sufficiently to feed a line, maximum possible gain was essential. If the gain of the first stage was fixed at a high value to accommodate weak speech, overloading would occur on (for example) the full output from an orchestra. By the use of feedback control, not only can gain be reduced at high volume,

but the valve can accept a higher net input before the over-all distortion reaches a limit that is judged intolerable.

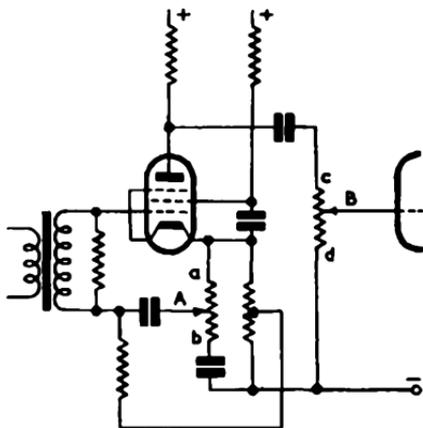


FIG. 31—Volume control by variable feedback.

*A* and *B* are ganged so that *A* moves from *a* to *b* before *B* starts to move from *c* to *d*

#### 4.4.2. War-time Amplifiers

The need for simplicity of manufacture placed severe restriction on the design of amplifiers brought into operation during the war years. This meant even employing plywood chassis and, in the circuit design, the elimination of any components difficult to make or obtain. The AL/6 output transformer, used in the standard output stage already mentioned, fell into this category. So a new range of amplifiers was designed, namely, D/11; TV/20, TV/21; LSM/6. In these amplifiers, resistance networks for feedback, and transformers not carrying direct current (or balanced in the case of push-pull), were the order of the day. In the loudspeaker amplifier LSM/6, the output transformer does carry d.c. but the rather special loading conditions and a simple form of negative feedback made a very simple transformer design possible.

#### 4.4.3. Post-war Amplifiers

As soon as war-time restrictions were eased the need for better amplifier equipment became urgent. The General Purpose Amplifier GPA/1, the line-feeding amplifier C/8, and the Miscellaneous Power Amplifier MPA/1 were, therefore, developed to satisfy the three general requirements mentioned in paragraph 4.2. As these are now the major designs it may be well to discuss each in a little more detail.

## AMPLIFIER EQUIPMENT

### (a) GPA/1

Experience with the D/11 showed that an amplifier capable of being set in three or four specific maximum gain conditions would satisfy all normal requirements for a lever-raising amplifier. By employing varying degrees of negative feedback on the first stage of a two-stage amplifier, three maximum-gain ranges are available of 80, 65 and 50 db respectively. With this arrangement background noise is reduced and the best performance ensured for each successive gain range. Since negative feedback cannot reduce the gain of a stage below unity a fourth gain range of 35 db max. is obtained by switching out the first stage altogether. Fig. 32 is a simplified diagram of the GPA/1 circuit and it will be observed that the three switches  $S1a$ ,

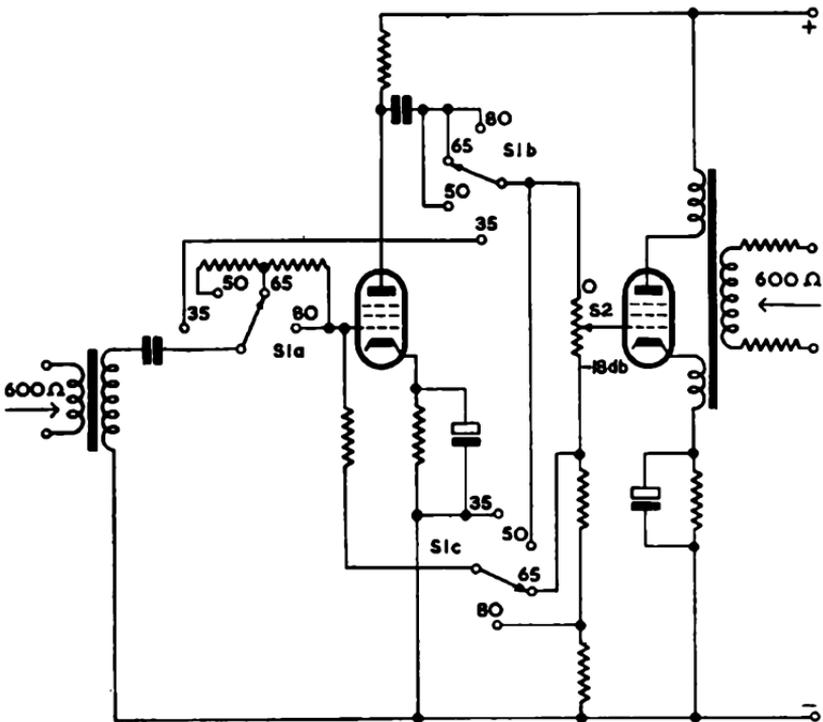


FIG. 32—Simplified circuit of GPA/1

$S1b$  and  $S1c$ , provide the adjustment of gain range. The control  $S2$  (which in practice is a stud potentiometer) provides variable gain control of 2 db per step over approximately 20 db, giving adequate adjustment on each pre-set range.

The pre-war standard output stage was reintroduced as it had proved so simple and satisfactory generally; it enables a peak output of + 20 db to be obtained in a 600-ohm load using the preferred

AC/SP3 valve and employing approximately 20 db negative feedback. The output impedance, which at the transformer terminals is low by virtue of the voltage feedback, is raised to 600 ohms and stabilised by two close-tolerance padding resistances. The impedance of the input circuit is stabilised when required (e.g., following a line equaliser) by the inclusion of a 10-db attenuator pad of high-stability resistors.

### (b) AMC/2

The need for a small, cheap, single-stage low-level amplifier was felt when the post-war design of studio equipment was undertaken. This was a case where the standard GPA/1 amplifier used in considerable quantities would prove too expensive, and the microphone amplifier AMC/2 is virtually the first stage of a GPA/1 with some modifications (see Fig. 33). Originally the only requirement was that

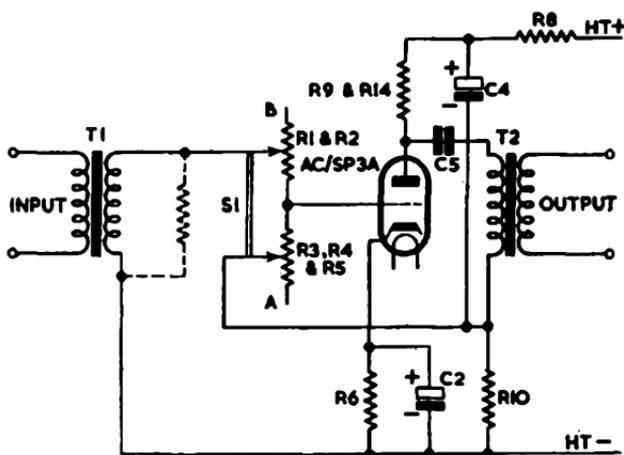


FIG. 33—Simplified circuit of AMC/2

the unit should follow a microphone and have a gain not exceeding 50 db so that a peak output of more than zero db was unlikely to be encountered. Subsequently, however, the amplifier was called upon to deliver increased power even up to zero volume, which it will do provided the 50-db gain setting is not used. The 40-db and 30-db gain settings are obtained by introducing feedback of 10 and 20 db respectively, thereby enabling the higher output to be delivered for a given amount of distortion.

### (c) C/8

The specification for a line-feed amplifier is that it should be capable of volume output of +10 db (peak output +18 db) with a

## AMPLIFIER EQUIPMENT

margin to allow for errors of control and/or line-up. This output should be distortionless even when the line impedance is reactive and falls to as low as 75 ohms or rises to 2,000 ohms. The output impedance must be resistive within close limits so that amplifiers can be interchanged without upsetting the equalised characteristic of the line. The input impedance should be of the order of 50 kilohms so that a number can be shunted across a 600-ohm circuit without introducing appreciable loss, and the gain should be fixed at 10 db. The C/8 fulfils these somewhat stringent conditions; each unit consists of two separate amplifiers fitted on one panel. In use, every line is preceded by an attenuator whose value is adjusted according to the level required to feed the particular line. Therefore, no matter

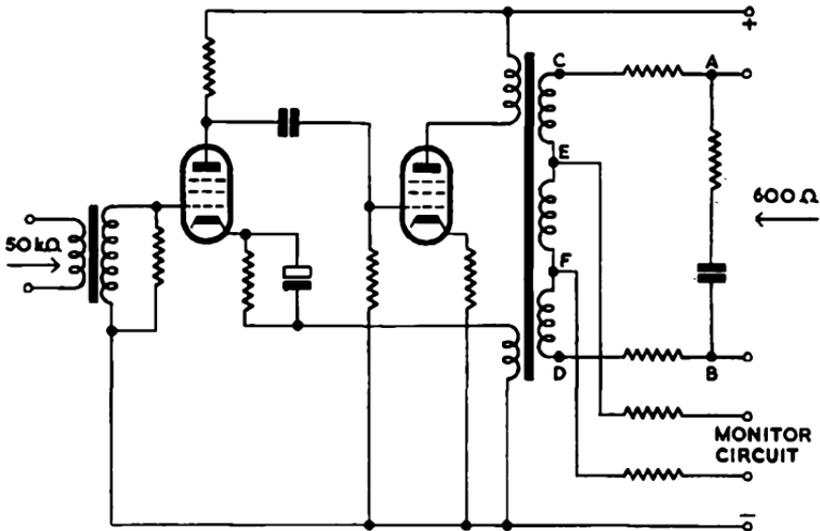


FIG. 34—Simplified circuit of C/8

what the preferred level for the line may be, the input volume to the attenuator is +10 db.

It is dangerous to monitor across the outgoing circuit because the quality is affected by the impedance of the line itself; a novel feature has therefore been included in this amplifier consisting of a separate output circuit for monitoring which, while derived from the secondary of the output transformer, is yet unaffected by variations in impedance of the load on the main output circuit.

The arrangement is shown in Fig. 34. The effects produced by taking the monitoring circuit from points *EF* instead of *AB* are:

- (i) The source impedance common to both output circuits is reduced, thereby minimising interaction between them;

- (ii) The input level to the monitoring circuit is reduced from +10 db to zero;
- (iii) The monitoring loudspeaker gives a true indication of quality at the point immediately following the last valve in the chain, because the circuit which feeds it is unaffected by the impedance of the line connected across A and B.

(d) MPA/1

This is a power amplifier which was originally intended for rack mounting. In common with the GPA/1 and C/8 it was designed to be suitable for front and back mounting on the same bay. Its main use, however, has been for feeding loudspeakers and for this purpose it is mounted in the loudspeaker cabinet. It will give an output of 10 watts to a 3-ohm or 12-ohm load with an input level of -5 db or above, and is provided with gain controls covering a range of 38 db. The circuit is quite conventional and uses push-pull output valves in Class AB; because of the magnitude and variation of h.t. power consumption of this arrangement the mains supply unit is built in as part of the amplifier.

(e) LSM/8

Individual loudspeaker units call for a self-contained amplifier which need not fulfil any special conditions peculiar to the BBC. The LSM/8 therefore is a commercial unit which has largely replaced the MPA/1; it forms part of the Loudspeaker Unit LSU/10 (p. 79) where it feeds a dual loudspeaker via a crossover filter which ensures that the power delivered in the lower part of the audio range is fed to one section of the loudspeaker unit and power in the upper part to the other.

Recording amplifiers fall into the class of power amplifiers but are specially designed for recording purposes.<sup>3</sup>

#### 4.4.5. *Amplifiers for Type B Studio Equipment*

Very recently new designs of sound equipment have been undertaken in order to take advantage of modern trends in reduced sizes of components, to enable 19-in bays to be adopted as standard, to improve the flexibility of equipment, and to replace obsolete apparatus. As part of these new designs three new amplifier types have been developed which will be suitable and sufficient for most of the needs of sound equipment (see Plate 1).

These new amplifiers differ markedly from previous designs. They are all constructed on the same basic chassis designed for quick insertion or removal; as the unit is pushed into position a multi-



PROGRESS IN STUDIO AMPLIFIER DESIGN:  
(Top to bottom) 1930, 1940, 1954

The set of three small units forms the basis  
of the new BBC Studio Equipment Type B

point plug engages with a corresponding socket on the bay, connecting both programme and power supplies. The chassis is approximately  $4\frac{1}{8} \times 6\frac{1}{8} \times 4\frac{1}{8}$  in over-all and the units are suitable for mounting singly or in groups on 19-in or 22-in bays, and are designed to satisfy both control room and studio equipment needs. The three basic types are as follows:

#### (1) GPA/4

This amplifier has an over-all gain of 73 db which is adjustable in steps of 10 db to 63, 53 or 43 db as required. It is therefore suitable for any position in the chain where a fixed amount of gain is required and virtually replaces both the GPA/1 and the AMC/2. A sub-type GPA/4A is also available which differs from the parent type only in the fact that a 10-db gain control in  $\frac{1}{2}$ -db steps is added and also in that the gain is reduced by a 3-db pad which stabilises the input impedance when the amplifier follows line equalisers.

The circuit of the GPA/4 however, differs considerably from the GPA/1 and similar amplifiers. The output stage is push-pull to reduce the size of the output transformer and a double-triode valve is used for circuit simplification. This same valve (CV455) is also used with the two halves in cascade to provide two stages of amplification prior to the output stage, and negative feedback is applied over-all. Thus a new universal standard valve is introduced which will replace the AC/SP3, a stable amplifier with considerable negative feedback is obtained, and small physical size and few components are required for the complete unit. In all respects the performance is at least as good as the GPA/1.

Two points in the design may be mentioned. The first is that a special circuit is used in the output stage whereby out-of-balance between the two sections of the output valve is automatically compensated. The other is that the coarse gain control is obtained by simultaneously reducing the forward gain and the feedback ratio so that their product is held constant; in this way the performance and stability of the amplifier is constant at all gain settings.

#### (2) C/9

The C/9 has performance characteristics very similar to those of the C/8, with one important exception. The output impedance of the C/9 is 125 ohms and the peak output available here is +28 db. When used for its normal purpose of feeding to line, padding resistances to give 600-ohm output impedance and a suitable attenuator are provided as part of the line termination equipment. This ensures that reasonably good quality monitoring can be carried out at the amplifier output irrespective of the line characteristics and without the need of a separate monitoring circuit.

Because of its low output impedance the amplifier can be used for other medium-power applications such as feeding ringmains.

The output circuit is identical in all other respects with that of the GPA/4 and a CV455 valve (one half only) is used as an input stage, again with feedback applied over-all from a winding on the output transformer to the cathode of the first stage.

### (3) MNA/3

The third amplifier in this series is the monitoring amplifier MNA/3, which is very similar in performance to the MNA/1 referred to in section 4.8.1.

The foregoing section indicates that progress in amplifier design has progressively reduced the mounting space required. This is effectively illustrated in Plate 1.

## 4.5 MECHANICAL FEATURES

When all technical equipment was centralised in one control room, that is until about 1940, all BBC amplifier equipment used under static conditions was mounted on equipment bays. The units were constructed on solid horizontal panels with components mounted on both sides, with a dust cover on each side. The standard bay was designed to accommodate panels  $22\frac{1}{8}$  in wide, the panel depths being multiples of  $2\frac{1}{4}$  in as required. Unfortunately this standard, which was once used by the Post Office, is now obsolete, and the future may show a very gradual change-over to the universal standard width of 19 in.

Since 1940 there has been a tendency to decentralise studio equipment and this has led to a change in mounting arrangements. Individual units have become smaller and have been arranged for mounting in a lockable cabinet.

At the same time, on account of the premium set on space in the post-war period, a tendency to reduce unit sizes and to economise on mounting space can be seen. Units arranged with nearly all the components on one side of the panel enable a second unit to be mounted back-to-back with the first on the same bay. The General Purpose Amplifier GPA/1 is the earliest example of this technique while the C/8 and MPA/1 are further examples; it will be noted that they are fitted with plug-and-socket terminal connections to facilitate their removal from the bay for servicing.

## 4.6 STANDARDISATION OF VALVES

In 1938, when the use of mains supply units became general, an attempt was made to reduce the valve types used in BBC apparatus

## AMPLIFIER EQUIPMENT

and so simplify the spares problem on stations. Two valves were therefore selected which would satisfy 80 per cent of the requirements of all regular equipment. One, the AC/SP3, was a high-slope pentode fitted with a reversed-helix heater; this was found suitable for both a high-gain low-level stage and for a medium-power output stage up to about 250 mW. The other, the AL60, was a power output valve capable of delivering an output of 5 watts. These two valves were therefore standardised and all designs of equipment used them wherever possible.

The introduction of miniature valves, together with the smaller sizes of components and the increasing obsolescence of the older types, now justifies the adoption of a new preferred type and the redesign of the amplifier equipment on smaller lines. These new units have been described in section 4.4.5.

### 4.7 POWER SUPPLIES

All amplifier equipment requires two basic forms of power supply, one at low voltage for heating valve filaments or cathodes and one at 300 volts or so for supply to the other valve electrodes. It has been seen that in all pre-war BBC installations these supplies were obtained from batteries and because of the difficulties of regulation, particularly in the low-voltage supply, all apparatus used on the station was centralised in a control room where large bus-bars could be accommodated. This apparatus was gradually replaced by equipment utilising the public supply mains as a source of power, using conversion units which are referred to as mains units. Each amplifier or group of amplifiers is supplied from a mains unit, and in the control room many such small units have been employed. This arrangement satisfies the dual needs of flexibility on the one hand without problems of regulation, and of reliability on the other since the failure of any one unit disturbs but a small number of amplifiers.

The size of the standardised mains unit was governed by the current-carrying capacity of the largest hard-valve rectifier, and by a process of evolution a general purpose unit Type MU/16H is now used for 90 per cent of the supply requirements. It provides up to 80 milliamps smoothed d.c. output at 300 volts together with about 30 watts at either 4 volts or 6.3 volts for filament heating.

While discussing power supplies mention should be made of the requirements for relay and other switching equipment. In the past, a 24-volt supply has been provided but modern equipment uses the more common standard of 50 volts. This is always provided by a battery float-charged from the supply mains to ensure complete reliability. Even a momentary cessation of supply might release

certain self-held relays and destroy a set of conditions which must then be re-established, with inevitable delay.

Continuity of mains supply at main centres is assured by a 230-volt 50 c/s generator driven by a diesel engine which self-starts if the public supply mains fall. Essential equipment is automatically switched over to this reserve supply.

For outside broadcasts use is made of the public supply mains wherever possible, but a spare amplifier fed from batteries is always provided as a safeguard against mains failure. On locations where no mains are available both the regular and stand-by amplifiers are battery-fed. The batteries consist of 6-volt accumulators for l.t. and 250-volt dry batteries for h.t. made up from a number of smaller units.

#### 4.8 SPECIAL AMPLIFIERS

Several special amplifier types require separate mention. In this category fall programme meter and monitor amplifiers, limiter amplifiers and testing equipment.

##### 4.8.1. *Programme Meter Amplifiers*

The controlling of programmes so that the permitted dynamic range is not exceeded is not possible with the ear alone as a guide. Some visual indication of programme volume is therefore required, preferably with a logarithmic scale. The BBC uses a peak programme meter for this purpose; it consists of a fast-acting meter which must be driven by a programme meter amplifier and it is this amplifier which must produce at its output a direct current which is proportional to the logarithm of the peak input programme amplitude. The ratio of *time of rise* and *time of fall* of this current has been settled by experiment (Chapter 8).

The basic circuit for amplifiers used to drive a programme meter is shown in Fig. 35.

An amplifier stage feeds a full-wave diode rectifier circuit, the source impedance of the stage and the integrating capacitance being controlled to give a charging time constant of 2.5 milliseconds, while the leak resistance  $R$  across the capacitance  $C$  gives a discharge time constant of 1 second. Therefore a potential which is proportional to the programme amplitude appears across the capacitance, and this potential rises relatively quickly but decays slowly. A current, varying approximately as the logarithm of this voltage, is produced by means of a variable- $\mu$  valve  $V3$ ; to get the logarithmic characteristic, however, the potential is applied to the grid in a negative sense and the anode current falls from a pre-set value as the signal

increases. For this reason the programme meter itself must register zero signal at full scale deflection and maximum signal at zero current; to meet this condition the BBC uses a programme meter having its mechanical zero at the right-hand side. Adjustment of the zero current and of the valve characteristic is obtained by variation of the electrode potentials.<sup>4</sup>

This basic peak programme meter circuit is to be found in the PPM/2, MNA/1, OBA/8, OBA/9 and TPM/3 as well as some other amplifiers. The second only is a genuine monitoring amplifier and is a development of the PPM/2 in that an audio output circuit is provided. In the OBA/8 and OBA/9, the programme meter and

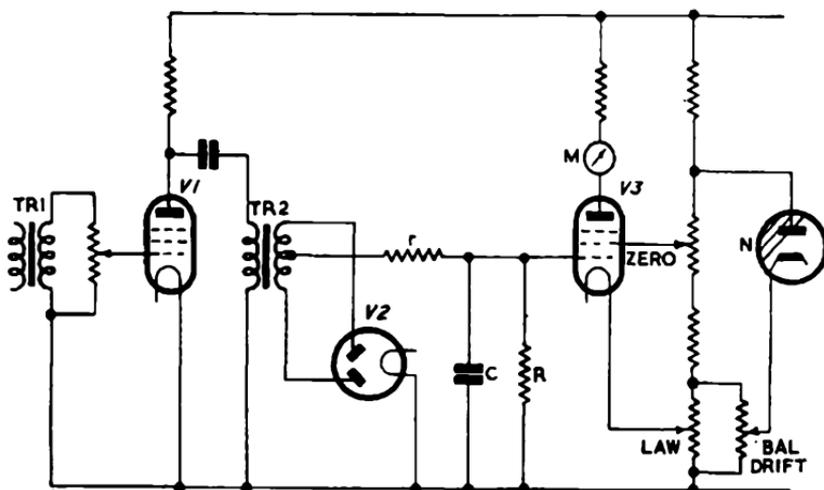


FIG. 35—Basic programme meter circuit as finally developed

its amplifier are incorporated in the main amplifier unit to save space and weight. The TPM/3 is really a test unit and is virtually a PPM/2 with a high-gain amplifier of high input impedance preceding it, so that it can be used for bridging across low-level points for test purposes. Other test equipment is described in Chapter 7.

#### 4.8.2. Limiter Amplifiers

However carefully a programme is controlled there is always the risk that a sudden peak may exceed the permitted maximum level. To protect the transmitter and to prevent distortion a limiter amplifier is included in the chain, generally at the transmitter. This amplifier normally works at a fixed gain but if an excessive peak occurs the gain is almost instantaneously reduced and recovers either quickly or slowly thereafter, according to the operating conditions.

The effect is achieved by employing a gain stage using a variable- $\mu$  valve from the output of which is derived a rectified potential proportional to the output signal when the latter exceeds a certain pre-set maximum. This potential is used to bias the gain stage so that when a signal above the predetermined level is received, the gain is rapidly reduced until the output signal only very slightly exceeds the pre-set value. The anode current falls when bias is

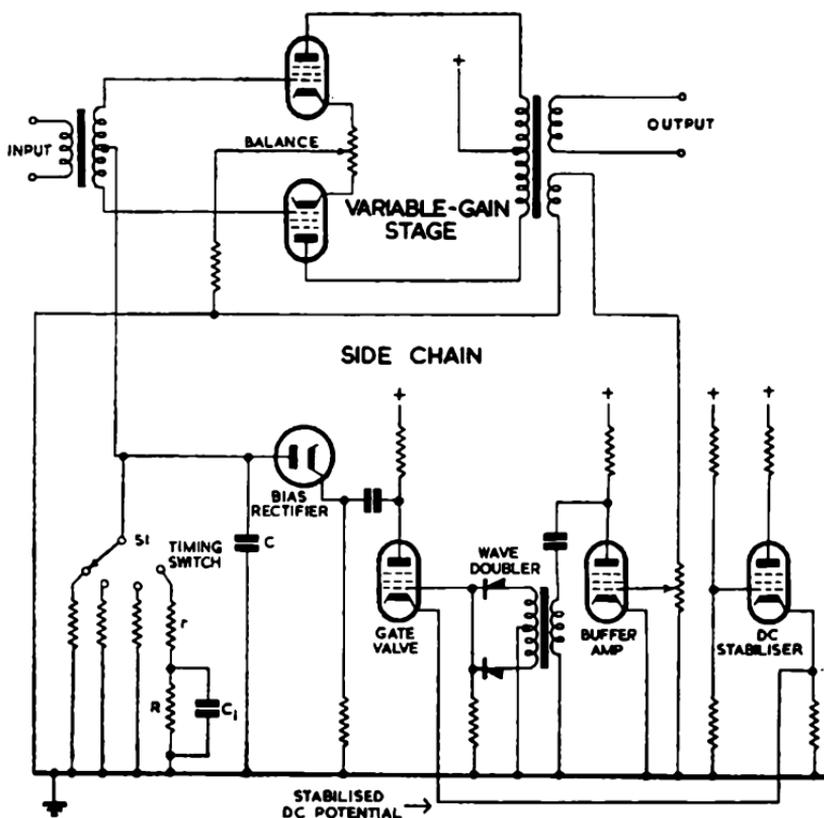


FIG. 36—Simplified circuit of limiter

applied and to prevent this change of current causing noise on the programme the gain stage consists of a carefully balanced push-pull pair (see Fig. 36).

The heart of the side chain which provides the bias is the gate valve which passes nothing to the bias rectifier until the signal exceeds the cathode bias of some 20 volts provided by the d.c. stabiliser valve. A portion of the signal at the output of the main (variable-gain) amplifier is fed via a buffer valve to a full-wave

rectifier with no integrating capacitor. The gate valve is therefore fed with half-waves which are proportional to the positive and negative cycles respectively in the signal. The proportion of the output signal which is fed to the side chain determines the output above which the gain of the main amplifier is reduced. The bias rectifier charges its integrating capacitor rapidly (as in the case of the programme meter amplifier) and this potential is used to control the gain of the main amplifier. The potential leaks away at a rate determined by the setting of the timing switch *S1*.

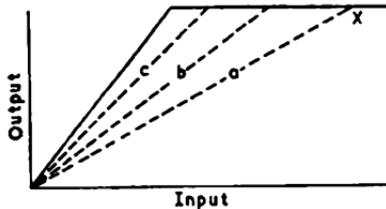


FIG. 37—Characteristic gain of limiter

The input/output characteristic of the device is as shown in Fig. 37. As the input is slowly raised from zero the output is at first proportional to it as shown by the sloping part of the full line. Above the critical level, however, no further appreciable increase in output occurs, even if the input is increased to the point *X*. If the input is now reduced, the dotted characteristic marked *a* is followed, which corresponds to a linear characteristic of lower gain than originally. If the input is now maintained at a low value, the characteristic is slowly restored (via the conditions marked *b* and *c*) to the original state. It will be understood that no harmonic distortion is introduced except during the time when the gain is being reduced; this condition lasts but a small fraction of a second and so distortion is inaudible.

The effect of the limiter depends very much upon the speed with which the gain is restored. If a very high leak resistance is selected by the timing switch, resulting in a very slow restoration, the unit acts almost as a level-setting device adjusting the gain so that on peaks the modulation of the following transmitter just reaches 100 per cent, or the maximum modulation level of which that transmitter is capable. On the other hand, if the restoration time is fast (of the order of a second or so) and if the input to the device is raised above normal, it acts as a compressor reducing the peak syllables on speech and raising the level between them.

Various problems occur in the design of such a unit but mention will be made of one only. A single unintentional overload such as a drum-beat has the effect of reducing seriously the subsequent programme level with disastrous effects on an orchestral work. This

can be overcome by providing a special discharge circuit  $rRC_1$  (Fig. 36) for the main bias integrating capacitor  $C_1$  is about 20 times larger than  $C$ , and  $r$  allows any potential across  $C$  to discharge quickly into  $C_1$ , thereby restoring the gain quickly; if, however,  $C$  gets repeatedly charged, a potential will ultimately be developed across  $C_1$ , reducing the gain which will now only be restored slowly.

Typical examples of the above type of amplifier are the LIM/2 and its modern counterpart the LIM/5. Other types of limiting (or peak-chopping) amplifier are instantaneous in operation and cut off the peaks of all waves above a certain level. These are sometimes used as part of a transmitter to give complete protection from instantaneous over-modulation, which the limiter described above does not always do.

A special type of peak-chopping amplifier is used on some of the European Services in order to improve the intelligibility of speech in the face of noise. The frequency response of this circuit is deliberately distorted and the programme is then fed through the peak-chopping amplifier at such a level that very severe limiting occurs. The result, surprisingly, is more intelligible in the special conditions postulated than a normal broadcast would be.

#### 4.8.3. *O.B. Amplifiers*

Outside broadcast amplifiers differ from those used at permanent centres, mainly because they must be easily portable and must embody a number of features which are normally provided by other items of equipment in studio premises. For example, the amplifier must not only raise the level from microphones, but it must be capable of feeding direct to line. Moreover, it must contain a (virtually) continuously adjustable gain control for programme operation and incorporate a programme meter.

The OBA/8 has already been referred to and is an example of a design to meet this specification which has remained unchanged for some 15 years. The requirements at many O.B.s have expanded in this period and much auxiliary equipment is required to provide the facilities needed. Accordingly a new design of equipment, the OBA/9 and associated units, has recently been introduced which replaces the OBA/8 (see Plate 2).

#### OBA/9 and Associated Equipment

The facilities required for O.B. working vary from one occasion to another and (in addition to microphone fading and programme control) embrace such matters as cueing to commentators and artists, telephone communication to the studio centre, programme

feeding to public address installations and other circuits, the use of mixed types of microphone, and programme cueing usually obtained by radio pickup.

The OBA/9 amplifier and its associated units are designed to meet these requirements; they are constructed so that they can be carried *en bloc* on a special two-wheeled trolley for easy transport and very easily set up and operated in this condition. The complete set comprises one or two four-channel mixer units with alternative input impedance for ribbon or moving-coil microphones, three 50-ft. drums of microphone cable, two OBA/9 amplifiers with built-in programme meter and main gain control, and a switching unit providing alternative programme routing, telephone switching, and cueing facilities.

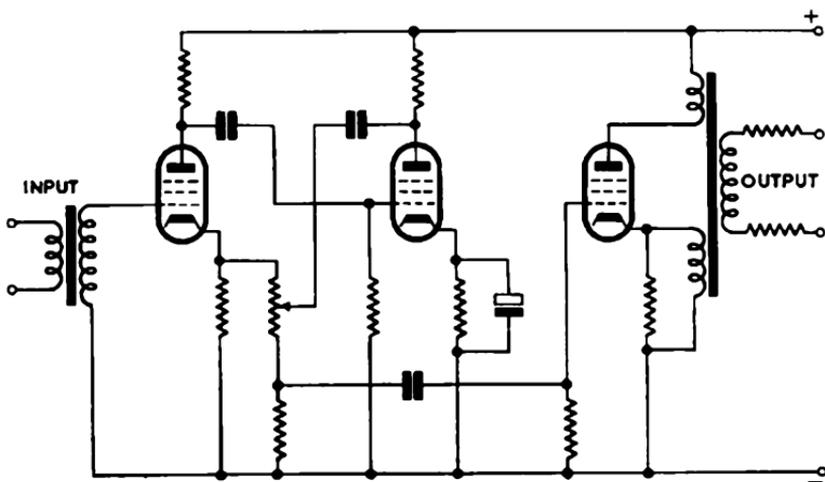


FIG. 38—Simplified circuit of OBA/9  
Showing ingenious gain-control arrangement

A trap valve and L.S. amplifier unit which provides three subsidiary programme feeds and a medium-power output to feed the loud-speaker is also provided, and finally a power supply unit provides both mains and battery supplies.

The equipment is arranged so that interconnection of units is quickly made by plugs and cords and so that additional equipment can be taken on those relatively few occasions when additional features are necessary. The bulk and weight of the equipment is such that it can be carried in a saloon car with wide doors.

Two points of the design call for comment. The main gain control of the OBA/9 uses a special circuit (see Fig. 38) which not only applies negative feedback to the first stage but also gives a potentio-

## STUDIO ENGINEERING FOR SOUND BROADCASTING

meter effect on the final stage without the necessity of a ganged control. The other point is that arrangements are made to "reactivate" the dry h.t. battery at times when the mains supply is in use but the batteries are not, and it is found that this system prolongs the life of such batteries very greatly.

### REFERENCES

1. BBC Engineering Training Supplement No. 3. Harmonic Distortion and Negative Feedback in Audio-frequency Amplifiers.
2. BBC Technical Instruction S.3. Audio-frequency Amplifiers.
3. BBC Technical Instruction R.1. Disk Recording Equipment, Type D.
4. BBC Engineering Training Supplement No. 6. Programme Meters.

## CHAPTER 5

### BROADCAST PROGRAMME SOURCES

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#### 5.1 INTRODUCTION

THE sources of programme feeding the broadcasting chain fall broadly into three main groups. First, there are the rehearsed programmes which take place in studios designed for the purpose; secondly, there are topical transmissions which are generally outside broadcasts; and thirdly, there are programmes which have been previously performed and recorded. It often happens, however, that two or more different kinds of source are used in one programme; for example, recorded material may well form part of a production the bulk of which is being transmitted live, or an outside broadcast may well consist of a carefully rehearsed programme which for various reasons cannot be performed in a studio. Although each of the sources classified above requires distinctive equipment peculiar to its needs, there is considerable overlapping of function so that the design of the equipment must be versatile in application.

#### 5.2 BASIC SOURCES OF PROGRAMME

Probably the major part of broadcast material originates in the first place from studios which are designed and built specially for broadcasting. These studios, however, have to satisfy a variety of requirements which depend upon the type of programme to be transmitted and these requirements are often mutually opposed. For example, the size of a studio required for broadcasting a small orchestra would be excellent for a talk so far as acoustic conditions are concerned, but for the psychological comfort of the speaker and for the talk to achieve its best effect the intimate atmosphere of a small room must be obtained.

The types of studios fall, therefore, into four main classes.

##### 5.2.1. *Talks Studios*

As stated in a previous chapter, these are generally small, intimate studios, and the technical facilities required in them are usually quite simple. They generally contain no more than a table, two microphones, and cue lights, and the acoustics are designed for their particular purpose (Chapter 3).

### 5.2.2. *General Purpose Studios*

Studios larger than the normal talks studio are required for such programmes as chamber music, Children's Hour, or large discussions, and still larger studios for small orchestras, dance bands and some plays. In the 1930's the policy was to build and equip a studio for the particular purpose for which it was required and *Variety Studio* or *Military Band Studio* were commonplace titles. When studio space was at a premium during the war all the medium-sized studios had to be used for various types of programme and they therefore became known as General Purpose Studios; this use of studios for different purposes has continued ever since because of the obvious economy in space.

### 5.2.3. *Dramatic Production Studios*

The technique of dramatic production has been discussed fully in Chapter 3. The original practice of using a number of studios for each dramatic production has given way to a single studio, with an echo room when necessary.

### 5.2.4. *Audience Studios*

It has become the practice, especially in the case of variety productions, to include an audience in the studio in order to create an "atmosphere". Such an atmosphere is of very great assistance to some artists, particularly comedians, and it also gives the listener a sense of participation and excitement. The presence of an audience means that studios approximating in size to concert-halls or theatres are needed. In many cases disused halls have been taken over and equipped for broadcasting on a permanent basis, while sometimes public halls normally used for other purposes are similarly equipped on a semi-permanent basis.

### 5.2.5. *Subsidiary Sources of Programme*

Besides the output from the various studio microphones it is often necessary to mix into the programme excerpts from O.B. points or from recorded disks. The former are obtained by routing the O.B. contributions to the studio cubicle and treating them virtually as additional microphone channels, while the latter are obtained from a gramophone bank located in the studio cubicle.

The gramophone bank usually consists of two or three turntable desks; each desk (known as a TD/7) has two turntables and pickups. The output from each pickup is controlled by a fader on the gramophone desk and the mixed output is fed to the main control desk and treated as another channel.

### 5.3 STUDIO CUBICLES AND TECHNICAL FACILITIES

It is essential that means be provided to produce and present artistically the programme originating in a studio, and to that end each studio is provided with a sound-proof cubicle adjacent to it.

The cubicle is treated acoustically so that the programme may be monitored under good listening conditions, and a control desk is provided on which all the necessary technical controls are grouped. A large double glass window is set in the wall between the studio and the cubicle, and the control desk is sited so that an operator and a producer can watch the performance through the window while operating the controls. In addition to facilities for balancing the outputs from the studio microphones and the control of volume level (see Chapter 8), means are provided for cueing the artists, controlling artificial echo, introducing effects, pre-fade listening to outside contributions, and communication to the central control room and to the outside sources of programme.

Complete artistic and technical control of the programme at this point is an axiom of BBC technical policy and is referred to as "control at source". The basic facilities which must be provided to carry out these functions may be grouped as follows.

#### 5.3.1. *Fading of Microphone Sources*

In the very simplest talks studio it might be possible to operate with one or two microphones only which could be switched in or out of circuit as required. In practice it is usually essential to be able to "fade in" the microphone or other channel gradually, and a variable attenuator or *fader* is provided for this purpose. Each microphone feeds into such a fader and in some cases interconnection facilities are provided so that either the relative association of microphones and faders can be interchanged, or other circuits such as alternative microphones or O.B. sources can be substituted. The faders themselves are stud-type variable-attenuator units whose rotation is limited to 180 deg to permit a quick fade-out with one easy movement of the wrist.

Besides the simple operation of fading microphones in or out it is often necessary to mix the outputs of various microphones in different proportions, the resulting mixture being controlled as a whole on a main fader. The latter is most often the main gain control on which the over-all dynamic range is adjusted as mentioned later, but in some large installations a *group fader* is provided which enables a pre-set mixture to be faded in and out at will.

For many years, the fading and mixing of microphones was done immediately following the microphones in the circuit and therefore

at a very low level. For this reason simple series resistance faders were used since no additional basic loss could be tolerated, and this resulted in changes in level on one microphone circuit when another microphone was faded up; these operations therefore must be kept to a minimum on this type of equipment and all dynamic control carried out on the main control. In later equipments, such as the Type A, an amplifier is introduced in each microphone circuit and the mixing is therefore done at higher level where some basic loss is not important. This permits constant-impedance mixing which overcomes the difficulty of changes in levels mentioned above, and makes less stringent demands on the faders to maintain a good signal-to-noise ratio.

### 5.3.2. *Dynamic Control*

To achieve the dynamic control mentioned earlier, the studio manager must adjust the over-all gain of the chain in a manner indicated by experience as shown in Chapter 3. For this purpose he is provided with a peak programme meter which indicates the momentary peaks of programme and enables him to adjust the main gain control so that they do not exceed the maximum level permitted (6 on the meter). The main gain control is similar to a fader but covers a greater range of attenuation and has a greater angular movement. It is so graded that a smooth fade-out to inaudibility can be obtained even on a very loud programme.

### 5.3.3. *Monitoring*

The basic requirements for source monitoring are covered in detail in Chapter 8 and it is only necessary here to state that a peak programme meter and a good-quality loudspeaker unit are provided in the cubicle for visual and aural monitoring respectively. In addition to this main check on the programme, facilities are required so that any contribution from outside the studio may be listened to before it is faded into the programme. This is usually called *pre-fade listening*; in many cases it is done by the provision of a special monitoring circuit from the control room, but in modern equipment more convenient arrangements are provided.

The loudspeaker which is used for aural monitoring is one of the most important items of studio equipment. Ideally, it is desirable to use one type of loudspeaker only in all monitoring areas so that variations in quality do not occur due to differences in loudspeaker types. If advantage is to be taken of any improvement in loudspeakers, therefore, it means the changing of a large number of units; such a change is not only costly but can only be done gradually, so that more than one type will be in use concurrently.

For a long time a single 12-in cone loudspeaker was used in an open-back baffle as a standard unit for source checking. Advances in loudspeaker and loudspeaker cabinet design have led to the introduction of a new unit known as the LSU/10. This unit is capable of wider and smoother frequency response, lower intermodulation distortion, and more effective distribution of the higher audio frequencies radiated.

The assembly consists of the cabinet, the loudspeaker, a power amplifier, and a frequency-dividing filter. The cabinet is a ported unit and enables a smooth response to be obtained without cabinet resonances down to the lowest frequencies of interest. The loudspeaker itself consists of two parts; one is a 15-in cone unit capable of good reproduction over the lower part of the audible range; the other is a cellular horn unit mounted at the centre of the cone and reproduces the higher frequency spectrum from about 1,200 c/s to 7,500 c/s. The amplifier, which is housed in the cabinet, is capable of a peak output of 10 watts and its output circuit is divided by the filter unit so that the low frequencies are fed to the cone section of the loudspeaker and the upper frequencies to the horn section.

These heavy equipments are, of course, quite unsuitable for O.B. use, so that in this field a small commercial 8-in cone loudspeaker unit is used. This is mounted in various types of baffle or cabinet and has provided a satisfactory compromise between quality on the one hand and size and weight on the other. During the war years a very simple unit consisting of a 10-in cone loudspeaker mounted on a flat baffle was introduced in considerable numbers for austerity reasons. Many of these units are still in use in studios and elsewhere.

### 5.3.4. *Communications and Cueing*

An important group of facilities is covered by this heading; in fact, impairment of these features may cause a programme failure no less serious than the failure of the programme circuits themselves. The primary requirement is a simple arrangement whereby the artists in the studio can be given a signal to start or stop. Clearly this cannot be done aurally so a system of green cue lights is used; the lights, which are usually mounted on a stand adjacent to each artist or group of artists, are controlled by keys on the cubicle desk operated either by the studio manager or by the producer. In the larger studios several cue lights are fitted and operated either individually or in a group. The signalling codes vary from one production to another.

During rehearsal the producer will frequently wish to give instructions to the artists and, to save continual excursions into the studio, a talk-back circuit is provided, comprising a microphone in the cubicle and a loudspeaker in the studio. This circuit is energised by

a non-locking key and safeguards are provided so that it cannot be inadvertently used when the studio is on transmission.

Frequently important parts of a programme obtained from recordings or outside sources are inserted between excerpts from the studio itself. In order that the artists shall be able to appreciate the programme as a whole and to assess the "atmosphere" (a matter acknowledged to have a great bearing on the quality of an artist's performance) these inserts are reproduced in the studio itself through the talk-back loudspeaker. It is, of course, essential that the loudspeaker is silenced when any microphone in the studio is "live" or a howl would immediately develop, so all the microphone faders are fitted with an auxiliary switch to ensure that this condition is satisfied. The arrangement is shown theoretically in Fig. 39; the *L.S.*

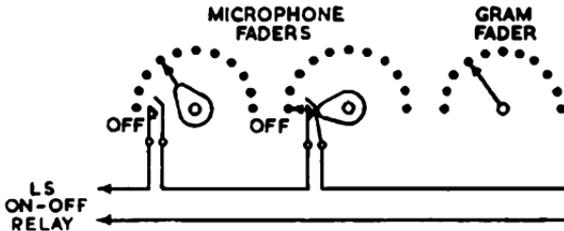


FIG. 39—Detail of loudspeaker muting circuit.  
When the fader is moved from the "off" position, the programme input to the speaker is automatically broken

circuit is completed only when all the microphone faders are in the "off" position.

Very recently facilities have been introduced in certain studios enabling effects and other material on gramophone disks to be reproduced in the studio itself and picked up by a studio microphone. In this way the artists hear these effects even when their own microphone is "live". This tends to help them in their performance, but sometimes, it should be said, to the detriment of the "effect" as heard by the listener. On the other hand, some acoustic effects can be obtained by this means which cannot be obtained by direct reproduction, e.g. the illusion of a band marching into the distance.

It is often desirable that artists in the studio should hear the close of the previous programme in the service to which they are about to contribute. This form of cueing is achieved by providing a feed to all studios from all the major transmission chains to which the studio may contribute. By selecting the appropriate circuit and feeding it to the studio and/or cubicle loudspeaker, all are fully aware of the progress of events.

Communication between the studio cubicle and the control room, continuity suite or recording room to which the studio is feeding its

programme is also essential. This is provided in two forms, namely a programme starting light cue and a telephone circuit. The light cue system provides a circuit whereby the continuity suite or control position can illuminate the red lamps in the studio suite and another circuit whereby the studio manager can light a green lamp and sound a buzzer at the control point. The system is operated as follows. Immediately before programme from the studio is due to commence, the continuity engineer gives a flashing red light signal to the studio, warning all in the studio and cubicle to be quiet and ready to start. The studio manager acknowledges this by pressing his "buzz-in" button, indicating that the studio is ready; this operates a buzzer momentarily and lights the green light associated with the particular channel concerned in the continuity room. The lamp stays lit till the engineer extinguishes it by an operation of his key; then, as a signal to the studio to start, he switches on the studio red light. The latter remains lit so long as the studio is "on the air" and is only extinguished by the continuity engineer when he has faded the studio out.

For all other communication with the studio the engineering telephone circuit is used. This normally terminates on a central position in the control room but when the studio output is selected by a continuity or control position, the telephone circuit is transferred to that position also. In some large studios a second telephone circuit is provided which can be extended via the control room to any O.B. or other source which is being routed into the studio concerned as a contribution to its programme.

When a studio is being used solely for recording, the producer often finds it advantageous to interrupt and re-record portions of the programme not considered satisfactory. To reduce time wastage and the consequent dislocation of the operational recording schedule, it is always arranged that the studio manager or producer can interrupt the programme by superimposing talk-back upon the programme circuit. The recording engineer will hear this instantly and can take immediate action according to instructions.

### 5.3.5. *Echo and Sound Reinforcement*

One of the effects most commonly required in dramatic production, and often also in other types of programme, is artificial reverberation. As has been explained in Chapter 3, this is commonly referred to as echo. Strictly speaking this is a misnomer because echo implies a focused reflected sound which reaches the hearer after the same sound has been heard directly; the time lag between the original sound and its echo depends on the difference in path length between direct and reflected sounds. The effect required is

*reverberation*, which implies a succession of reflections from the walls of a room which gradually die away at a rate depending upon the total absorption of the structure and its furnishing. The term echo has, however, become established by long usage and is more convenient for our purpose.

Echo then, or reverberant sound, is produced by routing the output from the particular microphone or microphones to a loudspeaker in an echo chamber, i.e. a bare room with non-absorbent walls, ceiling and floor. A microphone in this room picks up the reverberant sound which is reintroduced into the programme chain; the original and the echo signals are blended on the studio control desk in the proportion required to produce the desired effect. Precautions must be taken to prevent the mixed signal from being fed to the echo chamber, otherwise coupling between the echo microphone and loudspeaker would produce a howl.

It is often necessary to provide facilities for sound reinforcement in audience studios. This implies feeding the programme to loudspeakers fitted at strategic points in the auditorium to improve the listening conditions, in the manner of a public address system. Here again there is danger of setting up a howl due to feedback from these loudspeakers to the microphones themselves. It can be avoided by judicious siting of the loudspeakers and regulation of the volume level fed to them.

#### 5.4. STUDIO TECHNICAL EQUIPMENT

The technical equipment used by the BBC for providing the facilities outlined above can be divided into three main categories in accordance with historical sequence.

In the years prior to 1940 the only technical equipment located in the studio and cubicle were the microphones, the loudspeakers and the controls mounted on the control desk. All the amplifier equipment was centralised in the control room and connected to the appropriate studio as required.

In the war years austerity studio installations were undertaken and for this purpose outside broadcast equipment was used; this incorporated amplifying, mixing, monitoring and communication equipment and was mounted in the studio control cubicle. Each studio, therefore, became a self-contained unit, and was virtually independent of a control room.

From 1945 onwards a new design was introduced which provided many more facilities than did the austerity equipment. The decentralised principle was maintained since war-time installations had shown the operational advantages of using studios for rehearsal without having to set up circuits in the control room.

5.4.1. *Pre-war Studio Equipment*

Except in certain regional premises very little of the pre-war type of equipment is still in use, and only brief reference will be made to it.

Microphone and gramophone sources were mixed at low level in the studio cubicle using unbalanced circuits; the output of the mixer was fed via an announcer's switch in the studio to an A amplifier in the control room and back to the main gain control in the cubicle. The output from this control was fed again to the con-

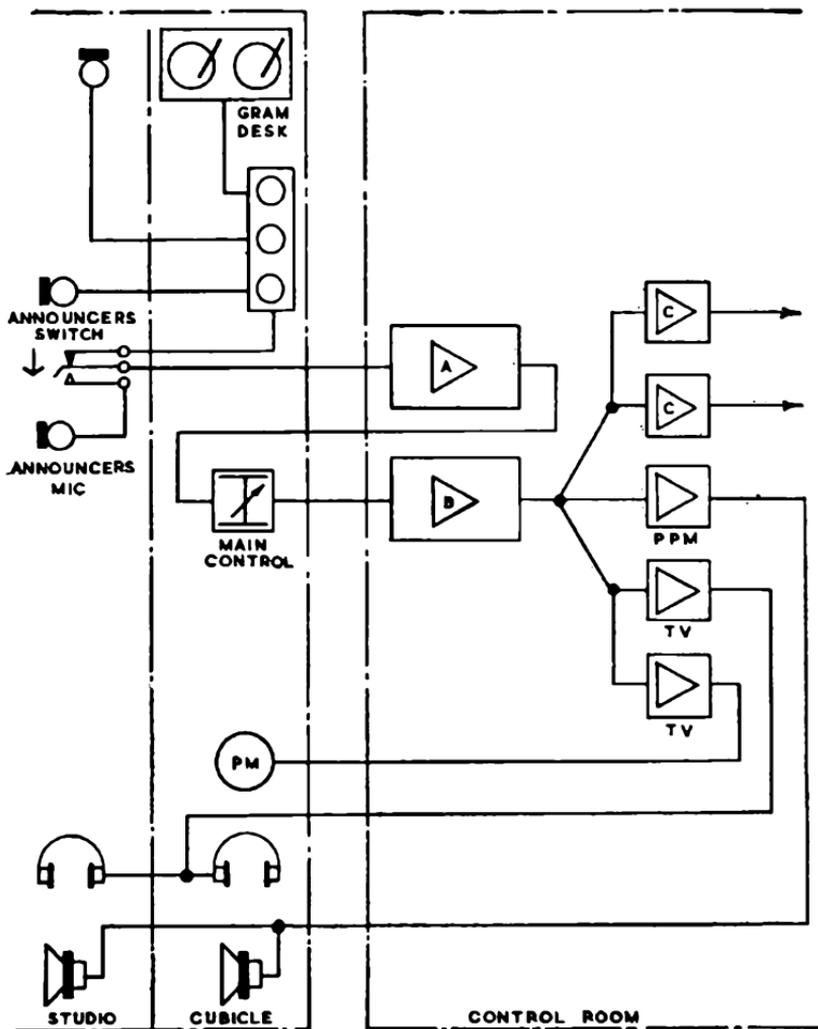


FIG. 40—Pre-war studio equipment: block schematic diagram

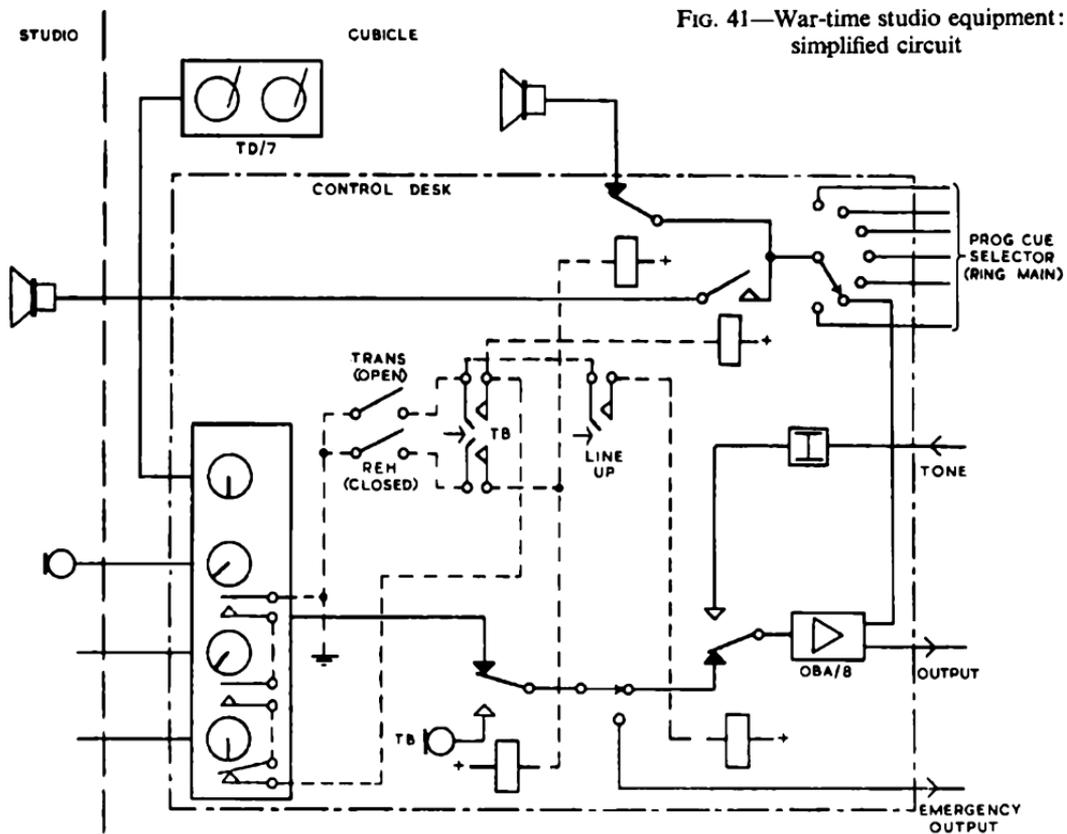


FIG. 41—War-time studio equipment:  
simplified circuit

trol room where it was connected to the input of a B amplifier which might be available. The output of the B amplifier divided into three main paths, viz. distribution to outgoing lines via C amplifiers, feeds to headphones and loudspeakers in the studio and cubicle via trap-valve amplifiers, and a feed to a programme meter in the cubicle via a programme meter amplifier (see Fig. 40). All these circuits had to be set up as required.

For the production of plays, a *dramatic control panel* was arranged so that the output of the A amplifier of any studio could be plugged to one of its several control channels; a second output circuit from the A amplifier was available for echo effects. Moreover, talk-back circuits were provided between the panel and any studio connected to it. These facilities satisfied the multi-studio technique of dramatic presentation in use at that time.

This arrangement of studio equipment required the assistance of the control-room staff every time a studio was wanted either for transmission or rehearsal and was operationally cumbersome.

#### 5.4.2. War-time Studio Equipment

The necessity of transferring studios to areas deemed relatively secure during the war required an installation procedure which could be carried out extremely rapidly. To satisfy this demand, O.B. apparatus provided a ready answer since this equipment was designed to provide all the technical requirements for a programme source under temporary conditions.

The OBA/8 amplifier and its associated Mixer, MX/18, and Supply Unit MU/3A, which had been designed just prior to the war, were therefore mounted on a simple wooden desk with suitable key and relay circuits to provide additional features such as talk-back, cue circuits and other special switching. The circuit arrangement is shown in Fig. 41. This desk, together with one or two gramophone turntable desks TD/7 and a simple baffle-mounted loudspeaker, provided all the essential technical furniture for a studio cubicle.

It will be seen from Fig. 41 that with the above equipment mixing is carried out at microphone level before any amplification is introduced, in accordance with previous practice. Other points to which attention may be drawn are, first, that the loudspeaker in both studio and cubicle may be used to receive cue-programme from the ring main prior to transmission and be switched to the studio output when the transmission has started; secondly, that talk-back, which uses the main amplifier, is rendered inoperative on transmission by operation of the transmission/rehearsal key; and finally that an emergency output line to the control room fed from the mixer output provides some insurance against failure of the main amplifier or its power

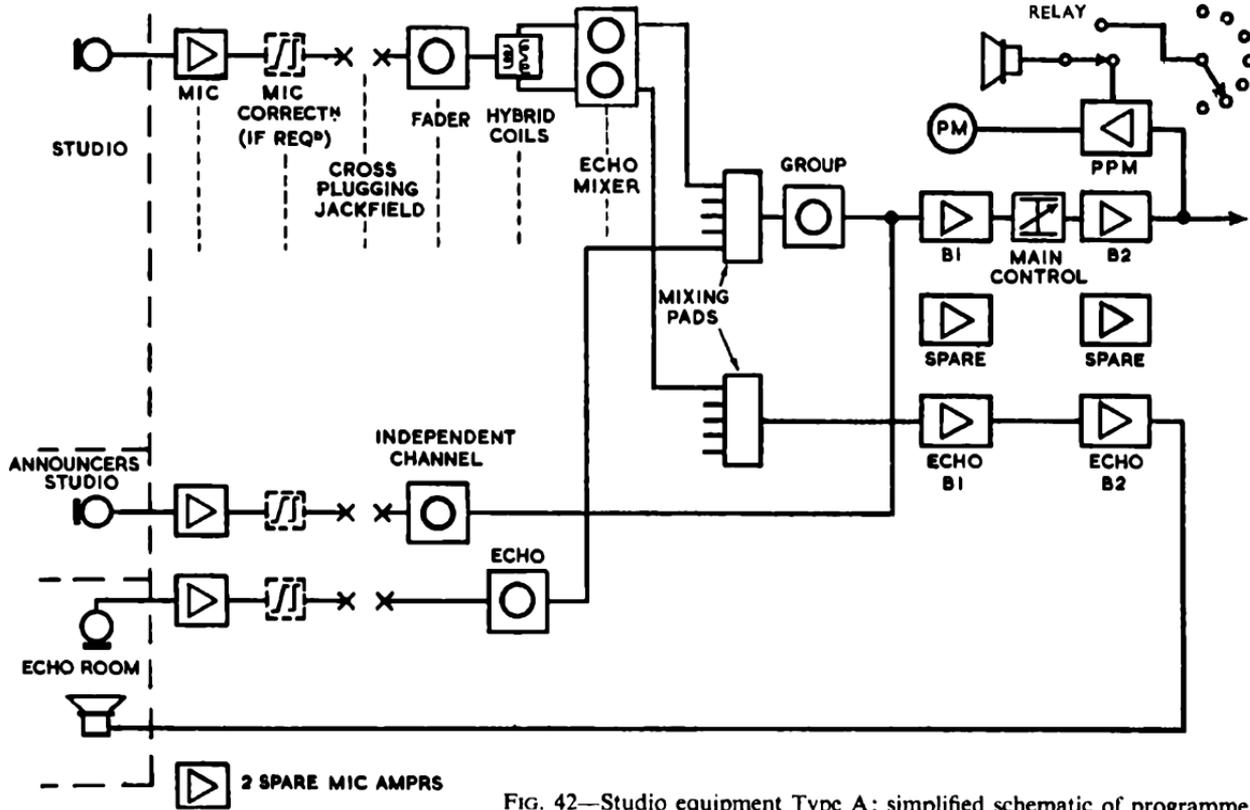


FIG. 42—Studio equipment Type A: simplified schematic of programme circuit

supply. The equipment is described more fully in Technical Instruction ST.1.

The major advantage of this austerity equipment is that it is self-contained and the studio can be brought into use for rehearsal without reference to the control room at all. The disadvantages are the variation in level inherent with low-level mixing, the inadequate emergency facilities, the inconvenient position of the controls, the limited facilities available, and finally the somewhat untidy appearance.

#### 5.4.3. *Studio Equipment Type A*

In this equipment an attempt was made not only to provide the facilities which were felt to be lacking and to improve the performance generally, but also to standardise the equipment so that big or small studios would be catered for by variations in assembly of otherwise standard units. This arrangement was designed to speed manufacture and to reduce delay in equipping any particular studio once the requirements were known.

From a circuit point of view, the major feature is that an amplifier is provided in every microphone or gramophone channel prior to the mixer. Since the mixing can then be done at high level the inherent loss when constant-impedance faders are used can be tolerated and the problem of level variations therefore removed. Secondly, since the equipment is decentralised from the control room, considerable care is taken to provide spare equipment which can be quickly brought into circuit by a studio manager. Each amplifier can be replaced by a spare merely by the operation of a key and all the faders are quickly demountable for interchange or replacement.

The technique of single studio working which had been adopted to meet war-time circumstances required echo facilities in varying degree on the output of each microphone. This is provided by the use of hybrid coils to avoid the possibility of howl-round previously referred to, and by the provision of a pre-set control by which the ratio of echo can be controlled on each microphone circuit. The feature of the hybrid coil which enables this to be done is that two output circuits can be fed from a single input but each output is immune from voltages fed into the other (see Chapter 9).

Facilities for cross plugging any microphone or gramophone to any fader are provided; these enable a logical sequence of faders to be used on a complicated programme, and also prevent having microphone cables strewn all over the studio floor. This also involves the automatic change-over of the muting circuit for the studio loud-speaker, and where individual cue-light circuits are provided for each microphone, these cue-light circuits must be changed over as

well. Such requirements cause great unsuspected complexity in design.

In order to accommodate all the equipment required for the largest studio the control desk itself would have been unjustifiably large, particularly as some bay-mounting equipment is employed. Moreover, there would have been danger of valve microphony if the desk were kicked. All the amplifiers and relay equipment, therefore, are housed in a separate steel cabinet which is locked to prevent unauthorised interference. The amplifiers have resilient mountings to guard against mechanical shocks.

Several sizes of standard assembly were originally planned and coded Mark I-V, providing varying combinations of facilities, but subsequently two only were adopted, namely Marks II and V. Later two larger sizes were introduced in limited quantities for special requirements. Complete technical descriptions of these will be found in Technical Instruction S5. A simplified schematic of the programme circuits of Type-A equipment is shown in Fig. 42.

#### 5.4.4. *Miscellaneous and Future Studio Equipment*

The Type-A equipment described above provided most of the facilities required in the majority of post-war studios and represented a considerable advance on previous equipment. It suffers, however, from two major disadvantages; in the first place it is rather costly to install and, particularly in the case of small talks studios, this high cost is not always warranted. Secondly, it is somewhat inflexible when once installed and modifications necessitated by changing techniques are difficult to carry out. These changing techniques, too, have in a few cases called for complex installations which cannot be accommodated in the standard-size Type-A desk and cabinet.

The first difficulty has been met on an interim basis by the adoption of a commercial design made primarily for overseas markets. The simple requirements of a talks studio are much the same the world over, but the complex facilities needed for the larger studios vary considerably and no commercially designed equipment is available for these applications. A new BBC design, Type B, is therefore being prepared which will provide all the essential facilities provided by Type A, will be sufficiently flexible to enable modifications and extensions to be carried out easily, and yet will be considerably less costly.

The basis of this design is that it will consist of an assembly of sub-units which, while standard in themselves, can be grouped in various combinations, thereby giving both flexibility and uniformity. These units will for example be a high-grain amplifier, a monitoring amplifier, a range of three types of small control sub-panels, a PPM

panel, etc. The control sub-panels will each carry a fader channel and will have various combinations of associated controls such as cue keys, etc. These units will be assembled in two or three preferred arrangements but there will be much room left for modification when required.

### 5.5 O.B. FACILITIES

In general the presentation facilities required in an outside broadcast are necessarily less stringent than in a studio production, and the emphasis is on the temporary nature of the equipment set-up. Nevertheless, the BBC carries out a large number of O.B.s that call for quite elaborate, if not permanent, equipment. A few occasions arise when sufficient equipment could be carried in a suitcase, but the number is too small to warrant its provision. In BBC practice, portable recording equipment generally meets this need.

The following facilities are, therefore, necessary in an equipment which is to fulfil its purpose adequately in a majority of O.B.s.

- (a) Provision for up to four microphones and easy extension up to eight, with table and floor stands as necessary.
- (b) A main programme amplifier with main gain control and peak programme meter.
- (c) Loudspeaker monitoring.
- (d) Light-cue circuits to commentators or artists.
- (e) Telephone communications with the studio centre.
- (f) Mains-operated power supply and battery stand-by.
- (g) Spare facilities to ensure continuity of programme output under all foreseeable circumstances. This demands, amongst other things, a spare main amplifier and power supply and quick means for interchange of programme and control line to the studio centre.
- (h) Programme cue, generally by radio pick-up from the programme service being contributed to.
- (j) Minimum weight and size.

Less essential yet highly desirable facilities include means for independent feed to a local public address system, telephone facilities to remote commentators etc., provision for microphones of various impedances, and arrangements for rapid transport and set-up.

#### 5.5.1. *OBA/8 Equipment*

Before 1938 the requirements were perhaps not so stringent but the equipment then in use was not well adapted to them by modern standards, particularly in the matter of bulk and weight. In that year

the OBA/8 amplifier was introduced together with its associated units, i.e., Mixer MX/18, Mains Unit MU/3A, Loudspeaker Unit LSU/1, and other miscellaneous items. It represented a considerable technical advance at the time since pentode valves, the peak programme meter, and mains supply were used for the first time in O.B. equipment; moreover, consideration was given in the design to ease of transportation.

Full details of this equipment will be found in Technical Instruction S3, but a brief description may not be out of place. A circuit schematic of a simple O.B. set-up is shown in Fig. 43. The Mixer

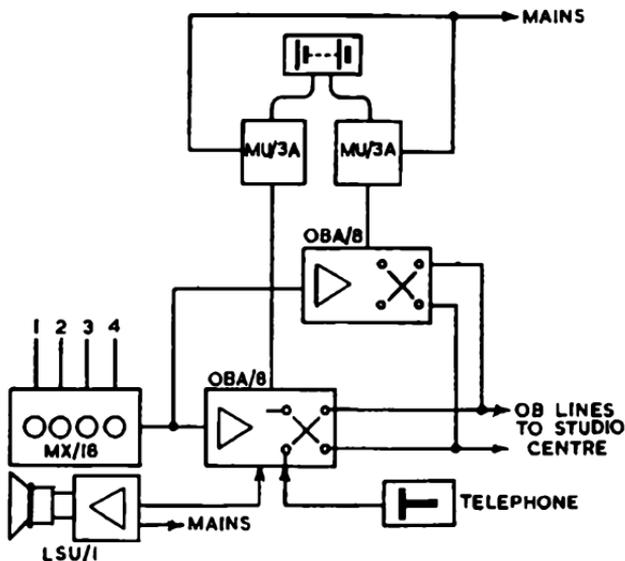


FIG. 43—Simple O.B. circuit

MX/18 provides balanced low-level mixing for four sources; the OBA/8 amplifier itself has an over-all gain of 90 db and feeds +4 db volume to line with a source impedance of 75 ohms. It consists of a two-stage amplifier using pentode valves and the main gain control is a ganged unit which introduces negative feedback and so enables the first stage to accept relatively large signals without overloading. The programme meter circuit is incorporated in the amplifier and a 50-c/s tone from the mains enables the sending level to be standardised on the site. Keys are provided so that the outgoing programme line and the control line can be quickly interchanged; these are indicated by crosses on the diagram. The Loudspeaker Unit LSU/1 contains a small amplifier giving about 1 watt to the loudspeaker,

and the baffle folds up to provide a carrying case. The mains unit has provision for the connection and quick substitution of batteries.

This equipment satisfied requirements for many years but gradually additional items of equipment were found necessary on many O.B.s and by 1950 the amount of this auxiliary equipment generally exceeded the bulk of the basic equipment itself. Accordingly a new design was undertaken which resulted in the OBA/9 amplifier and its associated units.

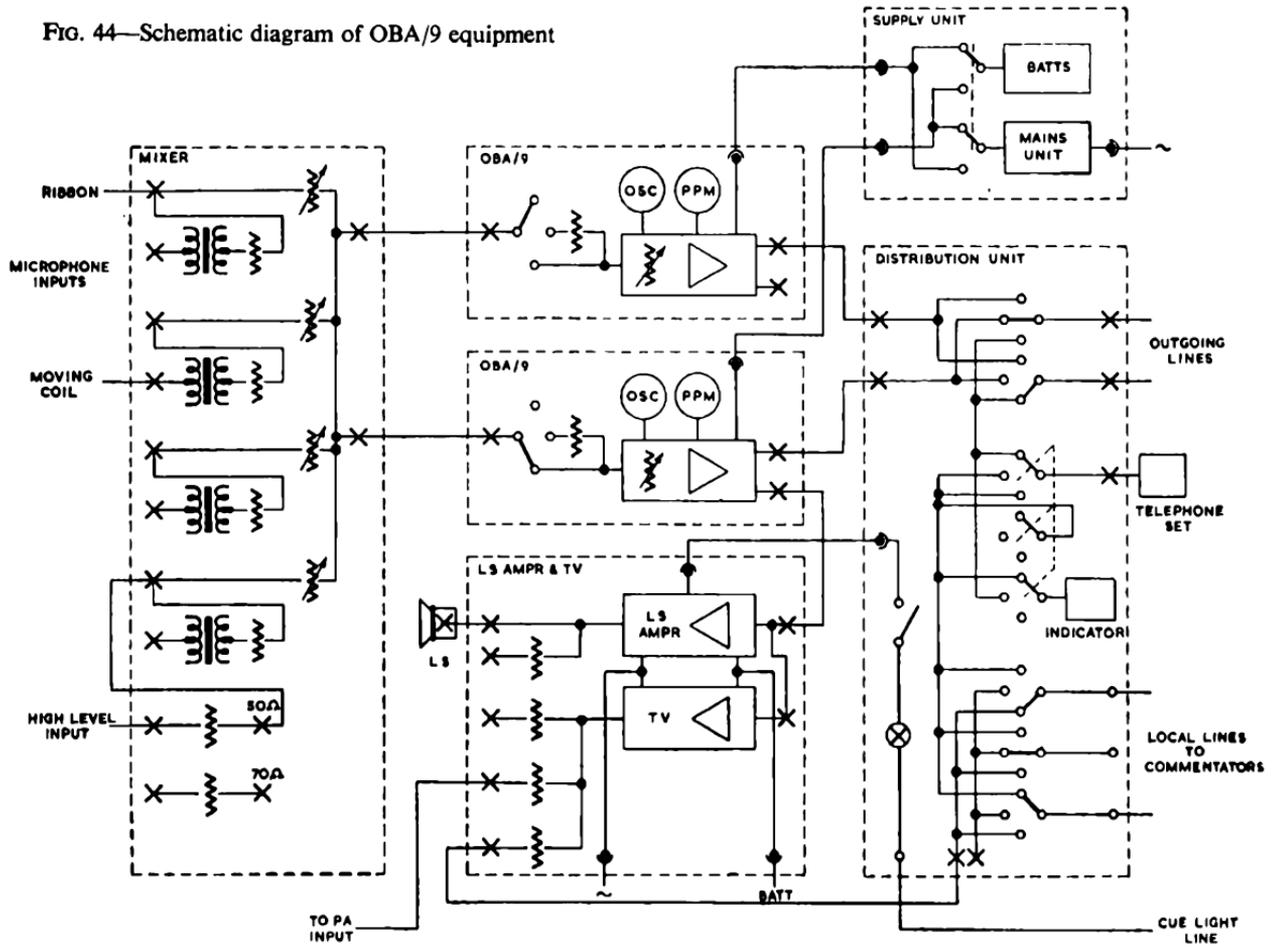
### 5.5.2. *OBA/9 Equipment*

In the design of this equipment it was not intended that all O.B. requirements should be catered for in the basic assembly. These are too varied and would result in the equipment being too cumbersome for general use; the requirements set out on page 89 which meet a large percentage of the O.B. situations were, therefore, taken as a basis and the equipment designed so that additions were easily possible.

In order to simplify transport, the equipment is designed to fit on a trolley (Plate 2, p. 96) similar to that used by an hotel porter which could be carried in a car with suitably shaped doors. The equipment is constructed in units which can be dismantled and carried separately if necessary, and the interconnection of the units is quickly carried out by plugs and cords. Three drums of microphone cable are carried on the back of the trolley and the cables can be run out and the inner ends plugged to the equipment without dismantling the drums; setting-up the equipment is, therefore, a relatively simple operation.

The facilities provided consist of a Mixer Unit MX/29, two amplifiers OBA/9 (one spare), a Distribution Unit DU/1, a trap-valve and loudspeaker-amplifier unit LSM/9, and a power supply unit SUP/6 (Fig. 44). The mixer provides for four input channels of 300 ohms or 30 ohms impedance together with attenuators in case one input is at high level; a second mixer unit can be accommodated on the standard assembly. The amplifier (whose basic circuit is on page 73) provides similar facilities and gives a performance equivalent to the OBA/8 with the exception of the output line switching, which is provided in the distribution unit. The latter unit gives great flexibility in the use and connection of programme, telephone, and cue lines, and also in the feeds both to the studio centre and to commentators or artists. The loudspeaker amplifier is mounted in a separate enclosed unit, and the trap-valve circuit provides programme feeds to public address systems and other destinations so often required. The supply unit contains both batteries and a mains unit which can be

FIG. 44—Schematic diagram of OBA/9 equipment



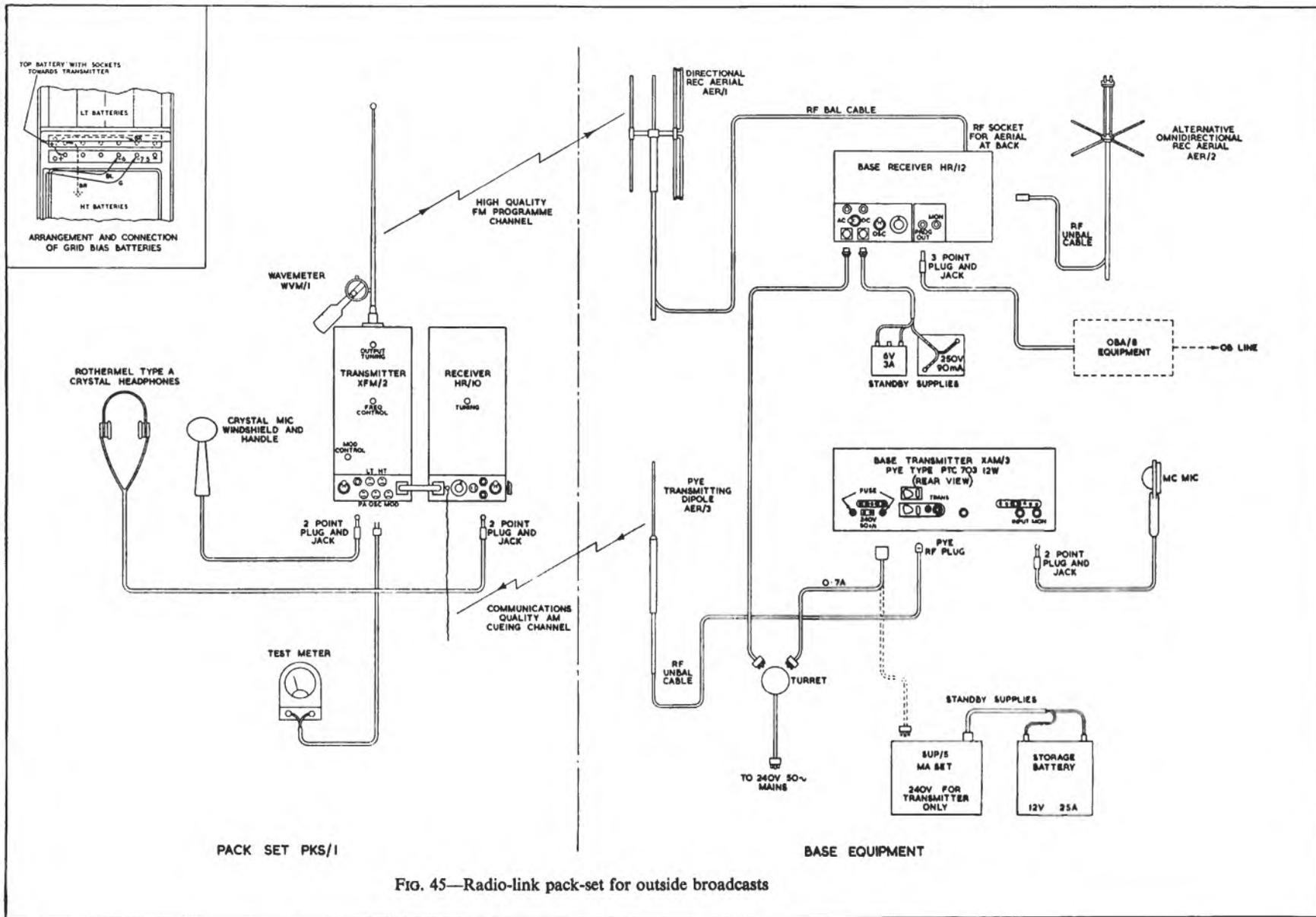


FIG. 45—Radio-link pack-set for outside broadcasts

connected to either of the two main amplifiers; normally one is mains-fed while the spare is connected to batteries during transmission only.

In order to prolong the h.t. battery life, a measure of battery reactivation has been introduced. Whenever one amplifier only is in use and fed from the mains, a small charging current is passed through the h.t. battery. This treatment prolongs its useful life many times, and takes place usually during the time of rehearsal when the spare amplifier is not required.

This equipment is described in detail in Technical Instruction S3.

### 5.5.3. *Miscellaneous O.B. Equipment*

The apparatus described above satisfies requirements when physical line circuits to the studio centre are available. When such circuits are not available use is sometimes made of radio circuits although the requirements in this field are limited. Two types of radio equipment have been used for this purpose; the first of these consists of transportable VHF communication equipment. This equipment has been used in two forms, one of which is of American design working on 26 Mc/s, and the other a British design working on about 90 Mc/s; neither of these has a power greater than 30 watts so that a short range over a line-of-sight path is the limit of their performance. Both of these sets are designed primarily for communication but the circuits have been modified so that they give quality suitable for broadcasting. Both sets use frequency modulation and therefore give good signal-to-noise ratio, and they provide for two-way communication so that a talk-back circuit from the studio centre is available. They are not, however, portable in the true sense of the word.

To meet the need for a really portable short-range radio programme circuit a pack-set having the dimensions of walkie-talkie equipment has been designed by the BBC and is shown in Fig. 45. This consists of a transmitter and receiver, battery operated, weighing 26 lb. It operates on 92 Mc/s with an output power of 1 watt, giving a range of up to 100 yards under poor conditions and up to about a mile in open country; it is used in conjunction with a base receiver and transmitter giving talk-back facilities. It is, therefore, valuable when the commentator must move about in crowds or, for example, follow a golf match, where a vehicle would be impracticable and the trailing around of a long cable inconvenient. This equipment differs from the usual walkie-talkie in the quality of speech obtainable; it is designed to provide speech quality approaching that of broadcasting standards, rather than mere intelligibility.

## 5.6 RECORDED PROGRAMMES

Another source of programme much used by the BBC is the reproduction from recordings made outside broadcasting hours, or at a time and place not fitting in with the programme schedule of the service for which the recordings are required. Technically, the source consists of sound-reproducing equipment and may be a gramophone pickup, the reproducing head of a magnetic recorder or the sound-head of a film projector, all with associated amplifiers and fading equipment. Programme planners, however, refer to these programme sources as "recordings"; they also use the term "recorded programmes". There is an important distinction between these terms; recordings may well be isolated excerpts inserted in a feature programme containing mostly "live" material, whereas a "recorded programme" implies that the whole of the programme is being broadcast from disks, tape or film.

Practically all BBC sound recordings are made on either disks or magnetic tape, with a considerable amount of "dubbing" (i.e., copying) from one system to another; this editing process is similar to the editing of films in that whole programmes can be built up from extracts of various recordings, interspersed with commentary as necessary. Film projectors are used to a limited extent, but they are not used as programme sources, the extracts being dubbed to disks or magnetic tapes.

Recorded-programme sources therefore comprise gramophone desks, or magnetic recorders. Gramophone records, which may be commercial disks or 78 r.p.m. disks recorded by the BBC, are reproduced on turntable desks known as TD/7s, which are referred to on page 76; for certain types of programme six or more turntables may be needed to facilitate selection of numerous recorded extracts. At the larger studio centres at least one studio is equipped with a bank of three TD/7 desks.

Many disks are recorded at  $33\frac{1}{3}$  r.p.m. and to reproduce these, special turntable desks are used. Commercial units of American manufacture were employed during the war years but these are being superseded by BBC-designed turntables, Type DRD/1, which can be used for either 78 or  $33\frac{1}{3}$  r.p.m. disks. In certain circumstances,  $33\frac{1}{3}$  r.p.m. disks are reproduced on the recording machines, most of which have full facilities for the purpose.

For magnetic tape recording, it is not economical to provide special reproducing machines for these the recording machines are used for reproduction. The general policy is to have two machines working together to provide for continuous programmes. Whether for recording or reproduction, two such machines, with their associated equipment, constitute a *channel*.

## 5.7 MISCELLANEOUS PROGRAMME SOURCES

Under this heading may be grouped a number of rather special programme sources as follows.

### 5.7.1. *Time Signals and Interval Signals*

One of the services provided by the BBC is the broadcasting of accurate time signals. These take one of two forms; they are either the chimes and strokes of Big Ben, or the six well-known "pips". Big Ben signals are derived from a microphone and amplifier connected by line to Broadcasting House. Alternative battery supply is provided for the amplifier, and change-over is automatic if for any reason the mains-derived supply fails. It is of interest to note that it is the first stroke of the hour which indicates the correct time; the chimes are quite erratic.

The six "pips" are controlled by the master clock at Abinger; this clock used to be situated at Greenwich and the signal is therefore referred to as the Greenwich Time Signal (G.T.S.). The signal consists of bursts of 1-kc/s tone; this tone, generated in London control room, is normally greatly attenuated by means of a rectifier bridge circuit; d.c. control pulses initiated by Abinger remove the attenuation and the six well-known "pips" are the result. The pips are generated every  $\frac{1}{4}$  hr and are selected for transmission in the appropriate continuity suites. The final "pip" denotes the exact  $\frac{1}{4}$ -hr instant.

From time to time gaps occur in the programme scheduled for transmission due to errors in timing or other causes. Where these gaps are of appreciable length they are filled by announcements or recorded items from a gramophone record selected by the Continuity Announcer, but short gaps are filled by a recorded interval signal such as the sound of Bow Bells.

### 5.7.2. *Unattended Studios*

There are circumstances where it is desirable to broadcast occasionally from a location remote from a studio centre where the item does not warrant, or allow time for, the provision of O.B. or mobile recording facilities; an example is at a sea or air port, where interviews with important travellers could be carried out at short notice. This need is met by the provision of a studio permanently linked with the nearest studio centre and provided with the simplest facilities. No dynamic control is undertaken and the equipment can be set in its operational condition by the interviewer simply by closing a single switch.

## 5.8 CONTINUITY SUITES

The responsibility for the production, artistic value and duration of an individual programme item has always been centred on the originating studio or source. For many years dynamic control of the programme has also been delegated to the source and the control-room staff have been responsible only for the correct routing of the programme, for monitoring the service and for emergency measures to correct any irregularity. To facilitate these duties in the control room a control position was allotted to each continuous programme service and to each subsidiary programme destined for recording or as a contribution to another centre. The duty of the control position operator was to select the source and the destination of the programme, to monitor it continuously on headphones and by peak programme meter, and to exercise over-riding dynamic control if this were technically necessary. This system is still used for miscellaneous programmes of limited duration, e.g. special recordings.

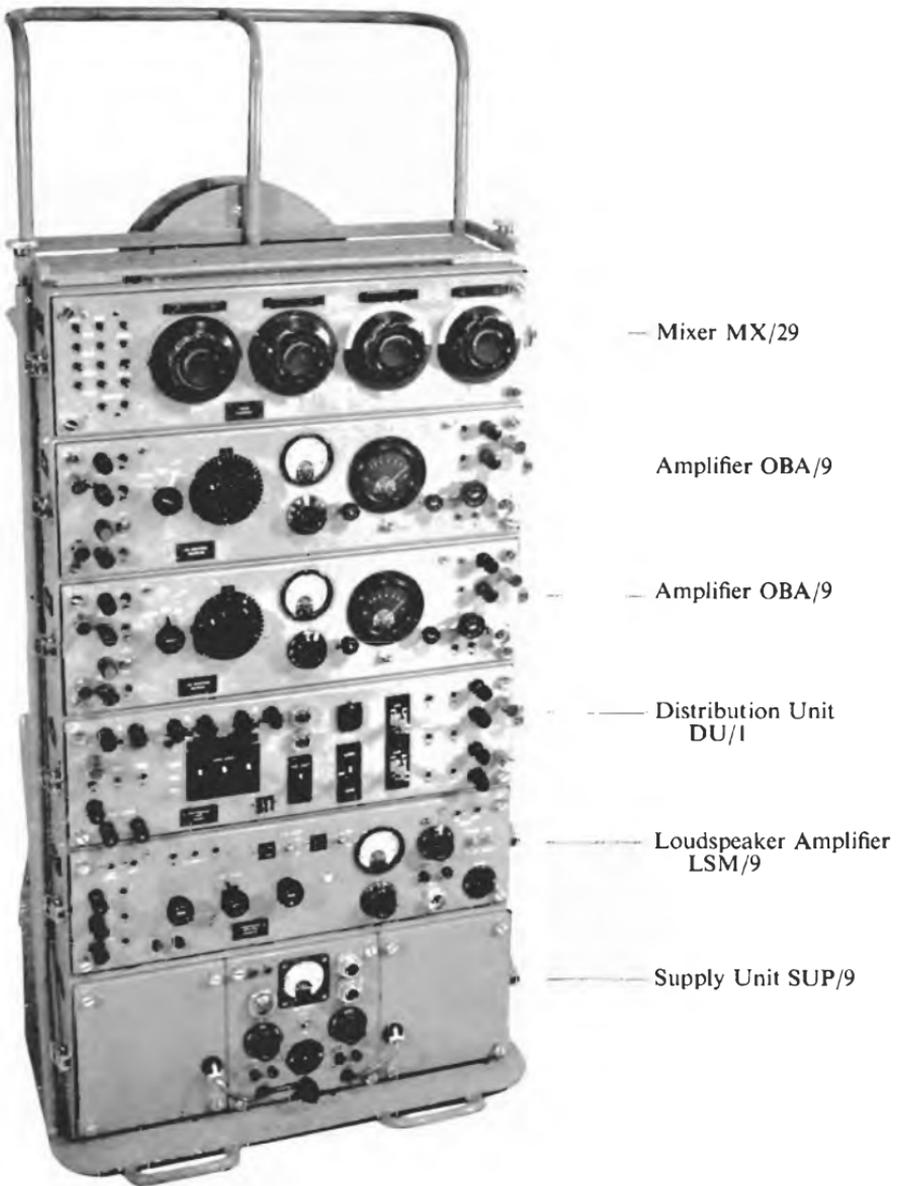
Provided everything went according to schedule, this system worked satisfactorily, but if a programme over-ran its time, or there was some technical failure, it was difficult for the engineer to organise remedial measures. Sometimes serious gaps appeared in the programme, or perhaps mutually incompatible types of programme followed each other with the briefest of intervals.

To overcome these difficulties, and generally to smooth out the presentation of the daily programme for a given service, the system of continuity working was introduced in 1942. Briefly, this consists of creating a focal point for each programme at which a presentation official (announcer) and an engineering operator are located. The duty of the former is to make snap decisions as to the action to be taken to cover programme over-runs, under-runs, or failures, and to insert interval music or linking announcements between programme items or to cover faults. The engineer's duty is to select and prepare the subsequent source circuits for transmission, to monitor the programme as a whole technically, and to take such engineering action as necessary in case of fault.

In order that the engineer can monitor continuously on a high-quality loudspeaker and carry out telephonic communications when the announcer is using his microphone, a studio is provided for the latter and an associated cubicle or room for the former. The usual window is provided between the two, and two-way talk-back circuits are provided for close liaison.

5.8.1. *Continuity Studio Equipment*

The continuity announcer is provided with a microphone and two gramophone turntables complete with fader controls. He is thus



OUTSIDE BROADCAST EQUIPMENT TYPE OBA/9

## BROADCAST PROGRAMME SOURCES

able to insert his own announcements or interval music, and to allow him to reduce the volume of the main programme when it is necessary to superimpose these additions, a main fader is also provided for his use. A loudspeaker, on which he keeps a continuous check on his programme and which is automatically silenced when the microphone is alive, is also fitted in this studio. Conditions may arise, either because of technical faults or because of unsuitable programme material, when it is necessary to abandon the programme from a particular source temporarily. The announcer will substitute a fill-up programme but he will wish to be aware immediately conditions return to normal. Accordingly, he is provided with means to switch either his loudspeaker or headphones across the input from the offending source; he can then reinstate the normal programme when the fault is cleared.

### 5.8.2. Continuity-room Equipment

The engineer is provided with means for selecting both the programme and the cueing and telephone channels to any of the sources within, or adjacent to, the studio premises, and up to four sources can be selected on to faders simultaneously. A loudspeaker and peak programme meter enable technical monitoring to be carried out and a main control attenuator permits over-riding control if this is found

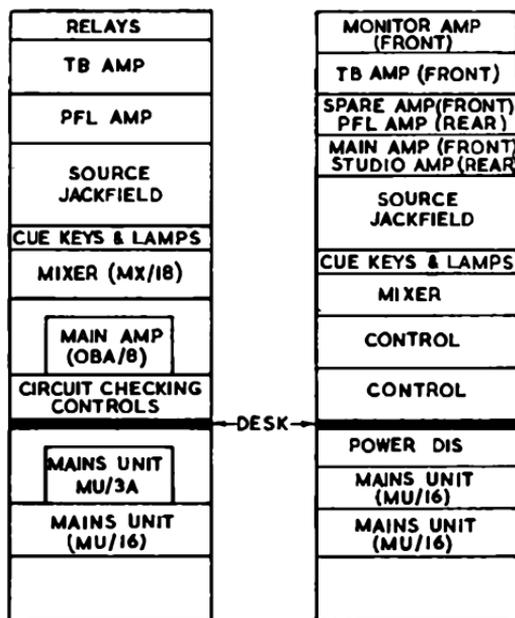


FIG. 46—Continuity room equipment bays: alternative layouts

## STUDIO ENGINEERING FOR SOUND BROADCASTING

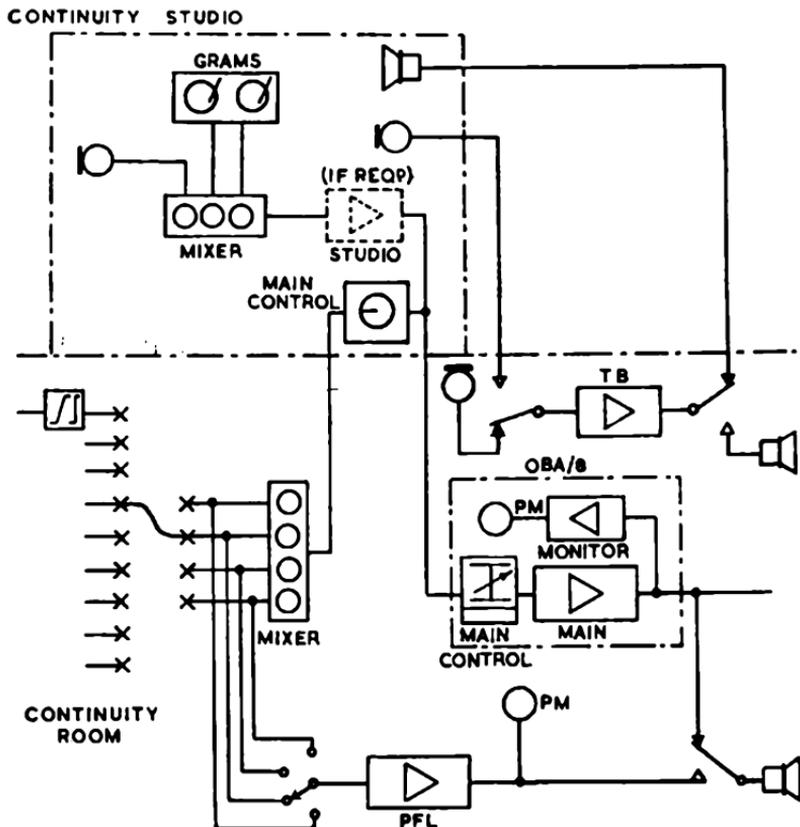


FIG. 47—Continuity working: schematic circuit

to be technically necessary; in normal circumstances, however, no control whatever is exercised. Line-up tone is always sent from a source prior to transmission and a test meter is provided for the continuity engineer to check its level. The keys which connect this checking circuit to the appropriate channel prior to the fader also enable a listening circuit to be established, thus giving pre-fade listening facilities on the various sources.

The actual equipment used is based on the war-time austerity arrangements adopted for control rooms and described in the next chapter. The amplifier mounted on a bay in front of the operator provides the nucleus of the equipment and the source selection function is performed by plugs and jacks. A sketch of such a bay, together with one of the recent modifications of it, is shown in Fig. 46 and a schematic for a continuity suite is shown in Fig. 47.

### 5.8.3. *Continuity Suites for European Services*

The function of the continuity suites in the external services is slightly different, partly because of the stricter time schedule which is maintained and the shorter average duration of programme items, and partly because of language difficulties. In the Overseas Service, the general form of the continuity suite is similar to that in the Home Sound Services, but the function of the announcer is more to announce changes in transmission frequencies, future programme items, and other special announcements, or to originate simple recorded programmes.

In the European Services, however, the changes of language in any one service are extremely frequent and, except from the technical point of view, continuity is meaningless. In these services the suite takes the form of a continuity room which is the focal point of the programme service. Two studios are attached to this room and it was the original intention that alternate programme periods should be allocated to each studio. In this way, each studio would have to deal with only one language at a time and at least a  $\frac{1}{4}$ -hr gap would be provided for a change of personnel and language. While this principle is maintained in essence, there is much interchange of studios between the various continuity rooms for reasons of operational convenience; technically, however, the suites are generally similar to those of the Home Sound Services.

## CHAPTER 6

### THE CONTROL ROOM

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#### 6.1 FUNCTIONS OF THE CONTROL ROOM

IN the previous chapter on Broadcast Sources some reference was made to the rôle of the studio manager, the producer and the continuity announcer, for it is at the programme source that the engineer comes into the closest touch with these people; moreover it is they who, in the long run, largely dictate the facilities which are required in the technical equipment associated with broadcast sources. In the control room, however, the artistic side of broadcasting is less in evidence and the operations which must be carried out there are correspondingly more technical.

The control room may be regarded as the nerve centre of the broadcasting chain. Its function is not unlike that of a telephone exchange, since it receives programme contributions from various sources and directs them to appropriate destinations. It is also the central switching point for communications relating to the transmission of programmes from source to destination.

Fundamentally these functions have changed little since the early days of broadcasting, but in detail several important differences may be observed between the modern control room and the old; these will become evident later, but perhaps the most important arises from the introduction of continuity working. This technique provides an interception point between the programme source and the control room proper for the specific purpose of direct co-ordination between programme officials and engineers.

##### 6.1.1. *Routing of Programme*

The routing of programme implies "switching programme sources to predetermined destinations". These destinations fall broadly into three categories, namely transmitters, recording channels, and miscellaneous destinations; the latter include wired-programme distribution, foreign re-broadcast networks, check and test listening points, and programme cue circuits. Depending upon circumstances the programme may be directed to any one of these destinations or to combinations of them simultaneously. The control room is respon-

## THE CONTROL ROOM

sible for setting up the appropriate routes according to the programme schedule, and also for keeping a check on the various circuits in order to trace and correct any fault with the least possible delay.

In the earlier control rooms it was also necessary to set up circuits for studio rehearsals because there were not enough amplifiers for each studio to have a permanent allocation. Not only was it necessary to switch the amplifiers to the appropriate studios and to connect up the auxiliary circuits but in some centres rehearsal control positions were provided at which all the rehearsal arrangements were supervised. Modern practice avoids these operations by providing each studio with its own local apparatus and with its auxiliary circuits permanently set up (e.g. programme meter, headphones, loudspeakers and talk-back circuits).

### 6.1.2. Line Terminating Equipment

Because broadcast transmitters are usually located on sites remote from centres of population, and also because programmes do not always originate in the centre which normally feeds the appropriate transmitter, much use is made of landlines to convey the programme up and down the country. These lines are rented from and maintained by the Post Office, and their ends terminate in BBC premises, at which point the Post Office responsibility finishes (Chapter 7).

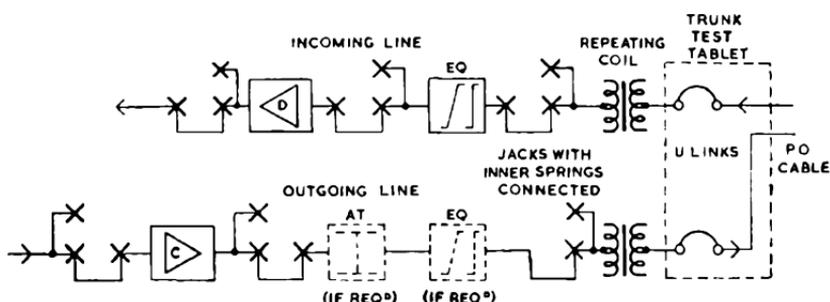


FIG. 48—Line terminating equipment: schematic

Losses occur on these lines which in theory should be due only to the initial and final lengths of line connecting the BBC premises to the nearest Post Office repeater stations, but in practice additional losses occur which must be made good.

The equipment required in connection with these lines is all located in the control room, a typical arrangement being shown in Fig. 48. It consists of line terminating equipment for both the outgoing and the incoming lines (i.e. isolating U-links, repeating coils and cross-connecting facilities) while for the incoming lines equalisers for

frequency correction and D amplifiers for raising the level are required. For the outgoing lines C amplifiers are necessary for isolating and feeding them and sometimes equalisers are required also.

In addition to the permanent circuits, which are known as *Simultaneous Broadcast (S.B.)* circuits, a number of lines are rented from the Post Office on a semi-permanent basis; these are generally lines to points which are often used for O.B. programme contributions. Such lines, although permanently available, are only equipped with equalisers and amplifiers when required and these are set up on a special O.B. bay. Equalisers and attenuators with variable settings are connected to the appropriate lines shortly before they are wanted for programme and are adjusted to pre-arranged settings.

### 6.1.3. *Control of Miscellaneous Programmes*

The basic transmission networks are supervised in *continuity suites* which, apart from the Home Service, are all concentrated in London. The Home Service is operated on a regional basis and although during most of the day the Basic Home Service from London is radiated from all transmitters, each regional centre has a continuity suite which is manned during those periods of the day when the regional programme differs from, or contributes to, the Basic Home Service.

In addition to the programmes which are being handled in this way, other programme items must be dealt with by the control room. The most important of these are programmes which are being recorded for subsequent transmission, and contributions to the other services whose continuity suites are at other centres or premises. For this purpose the control room is provided with one or more control positions where an engineer can supervise these programmes and carry out such technical liaison as is necessary from time to time. Experience has shown that this arrangement is preferable to making a direct connection from source to destination, first because line-up and starting procedure is thereby much simplified, and secondly because minor irregularities can be expeditiously dealt with so that the dislocation of the operational schedule is avoided. Such dislocation would cause grave inconvenience and disturbance not only to the particular programme concerned but to all other programmes handled.

### 6.1.4. *Technical Supervision and Maintenance*

Besides the routing of programmes and their supervision, an essential duty of the control-room staff is to check the technical apparatus and circuits, and to correct faults and rectify departures from optimum performance which may occur. For example, each

studio is inspected and the apparatus checked daily, whilst comprehensive tests are made periodically. Similar regular testing is carried out on control-room apparatus, S.B. circuits between centres, and on other lines between premises which are in constant use.

The factors to which closest attention must be paid, both on individual items of equipment and on over-all links, are the voltage gain, freedom from overload distortion, smooth frequency characteristic, and freedom from background noise. To enable these features to be easily checked, test apparatus is provided in every major control room; this consists of a variable-frequency tone source, an amplifier detector on which voltage levels may be measured, a test peak programme meter for programme and noise tests, and other auxiliary apparatus such as a high-pass filter, a variable attenuator, and a Wheatstone bridge (Chapter 7).

#### 6.1.5. *Communications*

For the efficient handling of the programmes it is essential to arrange for adequate and rapid communication between the various people concerned. Notwithstanding the simple cue-light system which exists between the continuity or control positions and the studios, which was referred to in Chapter 5, much use is made of telephone circuits, both between the above two points and all over the broadcasting system. In fact almost without exception where a programme circuit exists between two points, there will also be telephone circuits. These telephone circuits are for engineering communication only and are quite distinct from any inter-office or normal Post Office public service systems. They all terminate in the control room which functions as a local exchange for the traffic on them.

In regional control rooms and others of similar size all the control lines, as they are called, terminate on what is referred to as the S.B. position, but in the larger London centres the internal lines to studios, recording rooms, and the like are allocated to a separate position in order to reduce the quantity of traffic at one point (O.B. circuits are treated separately). These internal-source telephone circuits and those associated with O.B. points are extended to the continuity or control position for the period during which they are originating programme, to expedite direct communication; at other times the control room personnel handle telephone communication to and from these points.

Communication circuits are almost as important as the programme circuits themselves from the point of view of the smooth handling of the programme. The telephone is the normal method used but at certain key points more expeditious arrangements than the telephone are made by the provision of talk-back circuits. Thus

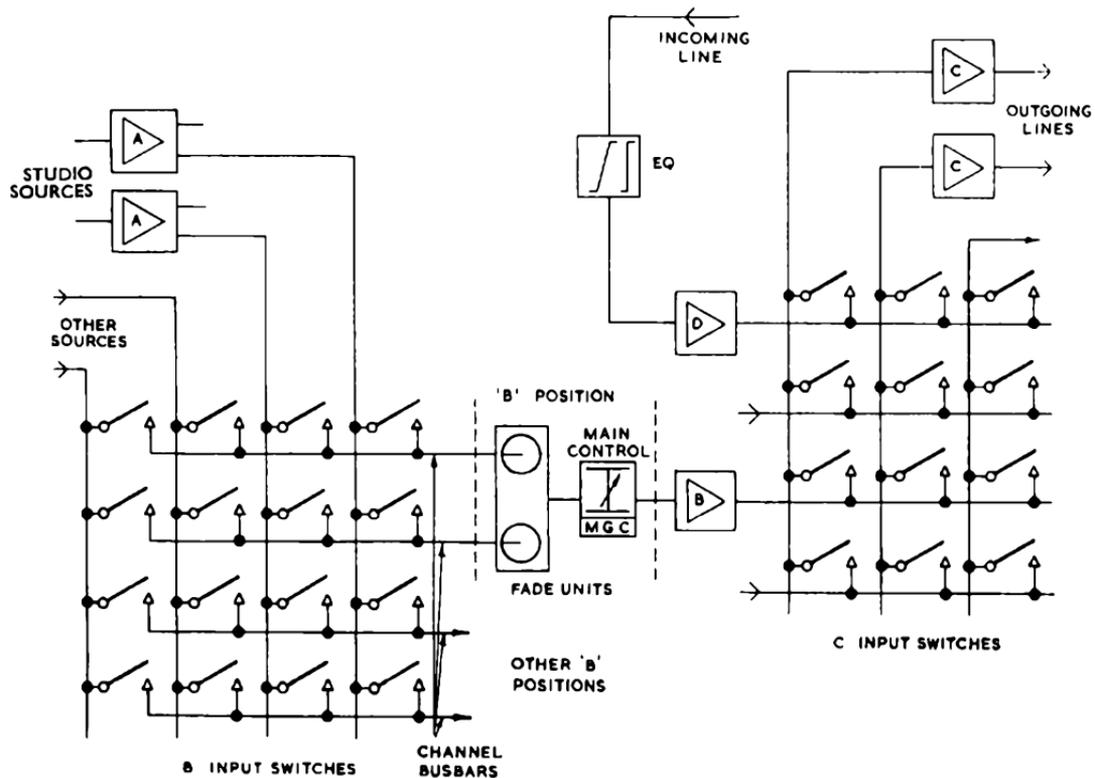


FIG. 49—Relay input switching for B and C amplifiers

the continuity engineer and announcer are in touch with each other by talk-back and in Broadcasting House, London, the senior control-room engineer can communicate with various key positions in this way.

In addition to the communication circuits mentioned in this section, there is a comprehensive communication system for long-distance telephony and telegraphy. This system is described in Chapter 9.

## 6.2 APPARATUS FOR PROGRAMME ROUTING

It has been observed in a previous chapter that in general the type of equipment used by the BBC falls into three groups in historical sequence; this applies also to switching equipment.

- (i) The equipment in use up to 1940 was generally a remotely-operated system using banks of relays and was referred to as the "B- and C-input Switching System" (B and C indicating control and distribution amplifiers).
- (ii) During the war a manual system was employed using plugs and jacks because of its flexibility, speed of installation, and economy.
- (iii) Plans are now being made to introduce a remotely operated system using automatic telephone selector switches.

### 6.2.1. *B- and C-input Switching System*

This system, which is shown in schematic form in Fig. 49, consists of two quite separate parts. In the B-input switching section all the main sources of programme are connected to a number of bus-bars (shown vertically in Fig. 49), while all the input channels to the fade-units on the control positions are connected to a similar series of bus-bars (shown horizontally). The term bus-bar here implies a common connection rather than a physical component. At each point of intersection of these two bus-bar systems a relay is located so that any source can be connected to any channel. To make this connection, the appropriate relay is actuated by two successive operations. First a locking channel key (Fig. 50) on the control desk appropriate to the channel required is thrown, this action preparing all the relays corresponding to this bus-bar to receive an operating stimulus. This stimulus is provided by pressing the appropriate source key; similar groups of keys are provided on all the other control positions. The stimulus is applied to all the relays associated with the appropriate source bus-bar but the only one to operate is the relay which has been prepared by means of the channel key.

The way this is brought about is shown in Fig. 50 and for convenience of explanation it will be supposed that source 3 is to be

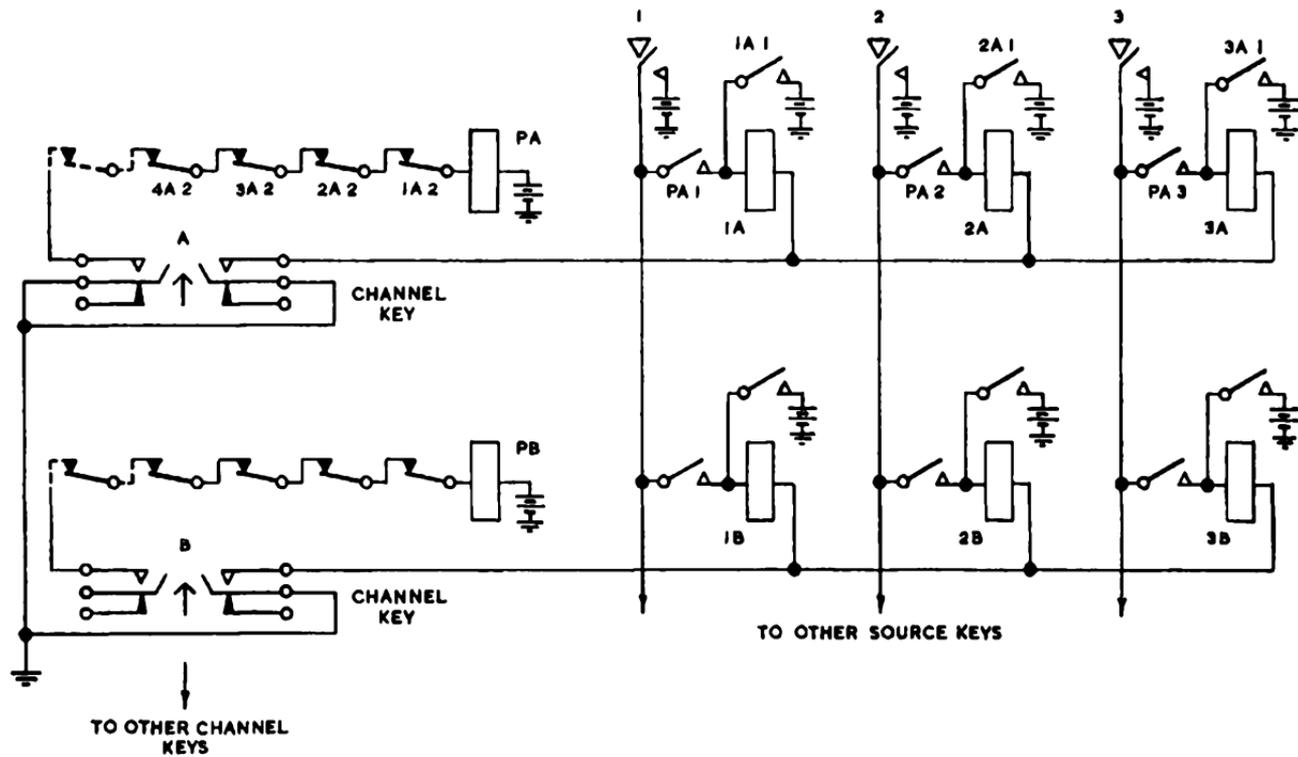


FIG. 50—Relay input switching: switching sequence

connected to channel *A*. Operation of the channel key not only connects one side of all the channel relays 1*A*, 2*A*, 3*A*, 4*A*, . . . to earth but also operates the relay *PA* which in turn connects the other ends of the channel relay windings to the appropriate source keys. Momentary depression (or "punching") of source key 3 then operates relay 3*A* which holds itself operated by contacts 3*A*.1; but contact 3*A*.2 breaks the circuit for relay *PA* and disconnects all the channel relays from their source keys again. The function of relay *PA* is first to ensure that no other channel relay is actuated by the source key by any "back door" circuit, and secondly to prevent a second source being inadvertently punched to a channel already connected to a source.

In the C-input switching section the C amplifiers which feed the destination circuits are switched to the B outputs by an exactly similar system. The bus-bars corresponding to the sources, however, in this case carry the B amplifier outputs or the outputs of the D amplifiers which are connected to the incoming programme lines.

There are some centres, e.g. Belfast, at which this basic system has been retained but modified and simplified to conform with modern technique. The arrangement of the equipment, however, is similar in all cases in that the relays are mounted on a bay while the operating keys are fitted remotely on the operating desk, whether it be the control positions of the older technique or a centralised switching position in the newer installations.

### 6.2.2. *Plug and Jack Switching*

At the outbreak of war it was essential to move the vital control-room equipment into rooms considered safe from air attack. Moreover, this reconstruction had to be carried out with great urgency in several places at once, so that the simplest possible switching arrangements had to be adopted. For this reason standard plugs and jacks as employed by the Post Office were used on a large scale. These were already used by the BBC for the substitution of faulty equipment and for other miscellaneous switching purposes so that their reliability and operating convenience were known.

The basis of the system adopted was the Listen/Apparatus/Line combination shown in the sketch Fig. 51 (*a*). In this system it is possible to substitute a faulty unit or circuit by a spare without interrupting the programme, and the system is used on amplifier bays and on incoming and outgoing lines bays for this purpose. For example, Fig. 51 (*a*) shows at the top an amplifier normally fed from a particular line and feeding a particular circuit via the appropriate apparatus and line jacks and the interconnection of their inner springs. The lettering above each vertical row of jacks indicates the jack-field

labelling, and is the key to the function of each jack. Thus, for Amp. No. 1 In., the *apparatus* jack outlets are connected direct to the input of the amplifier, whilst the *line* jack outlets are connected to the line which normally feeds that amplifier input. The insertion of a plug into either of these jacks breaks the connection between the two. By inserting the plugs of double-ended cords in the positions shown, the lower amplifier is automatically connected to the line in place of the upper one (supposed faulty) and the latter disconnected from circuit. Fig. 51 (b) shows a development of the system for distribution switching requirements. In this arrangement a number of C amplifier inputs (for example) can be selected individually or in a group. Thus if the connection shown in full line is made, the B group can be disconnected from its normal source and connected to source A, the latter being disconnected from the normal group which it feeds. On the other hand by using the connection shown dotted a particular outgoing circuit can be transferred from its normal source and added to the distribution from source C.

In a large control room using the jack switching system many hundreds of jacks, mounted on apparatus bays, are needed. These bays take up considerable space and however carefully the layout is planned the system tends to become unwieldy in operation.

### 6.2.3. Selector Switching Systems

The advantage of a selector-switching system is that it permits several switching operations to be carried out simultaneously instead of consecutively. A system using these switches enables both programme and communication circuits (together with indicators, etc.) to be established simultaneously; it also permits pre-selection of a number of routes to be made, and the actual switching changes to be triggered simultaneously either manually or by a clock-controlled operation. Such a system is also capable of handling a large number of programme sources and destinations and provides the operational convenience of a remote relay system, yet it is economical both in terms of space and cost. Future switching installations therefore are likely to employ selector switches and the forerunner of such installations is to be found in Bristol.

Selector switches are much used in automatic telephone engineering and a particular type known as a motor uniselector is used in this system. This switch is really a multi-pole rotary switch which can connect 16 wires simultaneously to any one of 50 positions, and in operation this switch drives itself round to a position which has been electrically "marked". It will be appreciated therefore that such a switch can replace a very considerable number of relays, and much greater convenience and flexibility is provided while the cost and

# THE CONTROL ROOM

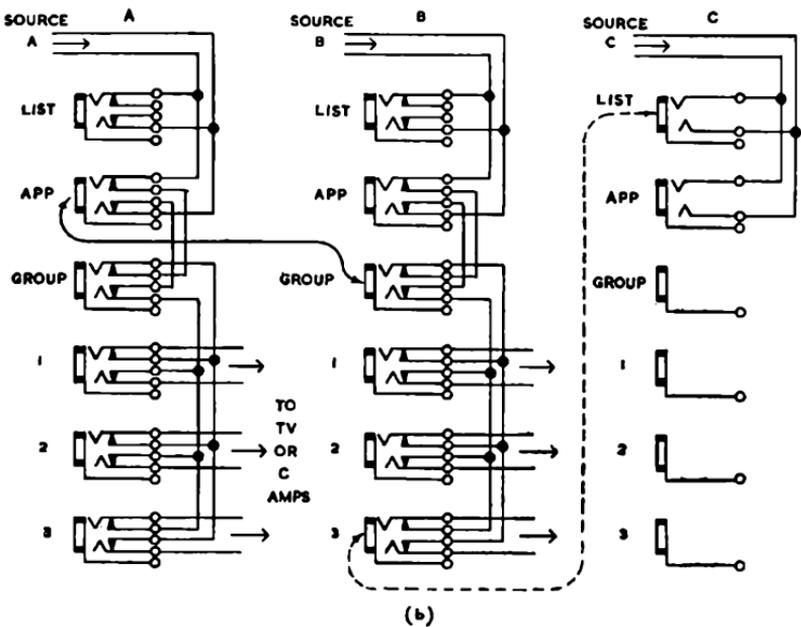
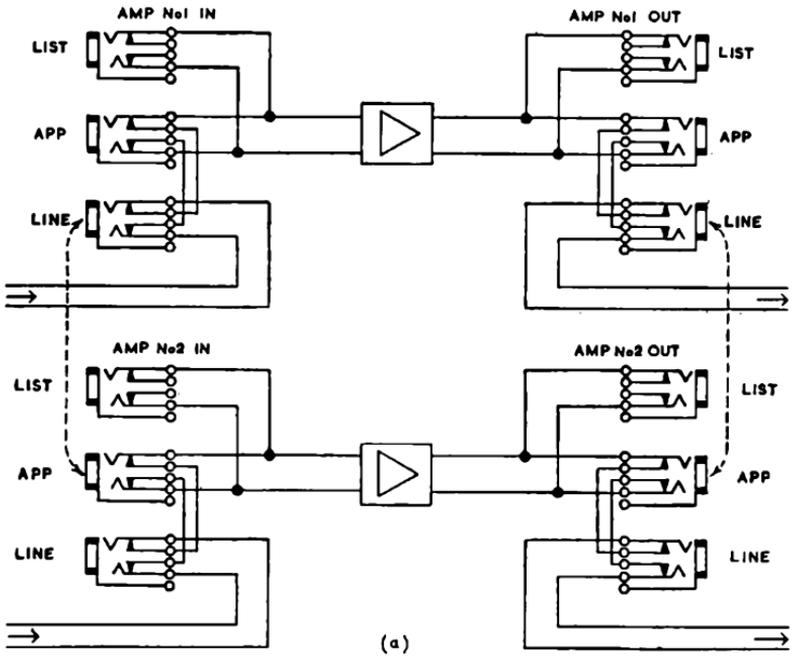


FIG. 51—The three-jack system

space required is much less, particularly for large installations. The switching system developed around this switch is based on the fact that whereas access to many sources and destination circuits must be provided, yet only a relatively small number is required at any one time. Therefore a limited number of channels is provided which, together with two or three C amplifiers permanently connected, form a link between the sources and destinations.

The switching system is shown in block schematic form in Fig. 52. All the source circuits are "multiplied" to corresponding positions on a number of switch banks while the wipers of the switch are connected to the inputs of two or three C amplifiers or to the channel faders of the continuity or control positions. The output circuits of the C amplifiers are connected to similar selector switches and the destination circuits are multiplied to their banks in the same way as the sources. A channel consisting of two or three C amplifiers can therefore be used to connect any programme source to two or three destinations simultaneously and the telephone and cue circuits are established at the same time. The selectors are controlled remotely from an operating desk and it is possible to pre-select the next positions that any number of selectors shall occupy so that several route changes can be made simultaneously. Moreover facilities are provided so that the operator can monitor at various points in the chain, both aurally and by programme meter, all the programmes being handled.

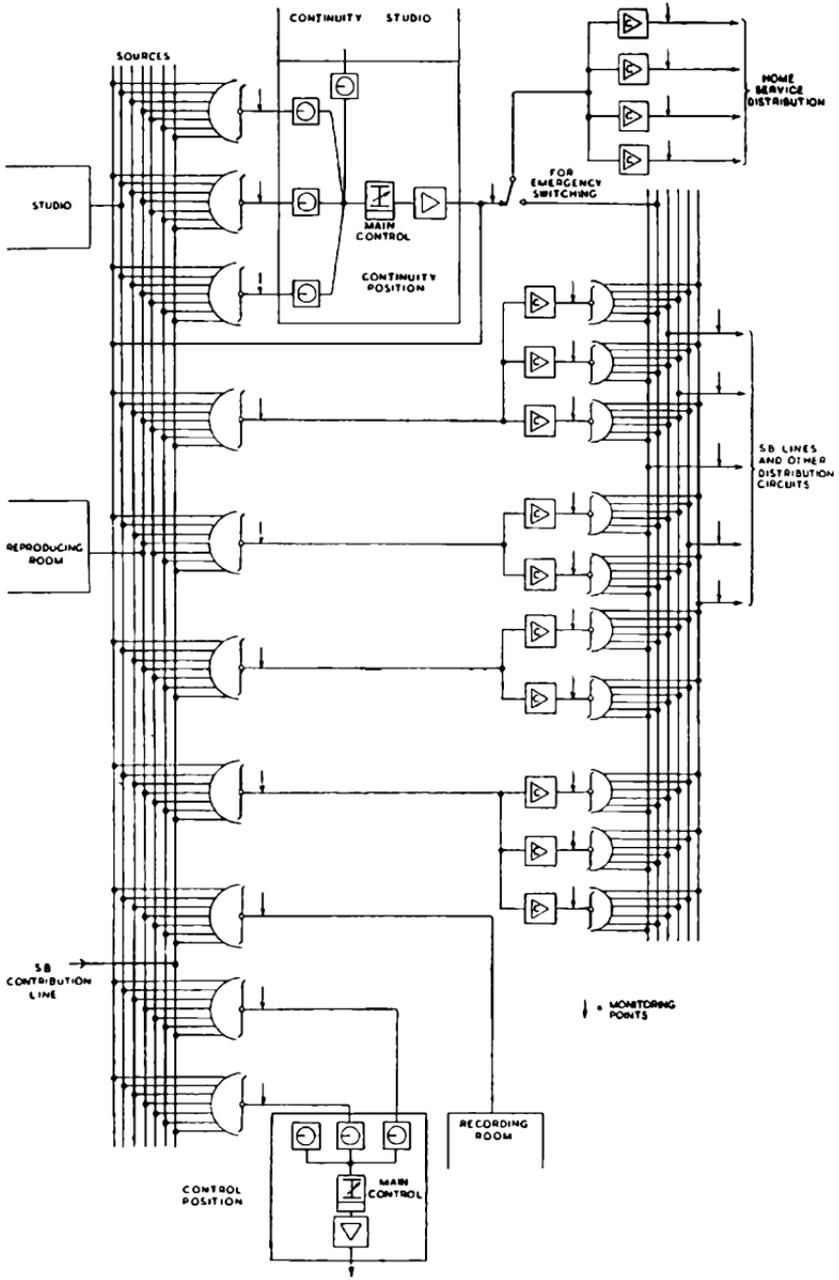
### 6.3 LINE TERMINATING AND AMPLIFYING EQUIPMENT

Reference has already been made to the equipment necessary to terminate the Post Office lines in BBC premises. This equipment is all provided by the BBC and is shown in the diagram Fig. 48 which will now be treated in more detail.

In order to minimise cross-talk in the Post Office cables each line must be maintained carefully balanced about earth, yet to enable tests to be made on the line a break-point must be provided. Accordingly the Post Office cable is terminated on a *Trunk Test Tablet* which provides both a U-link interruption point for each circuit and a convenient termination for the cable without disturbing the accurate earth balance. These tablets are mounted on a bay\* which also carries a number of repeating coils (line transformers), one for each circuit; the latter ensure the accurate balance of the Post Office

\*A confusion in terms has arisen here. In the Post Office and elsewhere the upright structure carrying equipment panels is called a *rack* while an assembly of racks is called a *bay*. In the BBC these terms have always had the reverse meaning for a reason lost in antiquity, and to avoid confusion the BBC nomenclature is used in this chapter.

# THE CONTROL ROOM



**FIG. 52—Selector switching system**

circuits no matter what unbalance may exist on the circuits connected to the other winding.

At this point the circuits are segregated into incoming and outgoing groups and are run to appropriate bays carrying jacks and, in the case of incoming lines, equalisers. It is here that the function of the *Distribution Frame*, which is used in every control room, can be best explained.

The Post Office are always at liberty to use any one of the circuits in their cable as an incoming or as an outgoing circuit and on some circuits the allocations are changed from time to time. In order that the correct grouping of circuits can be maintained on the equipment and jack-field bays without repeated need for costly re-wiring, each circuit is taken from the U-link and repeating coil bay, not direct to the appropriate equipment bay, but to a tag-block on the distribution frame. Two other adjacent tag-blocks on this frame carry circuits to the incoming or outgoing equipment bays respectively and a short piece of twisted wire (known as *jumper wire*) is used to connect from the first block to one of the other two as necessary to complete the route of each circuit. Any change of route may therefore be made in a few minutes with the aid of a soldering iron.

The convenience which this system provides in the semi-permanent re-routing of circuits is so valuable that all similar interconnection circuits in the control room are also run via tag-blocks on the distribution frame. In addition the distribution frame provides a convenient point at which to terminate those Post Office circuits which do not warrant the elaborate trunk test-tablet terminations described above, e.g. O.B. circuits, which are terminated on fuse mountings.

### 6.3.1. *Standard Bay Assemblies*

Before tracing the programme circuits farther through the control room it is convenient to indicate how the bay-mounted apparatus therein is disposed. The apparatus is constructed on panels 22 in wide and a multiple of  $2\frac{1}{4}$  in deep which are mounted on the bay uprights. In general each bay accommodates the apparatus concerned with a particular function, e.g. incoming line termination equipment, or source-switching relay equipment. In the early control rooms, which were individually designed and where appearance was studied, considerable overlapping of function sometimes occurred between the individual bays. To accomplish the rapid war-time installations, however, a number of standard bay designs were introduced on a strictly functional basis and the majority of present control rooms contain these standard bay assemblies (sometimes only partially filled).

Thus there is a standard U-link and repeating-coil bay, an incoming

lines bay, an outgoing lines bay, an O.B. bay, and various amplifier bays. It is convenient in the further description of the control room to speak in terms of these standard assemblies although the principles are the same in the older control room installations.

### 6.3.2. *Incoming, Outgoing and O.B. Line Bays*

Incoming circuits are routed via the distribution frame to the incoming lines bay where they are provided with break-jack interception before being fed to the equalisers mounted on that bay. The output of each equaliser is fed to a D amplifier on an amplifier bay and returned to break-jacks on the incoming lines bay where the equalised and amplified output is available at zero level. From this point the circuit is directed either as a source to any control position by plugging, normal inner-spring connection, or relay switching. If it is part of a circuit passing through the premises it is switched via a C amplifier to the outgoing line (Fig. 49).

The outgoing lines bay consists of a large number of break-jack interception points, together with attenuators if necessary, for all outgoing and distribution circuits. The output circuits of the C or trap-valve amplifiers are connected to the various lines via one set of listen/line/apparatus jacks, while the input circuits of these amplifiers are connected to the outputs of the control positions, D amplifiers, and other selected sources via a set of break-jacks on another bay. Thus complete flexibility is obtained for setting up destination routes and substituting spare apparatus or lines in cases of failure.

The O.B. line termination bay consists of a jack-field on which the temporary O.B. circuits are terminated in pairs. Every O.B. point is provided with two circuits which are usually identical and therefore interchangeable; one is used for programme and the other as a telephone control circuit. The O.B. bay also carries a number of repeating coils, variable equalisers of different types, and variable attenuators to which the programme circuits can be plugged and which are pre-set to equalise the known characteristics of individual lines. The output of the appropriate combination is plugged to a D amplifier input and treated like any other source. Telephone drop-indicators and an operator's telephone are also provided to which the control lines can be plugged prior to their extension to control positions.

### 6.3.3. *Amplifier Bays*

It has been found convenient to group all amplifiers together on separate bays according to their type and not to distribute them amongst other bays. The reason for this is that it allows more expeditious testing and replacement to be carried out. In the older

control rooms the following bays will be found; A amplifier bays for studio sources, D amplifier bays for line sources, B amplifier bays for control channels, C amplifier bays for line destinations, and T.V. amplifier bays for internal destinations. In current installations the corresponding bays are GP (General Purpose) amplifier bays for studio sources, line sources, or control channels, and T.V. or C amplifier bays for destinations. A jack-field is provided giving cross-plugging facilities for the input and output circuits of each amplifier; this enables rapid access to the circuits to be obtained for testing and for the substitution of a spare amplifier without interrupting programme.

### 6.3.4. *Power Supply*

Power supplies for amplifiers and test equipment used in control rooms are obtained from mains units, a number of which are fitted to each apparatus bay. Each of these units supplies a group of amplifiers, and a spare unit is carried on each bay to facilitate replacement of a faulty unit. To supply power for operating relay-switching and for signalling circuits, 24- or 50-volt batteries are used. The reason for not using mains units for these purposes is that should the mains supply fail, the relay contacts may not be restored to their previous circuit positions when power is restored. An independent battery supply is therefore of first importance.

## 6.4 OPERATIONAL POSITIONS FOR CONTROL AND S.B.

BBC operational practice demands that all programmes shall be routed and supervised by control-room operators. The regular programmes for transmission are handled in a continuity suite, while miscellaneous programmes for recording or for contribution to other centres are dealt with in the control room. The technical apparatus for continuity suites and control positions is identical and is generally mounted on a control position bay together with all the controls, but in some cases a more convenient control desk is provided.

Each programme passing through its control position must be routed to its appropriate destinations. This is done at the S.B. (simultaneous broadcasting) position either by the C-input relay switching system or by plugs and jacks on outgoing lines bays. The operator on this position (or his assistant in large premises) also deals with the engineering telephone communication traffic and all the S.B. and other control lines are centred on him.

### 6.4.1. *Control Position Bays*

These bays provide the basis of both continuity and control positions. The apparatus consists of a jack-field carrying the source circuits, any four of which can be plugged to four input channel

## THE CONTROL ROOM

faders. A telephone circuit is also provided either by a plugging operation on the S.B. telephone position in the control room or automatically with the source selection operation. In older installations cue-light circuits are provided and operated from a strip of telephone pattern keys above which are indicating lamps. The circuits are multipled to a similar set of keys and lamps on each control bay. More recently the circuit from the source is arranged to be selected to a single key and lamp appropriate to each channel via the sleeve of the source selection jack. The circuit adopted is interesting in that it provides for signalling in both directions yet only a single connection is used via the jack sleeve.<sup>1</sup>

Reverting to the programme circuit, the various sources are multipled to all the control positions so that each position can select any source and each source can, if necessary, feed more than one position. When a source is plugged to a channel (the input impedance of which is high so as not to disturb the programme volume) the programme is attenuated from zero to about  $-70$  db before the mixing point is reached to prevent cross-talk should two sources be mixed together on any one position. An amplifier is provided to raise the volume again to zero and a main control enables emergency

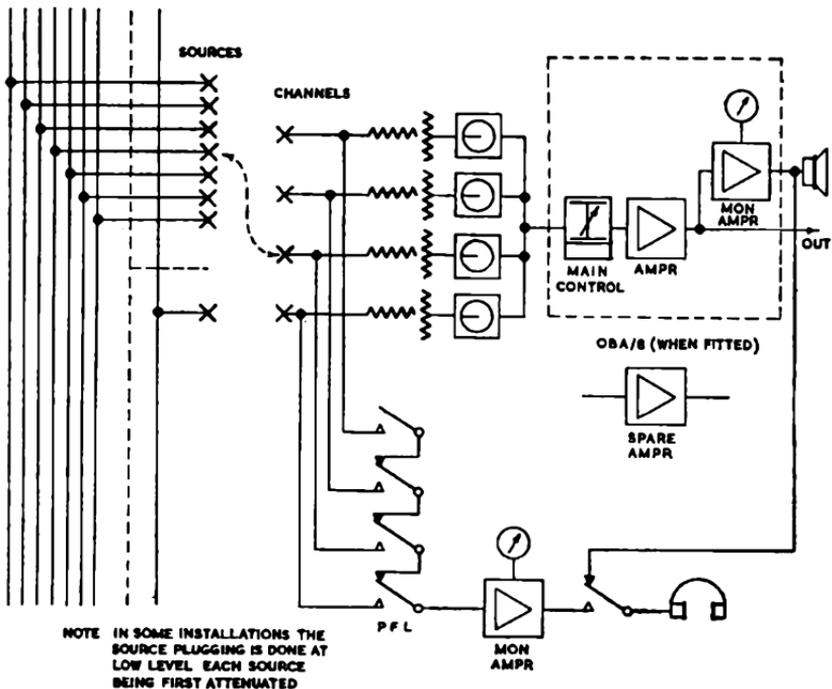


FIG. 53—Control position: programme circuit

adjustments of volume to be made if necessary. A monitoring circuit including an amplifier is also provided so that pre-fade listening can be carried out on each source circuit before the programme is faded up. The input impedance to this amplifier, like the input to the channel faders, is high so that no change in volume on the source circuit occurs when it is connected to it. A block schematic of the programme circuit is shown in Fig. 53.

### 6.4.2. *S.B. Operational Positions*

There are considerable variations in the S.B. operational positions depending chiefly upon the type of switching system in use at each centre. In some regional centres where C-input switching by relay is in use the S.B. position consists of a desk on which the switching keys, buttons and lamps are mounted, together with a manual telephone exchange for the S.B. and other control circuits. Where a plug and jack system is in use, however, the S.B. position is formed by the jack-field bays with a simple bay-desk in front. In this case both programme line routing and telephone line traffic are dealt with by manual operation of plugs and cords; in large control rooms this sometimes necessitates a complex arrangement of plug cords.

As explained earlier, a selector switching system has been developed. This enables a single operator to route many programmes to their destinations, to monitor them at various points in the chain, to pre-select several changes of route which can be implemented simultaneously, and to handle the telephone communication traffic expeditiously by connecting control circuits direct from source to destination wherever possible.

Fig. 53 shows the principles of the system, but experience indicates that this arrangement can probably be simplified.

### 6.4.3. *Overseas Programme Chain Switching*

The routing of the Overseas programmes requires many more switching operations than are generally required on the Home programmes. The reason for this is the necessity of changing periodically the frequency on which the programme is broadcast in order to take advantage of the best transmission conditions prevailing at any particular time. To maintain the service without a break, the programme is transferred from one transmitter to another at these times and a period of at least 15 minutes is allowed for transmitters to change their frequency and aerial ready to accept another service.

The transmitters are grouped at a number of sites in various parts of the country and each group is fed by a small number of S.B. lines. Although the transmitter changes are confined within each site as far as possible, there are many occasions when route changes are

## THE CONTROL ROOM

essential at the sending end. Moreover, to economise in lines, one circuit may carry one programme at one period of the day and different programmes at other times.

The programmes themselves are grouped at the source into networks and allocated a colour. Thus the green network represents the General Overseas Programme in English, while the blue is the basic European Programme using various languages at various periods of the day; each colour has its own continuity suite. The destination circuits to which the programmes are fed are called chains and the switching operation consists of connecting the various colour networks to the various chains in accordance with a schedule. This switching schedule could not possibly be put into effect without rigid timing of the programmes. The 24 hours is divided into  $\frac{1}{4}$ -hr periods and each programme must be limited in duration to a multiple of this basic unit. The switching operations are carried out at the end of each period and the switching time allocated is only 5 seconds, which must cover all variations. To enable this to be done the arrangement of routes for the succeeding period must be pre-set prior to the actual moment of switching.

This function is at present carried out on two identical jack-field bays, *X* and *Y*. Each bay carries a group of jacks, several of which are connected in parallel and fed from the output of the continuity room corresponding to each colour. A single jack on each bay is connected via a relay to each of the outgoing chains. When all the relays are unoperated the chains are all connected to the jacks on bay *X* and when they are operated the chains terminate on bay *Y* (see Fig. 54). During one period, therefore, the programme routing is being carried by one bay while the routing for the next period can be set up on the other. At the moment of change all the relays are operated and the second bay takes over the programme routing leaving the first bay for pre-setting for the next period.

Means of cross-checking the routes set up are provided both by indication of all the routes being fed by one colour when a colour button is depressed, and also by indication of the pre-set routes to be fed in the next period when a similar pre-set colour button is pressed. Comparison of the two indications shows at once what route changes will occur.

Automatic equipment has been developed so that the routing can be pre-set for each period of the 24 hours; the switching is carried out automatically to this schedule which is repeated every day until some change is made to the schedule. Overriding controls are provided to enable special or emergency routings differing from the schedule to be carried out. This system has not yet been put into operation.

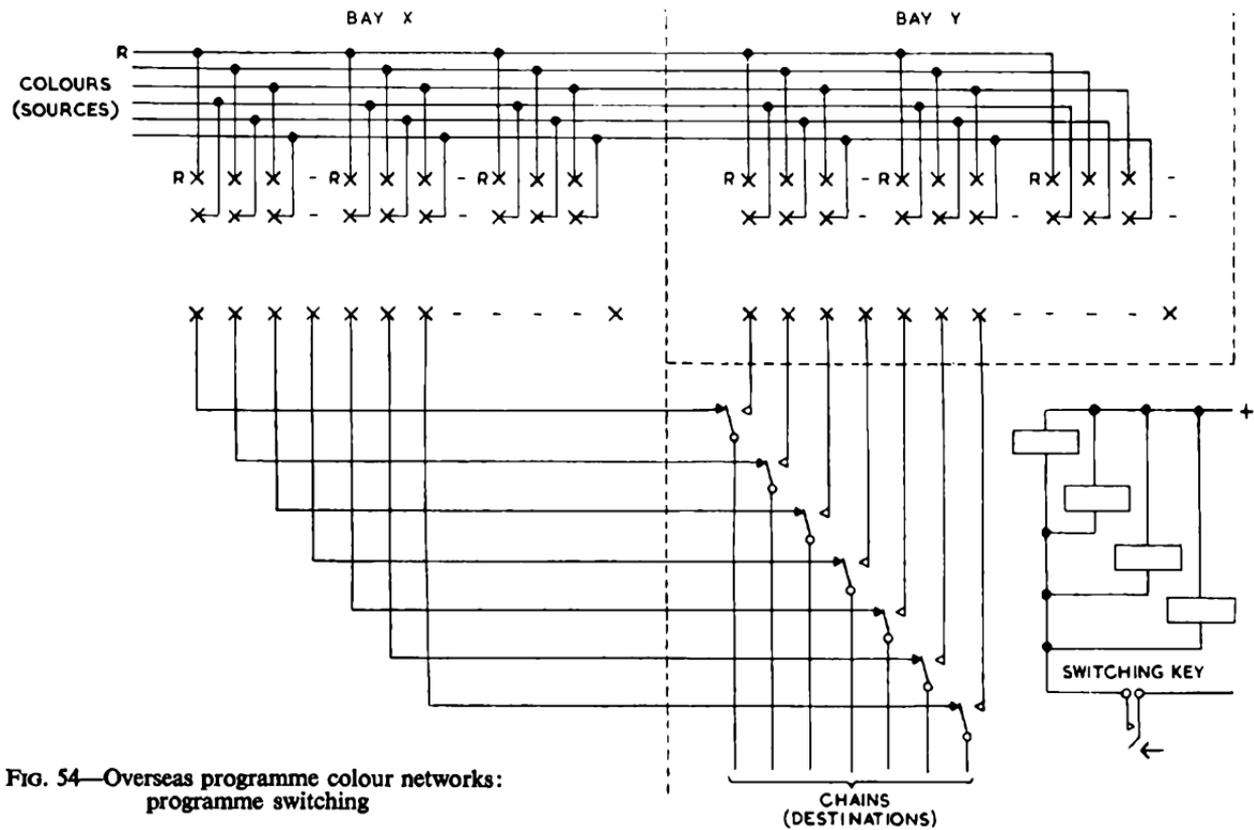


FIG. 54—Overseas programme colour networks:  
programme switching

### 6.5 TEST APPARATUS AND TESTING

It is essential that the performance of control-room as well as studio equipment should be checked periodically to ensure that no deterioration in characteristics shall occur. In general the characteristics of the apparatus which are checked fall under the following headings:

- (a) Over-all gain in the centre of the audio-frequency band.
- (b) Frequency response relative to the above.
- (c) Background noise.
- (d) Overload distortion.
- (e) Miscellaneous tests such as feed current in amplifiers and d.c. resistance of lines.

The apparatus used for these tests is described in Chapter 7 and consists of a variable tone source to supply an a.c. voltage at various frequencies, a calibrated attenuator to adjust the output level of the tone source, an amplifier-detector to measure the test levels, and a test programme meter to measure programme volume and noise. In addition, a harmonic tester, consisting of a high-pass filter which can be switched in or out of the test circuit at will, an Avometer for measuring valve feeds, and a Wheatstone bridge or bridge Megger for testing the d.c. resistance of lines are all essential items of test equipment.

This equipment is all grouped together on an A.C. Test Bay which also carries a jack-field and other miscellaneous equipment, such as a calibrated attenuator and repeating coils, and enables all routine testing in the control room to be carried out at one point. Tie-lines are used to route the test signals to the various pieces of apparatus and to studios when required.

### 6.6 CONTROL ROOM LAYOUTS

In general, a control room comprises a number of operating positions and an area in which static equipment is conveniently located. A typical arrangement of these features is as shown in Fig. 55. The S.B. position and other control positions are arranged along one side of the room, the equipment bays being placed behind the operators. These bays are often arranged in two rows, those in front carrying the line terminating jack-fields and equipment to which access is frequently required for adjustment, while those in the rear carry apparatus which requires less attention such as amplifiers, repeating coils, trunk test tablets, and the distribution frame.

In premises equipped during the war, notably in London, simple segregation of the apparatus and control positions was not easy because the operating positions are themselves bays and carry the

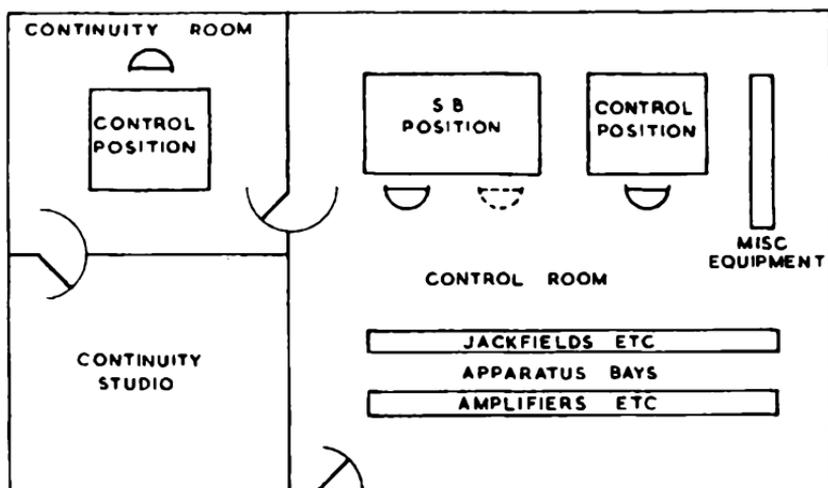


FIG. 55—Typical layout of control room equipment

line jack terminations. Different arrangements will be found in various places depending upon requirements of the premises concerned, the space available, and the previous history. The trend of modern design is to simplify the operational procedure and introduce again remote operation so that the programme traffic can be handled with the minimum number of operators at convenient control desks, and the static and remotely operated apparatus is segregated from them in an area where maintenance work only is carried out.

The continuity suite or suites carry out an essential part of the control room's function and ideally they should be located adjacent to it for efficient liaison. In fact in some regional centres the continuity position is situated in the control room as a continuation of the S.B. position, but the ideal situations are not always possible owing to the necessity of adapting existing premises for broadcasting needs.

### 6.7 TELEPHONES AND COMMUNICATIONS

The importance of telephonic communication over the engineering telephone system has already been stressed. Almost without exception, therefore, a telephone circuit is provided wherever one or more programme circuits exist. These control circuits, as they are called, are most vital where they link Regional S.B. centres since all the Engineering control traffic relating to all the programmes carried must pass over them. In order to keep them free for urgent messages, therefore, these circuits are seldom extended beyond the S.B. operator.

Telephone circuits fall into three classes, namely magneto systems, central battery signalling (c.b.s.) systems, and central battery (c.b.)

systems. In the magneto system two wires only are necessary for the circuit and long high-resistance circuits can be operated; a source of 17-c/s ringing current must be available, however, at each subscriber's instrument. In the c.b.s. system this requirement is not necessary but a local dry battery is required to operate the speech transmitter. In the c.b. system all the power necessary is derived from a central source at the exchange but the earth return circuit is also required.

In the BBC the magneto system is used in general because of its simplicity. At remote points such as O.B.s, a telephone instrument with a magneto generator for ringing is required, but at other points a supply from a central 17-c/s source is provided and a ringing key is all that is necessary. In the range of standard bays a control-line bay provides jack terminations for a number of lines with the inner springs of the jacks connected to drop-indicators. A ringing signal on the line therefore operates the appropriate indicator while insertion of the answering plug cuts off the indicator. In previous and later installations this call indication is provided by lamps operated by a relay which is automatically cancelled when the call is answered.

Some telephone circuits in the engineering system are on the c.b.s. system but these are gradually giving way to the simpler magneto system. A fuller account of the BBC Communication System is given in Chapter 9.

## 6.8 PROGRAMME VOLUME

In its progress through the equipment in studio premises the programme volume undergoes many changes for various reasons. A typical level diagram is given in Fig. 56 corresponding to post-war installations, and some explanation of this diagram is necessary.

It should first be understood clearly that the term "programme volume" as explained in Chapter 2 has no precise meaning until the programme has been controlled. Prior to this point (i.e., the main control) reference can strictly only be made to maximum peak level and minimum peak level, these indicating the programme peaks occurring when (say) an orchestra is playing a fortissimo or a pianissimo passage respectively. However, an *average programme level* in this part of the chain is often referred to and is taken to represent the programme volume produced by a trained broadcasting speaker with a strong voice; the programme volume of such a speaker once set seldom needs any further control.

As explained above, the normal volume to be expected from the microphone will vary over wide limits and these variations are even greater due to the differing sensitivity of various types of microphone. The average levels shown on the diagram refer principally to the

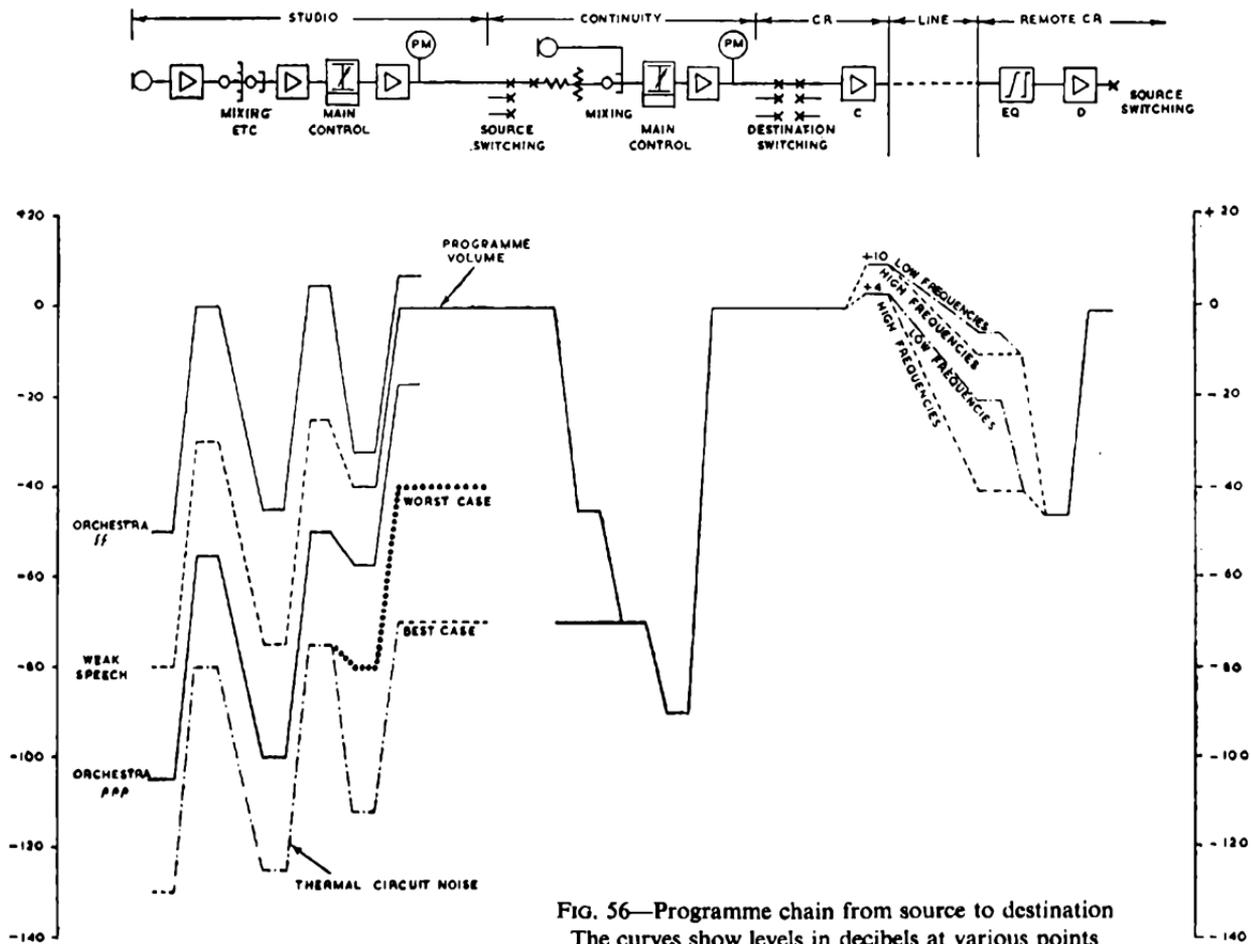


Fig. 56—Programme chain from source to destination  
The curves show levels in decibels at various points

## THE CONTROL ROOM

ribbon microphone extensively used by the BBC, and although  $-70$  db may be taken as a mean volume for a practised speaker, a level of  $-80$  db is shown for a weak speaker for which sufficient gain must be provided. This level is raised before or after mixing of the microphone circuits, according to the type of mixing circuit employed, and arrives at the main control at a level of the order of  $-20$  db.

The main control consists of a variable attenuator, and must be set at an optimum position which permits the introduction of additional loss on loud passages in a programme or less loss for weak items. The mean setting is that which gives the correct volume on average speech as defined in the preceding paragraph.

In modern practice all programme switching is carried out at zero volume, but the volume sent to S.B. and other outside lines is usually  $+4$  db with some variation in individual cases. Since the impedance of a line varies with frequency and the sending impedance is 600 ohms resistive, the level sent to line is somewhat arbitrary; it is therefore specified as the voltage which would be developed across a 600-ohm resistance connected in place of the line, and is usually measured in this way.

## REFERENCE

1. BBC Technical Instruction S5, Appendix 2.

## CHAPTER 7

### PROGRAMME CIRCUITS ON POST OFFICE LINES

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#### 7.1 INTERNAL AND EXTERNAL LINES

In previous chapters, we have discussed the broadcasting chain as a whole, without special reference to the lines which connect one part of the chain to another. These lines comprise (a) internal or "tie" lines between BBC local premises, e.g., studio and control room, and (b) external lines rented from the Post Office for long-distance connections. It is easy enough to connect a source to a central control room when both are in the same building. Nothing more is involved than a pair of insulated wires with an outer lead casing, which acts as a protection from mechanical damage or electrical interference. Such a pair consists of copper conductors each weighing 10 lb for a length of 1 mile and is known as 1 pair/10. If, however, the length of conductor for tie-lines between source and control room exceeds about a mile, steps must be taken to ensure that the original source signals are not distorted to an intolerable degree when reproduced on a good loudspeaker.

1 pair/10, and frequently 10 pair/10 internal tie-lines, are very largely installed by BBC staff, but when connections are made from outside premises they are the Post Office Engineering Department, which is responsible for controlling and maintaining telephone lines throughout the country.

For economy reasons, the Post Office does not run its tie-lines in either 1 pair/10 or 10 pair/10, but uses cables containing a large number of pairs; the actual number varies with local requirements and may be 50, 500 or more. Individual wires are insulated from each other by paper and the whole is surrounded by a lead sheath and referred to as a cable. For reasons which will be clearer later the weight of copper employed per wire per mile depends to a large extent on the distance of the locality from the main Post Office centre.

#### 7.2 GENERAL TRANSMISSION PROBLEMS

##### 7.2.1. *Non-loaded Lines*

A pair of wires in a cable consists of two conductors twisted to-

gether but separated from each other by a crimped paper insulator, which means that the dielectric between the conductors is effectively air.

There are four fundamental parameters which can affect the transmission of alternating currents along a line:

- (i) the copper conductor has a resistance per unit length,
- (ii) it has an inductance per unit length,
- (iii) the two wires, running close together, form a shunt capacitance, and
- (iv) there is an a.c. loss in the dielectric.

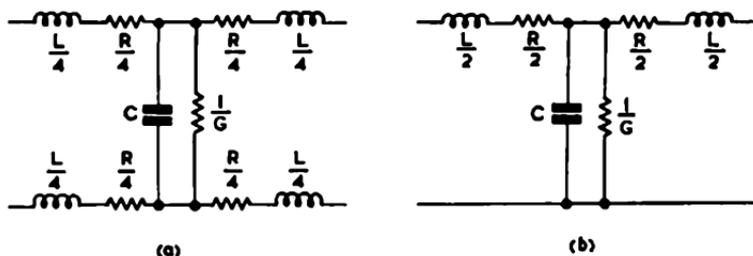


FIG. 57—Balanced and unbalanced networks representing small section of P.O. line

A small section of line can thus be represented by the balanced and symmetrical T network shown in Fig. 57 (a). Fig. 57 (b) is a simplified unbalanced form.

$L$  is the inductance in henrys per unit length

$R$  is the resistance in ohms per unit length

$C$  is the capacitance in farads per unit length

$G$  is the conductance in mhos per unit length

The unit of length in the British Isles is 1 mile. For a mathematical treatment of these networks, the reader should refer to the textbooks 1, 2, 3, listed on page 161. In this chapter, we shall only study their effects on the transmission of alternating currents.

If a network similar to Fig. 57 is followed by an infinite number of others identical with it, the current entering the network, and the impedance looking into it, will be unchanged at any frequency if the number of sections is increased or reduced at either end. This impedance condition is known as the *characteristic impedance* of a line or chain of networks of identical design.

Characteristic impedance is defined by the C.C.I.F. (Comité Consultatif International Téléphonique) as “the input impedance of a uniform line, or of an infinite chain or recurrent symmetrical networks”.

The natural logarithm of the vector ratio of the current entering a section to the current leaving the same section is known as the *propagation co-efficient*, i.e.,  $\log I_1/I_0 = P$ . This is a complex quantity, the real part of which is known as the *attenuation co-efficient*  $\alpha$  and the imaginary part as the *phase co-efficient*  $\beta$ . ( $P = \alpha + j\beta$ ).

$\alpha$  is in nepers (1 neper = 8.686 db) and  $\beta$  is in radians. The above quantities can be expressed in terms of the four line parameters as follows:

$$\text{Characteristic impedance, } Z_0 = \sqrt{\left[\frac{(R + j\omega L)}{(G + j\omega C)}\right]} \dots\dots\dots(1)$$

Here the unit of length cancels out and the characteristic impedance is therefore independent of the length.

$$\text{Propagation co-efficient, } P = \sqrt{[(R + j\omega L) \cdot (G + j\omega C)]} \dots\dots\dots(2)$$

In this case the unit of length does not cancel out and the propagation constant is a function of the unit of length in which the parameters are measured.

For a 10-lb cable to British Post Office specifications the resistance per loop mile is 176 ohms, and average quantities for the inductance, capacitance and conductance per mile, are 1 mH, 0.065  $\mu$ F, and 1  $\mu$ Mho at 800 c/s.

The conductance is directly proportional to frequency and is small enough compared with  $\omega C$  to be neglected as a first approximation.

$$\text{Thus } Z_0 \simeq \sqrt{\left(\frac{R + j\omega L}{j\omega C}\right)} \dots\dots\dots(3)$$

$$P \simeq \sqrt{[j\omega C(R + j\omega L)]} \dots\dots\dots(4)$$

At 1 kc/s  $\omega L = 6.28$  ohms  
and at 10 kc/s  $\omega L = 62.8$  ohms

For the case under consideration we can ignore  $\omega L$  also as even at 10 kc/s its effect when added in quadrature to  $R$  is insignificant compared with the effect of  $\omega C$ .

$$\text{We then get } Z_0 \simeq \sqrt{\left(\frac{R}{j\omega C}\right)} = \sqrt{\left(\frac{R}{\omega C}\right)} \cdot \sqrt{45^\circ} \dots\dots\dots(5)$$

$$\text{and } P = \alpha + j\beta \simeq \sqrt{(jR\omega C)} = \sqrt{\left(\frac{R\omega C}{2}\right)} + j\sqrt{\left(\frac{R\omega C}{2}\right)} \dots\dots\dots(6)$$

$$\text{i.e., the attenuation constant } \simeq \sqrt{\left(\frac{R\omega C}{2}\right)} \dots\dots\dots(7)$$

These approximations must be used with caution and  $L$  and  $G$  must necessarily be taken into account when accurate determination of the above quantities are required.

Proceeding, it is quite easy to see that:

PROGRAMME CIRCUITS ON POST OFFICE LINES

- (a) characteristic impedance increases with increase of resistance but decreases with increasing capacitance and frequency,
- (b) attenuation increases with increase of resistance, frequency or capacitance.

The resistance per mile depends on the weight of conductor, and the capacitance is kept as low as is economically possible and does not vary greatly for different types of cable. It covers the range  $0.054 \mu\text{F}/\text{mile}$ , in low-capacity cables used for long-distance transmission at high frequencies, to  $0.078 \mu\text{F}$  for short local telephone subscribers' cable.

So far we have considered only the infinitely long line and  $Z_0$  and  $\alpha$  are quite independent of the sending and receiving impedances. In BBC practice the receiving impedance is nominally 600 ohms. The sending impedance is 600 ohms at main stations and 75 ohms at outside broadcast points. We must now go a step farther and take these impedances into account.

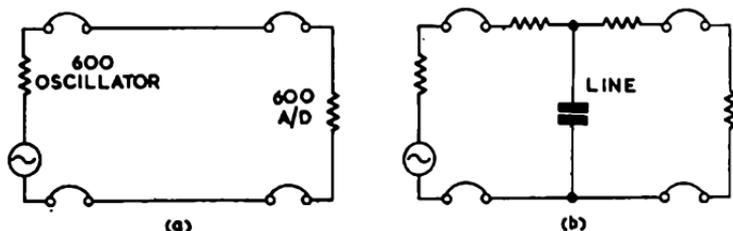


FIG. 58—Representation of insertion loss

Fig. 58 (a) shows a 600-ohm generator feeding into a 600-ohm amplifier detector, the circuit being lined up so that the amplifier-detector will read zero level (1 mW) at every frequency provided  $E$  (1.55 V) is independent of frequency.

In Fig. 58 (b) a line represented by a symmetrical T network has been inserted between the oscillator and amplifier-detector; the latter will no longer read zero for a constant value of  $E$ . Instead it will read the loss caused by the line at each frequency. This is known as the *insertion loss*. The insertion loss of a line is different from the attenuation of the line (i.e., the attenuation co-efficient multiplied by the length of the line) because the latter is quite independent of the line terminations whereas the former takes account of the fact that the terminating impedances are not the same as the line characteristic impedance. Now the characteristic impedance varies as the frequency varies, whereas the 600-ohm terminations used above do not; therefore the current entering the network divided by that

leaving the network cannot be the same in both cases. Consequently we cannot get full transference of power from generator to line or from line to receiving impedance at all frequencies. The degree of transference will vary considerably over the frequency range. For the type of line under consideration the impedance at high frequencies is considerably lower than 600 ohms and this mismatch augments the already increasing attenuation with increase in frequency.

Assuming the insulation resistance to d.c. is infinity, the attenuation of a line at zero frequency (i.e., d.c.) is zero, because the line, extended to infinity, merely consists of a series resistance, and the current leaving any mile section will be the same as that entering it. The insertion loss, however, will not be zero and if  $R_1$  is the d.c. resistance between two 600-ohm impedances the insertion loss in db will be

$$20 \log_{10} \left( 1 + \frac{R_1}{2 \times 600} \right)$$

This will be the minimum loss and it will remain constant up to about 50 c/s at which frequency the attenuation will have increased sufficiently to affect insertion loss.

### 7.2.2. Loaded Lines

It will have been noticed that the capacitance is the factor responsible for increasing the insertion loss with frequency, by virtue of its effect on the attenuation and characteristic impedance.

Let us take another look at the approximate line formulae

$$Z_0 = \sqrt{\left( \frac{R + j\omega L}{j\omega C} \right)} \quad \text{and} \quad P = \sqrt{[(R + j\omega L)j\omega C]}$$

Suppose the value of  $L$  is increased until  $R$  is so small compared with  $\omega L$  that it can be neglected, then

$$Z_0 = \sqrt{L/C} \quad \text{and} \quad P = j\omega\sqrt{LC}$$

i.e.,  $Z_0$  is independent of  $\omega$  and is a resistance. If  $Z_0$  is made equal to the terminating impedances complete transfer of power from the sending end results. Further, as  $Z_0$  is made up of pure reactances only, no power can be lost in the line and the attenuation is zero, as is shown in the formula for  $P$  which has only an imaginary component. This result can only be achieved by ignoring  $R$  and  $G$ , whereas in fact these parameters do have a sufficiently significant effect to cause a loss. As an illustration let us assume  $R$  is significant

and  $G$  negligible; it can be shown that  $\alpha = \frac{R}{2(Z_0)}$ . Thus increasing  $Z_0$

will reduce the attenuation. As  $Z_0$  is approximately equal to  $\sqrt{L/C}$  it would appear that the greater the inductance the smaller the attenuation. Unfortunately there is a limit to the increase of induc-

tance which will produce an economic reduction in attenuation, owing to the coil resistance.

It is quite plain nevertheless that the addition of inductance, termed loading, has beneficial results. There are two ways of adding inductance:

- (a) continuous loading,
- (b) lumped loading.

Continuous loading will be found in submarine cables and is achieved by winding a tape of magnetic material round each wire of the pair. This is an expensive way of doing it, and the amount of inductance added is limited by the permeability of the magnetic tape. Values of 10-20 mH/nautical mile were used in older cables, but with the advance in magnetic tapes, higher values have been achieved.

Continuous loading has been used on submarine cables only because it eliminated difficulties in making water-tight joints which would have been required for *lumped loading* (i.e., inductance coils in "loading pots" added at defined intervals along the cable). Lumped loading is always used on land cables.

There is one important effect of lumped loading that does not occur with continuous loading. Formulae (1) and (2) were derived on the assumption that  $R$ ,  $L$ ,  $C$  and  $G$  are uniformly distributed and are not actually applicable when some of the parameters are lumped as in the loaded cable. If we assume that  $R$  and  $G$  can be neglected, sections of the loaded cable will appear as in Fig. 59, the capacitance

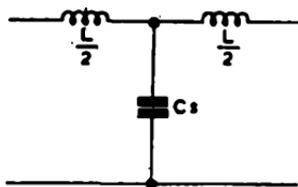


FIG. 59—Representation of loaded cable section

$C_s$  being the capacitance of the cable between loading points ( $C$  being the capacitance/mile and  $s$  the distance in miles) and  $L$  is the loading coil inductance. This looks like, and indeed it is, a low-pass filter with a cut-off frequency  $f_c = 1/\pi\sqrt{LCs}$ .

Thus lumped loading limits the bandwidth it is possible to transmit on the line and the higher the inductance the smaller the bandwidth will be. A table of commonly used loadings and the resultant cut-off frequencies is given below, together with the maximum usable bandwidth (80 per cent of  $f_c$ ). Fig. 60 compares the insertion loss characteristics of a non-loaded and a loaded line. On first inspection of Fig. 60, the upper curve would appear to be the better, but closer

STUDIO ENGINEERING FOR SOUND BROADCASTING

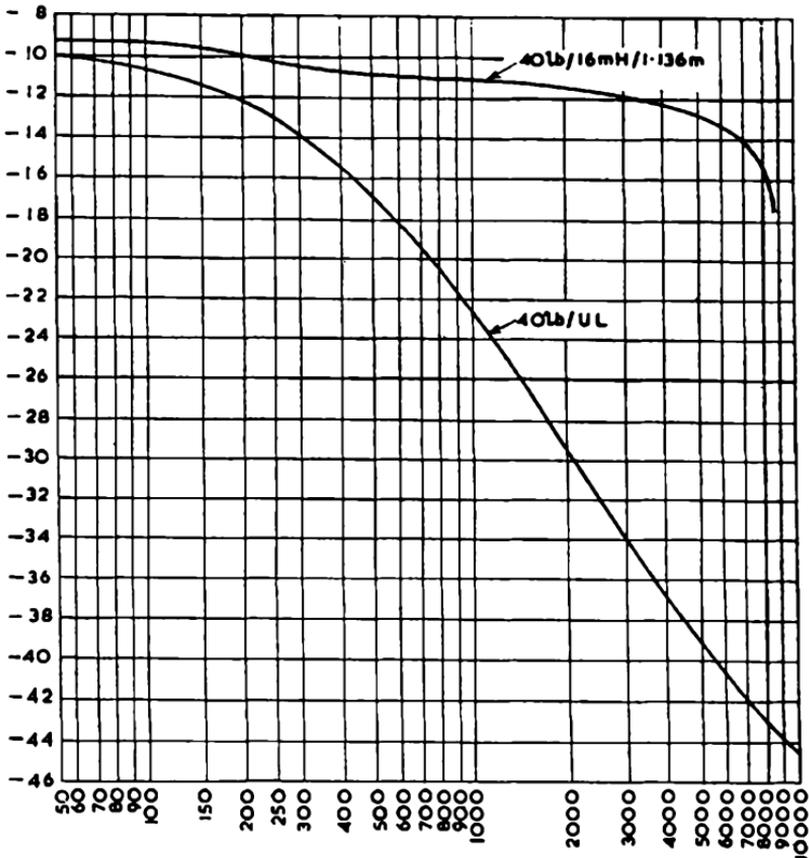


FIG. 60—Comparison of insertion loss on loaded and non-loaded lines

examination shows that whereas the upper curve begins to fall off rapidly at 8 kc/s, the lower curve has a gradual fall up to and beyond 10,000 cycles. If the loss producing the lower curve is correctly equalised, the non-loaded cable will give a better response at high frequencies than the loaded cable.

<i>Loading Coil Inductance (mH)</i>	<i>Spacing (miles)</i>	<i>Weight of con- ductor (lb/mile)</i>	<i>Approx. Characteristic Impedance (ohms)</i>	<i>Cut-off Frequency (kc/s)</i>	<i>Max. usable freq. (nom.) (kc/s)</i>
176	1.136	20	1,860	2.32	1.86
88	1.136	20	1,110	3.92	3.14
44	1.136	40	780	5.57	4.46
22	1.136	40	550	7.91	6.33
22	0.568	40	780	10.90	8.72
16	1.136	40	490	9.30	7.44

## 7.3 PROGRAMME CHANNELS ON CARRIER CABLES

We have considered so far unloaded and loaded cables which are, in general, used by the Post Office at audio frequencies. In 1938 the Post Office began a big expansion in their communication network by laying 12-channel carrier cables. These are low-capacitance cables specifically designed for the transmission of frequencies up to 60 kc/s and subsequently up to 120 kc/s. They consist of 24 pairs of 40-lb conductor and no specific provision is made for pairs to be used exclusively for programme transmission.

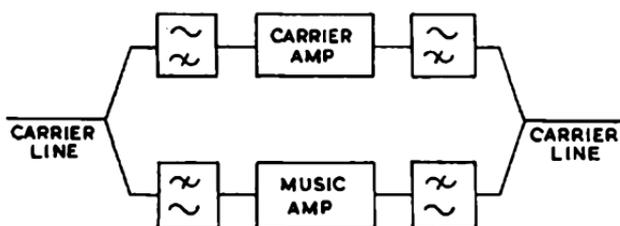


FIG. 61—Programme channel on carrier cable

It is, however, possible to derive music channels on the carrier pairs. Fig. 61 shows the first method adopted to produce what is known as a *sub-carrier* programme circuit. The 12-channel carrier system uses a band of frequencies from 12 kc/s to 60 kc/s, the band below 12 kc/s is thus available for a programme channel, but necessitates the use of filters at each repeater station as it is essential to amplify the two sets of signals separately. The cost of an amplifier is almost directly proportional to the number of octaves in the bandwidth it has to amplify, and it is obviously more economical to design the amplifier for the carrier bandwidth and supply an audio amplifier when required.

Filters are an expensive item and a development known as the *carrier phantom* programme channel was produced as shown in Fig. 62.

The carrier frequencies are carried on the physical (or side) circuits and the programme channel is on the Phantom derived from two carrier pairs.

Consider the path of the circulating current of the programme channel. Starting from the sending music amplifier, at *P* it divides equally between the two halves of the line transformer or repeating coil. Thus the fluxes generated by the two half currents are equal and opposite, no inductive reactance is offered to their passage and there is no flux linkage with the apparatus side, and therefore no interference with any carrier signals at that point. The current proceeds

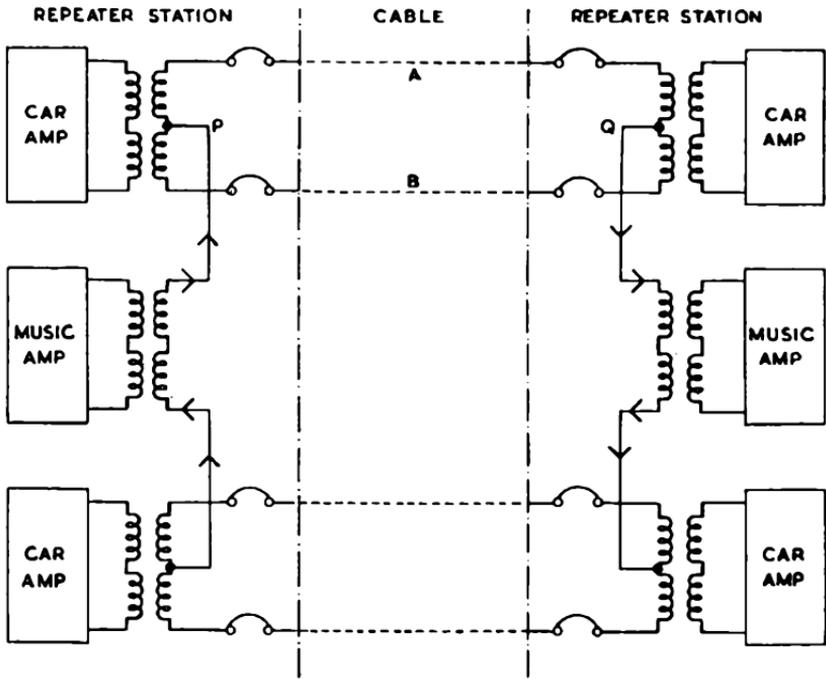


FIG. 62—Carrier phantom programme channel

in the same direction along both wires of the first pair up to the second line transformer. As the two wires of the side circuit are identical, the voltage drop along wire *A* will be the same as that along wire *B*. There will, therefore, be no potential difference across the line side of the line transformer. As before, the two currents are combined at the centre point *Q* of the transformer, again without interfering with the carrier signals as there is no flux linkage with the secondary. The two wires and the line transformers thus form one leg of the music channel and the return leg is formed by the second pair. If the side circuits are perfectly balanced there is no carrier voltage between the centre points *P* and *Q*, and no carrier current is induced in the phantom path. In practice, imperfections in the equality of the line transformer windings and slight unbalance in the pair of wires cause some of the carrier signals to appear in the phantom path, but as the minimum frequency is 12 kc/s they will not normally cause interference and filters in the phantom path are quite unnecessary, although of course separate amplifiers are still needed.

Use has been made of co-axial pairs for the transmission of audio-frequencies. Co-axial pairs were designed primarily for the transmission of frequencies from 60 kc/s to an approximate upper limit

of 4 Mc/s. On land, earth currents are induced in the outer sheath of the co-axial because the earth is not a perfect conductor, and the frequency band below 60 kc/s is unusable for programme purposes because of the induced noise. Submarine co-axials are immune from this type of induction owing to the high conductivity of the surrounding water, and it is possible to derive an audio channel. The ends must, however, be terminated in line transformers as soon as the cable leaves the sea, the circuit being continued in normal pair cable on land. Filters are required to separate the audio from the carrier frequencies at each terminal. The BBC rents three audio circuits of this type between Stranraer (Scotland) and Donaghadee (N. Ireland).

#### 7.4 NOISE INDUCTION

The term "noise" as used here implies any unwanted electrical currents likely to cause interference with the wanted signal; for instance, such noises might be caused by speech channels and other music circuits producing cross-talk. We must now investigate how unwanted noise is generated in a pair of wires and how it is prevented from causing serious interference with the wanted signals.

We have already seen that there is a capacitance between the two wires of a pair. There is, of course, in addition a capacitance between each wire and "earth" (the lead sheath). Further, each wire of this pair will have a capacitance to each wire of adjacent pairs in the same

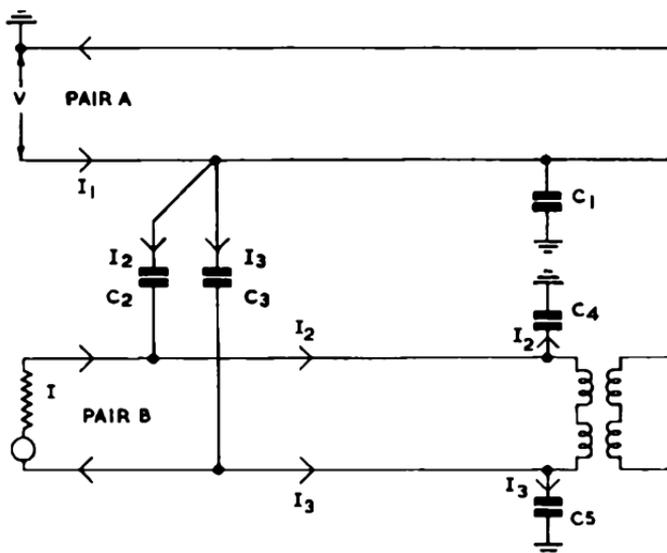


FIG. 63—Distribution of capacitance between adjacent lines

cable. For the sake of simplicity it will be assumed that adjacent to pair  $B$  there is a disturbing circuit consisting of a pair of wires, one leg of which is earthed. The capacitance system will be as shown in Fig. 63, and if the potential difference across pair  $A$  is  $V$  it will be noted that there are current paths which include the two wires of pair  $B$ . There are, of course, other current paths because the capacitances are in fact distributed throughout the length of the wires. They are only shown bunched at chosen points to simplify the argument. Currents such as  $I_2$  and  $I_3$  which flow in the same direction in both wires of a pair are known as *longitudinal currents*. If, for example,  $C_2 = C_3$  and  $C_4 = C_5$  then  $I_2 = I_3$  if the resistance of one leg is equal to that of the other leg of pair  $B$ , and there will be no potential difference due to  $V$  across pair  $B$ . If, however, the resistance of one leg of pair  $B$  should be greater than the other (known as *resistance unbalance*) and due perhaps to a dry joint, a potential difference will result. Moreover, if  $I_2$  should differ from  $I_3$  (due to *capacitance unbalance*) a potential difference will be created even if the two legs of pair  $B$  are of equal resistance. As, even at the higher audio frequencies, the impedance of the coupling capacities is high compared with the line resistance the values of  $I_2$  and  $I_3$  for a given potential  $V$  are, near enough, directly proportional to frequency in the audio-frequency range. Thus, as the frequency increases so  $I_2$  and  $I_3$  increase. Therefore, provided  $V$  is constant, the noise potential difference induced in pair  $B$  will be greater at the higher audio frequencies for both resistance and capacity unbalance. This form of induction is known as *electrostatic cross-talk*.

Cables can suffer mechanical damage which might fracture the lead sheath and let in moisture, causing a severe drop in insulation resistance between pairs and each wire and "earth". If the insulation resistances are unequal they also can be the cause of the generation of noise in the wanted signal path. Moreover, if the insulation resistance should drop to something well below the impedance of the capacitative paths,  $I_2$  and  $I_3$  will tend to be the same at all frequencies and low-frequency noise will be as prevalent as high-frequency noise.

A circuit can be screened from electrostatic induction by wrapping round it a helical lapping of metallised paper or thin metal foil. The current paths of  $I_2$  and  $I_3$  can then only be through the screen. The screen performs its function without being earthed. Earthing would in fact be disadvantageous as it would enable earth currents to circulate close to the screened pair and noise might be induced electromagnetically. There are a number of cables in which a few of the central pairs have been treated in this way for the express purpose of making them ideally suitable for the transmission of programme. It has frequently been found that when a cable has suffered damage,

these screened pairs have continued to give good service long after most other pairs have become unusable.

A circulating current in one pair sets up a magnetic field which induces longitudinal potentials in adjacent pairs. The relative magnitude of these potentials will depend on the geometrical positions of the wires with respect to each other, and the amount of cross-talk will depend on the resultant of these potentials. This is known as *electromagnetic cross-talk*. At audio frequencies it is negligible compared with electrostatic cross-talk for the currents normally permitted in cables. Pairs are twisted throughout their length and the effect of the twist is to bring the two legs of a pair alternately into magnetic fields of equal intensity, thus balancing the induced potentials. Even induction from near-by power supplies is minimised in this way, and it is important that only twisted-pair cable (single-pair cable included) should be used for programme.

Cross-talk has only been touched on here, and the reader is referred in the bibliography<sup>4</sup> to a paper which gives an excellent treatment of the subject.

In the majority of cables the pairs are "quadded", that is to say the cables are composed of a number of quads, each quad consisting of four wires. In star-quad cables the four insulated wires are twisted together symmetrically about a common axis, whilst in multiple-twin quadded cables the single insulated wires are first twisted in pairs, the two pairs then being twisted together to form a quad.

Both electrostatic and electromagnetic inductions are kept to a minimum in a manufacturing length (usually 176 yds) by the uniformity possible with modern manufacturing processes, and also by the way in which quads are individually twisted and laid up in the cable. A selection process is used when jointing lengths to keep capacitance unbalances at a minimum.

In spite of careful manufacture and selection of wires in the jointing of the cable there are bound to be residual unbalances, because economic limits must be fixed for cable manufacture, and subsequent changes can occur with slight mechanical distortion of the cable during handling, laying and creepage with time. Thus, a capacitance unbalance and the resultant high-frequency cross-talk are evils which must be accepted.

## 7.5 REPEATERS

We have already seen that on non-loaded cables, high-frequency attenuation is severe and something must be done well before the high-frequency signal drops to a level comparable with that of unavoidable cross-talk.

The obvious thing to do is to amplify the wanted signals at a

suitable point, where the signal-to-noise ratio is acceptable. In general this would have to be at some point before the high frequencies had suffered more than 40 db attenuation. This is not so simple as it appears, because at this point low frequencies may have been attenuated very little, if at all. Suppose the original sending level to have been 1 mW, and that high-frequency signals were attenuated 40 db; to restore the high-frequency signals to the original level, 40 db amplification is necessary. If the unattenuated low-frequency signals were amplified to the same degree, the power levels at the output of the amplifier would vary between 1 mW at high frequency and 10 watts at low frequency. It would, of course, be uneconomic to design amplifiers to handle levels of such great divergence; moreover, high levels of the order quoted would be unsuitable for transmission over Post Office lines. Before amplification, therefore, the levels at low frequencies must be attenuated to those of the high frequencies. This is achieved by preceding the amplifier by an *equaliser*\* (p. 142).

We have seen that the purpose of equalising and amplifying is to maintain an adequate signal-to-noise ratio; under certain circumstances, the amplifier itself may give rise to the introduction of line noise. In Fig. 64 is shown the grid of the first valve preceded by a

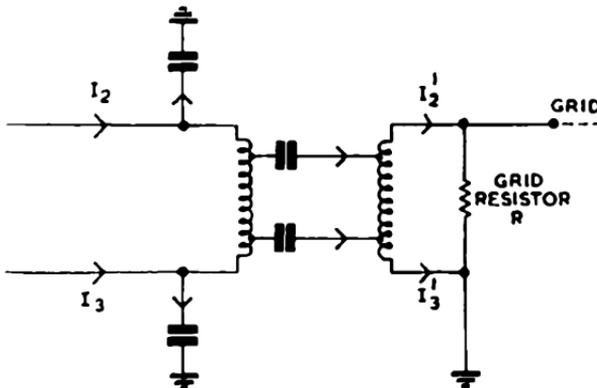
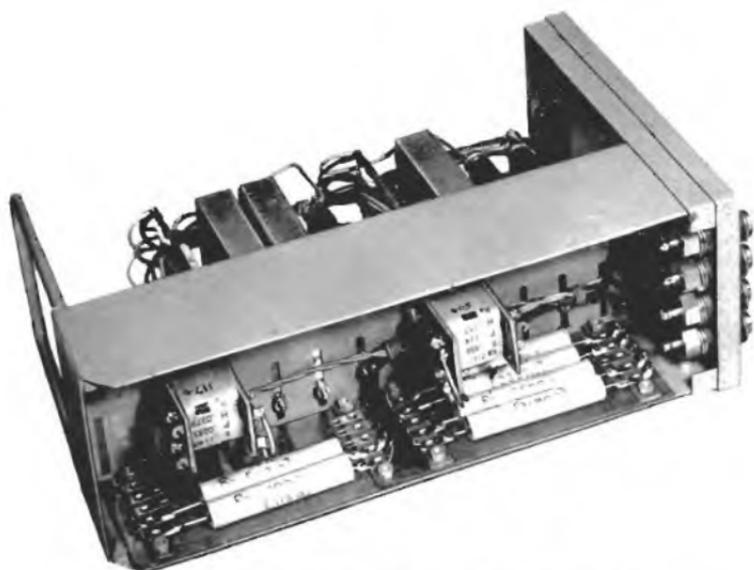
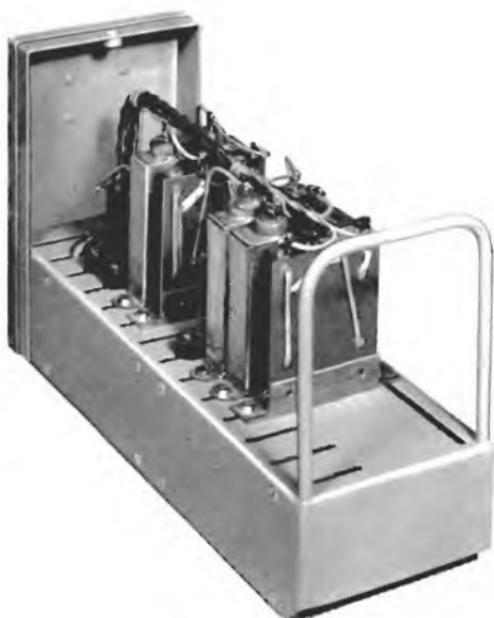


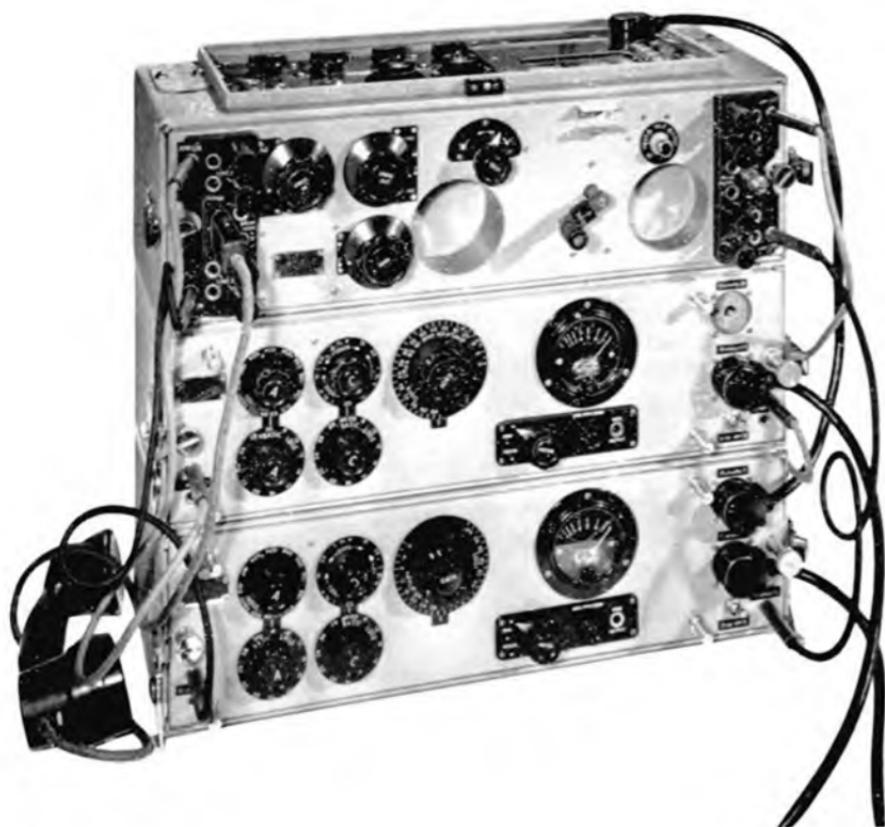
FIG. 64—Effect on amplifier noise using unscreened input transformer

simple form of input transformer and it will be assumed for the moment that the input transformer is used as the line transformer terminating the line. If there were no transformer at all, the earth of the amplifier would be applied directly to one leg of the line and  $I_2$  could produce the full noise potential across the grid resistor  $R$ . Interposing the input transformer only partially mitigates this because the transformer primary and secondary are wound in juxtaposition, and there is obviously a considerable coupling capacitance

\*A line treated in this manner is usually referred to as a *repeated* line.



EQUALISER BUILT TO ORDER ON STANDARD CHASSIS



PORTABLE REPEATER SET

between primary and secondary throughout the length of the winding. Thus  $I_2$  can still find a capacitive path to earth through the transformer and  $I_2$  is therefore still capable of producing a noise potential across  $R$ . The coupling capacitance should therefore be reduced.

This can be done by interposing a metallic screen, connected to earth, between the two windings (see Fig. 65).

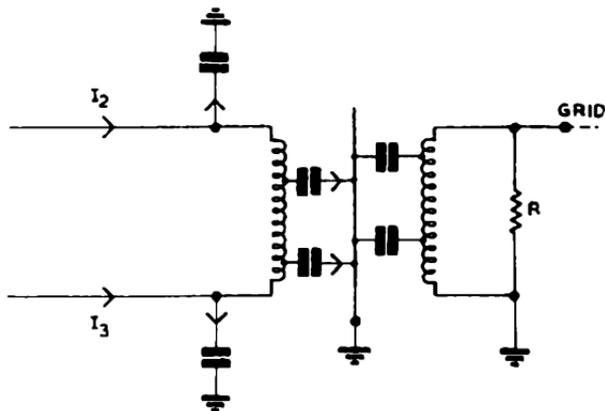


FIG. 65—Effect of screening amplifier input transformer

The effect of the screen is to tie both windings capacitatively to earth potential, and longitudinal currents find an easy path to earth before reaching the secondary of the transformer. The coupling capacitance is theoretically zero and no potential difference across  $R$  can be caused by  $I_2$  and  $I_3$ . The paths to earth for  $I_2$  and  $I_3$  must have the same impedances, otherwise  $I_2$  and  $I_3$  cannot be equal and a potential difference will arise across the line winding of the transformer.

## 7.6 REPEATING COILS

Although as a general rule, amplifier input transformers have a screen, the design requirements of a transformer terminating a line must be stringent; any imperfections in the effectiveness of the screen will result in a coupling capacitance between the two windings. The input transformer of an amplifier has usually other important requirements to meet such as a high step-up ratio, low leakage inductance and high shunt inductance, all of which increase the difficulties of meeting the line side requirements. Moreover, it is generally necessary to precede the amplifier by an equaliser, which for economic reasons and space requirements is generally unbalanced, i.e., all series-arm components are concentrated in one leg instead of being divided equally between the two legs. A separate low-ratio

transformer called a *repeating coil* is therefore invariably used for terminating a line.

The requirements it must meet are as follows:

- (i) Low coupling capacitance between line side windings and apparatus windings.
- (ii) The line side winding to be accurately balanced to screen with respect to capacitance throughout its length.
- (iii) The shunt inductance to be high to avoid low-frequency loss.
- (iv) The leakage inductance to be low to avoid high-frequency loss.
- (v) The coil used by the BBC has both line-side and apparatus-side windings split into two equal parts, thus enabling the use of phantoms on either the line or apparatus side. The two halves of each winding must be as near as possible identical for shunt inductance, resistance and capacitive balance to screen, otherwise, as described previously, there will be cross-talk between phantom and sides.

Balance to screen is only vital on the line side where substantial longitudinal currents are always present. The required balances are in general achieved by winding the line side in a twisted pair of wires. Each wire is then one half winding. The apparatus side is wound more conventionally and the balance is not as good. It is therefore essential to ensure the correct winding is connected to the line.

#### 7.7 REQUIREMENTS FOR GOOD TRANSMISSION ON LINES

Before proceeding farther, we must consider what the requirements are for the transmission of broadcast programmes. The sound pattern to which the human ear will respond extends over the frequency range 16 c/s to 20 kc/s, but it is by no means necessary to be able to produce the whole of this frequency range in order to achieve something almost indistinguishable from the original. It is probable that the majority of listeners cannot hear much over 14 kc/s. A number of subjective tests have been made with the object of learning just what bandwidth should be transmitted. The greater the bandwidth the greater the cost to achieve it; on the other hand something more is required than the conveyance of merely an intelligible signal. The latter is possible with quite a small bandwidth. Other requirements are that the harmonic or intermodulation distortion of the system shall not be such as to produce audible, unpleasant and unharmonious frequencies. Moreover, cross-talk and other noises must not be noticeably audible under ordinary listening conditions.

Apart from transmitting the required frequency bandwidth we must also ensure that any distortion caused by the line to the relative levels of individual frequencies in the sound signal are corrected by equalisation.

A line, as already noted, consists simply of a pair of copper wires, and although it causes unequal attenuation of individual frequencies, it should not produce harmonics. If, however, a bad joint is made in one leg of the pair, apart from causing cross-talk from adjacent pairs the surfaces in near contact at the joint might become oxidised, resulting in a small local rectifier which can cause partial rectification of programme signals and introduce distortion. It is unlikely to happen in a cable, as, although the wires are twisted together at the joints, there is not enough dampness to produce oxidation, but it can happen on jumper frames and other connection points. D.C. tests are made on circuits to check the perfection of jointing.

Care must be taken that amplifiers in the circuit are not overloaded at any frequency. To prevent this, and consequent distortion, two precautions must be taken: the line must be equalised before amplification, to avoid overloading at frequencies where attenuation is low and the gain must be adjusted so that the output volume is within the range for which the amplifier is designed.

The minimum requirements for a circuit to be suitable for the transmission of broadcast programmes are laid down by a body known as the Comité Consultatif International Téléphonique. They are as follows:

### *7.7.1. C.C.I.F. Specifications for Transmission Lines*

#### *Frequency Characteristics\**

Fig. 66 shows the maximum permissible variation of frequency characteristic referred to 800 c/s for programme circuits on the older types of line plant. Fig. 67 shows the same thing for what is now to be considered (since July 1949) a normal programme circuit.

It will be seen the latter reflects the general desire of broadcasting organisations to improve on what was previously considered acceptable. There is a good deal of old plant still in existence, incapable of meeting the more modern requirement, thus the various organisations cannot as yet put these limits (Fig. 67) into effect generally.

#### *Distortion*

At the maximum usable level, defined as +9 db at a point of zero relative level, the level of the harmonics is to be at least 20 db below the fundamental. It is noted this is a provisional figure only and it corresponds to 10 per cent of harmonic distortion. Broadcasting organisations have pointed out that harmonic distortion is noticeable for a separation of 28 db.

\*The paragraphs in small type are extracted from Vols. III and IV, XV Plenary Assembly of the C.C.I.F., Paris, July, 1949.

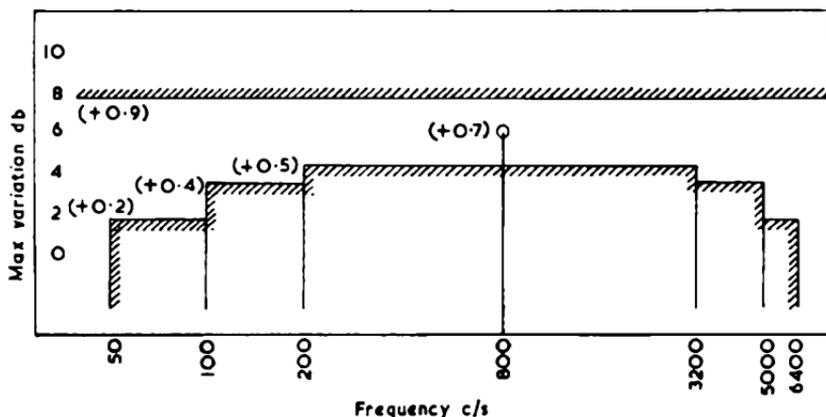


FIG. 66

(The figures in brackets are levels in nepers)

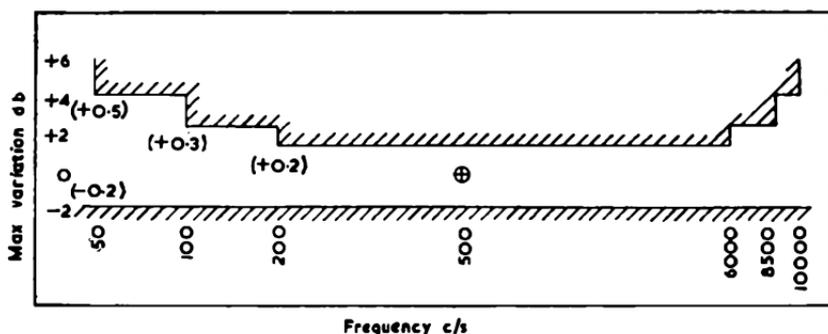


FIG. 67

The BBC considers single links are outside maintenance limits when the harmonic distortion is 37 db or less below the maximum usable level at 1 kc/s and 25 db or less at 100 c/s, corresponding to 0.6 per cent and 5 per cent respectively. Except for occasional peaks, the peak level of a programme is kept 2 db below the maximum usable level.

#### Noise

The separation in db between the maximum useful voltage and the psophometric\* voltage for noise and cross-talk is to be not less than 57 db with a weighting network in circuit and not less than 37 db without the network.

A specification for the psophometer is given in Volume IV of the *Paris Assembly*, July, 1949.

As described later, the Post Office and the BBC use peak pro-

\*From psophometer = a line-noise measuring device standardised by C.C.I.F.

## PROGRAMME CIRCUITS ON POST OFFICE LINES

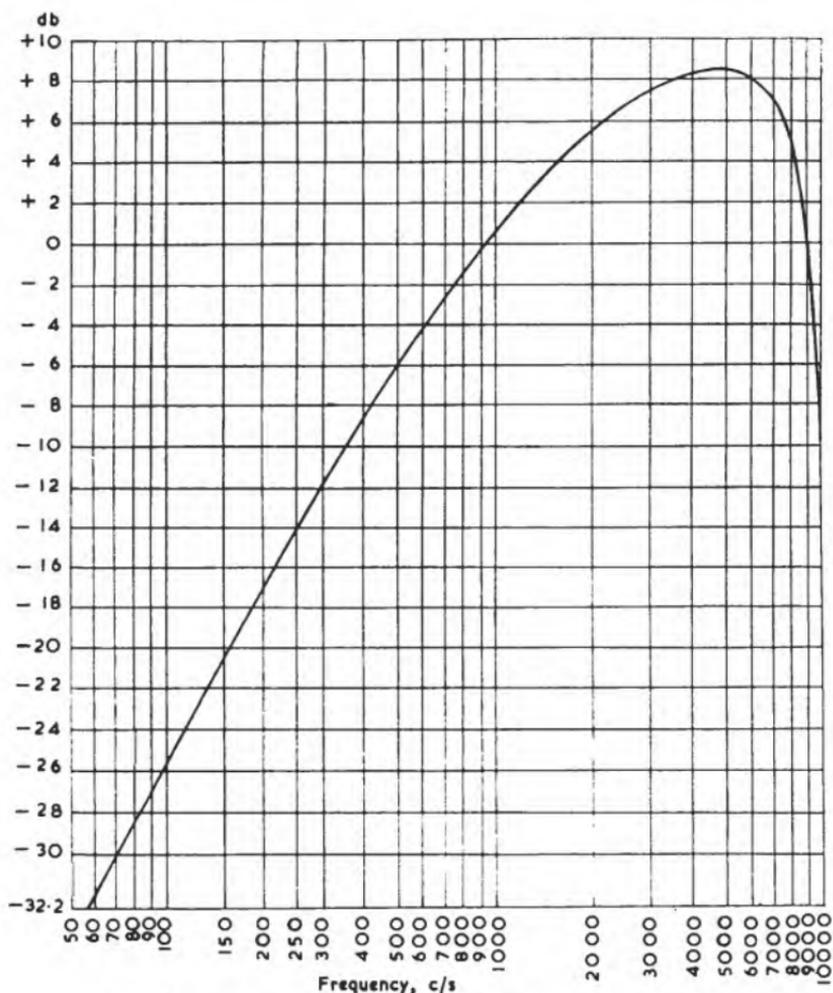


FIG. 68—Aural sensitivity network: frequency characteristic

gramme meters to measure noise and the BBC specifies it in terms of peak noise to peak programme separation.

Fig. 68 shows the frequency characteristic of the recommended weighting network (Aural Sensitivity Network) and its purpose is described in a later paragraph.

### Delay Distortion

The velocity of transmission on lines is not the same at all frequencies.

The effect is most marked on loaded lines, the high and low frequencies being delayed with respect to the middle frequencies, but

provided the relative delays are not excessive, it only becomes noticeable on programmes consisting of speech.

All frequencies are naturally delayed to some extent and the C.C.I.F. limits are specified in terms of maximum permissible delays relative to the minimum delay at the point of measurement.

*For Circuits on Old Plant*

50 c/s	70 m.sec	}	relative to delay at 800 c/s which is approximately the frequency of minimum delay
6,400 c/s	10 m.sec		

*For Normal Programme Circuits*

50 c/s	80 m.sec
100 c/s	20 m.sec
10,000	8 m.sec

**Example of Delay Distortion**

The distribution of BBC Light Programme to Burghead at the time of writing contains 693 miles of 16 mH/1.136 m. and the estimated delay distortion relative to 1 kc/s is:

50 c/s	50 m.sec
100 c/s	9.5 m.sec
7 kc/s	7.3 m.sec

Contributions towards London are routed in the main on non-loaded plant on which the velocity at the high frequencies is slightly higher than that at lower frequencies. Thus the effects of delay distortion at Burghead on contributions to the Light Programme from the North will be slightly reduced at the high frequencies.

**7.8 EQUALISERS**

Looking back into the line from the amplifier the line will look like a generator of e.m.f.  $E$  and impedance  $Z$  at any particular frequency.

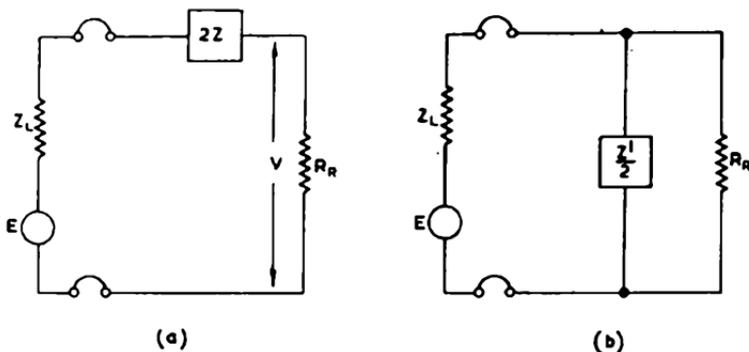


FIG. 69

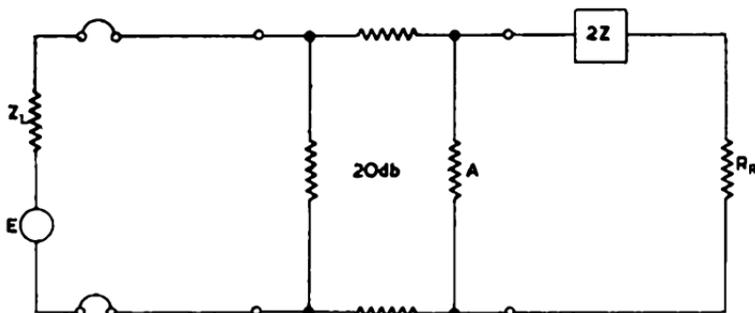


FIG. 70

If the input of the amplifier is  $R_R$ , then the voltage across  $R_R$  can be modified by either a series impedance or a shunt impedance as in Fig. 69.

The factors  $2Z$  and  $\frac{Z'}{2}$  are chosen for convenience in analysis.

The insertion loss caused by  $2Z$  (Fig. 69 (a)) is the ratio of the voltages ( $V_1$  and  $V_2$  respectively) across  $R_R$  before and after insertion of  $2Z$ .

$$\frac{V_1}{V_2} = \frac{Z_L + R_R + 2Z}{Z_L + R_R} \dots\dots\dots(10)$$

Similarly for Fig. 69 (b)

$$\frac{V_1}{V_2} = \frac{Z(R_R + \frac{Z'}{2}) + R_R \cdot \frac{Z'}{2}}{\frac{Z'}{2}(R_R + Z_L)} \dots\dots\dots(11)$$

It will be noticed that in both formulae the loss caused by the inserted impedance is a function of  $Z_L$ , the line impedance. Thus, unless the line impedance is known, it is impossible to predict the performance of the inserted network and it will be necessary to adjust it by trial and error to correct the over-all line characteristic, unless, of course, one is prepared to measure the line impedance at each frequency, a process even more laborious than trial and error.

Supposing, however, the line is separated from the equalising network by a 20-db attenuator of characteristic impedance  $R_R$  (Fig. 70). It can be shown that looking back towards the line from the point  $A$ , the impedance seen will not differ from  $R$  by more than 2 per cent even if  $Z_L$  is either nought or  $\infty$ . Thus the impedance  $2Z$  is effectively operating between two impedances  $R_R$ . The insertion

loss caused by it can be determined from formula (10) by putting  $Z_L = R_R$ , then

$$\frac{V_1}{V_2} = 1 + \frac{Z}{R_R} \dots\dots\dots(12)$$

i.e., it is independent of the line impedance and a function only of  $Z$  and the terminating impedance  $R_R$ . Its performance can, therefore, be predicted in advance, and a number of curves can be drawn for different component combinations making up the impedance  $2Z$ . A family of these curves can then be drawn on a piece of celluloid, insertion loss being plotted against frequency. If the line characteristic is drawn on graph paper, using the same scale in the opposite direction, the celluloid mask (as it is called) can be placed over the line curve, and its position adjusted until one of its curves most nearly coincides with the line curve. Calibrating marks on the mask curve can be made to give the appropriate design of the impedance  $2Z$ , and no trial and error adjustment is needed.

Similarly for formula (11) if  $Z_L = R_R$

$$\frac{V_1}{V_2} = 1 + \frac{R_R}{Z'} \dots\dots\dots(13)$$

and the same arguments apply.

The above can only be done at the expense of attenuating the incoming signal by something between 10 db and 20 db. If more than one equalising impedance is necessary to effect accurate correction, they must be separated by a further attenuator. If the resulting overall insertion loss cannot be recovered in the amplifier, some other form of equaliser which does not require the use of large attenuators must be used.

Referring again to Fig. 70: if instead of looking from point  $A$  towards the line we look from the line towards the attenuator it will be apparent that again the impedance seen will not differ from  $R_R$  by more than 2 per cent whatever the value of  $2Z$ . In fact, the combination of attenuator and  $2Z$  has a loss characteristic independent of the line, and therefore, predictable, simply because the line has no means of telling whether the corrector together with the attenuator are in circuit or not. Therefore, if we can design a corrector whose input impedance, as seen by the line, is always  $R_R$  whatever its combination of components, its characteristic curves will be independent of the line and there is no need for an attenuator to be included.

It is quite evident that such a corrector, which is known as a *constant-resistance equaliser*, must take the form of a combination of series and shunt elements and the component values of the corrector shown in Fig. 71 can be chosen to meet the required conditions, as will now be demonstrated.

$$\begin{aligned} \text{The input impedance } Z_E &= \frac{(R_R + Z)(R_1 + Z')}{R_R + Z + R_1 + Z'} \\ &= R_R \left( \frac{R_1 + ZZ' + \frac{R_1 Z}{R_R} + \frac{Z'}{R_R}}{R_R + Z + R_1 + Z'} \right) \end{aligned}$$

If we make  $\frac{ZZ'}{R_R} = R$  and  $\frac{R_1 Z}{R_R} = Z$ ,

then  $R_1 = R_R$  and  $Z_E = R_R$

Therefore, provided we choose  $Z$  and  $Z'$  so that  $ZZ' = R_R^2$  we can arrive at a network which, when connected between the line and  $R_R$ , will still present an impedance  $R_R$  to the line.

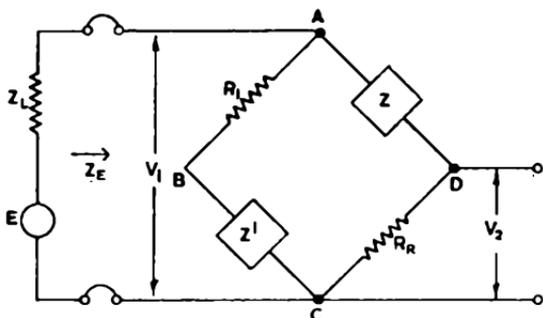


FIG. 71—Constant-resistance equaliser: theoretical

Thus, the voltage  $V_1$  for a given frequency is the same whether the equaliser is in or out of circuit, and the insertion loss of the equaliser is  $20 \log_{10} \frac{V_1}{V_2}$ .

It can be seen that  $\frac{V_1}{V_2} = \frac{R_R + Z}{R_R} = 1 + \frac{Z}{R_R} \dots \dots \dots (14)$

If  $R_1 = R_R$  and  $ZZ' = R_R^2$ ,  $\frac{Z}{R_R} = \frac{R_R}{Z}$  and the arrangement of impedances  $A B C D$  in Fig. 72 is a balanced Wheatstone bridge. Therefore, there is no potential difference between  $B$  and  $D$  and the voltage across  $B C$  also equals  $V_2$ .

$$\therefore 1 + \frac{Z}{R_R} = \frac{V_1}{V_2} = \frac{R_R + Z'}{Z'} = 1 + \frac{R_R}{Z'} \dots \dots \dots (15)$$

It is of interest now to compare the formulae above with formulae (12) and (13) and it will be clear at once that a given insertion loss characteristic can be met by:

- (1) A constant-resistance equaliser with a series-arm impedance  $Z$  and a shunt-arm impedance  $Z'$  such that  $ZZ' = R_R^2$  arranged as in Fig. 71.
- (2) An attenuator of characteristic impedance  $R_R$  and of at least 10 db followed by a series-arm impedance only, equal to  $2Z$ .
- (3) An attenuator of characteristic impedance  $R_R$  followed by a shunt-arm impedance only, equal to  $Z'/2$ .

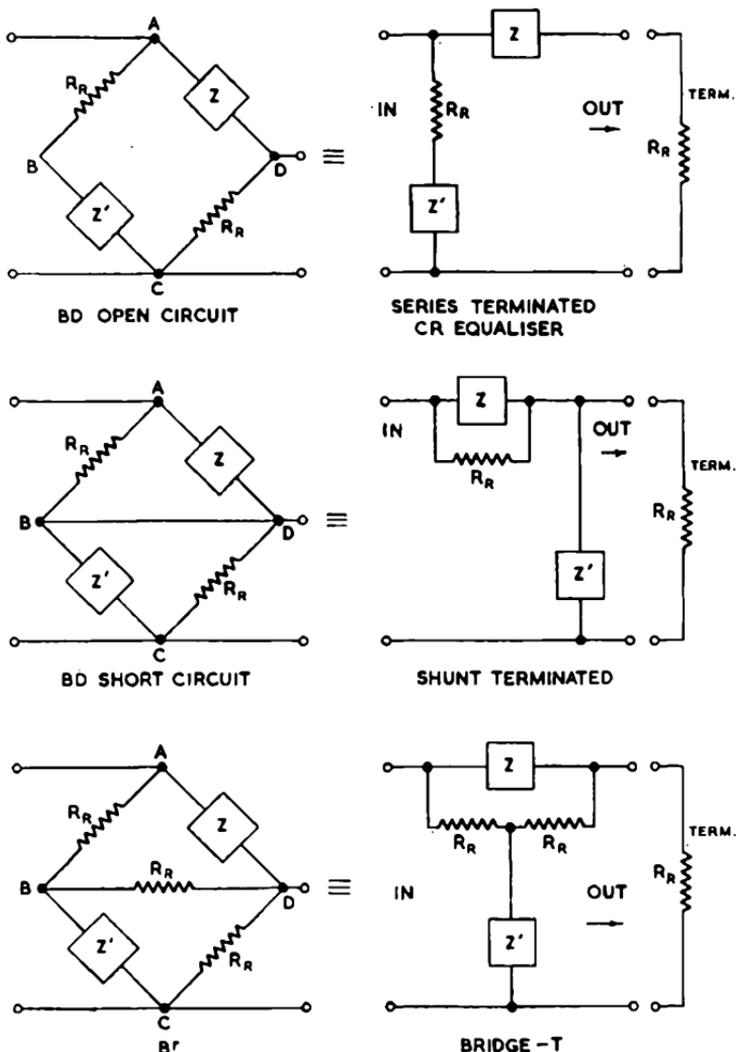


FIG. 72—Constant-resistance equalisers: practical

Thus a mask drawn for equaliser (1) can also be used for calculating equalisers of the types described in (2) and (3). The latter are known as non-constant resistance types, as they would not respond to anticipated performance if used without an attenuator unless they are terminated on both sides by constant resistances equal to  $R_R$ .

Referring again to Fig. 71, as there is no potential difference between  $B$  and  $D$  we can also short-circuit  $B D$  or insert a resistance (say  $R_R$ ) between  $B$  and  $D$  without in any way upsetting the insertion loss characteristic. We can then make the transformations shown in Fig. 72.

The three networks on the right-hand side are the more conventional forms of constant-resistance equalisers. It is important to realise that the resistance  $R_R$  no longer forms part of the network, and must be supplied in the form of the input impedance of the terminating apparatus. Failure to do so will completely upset the balance of the basic Wheatstone bridge and the equaliser will not exhibit the anticipated insertion loss. If we add to these three, the two non-constant resistance types, we have five configurations whose component values can be calculated from one mask for a given insertion-loss characteristic. A typical mask which can be used in this way is shown in Fig. 73.

It is of interest to note that all five configurations are used at one time or another in the design of simple and composite equalisers. A number of reasons influence the choice of a particular configuration for a required design, among them being space, economy, simplicity of construction and availability of suitable components.

It has been found that as almost every fixed equaliser has to be tailor-made, the process is simplified by assembling in tandem a number of simple sections, each of which can be predicted separately by masks. Each section is in itself simple, having no more than three components for  $Z$  and, therefore, simplifying the design of suitable masks and the methods for using them.

The impedance  $Z'$  will be the conjugate of the impedance  $Z$ , i.e., if  $Z$  consists of simply an inductance,  $Z'$  will be a capacitance. Stated generally, the rule is:

Components in parallel for  $Z$ , have conjugate components in series for  $Z'$ , and vice versa.

The configuration of the impedance  $Z$  can take many forms and the more commonly used varieties are shown in Fig. 74 together with the corresponding configuration of  $Z'$ . In every case  $R_R^2 = RR' = L/C' = L'/C$ .

The resistance  $R_R$  in the network (i.e., not the terminating resistance  $R_R$ ) is often combined with  $R'$  in the series terminated type

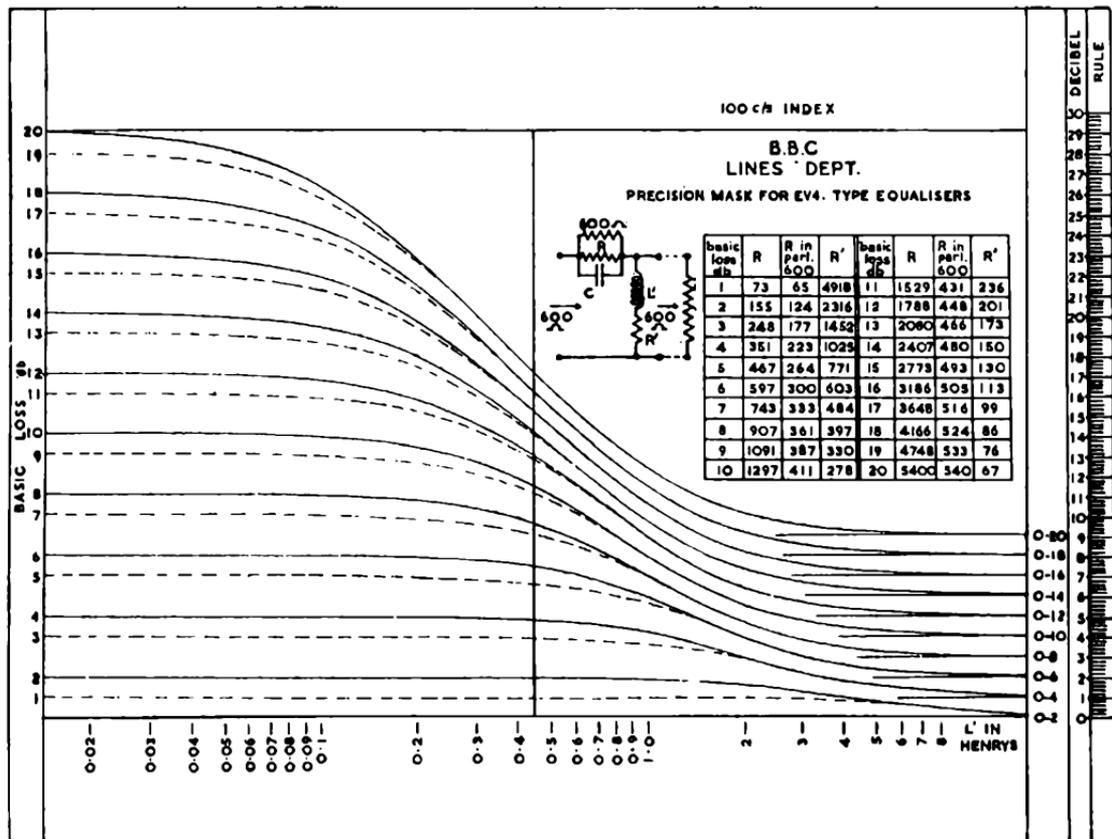


FIG. 73—Typical equaliser mask

PROGRAMME CIRCUITS ON POST OFFICE LINES

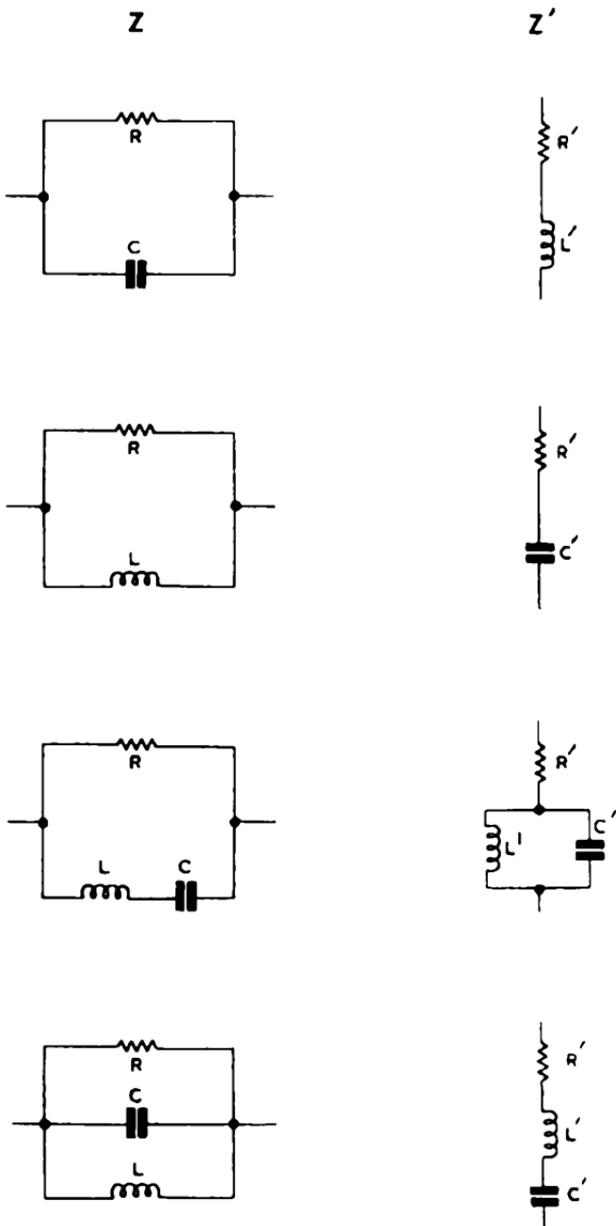


FIG. 74

or with  $R$  in the shunt terminated type, and this point must not be overlooked if a change of configuration is desired after construction.

### 7.9 BBC TIE-LINE CONNECTIONS

Summarising, the different classes of circuits which we are likely to require are as follows:

- (1) Lines to connect an outside studio within about 12 miles of the central control room.
- (2) Temporary lines for an outside broadcast.
  - (a) Unrepeated.
  - (b) Repeated.
- (3) A long-distance contribution or distribution circuit between two main centres.
- (4) A transmitter tie-line from a main centre.

Circuits used for transmitter tie-lines will fall either into category (1) or (3) and need not be considered separately.

#### 7.9.1. *Circuits from an Outside Studio*

As an outside studio is a fixed source, circuits can be connected permanently to the central control room. They will normally be terminated at the studio on a Post Office distribution case, starting as a small cable (say 7 pr. 10 lb) which is led into a Post Office manhole in the street. Here the requisite number of pairs in a main street cable can be picked up. The size of the latter depends on local Post Office requirements. The street cable, which is laid in earthenware ducts, eventually terminates in an exchange on a *jumper frame* which permits the interconnection of pairs from one cable with those of another. In this manner pairs are routed through perhaps a number of exchanges to that which is nearest the central control room. Here it is connected to a local-end cable which consists of a number of pairs connected permanently between the exchange and the control room, either via street cables as above, or, if the number of circuits justifies it, by a specially laid cable.

The local-end cable at the control room will in general be terminated on a Trunk Test Tablet (T.T.T.), which serves as a suitable interception point for test purposes. From the trunk test tablet the circuit is wired to a repeating coil, and thence to the equaliser. The equaliser is permanent and is built to requirements on a slotted chassis designed to take standard components. A typical equaliser of this type is shown in Plate 3 (p.136). The insertion loss is usually built out with a suitable attenuator so that, for the normal sending level at the studio, the output level from the equaliser is equal to that of other equalisers in the control room. This level is generally  $-45$  db.

From this point the circuit follows the control room chain of apparatus already described in a previous chapter.

### 7.9.2. *Outside Broadcast Circuits*

Use is made of street cables for outside broadcasts in much the same way as for outside studios. There are, however, certain important differences. The majority of O.B.s use only temporary connections, as broadcasts from any particular point may be very infrequent and may even be confined to a single occasion.

Before the broadcast a test is arranged to check the suitability of the circuit chosen. The following tests are made on a bridge Megger.

- (a) D.C. Tests
  - (a) Insulation
  - (b) Loop resistance
  - (c) Unbalance resistance

The importance of high insulation has already been stressed.

The loop resistance is measured for two reasons. First, the figure obtained is a record of the circuit and can be used for later identification. Secondly, the "goodness" of the connection can be observed, whilst measuring, by checking the steadiness of the meter reading. An unsteady reading would indicate a bad connection somewhere and this would have to be tracked down and eliminated at once.

It is generally accepted that the unbalance resistance should not exceed 4 ohms: this allows a small margin of error which might be caused by connecting leads, instrument errors, etc. If an unbalance is observed it is almost certain to be at some point where a temporary connection has been made for the occasion, and it should be possible to find and eliminate it with Post Office co-operation.

#### (b) A.C. Tests

A variable oscillator is taken to the O.B. point and tone sent to line at a number of selected frequencies covering the wanted range; the sending e.m.f. is maintained constant at all frequencies. At the receiving end, normally the control room at the studio centre, the incoming levels are measured on an amplifier detector. The readings indicate the insertion loss of the circuit at the various frequencies, and the figures are plotted on a graph. From this the equalisation required can be deduced. Since this type of circuit is for temporary use only, variable equalisers are used, and the settings noted. The equalisers may then be used for other purposes until the circuit is required for service; the settings are then adjusted to those selected on the test.

## (c) Noise

After the line has been correctly equalised the noise must be checked to ensure it will not intrude on the programme.

In many cases the O.B. point will be too far electrically from the control room to be dealt with in this simple way. The attenuation of the high frequencies may be so severe that line noise becomes obtrusive. An equaliser and amplifier at some intermediate point then becomes essential. If the routing of the circuit is through Post Office repeater stations the Post Office will insert the equalisers and amplifiers, although topping-up equalisers to account for the local ends and other discrepancies will still be necessary. In many cases, however, the circuit cannot be routed through Post Office repeater stations and the Post Office grants BBC engineers access to appropriate exchanges for the setting up of portable repeaters (see Plate 4, page 137).<sup>5</sup> The main features are:

- (i) A duplicate set of high-gain amplifiers equipped with peak programme meters.
- (ii) Power supplies (one battery unit and one mains unit).
- (iii) Variable equalisers designed to meet a wide range of insertion loss curves.
- (iv) A bridge Megger for d.c. tests, and
- (v) A telephone, together with a switching unit, for ease of operation.

A.C. tests are carried out on these amplified circuits as before with the inclusion of a harmonic distortion test to ensure the amplifiers are working properly and being used at the correct working level.

Complete d.c. tests can be taken where BBC repeaters are being used, but only terminal d.c. tests are feasible when the repeatering is done by the Post Office. This is considered an acceptable condition as the Post Office line up such circuits as permanent facilities and refer to them as O.P. (Occasional Programme) circuits.

A primary responsibility of the Post Office is to provide satisfactory communication between its subscribers. Thus the cables connecting subscribers who live some distance from a town centre are more likely to be loaded so as to keep the circuit loss from subscriber to subscriber reasonably low. The loading can be heavy as the bandwidth has only to be sufficient for the satisfactory transmission of speech, e.g., 88 mH with a cut-off frequency of about 4 kc/s. This is too low for broadcast purposes, the highest frequency that can reasonably be attained being 80 per cent of cut-off, i.e., 3.2 kc/s. Thus if an O.B. point should happen to be in a locality served by cables of this type (and it frequently is), it will be neces-

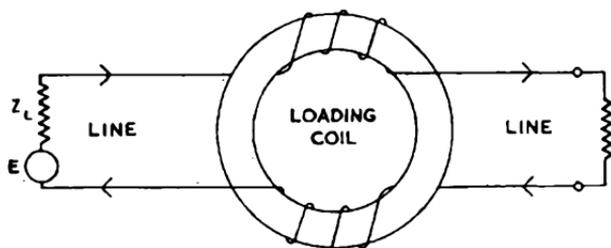


FIG. 75—Loading coil on single pair

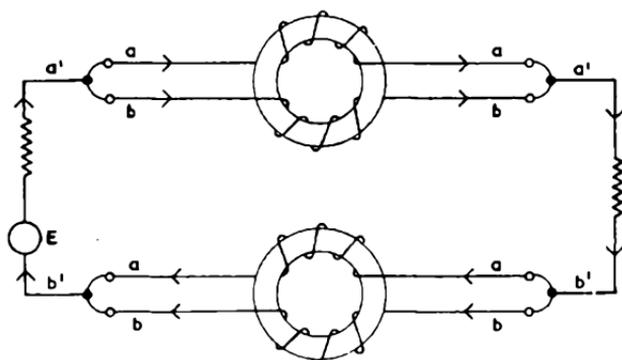


FIG. 76—Loading coils using two pairs for one line

sary to “bunch” two pairs together to eliminate the cut-off effect of the loading coils, as explained in the following paragraph. In some cases, these loaded cables have a few non-loaded pairs and bunching, of course, would be unnecessary if these were available.

Lines are loaded by inserting half of the loading coil in each leg of the line, the coil itself being wound on a toroidal core, and having a high coupling factor with a consequent low leakage inductance (see Fig. 75). In a single pair the current circulates, i.e., at a given instant it travels in opposite directions in each leg, and the coil is so connected that the two half windings are in series aiding, thus for a circulating current the full inductance is in series with the line. If we now reverse the direction of the current in one leg without affecting the other, i.e., make the current travel in the same direction in both legs, the two windings will be in series opposing, and the net inductance will be the leakage inductance which is very small for coils having a high coupling factor and is in fact zero for a factor of 1.

What is done, therefore, is to take two pairs and connect both legs

of each pair together ( $a$  to  $b$ ), as in Fig. 76. The music circuit will then be  $a'b'$  and the loading left, due to leakage inductance, is small enough to make the cut-off frequency well above the required bandwidth. Cables in this country are *quadded* and this bunching can only be done on pairs in the same quad. If pairs in different quads are used the cross-talk from other pairs, and general noise, will be serious as the pair balance is upset. Quadded pairs are easily identifiable, as the lower pair number is always odd, e.g., pairs 139 and 140 are quadded, pairs 140 and 141 are not. The effect of bunched quadded pairs is to lower the attenuation at low frequencies but increase it at high frequencies. Equalisation is, therefore, made more difficult and repeaters have to be used for shorter distances than when using unloaded cables of the same gauge.

In parts of Northern Ireland and Scotland long distances are traversed by 88-mH cable without there being a suitable interception point to insert a repeater. In these cases the use of a bunched pair is impossible as the noise level is far too high for broadcast purposes. Use is therefore made of apparatus referred to as *Split-band Carrier*.

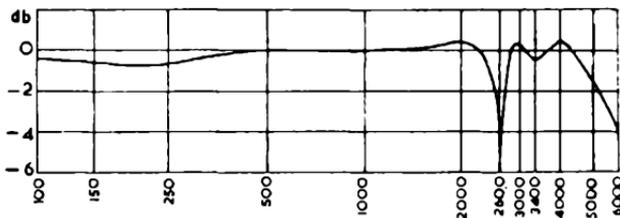


FIG. 77—Split-band carrier: frequency characteristic

There are two classes of this apparatus, but both divide the programme bandwidth into two parts by means of filters. In the first class, the lower part of the wave-band 50–2,700 c/s is transmitted on one 88-mH circuit (low-band channel). The upper band 2,700–5,100 c/s modulates a 5,400-c/s carrier and the lower side-band 300–2,700 c/s is filtered out from the modulation products and transmitted on the second 88-mH circuit (high-band channel). At the receiving end, which can be up to 80 miles away, both lines are equalised and amplified (the equalisers are quite simple) and the high band is demodulated by a 5,400-c/s carrier to restore it to its normal position in the frequency band. Both bands are then combined through combining filters. A total bandwidth of 5 kc/s is obtained in this way, though there is a sharp dip (not audibly noticeable) in the over-all frequency characteristic at the cross-over point. A typical frequency characteristic is shown in Fig. 77.

The second class puts both low and high bands through modulation processes, so that both are brought within the bandwidth 300–3,400 c/s. This is to enable the use of two Post Office carrier channels on a multi-channel carrier system. The bandwidth of such channels by international agreement is not less than 300–3,400 c/s. This system could be used, therefore, between points where there are no audio pairs, but where a Post Office multi-channel system on a repeated co-axial exists. The over-all frequency characteristic is similar to the first system described.

Circuits obtained in this way are inferior to those obtained by more conventional methods, but they are nevertheless superior to circuits with more restricted bandwidths or considerable noise, and certainly better than no circuit at all. These systems are only used where a circuit cannot be produced by any other means and on a few occasions where the number of repeaters required to establish a reasonable circuit on bunched pairs would be prohibitive from the point of view of equipment and staff. As an example, 80 miles of bunched pair would require at least five repeaters with an engineer at each, whilst a split-band would need only two engineers.

### 7.9.3. *Long-distance Distribution and Contribution Circuits*

Long-distance circuits, known as Simultaneous Broadcast Circuits (S.B.), will in general be routed in Post Office trunk cables which are of heavier gauge than local or toll cables.

In many trunk cables, screened pairs, lightly loaded and non-loaded, specifically intended for broadcast purposes, were included by the Post Office in the design of the cable. The majority of these, certainly along main cable routes, have already been absorbed into the BBC network.

Other classes of circuits available for broadcast transmissions, as already noted, are carrier phantoms and sub-carriers.

Owing to the length of these circuits it is obvious that repeaters have to be installed at frequent intervals. Where repeaters are necessary the circuits are lined up, that is equalised and amplified, by the Post Office between the Post Office centres nearest to the BBC at sending and receiving ends. From these two points they are connected to the BBC centres via the local-end cables. The circuit is then equalised over-all at the receiving BBC centre. The equalisers are tailor-made to suit the circuit and are a good deal more complex than those met with on other classes of circuit. In BBC parlance the circuit between two main centres, including a sending amplifier and a receiving amplifier, is known as a "link". To feed the Light Programme from London to Lisnagarvey there are four links, and if Belfast is making a contribution to the Light Programme, the total

number of links between Belfast and Lisnagarvey via London may be as many as nine.

The advantage of the link system is the operational flexibility which it offers for the rapid replacement of links which have become faulty, without any severe deterioration of standards.

Its disadvantage is that small acceptable frequency characteristic variations on individual links can sum up to significant variations when a number of links are connected together to form a chain. Such variations can be kept within reasonable bounds by specifying tight link limits. If links are equalised to within an over-all variation of 1 db from 50 c/s to the maximum frequency, it is possible to maintain them subsequently within an over-all variation of 2 db.

#### 7.10 TEMPERATURE EFFECTS ON LINE CHARACTERISTICS<sup>6</sup>

Every circuit suffers a variation of frequency characteristic due to the seasonal change of ground temperature. The amount of variation depends on the construction of the circuit, its length, and the temperature range of the area in which it runs.

It will be remembered that the attenuation in nepers of a non-loaded line is approximately  $\sqrt{[(R\omega C)/2]}$  and the attenuation increases with frequency.

But both  $R$  and  $C$  increase with increase in temperature and if it is assumed that  $R \times C$  is increased by a factor  $K$  for a given increase in temperature,

$$\text{then attenuation} = \sqrt{K} \cdot \sqrt{\left(\frac{R\omega C}{2}\right)}$$

As the attenuation at high frequencies is greater than that at low the *change* of attenuation at the former is greater than that at the latter, and for carrier phantoms the change of equalisation is of the order of 0.5 db/100 miles between 50 c/s and 8 kc/s for a change of temperature of 1 deg C. Now, the standard depth of Post Office underground cables is approximately 2 ft 6 in and the maximum change of seasonal temperature at this depth can be up to 20 deg C. For a carrier phantom from London to Manchester we might expect, therefore, a total change of equalisation of something like 18 to 20 db from winter to summer in extreme conditions. Something must be done to keep these variations within bounds or the over-all variation on a chain 1,000 miles long would be stupendous. This is done by including in the terminal equaliser a unit known as a *Temperature Correction Unit* (T.C.U.) which is easily adjustable to compensate for the frequency characteristic variations caused by temperature.

It is designed to limit the variations due to temperature to  $\pm 1$  db at 8 kc/s (for non-loaded circuits). Adjacent steps introduce equalisation changes differing by not more than 2 db at 8 kc/s. Thus when

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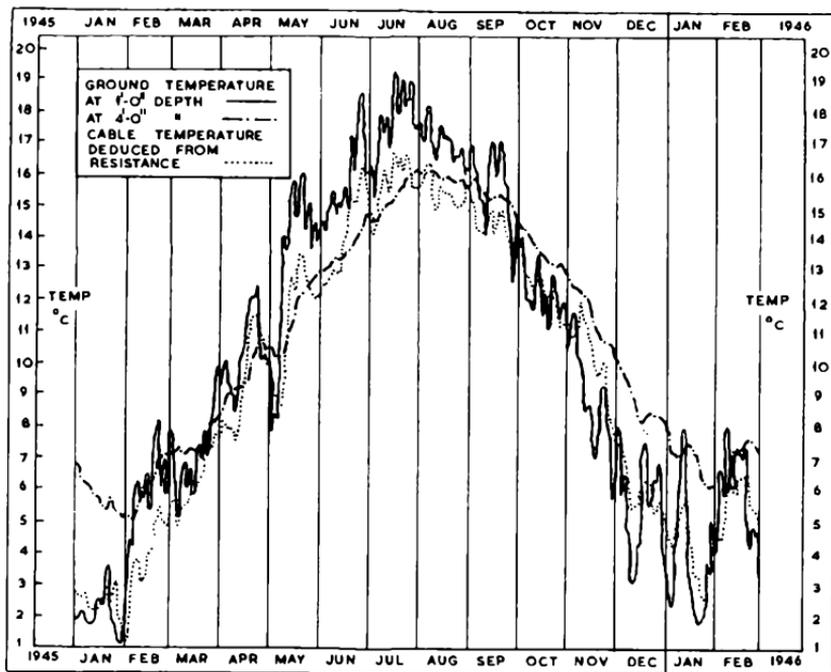


FIG. 78—Comparison of temperature effects

Comparison of deduced temperature of London-Birmingham cable with ground temperatures observed at Kew Observatory

the equalisation is in error by 1 db in one direction, a change of step is made which makes it 1 db wrong in the other and a continuing change of temperature would first make the equalisation correct and then later 1 db wrong again, necessitating another step change.

A routine had to be established for ascertaining when a change of the T.C.U. became necessary. To do this cable temperatures are measured in five areas of England and Scotland, and the temperature at which the equalisation will have changed from a basic characteristic by 1 db is determined for each circuit. By observing the trend of cable temperature changes, stations can be given advance notice of the date on which the T.C.U. on any particular circuit should be changed and the change can be made at a time when the circuit is not actually in service.

The cable temperature is read directly on a cable temperature bridge, a form of Wheatstone bridge calibrated in degrees centigrade. The measurement can be made without affecting any service the line may be carrying. The accuracy is dependent on the line resistance variations being due to temperature changes only. To guard against

serious errors caused by the insertion of interruption lengths necessitated by cable failure, the cable temperature is compared with ground temperature readings at an observatory in the vicinity of the cable route. These readings are at depths of 1 ft and 4 ft and one would expect the cable temperature (depth 2 ft 6 in) to fall roughly mid-way between—roughly only because observatory readings are taken at a point whilst the cable temperature is averaged over approximately 100 miles. The correlation obtained, however, is very satisfactory for operational purposes. Fig. 78 shows these measurements for a London-Birmingham route.

Many of the S.B. circuits are lightly loaded, and as the attenuation/mile of these is less than that of the non-loaded circuits the change of equalisation with temperature is less, as one would expect. It is quite sufficient, however, to warrant including them in the routine.

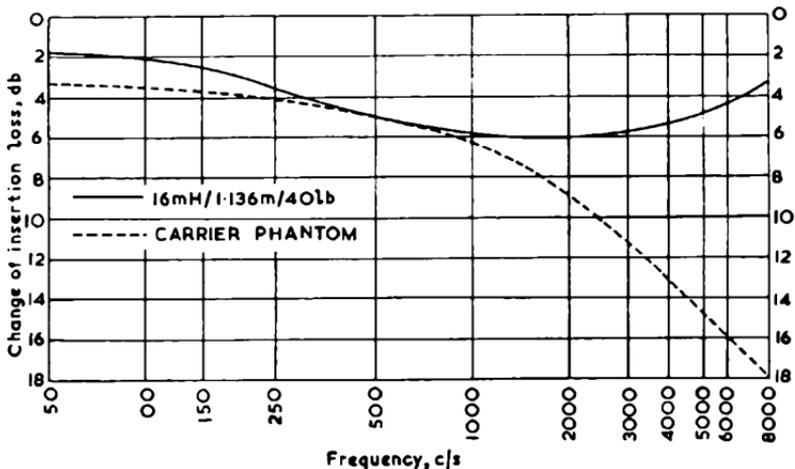


FIG. 79—Change of equalisation with temperature

The change of equalisation with temperature is given in Fig. 79 for non-loaded carrier phantom and 40-lb loaded circuits 16 mH/1.136 miles spacing for a change of 20 deg C on a 200-mile circuit.

Thus the design of the fixed equaliser for an S.B. circuit is considerably more involved than that for a shorter circuit. It consists of two parts, the first being the basic equaliser which is designed to equalise the frequency characteristic of the circuit as it would be at a reference temperature, known as the basic temperature. The second part is the T.C.U. (or may be even three T.C.U.s, depending on the length and construction of the circuit) which is set to keep the overall frequency characteristic within prescribed limits at the prevailing

temperature. Plate 5 is an example of such an equaliser, showing the T.C.U. mounted on the same chassis as the basic equaliser.

### 7.11 TEST APPARATUS

Equipment is necessary to carry out tests on the various types of circuits described above and a brief description is given here of the standard types used at BBC stations for this purpose. It is also used for testing other station equipment such as amplifiers and equalisers. Detailed descriptions will be found in BBC Technical Instructions.

The following test equipment is in general use:

Amplifier Detector	AD/-
Variable Oscillator	TS/-
Test Peak Programme Meter	TPM/-
Aural Sensitivity Network	ASN/-
Harmonic Routine Tester	FHP/-
Variable Attenuator	AT/-

All these items are mounted on a single A.C. Test Bay (Plate 6) together with one or two refinements to ensure the accurate testing of equalisers and amplifiers.

#### 7.11.1. *Amplifier-Detector*

As its name implies, a signal is first amplified and then rectified, the rectified current being used to deflect a moving-coil meter. Attenuators, calibrated in db, are incorporated. With a signal voltage of 0.775 volt across the input terminals and the attenuators set to read "0" db, the gain can be adjusted to make the meter read mid scale. The amplifier-detector, with an input termination of 600 ohms, will then measure signal levels in db referred to 1 mW in 600 ohms (standard zero level) from +10 to -55 by adjusting the attenuators till the meter reads mid scale and observing the settings of the attenuator dials. The attenuators can be adjusted to the nearest 0.5 db and a calibrated scale on the meter permits readings to 0.1 db.

#### 7.11.2. *Variable-frequency Oscillator*

There are two types of variable oscillator in common use, one a beat-frequency type with a usable frequency range from 30 c/s to 16 kc/s, and the other a resistance/capacitance type with a frequency range 40 c/s to 40 kc/s.

Both types will give a maximum output level from 600 ohms of 420 db in 600 ohms, but they are normally used with a built-in attenuator set to give an output level of "0" db.

Although there is little variation of e.m.f. with frequency a rectifier meter, which reads e.m.f. continuously, is provided to enable accurate adjustments to be made at all frequencies.

### 7.11.3. *Test Programme Meter*

The test peak programme meter is a standard BBC peak programme meter with an amplifying stage and calibrated gain controls, which can measure peak noise to peak programme separations of 70 db at points in the chain of zero programme volume.

### 7.11.4. *Aural Sensitivity Network*

This is a passive network made to C.C.I.F. Specification. It modifies the frequency characteristic of a noise in such a way that the peak reading on a test peak programme meter is approximately the same for most types of line noise causing equal annoyance in the presence of programme. That is to say, the meter registers the nuisance value rather than the precise level of a specific type of line noise.

The Post Office uses identical equipment for measuring noise and an agreed standard of acceptance based on measurements can be applied. This is more satisfactory than subjective aural tests which are too dependent on personal assessments and listening conditions, and which lead to disagreements often difficult to resolve.

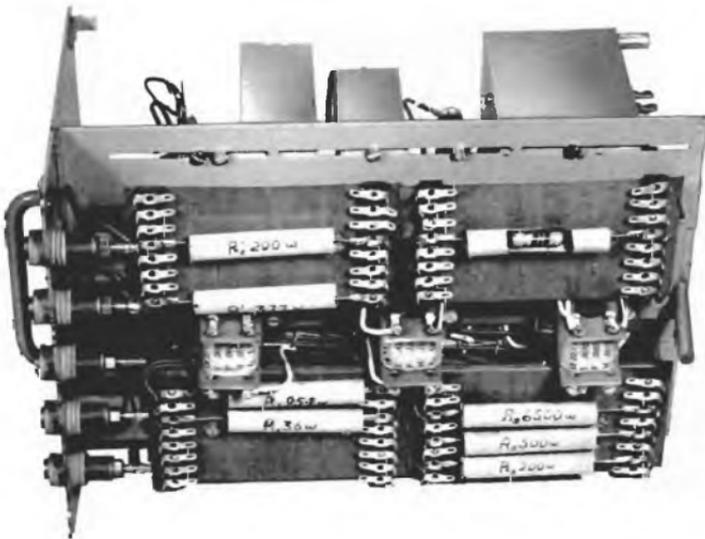
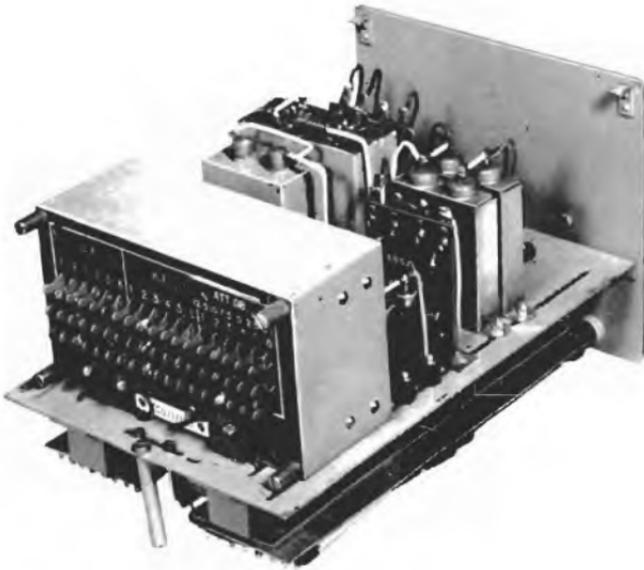
### 7.11.5. *Harmonic Routine Tester*

The harmonic routine tester is designed to measure total harmonic content at fundamental frequencies of 100 c/s and 1 kc/s. The equalised and amplified line output is connected to the tester and thence to an amplifier-detector. There is a three-position key on the tester. In the central position the fundamental (100 c/s or 1 kc/s) passes through the test panel without attenuation, and its level can be measured on the amplifier-detector. When the key is thrown (either up or down, depending on the frequency being tested) the fundamental is filtered out and the level of the total harmonics can be measured. The difference between the two readings gives the harmonic separation. The test is usually made at the normal sending level and at 10 db above normal.

It is not an ideal way of measuring harmonic distortion, nor does it measure the effect of intermodulation. It has the virtue, however, of being simple, robust and rapid to operate, and it has never been known not to show up a fault condition which would have resulted in audibly unpleasant effects on programme.

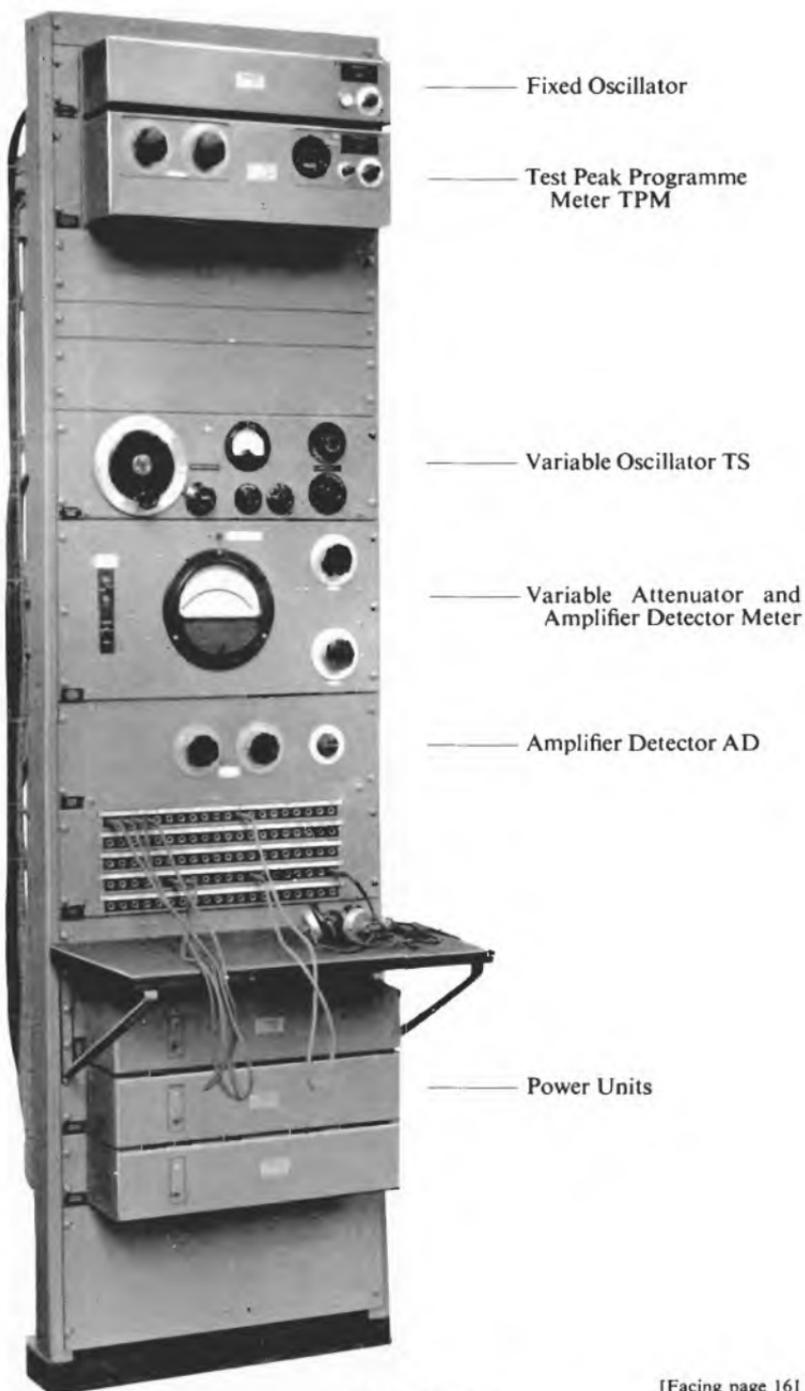
### 7.11.6. *Variable Attenuator*

This is a two-dial attenuator, variable in steps of 0.5 db from 0 db to 55 db. It is used to adjust the oscillator output level for special test purposes, and also for testing equalisers. For the latter purpose a change-over key is provided on the bay. It connects the oscillator



EQUALISER AND TEMPERATURE-CORRECTION UNIT FOR MAIN  
SIMULTANEOUS-BROADCAST (S.B.) CIRCUITS

PLATE 6



and amplifier-detector to the input and output respectively of the equaliser under test or to the attenuator. Thus the insertion loss of the equaliser at any frequency is obtained by comparing it with the attenuation needed to give the same reading on the amplifier-detector. In this way small inherent errors in the frequency characteristic of the oscillator and amplifier-detector are eliminated without any tiresome check calibrations at each frequency.

#### 7.11.7. Bridge Megger

The bridge Megger<sup>7</sup> is retained as a portable piece of apparatus, although jacks are provided on the bay to facilitate the various connections required for the following:

- (1) Insulation resistance between *A* leg and *B* leg
- (2) Insulation resistance between *A* leg and earth
- (3) Insulation resistance between *B* leg and earth
- (4) Loop resistance
- (5) Resistance unbalance

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## CHAPTER 8

### MONITORING

By H. D. M. ELLIS, M.A., M.I.E.E.

*Designs Department*

#### 8.1 GENERAL CONSIDERATIONS

In any broadcasting system, it is essential to ensure that the transmitter is radiating the right programme, and that the programme is of good quality technically and free from intentional or unintentional interference or background noise. In addition, the volume range of the programme must be kept within the limits set by the system without any noticeable effect on the artistic content of the programme material. These requirements necessitate keeping a constant watch on the programme at various points in the transmission system; this practice is called monitoring. The term monitoring has other applications in a broadcasting service, but in this chapter it specifically implies the functions of checking the technical quality and taking executive action when necessary. By the same reasoning, the person or automatic device performing these functions will be referred to as the *monitor*.

We can divide monitoring in this sense under two broad headings. In the first place, we must ensure that the programme which is passed into the transmission system at its source is as faithful a copy of the proceedings at the source (or as representative of the producer's wishes) as the limitations of the system will allow. This type of monitoring may be referred to as *source* monitoring and it is a function which, in general, necessitates a continuous listening watch. It can well be combined, or perhaps must be combined, with the programme control function and is essentially a matter for an operator having knowledge of musical and dramatic presentation.

Once having injected the programme into the system in a form artistically acceptable, further monitoring must take place to ensure that no significant change occurs during its passage through the chain. We may refer to this type of monitoring as *technical* monitoring and it is a much less difficult function. This will be clear when it is realised that, fundamentally, it is purely a matter of comparison whereas source monitoring to some extent calls for artistic discrimination.

## 8.2 SOURCE MONITORING

When source monitoring, the operator is generally faced with a problem similar to that of getting a quart into a pint pot. It is an unfortunate fact that the range of intensities which can be transmitted over the broadcasting chain is limited by the level of inherent background noise on the one hand and the power-handling capacity on the other. This range is much less than that normally experienced by the ear in everyday life; the normal human ear can accommodate without discomfort a sound-intensity range of some 100 decibels. That is to say if the weakest audible sound be said to have zero intensity, a sound intensity 100 decibels above this level will not cause physical pain (it might cause mental reaction!). In broadcasting studios, however, the intensity range normally encountered does not exceed about 60 decibels. In contrast, the range of intensity variation that a broadcast chain can conveniently transmit is of the order of 30 decibels. Therefore the operator must be able to adjust the gain of the chain so that very loud passages do not overload the system and very soft passages are not lost in the background noise. It should be observed here that this artificial compression of the dynamic range causes inconstancy in the ratio of input/output amplitudes; in other words, the ratio of input to output amplitude varies at different levels. This effect is known as *amplitude distortion*; it should not be confused with non-linear distortion which is much more objectionable.

The function of source monitoring is combined with that of microphone mixing and gain control and is carried out by a person who must have artistic sense and training; this is not only because artistic appreciation is often required to interpret the producer's aim, but because it is necessary to anticipate coming changes of level to ensure that the resultant controlled programme is not obviously different from the original (Chapter 3). Take for example an orchestral item containing a sustained pianissimo solo, followed by a sudden fortissimo from the whole orchestra. The monitor must so adjust his gain controls that background noise is masked during the solo but he must readjust them before the fortissimo to prevent overloading the chain; this he can only do by following a musical score. In making these adjustments care has to be exercised to maintain the illusion that the light and shade of the programme is not impaired. This the monitor cannot do if he has to rely solely on aural monitoring. Let us explain.

The ear is capable of judging loudness mainly by comparison, but it is extremely difficult to assess the absolute loudness of a sound. Moreover, the ear has a relatively short memory. For these reasons the loudness of a uniform passage of music, for example, can be

slowly reduced without the ear appreciating the fact, while a sudden crescendo is clearly noticed. In this way the monitor, anticipating a sudden loud passage, can prepare the way in advance so that it neither overloads the system nor affects unduly the listener's appreciation of the effect desired by the composer. But this inability to assess absolute level makes it impossible to rely on the ear to prevent the system being overloaded. It is true that the ear can detect the resulting distortion easily enough, but this is only possible after the crime has been committed. In order to give the monitor reliable information of the level he is actually sending, therefore, some more accurate indication must be provided. This is done by means of a programme meter, which gives visual indication of the precise volume at which the programme is being transmitted into the broadcasting chain at any instant (p. 166).

### 8.3 RELATIONSHIP BETWEEN AURAL AND VISUAL MONITORING

The provision for visual monitoring brings a number of problems in its train which are not immediately obvious. These arise from two fundamental causes. In the first place the ear, which is, after all, the final subjective measuring device, has a logarithmic characteristic with respect to intensity. This means that for a given increase in intensity the ear experiences an increase in loudness proportional to the logarithm of the increase in intensity. Moreover, the ear, as we have said, is capable only of assessing absolute intensity in very general terms and judges loudness largely by comparison. On the other hand, transmission systems in general and most measuring devices in particular have a linear characteristic with well-defined maximum and minimum limits. Unless, therefore, the visual measuring instrument is also given a logarithmic characteristic it is difficult to correlate what the eye sees with what the ear hears. And even if this is done, it is still quite difficult to control the programme so that the limits of the dynamic range of the system are not exceeded yet at the same time to load the system adequately and keep the average efficiency high.

The other difficulty arises from the fact that the impression of loudness is due not only to the peak sound pressure or intensity but is determined by the energy in the sound wave which is in turn a function both of the amplitude of the wave and of its shape. Some sounds, of which speech and percussion instruments provide good examples, contain a large number of short impulsive waves whose peak value is large and duration short (see Fig. 80 (a)). Other sounds, such as those produced by the stringed and wind instruments of the orchestra, have wave shapes which are much less "spiky" in appearance (see Fig. 80 (b)). In consequence the peak value is much

## MONITORING

less in a wave of the latter type, and although the two sounds illustrated in Fig. 80 have the same peak intensities, the apparent loudness of that shown in (b) is greater than that shown in (a).

The conflicting aspects of the problem now become apparent. On the one hand, in compressing the intensity range (for the reason already explained) it is necessary to preserve the apparent values of loudness of the various forms of programme in approximately the same relationship to each other as in the original sounds; on the

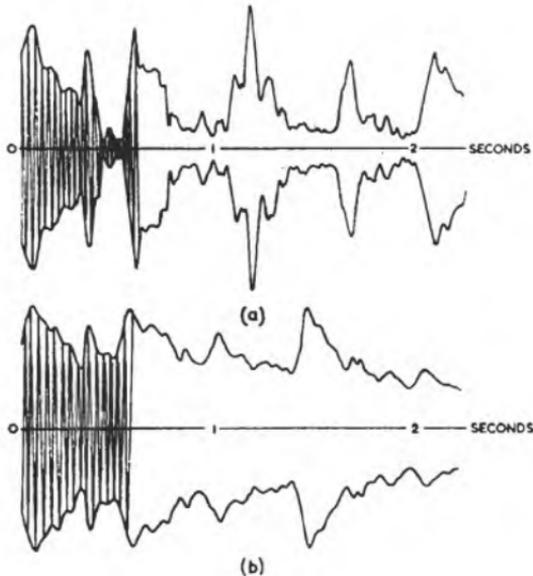


FIG. 80—Comparison of waveforms produced by different instruments

- (a) Impulsive sounds, such as percussion instruments
- (b) Smoother sounds, such as wind instruments

other hand it is desired to make the fullest economic use of the transmission system, which in broadcasting means that the fullest possible modulation depth of the most costly item, the transmitter, should be used at all times. Thus, as already stated, the problem cannot be solved by aural monitoring alone and it is necessary to resort to a more accurate system of measurement of either intensity or loudness. Visual indication is the obvious choice, which in the BBC takes the form of a programme meter; this, as will be shown, indicates instantaneous intensities.

In source monitoring, therefore, the monitor listens to ensure that the right programme is being radiated, that its quality is satisfactory and that it is free from interference; he controls the volume of the programme according to the indications given by the programme

meter which is calibrated in such a way that the full dynamic range of the transmission system is utilised.

#### 8.4 THE PROGRAMME METER

A programme meter differs from any ordinary meter in that it is required to indicate a quantity which is rapidly changing and here another arbitrary factor arises. Variations of both peak intensity and of loudness occur at a rate which is faster than the eye can follow; moreover, the impression of loudness itself is a subjective effect based on the integration of sound energy over some unspecified time. To give a visual indication, therefore, of either peak intensity or loudness, it is necessary to provide the indicator with some slow-acting or integrating characteristic which in the final analysis can only be arbitrary.

In view of what has been said, therefore, it is quite understandable that two general classes of programme meter have established themselves; indeed, it is perhaps surprising that there are not more. On the one hand, the *peak programme meter* broadly measures the peak value of the programme and enables a watch to be kept on the modulation amplitude handled by the system at the expense of the loudness indication. On the other hand, the *programme volume meter* measures the energy in the sound waves and gives a more reliable indication of loudness; its disadvantage is that it introduces a tendency to occasional overload and distortion, which can only be avoided by reducing the depth of modulation or by increasing the power-handling capacity of the transmitter. On this basis, the advantages of the programme volume meter are to some extent offset by a somewhat uneconomic use of the transmission system. The BBC uses the peak programme type of instrument.

Because programme amplitude is a constantly varying quantity, whose value is arbitrarily recorded by a meter with certain specified characteristics, some special term must be adopted for the unit which the programme meter registers. The intensity of steady tone is taken as a basis and is measured as a voltage relative to that corresponding to 1 mW in 600 ohms. This reference voltage (0.775) is called *zero level*. The level of any other continuous tone is expressed in decibel units above or below this value. When the programme meter is being used to indicate programme, however, intensities are expressed in terms of programme volume. *Zero volume* is defined as the intensity of programme material which causes the peak programme meter to kick frequently to a point 8 db above the reading for steady tone at zero level.

The peak programme meter used by the BBC has a scale which is approximately logarithmic. It rises to its reading very rapidly to

## MONITORING

ensure that short peaks are adequately recorded but falls slowly to reduce eye-strain. In order to reduce this strain as much as possible, a black scale with white pointer and lettering has been adopted as shown in Fig. 81. For the same reason, the number of marks on the scale was reduced to intercepts of 4 db. These line up well with the usual 2 db per step mixer and gain controls used in the Corporation, and purely arbitrary figures have been assigned to them. Present practice is to line up the meter on a standard tone so that "4" is

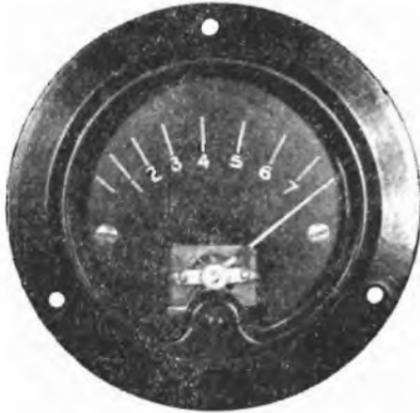


FIG. 81—BBC programme meter  
The normal scale shown is calibrated in 4-db divisions

registered when the tone level corresponds to approximately 40 per cent modulation of the transmitter. A peak to "6" therefore represents approximately 100 per cent modulation (8 db increase) and the range from "6" to "7" indicates the extent of inadvertent overloading. This precise indication is found to be very valuable in operational control. Normal control practice in the BBC is to adjust so that on quiet passages, peaks below "2" do not persist for long.

It may be of interest to mention here that when this programme meter was developed by the BBC a circuit was initially used which was capable of fully registering peaks lasting only 10 microseconds. This speed of response was found quite unnecessary and extended tests were carried out with a unit in which 80 per cent of the full peak value of a square wave was registered in 0.5 milliseconds. From these tests the interesting fact emerged that if the duration of a peak is very short, the ear does not get time to detect the distortion produced in the transmission system by momentary overmodulation, so that in general the shorter the peak the greater the percentage of overmodulation that can be tolerated. If too rapid a response is chosen for the indication meter, it has the effect of causing the operator to hold the gain of the chain rather lower than is necessary and so run

the transmitters at too low an average modulation level. Further tests showed that a circuit registering 80 per cent of the peak value of a square wave in 4 milliseconds provided the best compromise. Such a meter will therefore cause the operator to control his programme in such a way that momentary overloads will take place; but the resulting distortion will, by and large, pass unnoticed by the ear and the average modulation depth will be correspondingly raised.

As has been explained earlier, source monitoring is a function of the personnel at the origin of the programme in the studio. The BBC has, however, adopted a policy whereby there is a check source monitor, as it were, in the continuity suite associated with each programme. This duty is shared by two people (who have additional duties) namely, the continuity engineer and continuity announcer. The former is responsible for checking the general technical quality of the programme, while the latter is the arbiter as to whether the intrinsic programme value warrants the acceptance of inferior technical quality, as might occur for example in a transatlantic relay.

### 8.5 TECHNICAL MONITORING

Let us turn now to the other aspect of monitoring, which we have called technical monitoring. Here it is only necessary to ensure that what is coming out at the transmitter is similar to what is being put into the chain, but this is not always so easy as it sounds. The principal difficulty is that the output end of the system is generally far removed from the input end so that comparisons are not easy to achieve.

For many years the BBC has adopted the same system for this type of monitoring as for source monitoring, i.e. a listener with a loudspeaker assisted by a programme meter. Such monitoring has been carried out at all important junction points on the chain as well as at the transmitter itself and has resulted in much tedious occupation of many operators. Moreover, the job itself cannot be clearly defined. There is no means of telling whether a state of affairs judged unsatisfactory by the listener is in fact due to a deterioration of the system or to some idiosyncrasy on the part of the controlling operator or the producer at the source of the programme. Much fruitless communication results from this fact and in consequence the listening operator tends either to take note only of the more obvious faults or to worry about unimportant details according to his temperament. But by its very nature technical monitoring does not demand either a human being in constant attendance or a watch 100 per cent of the time at all points, if other means can be found of keeping a regular check and if faults on the system can be rapidly detected by any other system of checking.

## MONITORING

The BBC has recently developed, and introduced into service, equipment which enables an automatic comparison to be made of programme passing two points in the chain and which sounds an alarm if it is appreciably different at those points. Both quantitative and qualitative differences can be detected and the sensitivity is arranged to be on an approximately logarithmic basis. In this way the equipment is made to perform so that differences which are easily detected by ear are generally also detected by the equipment while those which are difficult to detect by ear are ignored.

It should be emphasised that while considerable sensitivity is required in relation to some qualities, for example non-linear distortion, only medium sensitivity is required for others, e.g. frequency-characteristic changes, and it is most important that the equipment should be not too sensitive; in fact, it is desirable that the sensitivity should depend upon the time of duration of an irregularity, since this is largely how a listener judges the seriousness of a fault.

### 8.6 AUTOMATIC MONITOR

This equipment is known as the *automatic monitor* and its method of operation is of some interest. At first sight it might be thought that it would only be necessary to balance the programme voltages

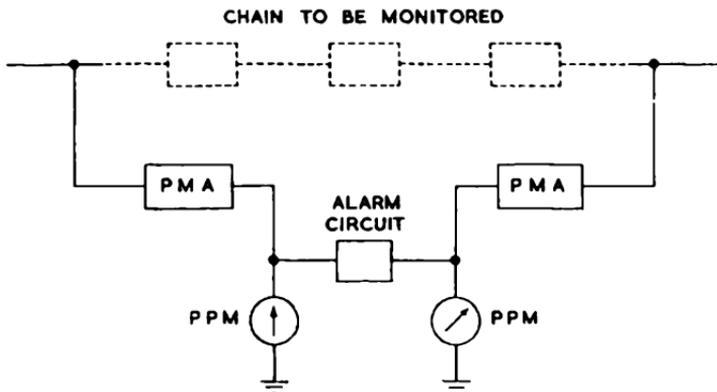


Fig. 82—Use of two programme meters to indicate unbalance between two points

at the two points of comparison against each other and apply the resultant unbalance to operate an alarm circuit. But in fact these voltages can differ by a large amount in certain respects (e.g. phase), yet the difference may be unnoticeable to the ear; on the other hand small differences in other respects may be very noticeable. A much more satisfactory result is achieved if two programme meters are

connected across the two check points and the difference in their readings used to indicate unbalance as shown in Fig. 82. Even this is not fundamentally satisfactory, for if the sensitivity is made such that an increase in background noise between the two points is sufficient to operate the alarm, the sensitivity will be so great that the smallest differences in level on programme, which is perhaps 100 times greater in volume, will operate the device. So non-linear circuits are employed which make the sensitivity dependent on level. Certain other arrangements are adopted to achieve the general result indicated above (e.g. the provision of integrating circuits) but the basic principle of the device is as outlined.

The automatic monitor has been developed in two main forms. The first, and simplest, relies on the assumption that the two programmes to be compared can be presented to the equipment in their normal form at a single locality, for example when it is required to check the programme arriving and leaving the same premises. Where a link covering a medium distance is also to be watched it may entail passing the programme back from a distant point by line or radio link; for example, the programme radiated by a given transmitter can be compared with the same programme feeding the line to that transmitter at the sending end of the line (Fig. 83).

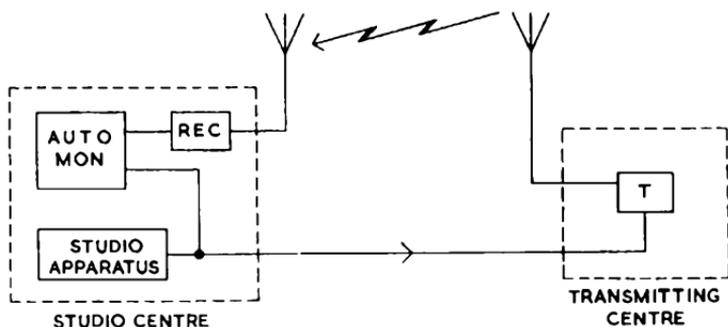


FIG. 83—Automatic monitoring between transmitter and studio centre

Certain precautions are taken in the equipment to avoid sounding the alarm, and therefore calling the maintenance engineer, when the discrepancy between the two programmes is due to some situation for which there is no immediate remedy and which is not the fault of the equipment. For example, overmodulation of the transmitter or atmospheric interference on the radio pick-up are matters which do not call for maintenance attention. This automatic monitor is described fully in BBC Technical Instruction S6, Section A, and is generally referred to as the *Automatic Monitor Minor*.

The other form taken by this equipment is known as the *Automatic Monitor Major* and is somewhat more complex; with it, the programme at two remote points can be compared. In this case the one connecting link between the remote points can be used to carry both the main programme and the information for comparison purposes. This is done by obtaining information about the programme at one of the comparison points and converting this information into d.c. signals which modulate a carrier. This carrier, which is above the audio band, carries the information back on the same link which carries the programme itself to the other point where it is compared with similar information obtained in a similar way. No further reference will be made here to this equipment since it is not widely used; more detailed information is available in *Proc. I.E.E.* Part III, September, 1951, page 329.

So far only applications of the automatic monitor have been considered in which the alarm is given if the circuit is proved faulty. A further development of the basic equipment has been made in which the monitor itself takes executive action, and this has made it possible to cover the needs of the small transmitter which can run unattended. In this case the monitor equipment not only samples the programme output continuously but also performs the following functions automatically according to circumstances:

- (i) Switches in reserve equipment to replace faulty equipment.
- (ii) Switches out a paralleled (faulty) unit.
- (iii) Closes down the station altogether if trouble persists.

Information to this effect is passed back to the nearest controlling station so that a maintenance engineer can be sent to clear the trouble. Some faults on the line can be corrected without a visit to the remote point, and in these circumstances after the trouble has disappeared the equipment starts up the station again.

Any automatic equipment of this type can only report a difference between programme volume or quality at two points; it cannot make any assessment of the programme material itself; for example it cannot tell whether a talks programme is in French or German. Ideally this is done by the source monitor but there remains the chance that some switching fault or other error can cause the wrong or an interfering programme to be injected into the chain at some point not under the control of the automatic equipment. It is therefore the practice to have somebody listening to the programme continuously on a loudspeaker at a point near the end of a long chain.

Thus a typical distribution network can be regarded as consisting of a number of links joining distribution centres as shown in Fig. 84, with a continuous aural check near the end of each link. From these

centres various spurs radiate, the spurs being covered by automatic monitoring when possible, though sometimes the function of maintaining an aural check on a main link and monitoring a spur are combined. If the spur terminates in an unattended low-power transmitter, the monitor relating to it can be of the executive action type which covers both the transmitter itself and major faults on the line feeding it. The figure shows only a hypothetical arrangement but it indicates the goal towards which BBC policy is working.

### 8.7 SEQUENTIAL MONITORING

Under certain circumstances it may not be necessary to maintain continuous monitoring on a specific circuit, for example on each of a number of adjacent transmitters radiating the same programme on different wavebands. In such cases it may be adequate to carry out intermittent checking; this method is called sequential monitoring. Although there is no reason why an automatic equipment should not be switched sequentially to a number of circuits, the BBC have so far used this system only in association with loudspeaker listening. It is generally employed at transmitter stations on the External Service networks. In operation the listening point is automatically switched, on a fixed-time basis, to the various check points in a given sequence. At the largest centre where this system is in use, there are as many as six different programmes which may be switched simultaneously to as many as 18 senders in any order depending on waveband changes. The combination may be changed every quarter of an hour throughout the 24 hours and the order can be completely arbitrary. A complex equipment has been devised to enable the

FIG. 84 (see opposite page)

NUMBER AND LETTER KEY	SYMBOL KEY
1 Main originating and distribution centre	
2 Secondary distribution centre	
3 Transmitter manned for maintenance only	 Transmitter
4 Transmitter with local aural monitoring	
5 Transmitter manned for maintenance and with local automatic monitoring (minor)	 Receiver
6 Transmitter manned for maintenance and with local automatic monitoring (major)	 Automatic monitor with alarm bell
7 Premises with several transmitters and local aural sequential monitoring	
8 Transmitters with local automatic monitoring and remote supervision	 Loudspeaker
A Low-power transmitters with automatic monitoring and control	 Operator on continuous monitoring duty

# MONITORING

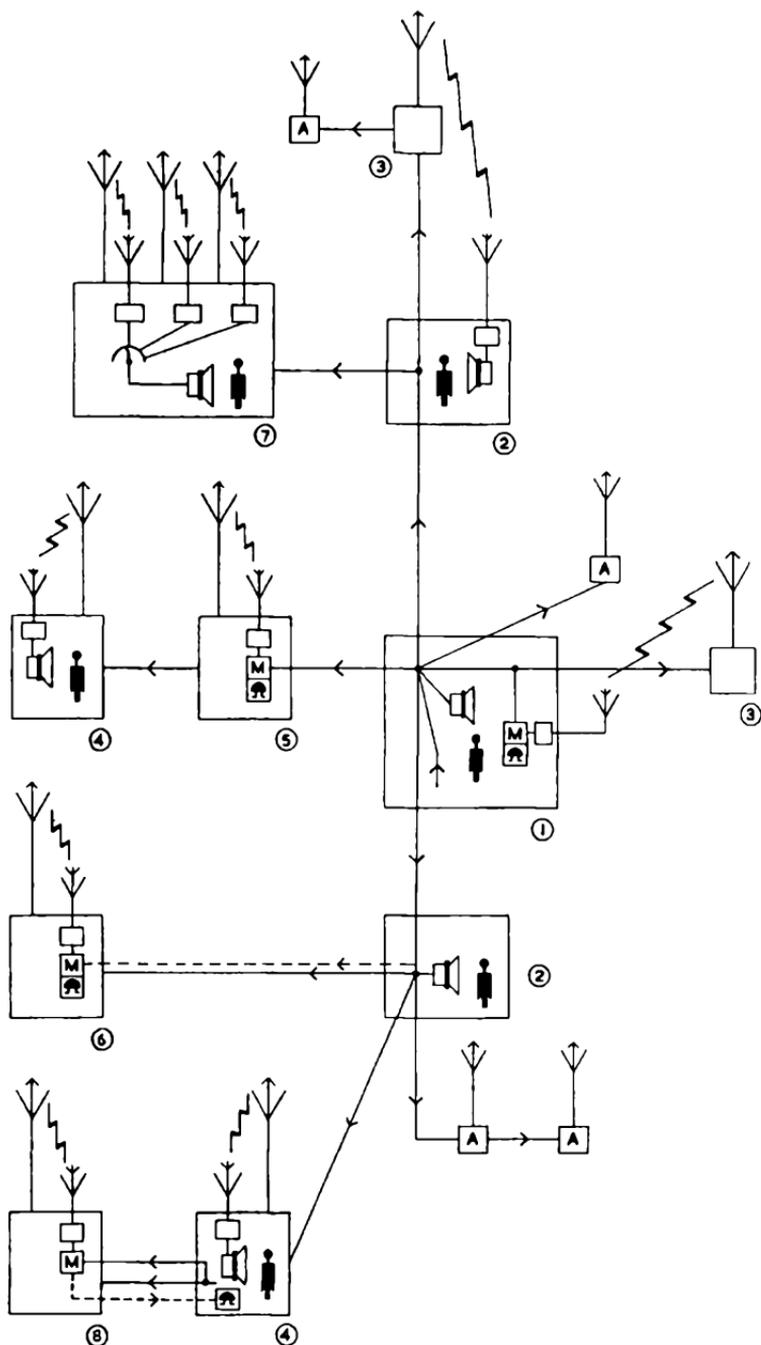


FIG. 84—Monitoring at various points in a distribution network  
(Dotted lines represent coded or d.c. signal line)

operator to listen first to all those points carrying one programme before passing to all those carrying another. In this way he is aware only of the changes of programme and not of individual check points and the strain of listening is considerably eased.

One final point should be mentioned. However vigilant the source monitor may be, there will inevitably be occasions when an unexpected peak will pass into the system. The tolerances allowed on most of the apparatus before overload causes serious distortion are generally sufficient to cover such errors, save in the transmitter itself where appreciable overmodulation cannot be tolerated. Each transmitter of any size is therefore fitted with a limiting amplifier which cuts down the gain of the chain momentarily when an excessive peak is received, restoring it gently thereafter. This unit therefore may be classed as a monitor since it watches the programme in a sense and takes executive action to prevent serious distortion or transmitter damage occurring when the load is excessive.

## CHAPTER 9

# THE BBC COMMUNICATION SYSTEM

*By J. S. NORWELL*

*Lines Department*

### 9.1 INTRODUCTION

THE communication system described in this chapter consists of a network of Post Office private lines, together with associated equipment used for telephone and telegraph communication between BBC centres throughout the country.

Although the communication system does not form an actual link in the transmission of programmes, it has a vital part to play—something like that of the lubricant without which a chain would creak very audibly, work inefficiently, and ultimately fail.

At the present time the BBC is organised in seven regions, each of which is more or less autonomous as far as programmes and administration are concerned. Nevertheless, many services must remain centralised if they are to work efficiently—e.g., the higher direction, finance, and engineering specialist departments—and all these must be in easy touch with each other and with the regions.

BBC centres are located in widely separated towns, for example, London, Manchester, Belfast, and within these towns there may be many separate premises. There are also “intermediate” satellite centres, located in lonely parts of the country. It is quite obvious that the internal network of communication must be comprehensive and varied to give satisfactory service to administrative, programme and engineering groups.

### 9.2 GENERAL REQUIREMENTS

#### 9.2.1. *Users of the Systems*

The seven regions of the BBC (see Fig. 85) all contribute to the three programme services which at present are designated Home, Light and Third. London, however, is the central headquarters for programme direction, administration and engineering, in the Home, Overseas and Television services.

The main users of the communication system are: (i) the engineers who control programme origin, routing and transmission, located in stations scattered all over the country; (ii) programme planners,

announcers, news staff, etc., with similar but less extensive distribution; (iii) general administrative officers in direction, engineering and general services.

### 9.2.2. *Telephone Requirements*

The telephone requirements of all groups are very similar; control room engineers are mainly concerned with programme operations and they, therefore, need little more than simple two-way speech circuits between adjacent centres and between these centres and the transmitters, but they must have absolute priority. Those in the second group (especially those in charge of programmes actually being broadcast) must obviously be in close touch, although much forward planning can be done through the post. The third administrative group needs a wide range of communication facilities, from the office next door to the farthest outpost.

### 9.2.3. *Telegraph Requirements*

Telegraphy is needed by all categories—the control room engineers for passing service messages; programme staff for transmissions of scripts, timings, and news; and general administration, whose use of telegraphy, too varied to detail, is mainly for material which requires more rapid transmission than by post, yet cannot be dealt with satisfactorily by telephone.

### 9.2.4. *Circuit Classification*

The communication system is somewhat similar to the public service provided by the Post Office in that it can be divided roughly into three groups: (i) short-distance, or “local” networks up to about 10 miles, connecting premises in the same town; (ii) medium-distance or “toll” areas, say 10–50 miles; and (iii) long-distance or “trunks”, connecting widely separated regions. We shall not consider further the local services, for they are usually obtained by a network of Private Wires (PW) and Private Branch Exchanges (PBX) of standard Post Office pattern, which also allow access to the public services; these are fully described in Post Office and other publications.

## 9.3 TECHNICAL REQUIREMENTS

### 9.3.1. *Basis of Assessment*

The design of a communication service must take into account these factors: (i) the users' needs; (ii) the quality of the service that will satisfy them; and (iii) probable future developments. We have already assessed the type of service wanted. Let us now see what sort of quality will be acceptable. This means broadly the volume and

## THE BBC COMMUNICATION SYSTEM



FIG. 85—The seven BBC regions

clarity of the telephone conversations, speed of connection, and the accuracy, speed and capacity of telegraph circuits.

The efficiency of a telephone circuit can be measured to a fair degree of accuracy by a statistical method known as "Immediate Appreciation". Sentences of six to ten words taken at random are read in sets of 20 over the circuit under test and a success is scored whenever a sentence has been completely understood at once. The percentage of such sentences immediately appreciated is used as a measure of the efficiency of the circuit. Volume, frequency

range, circuit loss and noise, room noise, are all varied one at a time, and from the figures obtained it is possible to arrive at a good estimate of the best values of volume and frequency range in relation to circuit loss. Loss is more important than upper frequency cut-off, and provided the loss does not fall below 15 db an upper cut-off of 2,300 c/s allows 90 per cent of all calls to enjoy an immediate appreciation of 90 per cent. No increase in efficiency results by transmitting frequencies below about 300 c/s.

### 9.3.2. *Speed in Connecting Circuits*

The speed of connection, while dependent to some extent on circuit and equipment reliability, depends far more on the number of circuits provided and on operating efficiency. Quite obviously, one line cannot carry more than 60 minutes of conversation an hour, and if each call, including the time taken by the operators to establish it, lasts 5 minutes, 12 conversations will be possible. This amount of conversation represents one "traffic unit". Put another way, if there are 48 5-minute calls between two places, then four lines would be needed. The number of traffic units, therefore, is given by the number of calls per hour multiplied by the average duration of a call in minutes, divided by 60. In the example,  $\frac{48 \times 5}{60} = 4 \text{ T.U.}$

### 9.3.3. *Accuracy of Telegraph Service*

Accuracy in telegraphy is mainly a function of the code used, and the stability and freedom from interference of the channel. As with telephony, operational ability is of great importance. The speed of the teleprinter system is internationally fixed at a maximum of about 66 words per minute; allowing for average manual operating efficiency of about 75 per cent, the speed obtainable is about 35 words per minute for short periods, coming down to 25 words per minute over a long period. Thus one circuit can handle about 12,000 words per day of 8 hours, sending in one direction at a time.

### 9.3.4. *Flexibility*

With possible future developments always in mind, flexibility is an important consideration—that is, the ability to change circuit routing to cope with changing traffic.

The equipment must be capable of working with existing installations—PBXs, private wires, local telephones, Post Office teleprinter machines and circuits.

### 9.3.5. *Maintenance*

Although it is of first importance that maintenance should be reduced to a minimum, the equipment is of such complex nature

that specially trained engineers are needed for its efficient maintenance.

#### 9.4 DESIGN CONSIDERATION

##### 9.4.1. *Circuits Required*

The circuits required in the BBC Communications System are broadly as follows: (i) permanent (24-hour) control circuits between adjacent control rooms; (ii) one or more permanent PBX circuits between London and each regional centre, and between some of these centres; (iii) supplementary PBX circuits over these routes during peak periods; (iv) permanent telegraph circuits between London and each regional centre; (v) possibility of expansion if telegraph traffic should increase.

##### 9.4.2. *Use of Existing Lines*

Two important factors influenced the design of the BBC system: (a) the existence of a number of music circuits which are used only in the evening, but rented 24 hours a day (because of the financial advantages of that arrangement); (b) reserve music circuits which are frequently idle. Since the great bulk of telephone and telegraph traffic occurs during office hours, ending just about the time when additional evening programmes start, these two uses dovetail fairly well and provide communication circuits without extra rental.

##### 9.4.3. *Basic Communication Unit: Carrier Circuits*

In view of all the foregoing considerations, a satisfactory communication system could be made up as follows: A basic "communication unit" consisting of a carrier system using two lines (one in each direction between adjacent centres), providing three two-way telephone and three two-way telegraph channels; and directly-rented circuits to supplement them. These channels should be capable of extension on other carriers to form low-loss direct links between London and the main centres. The carrier design should allow it to be used (i) with any music-type line, so that if the normal line failed, service could be restored on any music line which might be available; (ii) with music lines that are frequently idle during office hours.

##### 9.4.4. *Non-Carrier Circuits*

The carrier principle is not always the best to use, either for economic reasons, or because only one circuit is required, or merely because suitable circuits do not exist. Then direct "point-to-point" speech circuits can be rented from the Post Office, or special equipment designed to use a single short two-way music circuit to provide one high-quality low-loss speech circuit.

#### 9.4.5. *Location of Equipment*

The equipment is concentrated as far as possible in technical areas at each centre, having regard to convenient accommodation, access to cables, and facilities for maintenance, which is carried out by BBC engineers, specially selected and trained for the work.

#### 9.4.6. *Telegraph Equipment*

Telegraph equipment consists largely of teleprinters; operation of the keyboard of one machine causes a printed message to be delivered at the distant machine. These machines have been highly developed commercially, and are used extensively by the G.P.O.; the BBC has drawn largely on the experience of the Post Office in methods of using the system, but maintains its own machines.

### 9.5 PRINCIPLES AND METHODS

An exposition of transmission theory would be out of place here, but we must consider some of the applications of this theory to telecommunications practice.

We are concerned with frequencies not much higher than 10 kc/s, and although this is low by comparison with radio communication the range is large, for from the 25 c/s of teleprinter practice to 10 kc/s there are eight octaves. Within this band many communication channels can be accommodated by the use of frequency-band filters, often in conjunction with frequency-shifting devices.

#### 9.5.1. *Frequency Band Filter*

A filter is ideally a purely reactive device which allows the transmission of one or more bands of frequencies without loss and attenuates all others. Simple (unbalanced) prototypes of low-pass and high-pass filters are shown in Fig. 86. In a low-pass filter the loss is very small from zero frequency up to the *cut-off* frequency  $f.c.$ , whereupon it starts to increase up to infinity. Conversely in a high-pass filter the loss at infinite frequency is zero down to the cut-off frequency, whereupon it increases to infinity at zero frequency. *Band-pass* and *band-stop* filters have two cut-off frequencies, between which the transmission loss is zero and infinity respectively. Many ingenious variations of these prototypes have been devised to meet special requirements. An important filter characteristic is input and output impedance, which must bear a definite relationship to that of the apparatus to which it is connected. To avoid reflections and excessive loss in the pass range input and output impedances should be equal; in Fig. 86 (a),  $Z_{2_2}$  and the load resistance  $ZR_R$  must be so related.

THE BBC COMMUNICATION SYSTEM

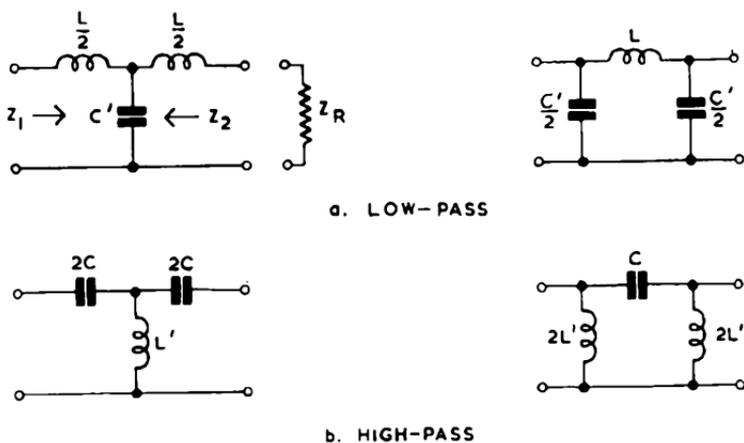


FIG. 86—Band filters

(a) Low-pass  $f_c = \frac{1}{\pi \cdot \sqrt{LC}}$

(b) High-pass  $f_c = \frac{1}{4\pi \cdot \sqrt{LC}}$

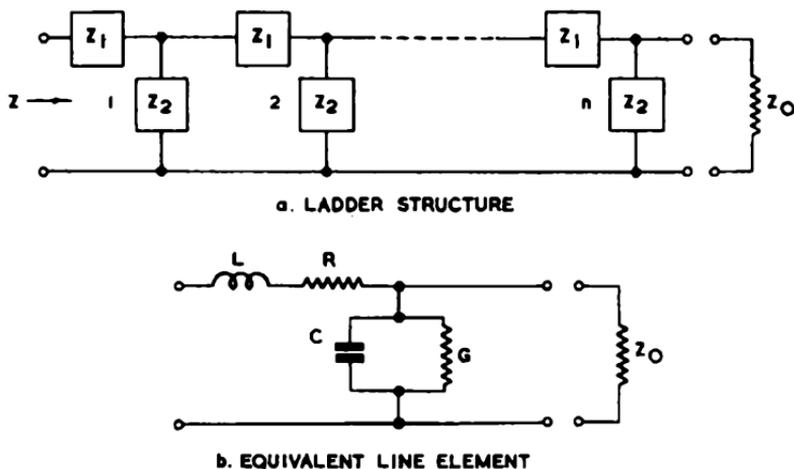


FIG. 87—Ladder-structure

A special case of interest is that of the ladder structure (Fig. 87 (a)). If  $Z_0$  is the input impedance of each of an infinite number of  $Z_1, Z_2$  sections, then the impedance looking into the network shown is also  $Z_0$ . Now a line—single wire or pair—terminated in its characteristic impedance (p. 125), can be represented by an infinite number of sections of such a ladder structure, each having this characteristic

impedance  $Z_0$ , made up of series elements  $Z_1$  and shunt elements  $Z_2$ ;  $Z_1$  and  $Z_2$  have the composition as shown in Fig. 87 (b),  $L$  being the inductance,  $R$  the resistance,  $C$  the capacitance (between wire and ground or wire and wire) and  $G$  the leakage.

9.5.2. Hybrid Transformer (Hybrid Coil)

Another important device is the hybrid transformer or hybrid coil, also called from its construction, the "three-winding" transformer. Referring to Fig. 88, the windings of the transformer are

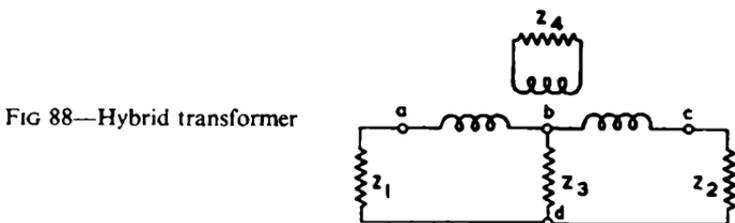


FIG 88—Hybrid transformer

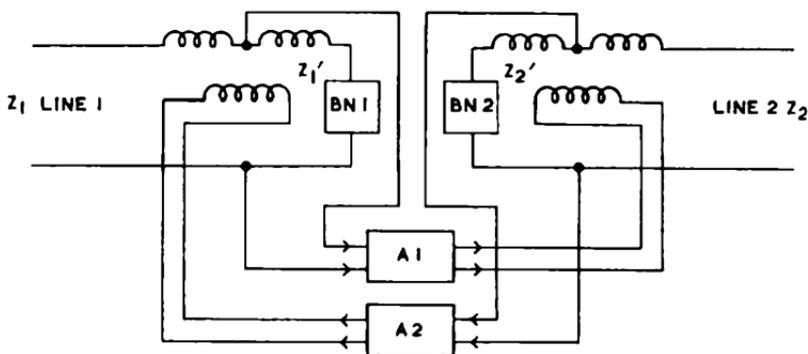


FIG. 89—Use of hybrid transformer on a repeated line

arranged with great accuracy so that winding  $ab =$  winding  $bc$ ;  $ab$  and  $bc$  are connected in series-aiding;  $Z_1 = Z_2$  and  $Z_4 = \frac{1}{2}Z_3$  (as seen through winding  $ac$ ). No transfer of power can take place between  $Z_1$  and  $Z_2$ , nor between  $Z_3$  and  $Z_4$ ; power from  $Z_1$  or  $Z_2$  can be transferred to  $Z_4$  and vice versa, but half of it is dissipated in  $Z_3$ , and there is a loss of 3 db in the process.

Two common applications of this device are of interest. The first (Fig. 89) is an early type of amplifier (repeater) used on a long telephone pair, in which amplification is applied in both directions simultaneously. The balancing network impedance must match that of the line; provided they are sufficiently well matched, very little of

the amplified speech currents in one anode circuit will appear across the grid of the other valve. Any unbalance allows a certain amount of energy to be fed back, and if sufficient is fed back a howl or "sing" will develop.

The "singing point"  $s = 20 \log_{10} \frac{Z_1 + Z_2}{Z_1 Z_2}$ , where  $Z_1$  and  $Z_2$  are the line and network impedances of such "two-way two-wire" circuits is dependent on the accuracy with which the balancing networks simulate the lines, and so in practice a limit is set to the amplification obtainable: this limitation cannot always be tolerated, so that on some circuits, the four-wire method has to be used.

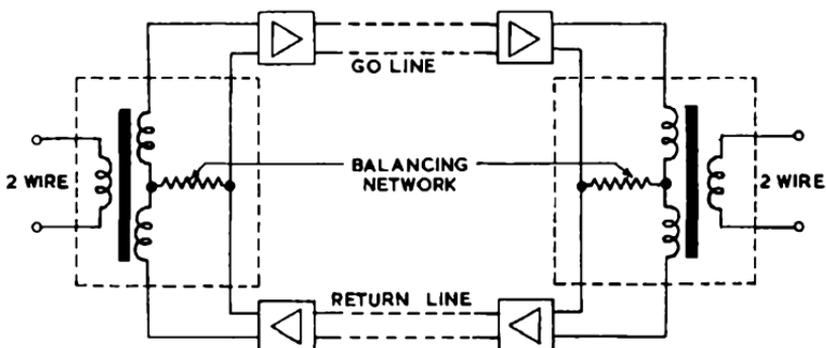


FIG. 90—Use of hybrid transformer to change from two-wire to four-wire circuit

The second use of the hybrid coil (Fig. 90) is to change two-wire working into four-wire working, and vice versa. Two pairs are required for the main circuit—one for the GO direction, the other for the RETURN, as they are commonly called—and amplifiers of normal type are inserted where needed to restore volume; restrictions of circuit length and construction are not apparent in this system. The two pairs must, of course, be combined at their ends to enable two-way conversation to take place, this connection being effected through the hybrid coil. Assuming that the GO and RETURN impedances are equal (as they can easily be), the degree of stability depends on the balance at each end between  $Z_4$ , the two-wire side, and  $Z_3$ , the balancing network. In practice, the balance is never perfect, but is good enough to provide a four-wire circuit which has only 3 db loss between one two-wire circuit and another, even though it may be several hundred miles long.

It sometimes happens that two such four-wire circuits have to be connected together; it is possible to do this without the usual loss in the hybrid, and therefore to provide, from two or more circuits in

tandem, one low-loss circuit of considerable length. The simplest method is, of course, to miss out the hybrids altogether, and connect the two pairs through; but as we have specified flexibility (section 3.4), and as we must preserve ease in maintenance, another way has been adopted, colourfully described as “tail-eating”, and is illustrated in Fig. 91. Briefly, instead of balancing the two-wire connection  $Z_4$ ,

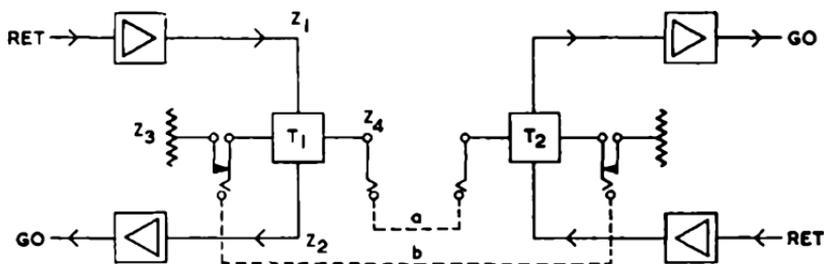


FIG. 91—Connecting two four-wire circuits without loss by the tail-eating method

by a network  $Z_3$  the latter network is replaced by the impedance of the other bridge-point, assumed to be similar because we are dealing with similar carrier circuits. With one or two modifications (means for phasing and adjusting voltages) this method produces the desired result, it being possible to test any portion of a long circuit or to switch it in a two-wire condition, merely by inserting plugs in the appropriate jacks as indicated.

### 9.5.3. Modulation and Demodulation in Communication Circuits

Modulation, and its complementary demodulation, are employed extensively in communications. The process is in essence the translation of a signal to a higher frequency range and its subsequent restoration to the original.

#### 9.5.3.1. The Static Modulator

The teleprinter instruments are operated by pulses of d.c. and to enable these pulses to pass over a circuit capable of transmitting audio frequencies but not d.c., they are used to modulate a tone of 2,600 c/s. The modulating device is called a *static modulator* and is shown in diagrammatic form in Fig. 92. The adjacent windings of the two 1 : 1 ratio transformers  $TR1$  and  $TR2$  are accurately centre-tapped;  $MR1$ – $MR4$  are metal rectifiers which exhibit a low resistance when the voltage is applied in one direction and a high resistance when the voltage is applied in the other. In one condition the voltage from the teleprinter makes the resistance of  $MR1$  and  $MR2$  low and that of  $MR3$  and  $MR4$  higher; 2,600 c/s from an oscillator will then

## THE BBC COMMUNICATION SYSTEM

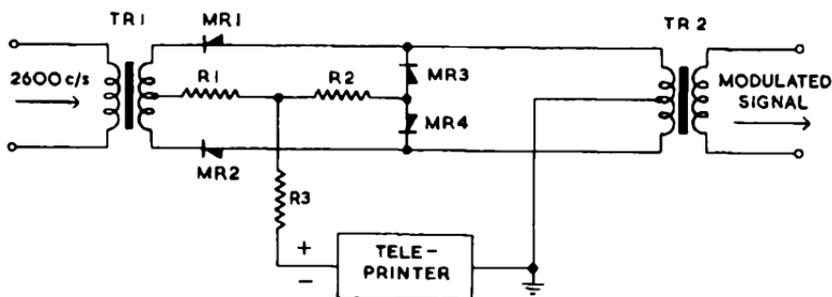


FIG. 92—The static modulator

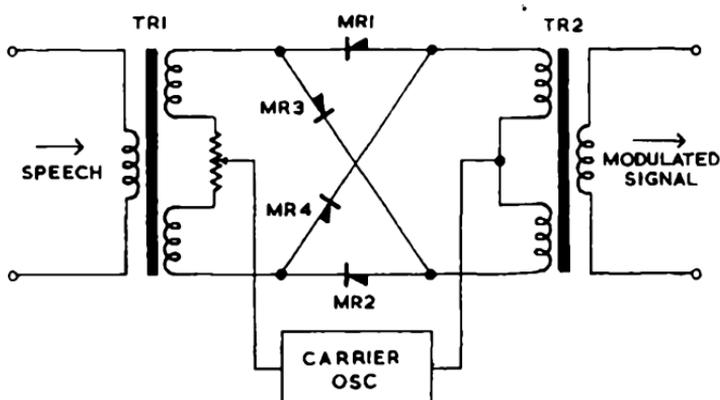


FIG. 93—The ring modulator

pass to line with little loss. On reversal of the voltage during signalling, *MR1* and *MR2* will be closed, and *MR3* and *MR4* open, and no oscillations will get through. This method is greatly superior to that using mechanical contacts which wear, corrode, and require adjustment. The result of the modulation process is a 2,600-c/s carrier  $\pm$  the teleprinter signal frequencies of up to about 80 c/s, the modulator acting rather like an ON-OFF switch.

The reverse process is not, however, used for restoring the d.c. signals at the receiving end, because the current and voltage requirements cannot easily be met by such process. Instead, as will be seen later, the 2,600-c/s pulses are demodulated by rectification, the d.c. pulses controlling a telegraph relay through a valve amplifier.

### 9.5.3.2. The Ring Modulator

In telephony the requirements are somewhat different; the modulating voltage is a complex wave and a *ring modulator* is used. As in the

static modulator, two transformers and four metal rectifiers are employed, but arranged as in Fig. 93. Exactly the same switching property of these rectifiers is used, but the effect on signals passing through is different, because both carrier and signal voltages are alternating, the carrier voltage having a higher frequency than the signal. Provided that the voltage of the carrier is much greater than that of the signal, each reversal of the carrier changes the direction of the signal current at the output. Suppose that a signal consisting of a 1,000-c/s sine-wave is applied to *TR1* primary and that a carrier frequency of 8 kc/s is applied to the centre-taps; no carrier will appear at the output since the primary of *TR2* is accurately balanced. A study of Fig. 94 will show that the 1,000 c/s is switched at every

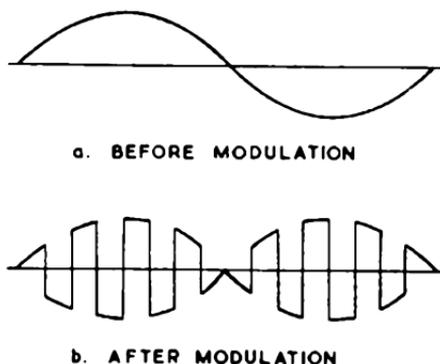


FIG. 94—Effect of ring modulation

half cycle of the 8,000-c/s carrier, the result at the output of the modulator being as shown by the thick line. This is in fact a double-sideband suppressed carrier—7,000 and 9,000 c/s with many harmonics, which are filtered out in subsequent circuits. A complex speech waveform is dealt with in the same way. The action of the ring is that of a reversing switch.

The demodulation process is very similar, but, of course, the result is the reproduction of the original signal frequencies and other frequencies of no significance which are subsequently filtered out. Sending and receiving carrier frequencies at each terminal must be within a few cycles of each other to avoid distortion; ideally oscillators should be locked by a pilot tone transmitted in some unused part of the frequency spectrum.

#### 9.5.4. Calling Signals

There are various methods for calling the attention of the distant station, or indicating the condition of the circuit, or switching. The usual way of calling is to send an a.c. at about 17 c/s along the

circuit in order to ring a bell, or, by rectification, to operate a relay. It is not convenient and usually not possible to signal by d.c. or even by low-frequency a.c. over circuits comprising amplifiers, and so frequencies in a higher range are employed—usually in the 500–1,000 c/s region, and commonly designated “voice-frequency”, abbreviated v.f. Within the BBC system the normal v.f. signal is 700 c/s but 600 c/s is occasionally employed if “selective” signalling is wanted.

## 9.6 THE THREE-CHANNEL CARRIER SYSTEM

We have seen that BBC communication requirements can be met by similar three-channel carrier systems between adjacent centres—some operative throughout the 24 hours, the others brought into service only during peak periods—supplemented by point-to-point circuits directly rented from the Post Office.

### 9.6.1. Derivation of Channels

A carrier system consists of a *carrier terminal* at each end, connected by two circuits (one in each direction) equalised and amplified to give equal transmission of all frequencies from 50 c/s to over 8,100 c/s, with zero loss.

Fig. 95 shows approximately the division of the frequency spectrum to form three telephone channels.

Telegraph channels are provided as required by using a small frequency band at the extremity of each telephone channel; this is possible without appreciably affecting intelligibility.

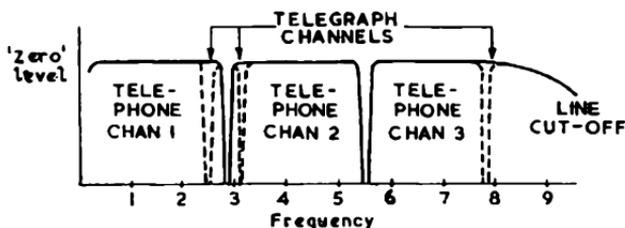


FIG. 95—Division of audio-frequency band for three telephone channels

### 9.6.2. Speech Paths on Individual Channels

Let us now trace the paths of speech through the equipment from terminal *A* to terminal *B* for all three channels in turn, Fig. 96, neglecting attenuators and telegraph channels for the moment. First, channel 1, the audio unmodulated, or physical channel. Speech coming from a telephone instrument or PBX switchboard is limited in level by *L1* and passed by Hybrid 1 to the send side, restricted to 2,730 c/s by *LP1* and after having its volume adjusted to the correct

level by the send amplifier, is sent to line. Arriving at the other terminal, it passes via the variable attenuator to *LP1*, thence via attenuators to channel 1 amplifier, into Hybrid 1 and so to the receiving PBX or instrument. Speech in the reverse direction takes a precisely similar path.

Channel 2 speech is limited by *L2* and passed by Hybrid 2 to Modulator *MR/2* No. 2, where it modulates a carrier of frequency 5,600 c/s supplied through a trap-valve. The resultant upper sideband 5,400 c/s–2,860 c/s passes through *LP2* and *HP1* and then joins the path taken by channel 1 until it reaches *LP1* and *HP1* at the receiving end. At this point there is only one path open, via *HP1* and *LP2*, through which it passes to be demodulated in *MR/3* by a 5,600 c/s tone. The unwanted upper sideband is suppressed by *LP4*, and the lower sideband which is in fact channel 2 speech reconstituted, is restored to correct level in channel 2 amplifier and goes via Hybrid 2 to the channel 2 subscriber. Again, return speech takes a precisely similar path in the opposite direction.

Channel 3 operates in much the same way; but let us suppose that we wish to derive a telegraph circuit on it. Speech from PBX is limited by *L3* and passed to Send through Hybrid 3, but now it is restricted to 2,370 c/s by *LP3*; there the modulated 2,600-c/s telegraph tone joins it, the resultant combination being again modulated by a 5,400-c/s carrier in *MR/2* No. 3. This time the upper sideband 5,600–8,100 c/s is selected by *HP2* and joining channel 2, passes with it to the receiving terminal until it reaches the junction of *LP2* and *HP2*. The latter is the only path open to it, by which it reaches *MR/3* to be demodulated by 5,400 c/s. The unwanted upper sideband is suppressed by *LP4*, the reconstituted speech passing through channel 3 amplifier, *LP3* and Hybrid 3 to the receiving subscriber; the telegraph tone can pass only through *HP3*. The return path is exactly similar. Once the combined signals reach the input of the Send amplifier, they can be treated as a single entity—equalised, amplified, and switched as one, until they reach the junction of *LP1* and *HP1* again.

Complete linearity of common amplifiers is necessary since the effects of intermodulation are very annoying—probably even more so than if the line were used for a single programme, when only a certain “roughness” would be noticed. Fuller details of the equipment used in the BBC Carrier Communication System are given in Technical Instructions L1 and L2.

## 9.7 SIGNALLING

Signalling over a multichannel telephone system is necessarily done by voice-frequency means, since neither d.c. nor low fre-

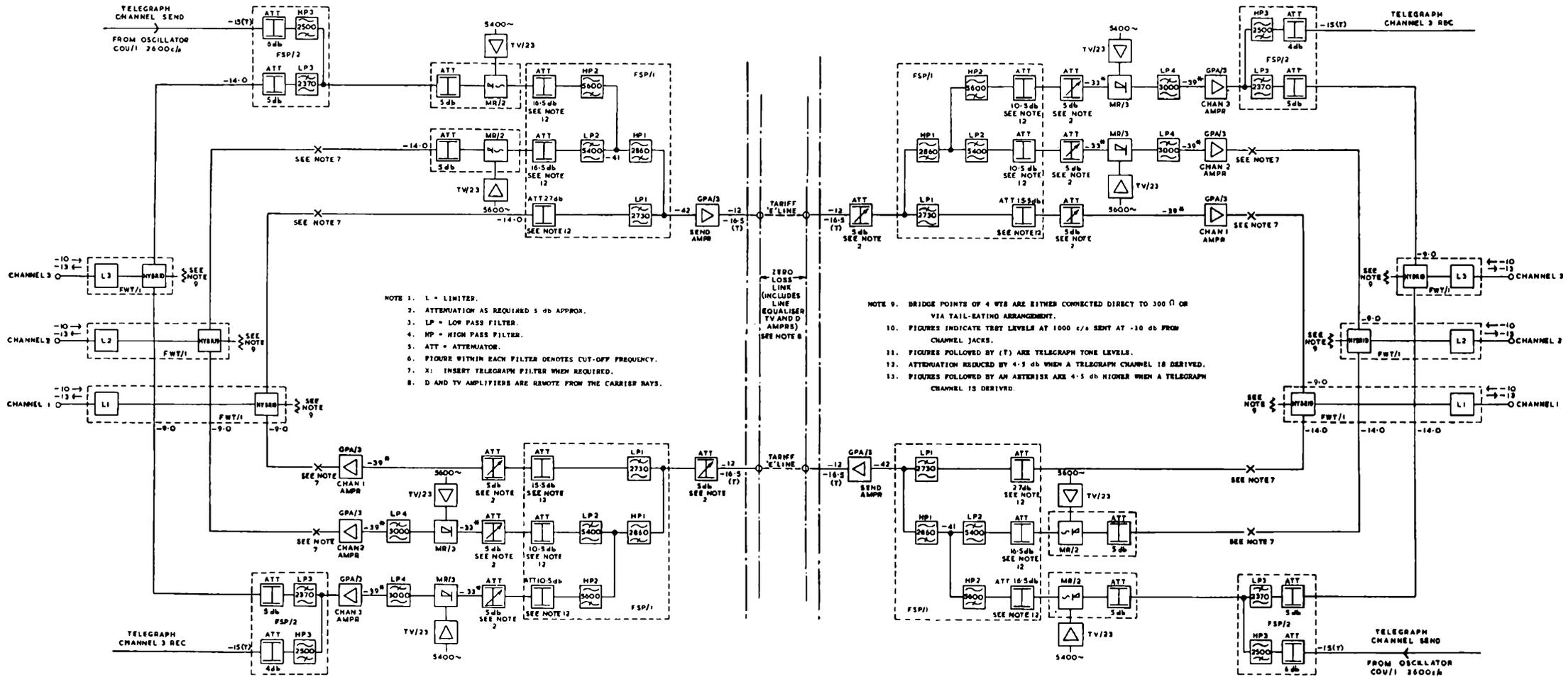


FIG. 96—Schematic of three-channel carrier system

quency a.c. (i.e., 17 c/s as used on local Post Office circuits) will pass through; and to avoid confusion with Post Office signalling tones of 500, 750, 900 c/s etc., the BBC makes use of 600 c/s and 700 c/s.

### 9.7.1. The Signal Oscillator

The 700-c/s or 600-c/s tone is generated by a single-valve oscillator having adequate frequency stability (within about 0.5 per cent), negligible harmonic, and an output of 1 mW or +4 db above 1 mW, with 600 ohms impedance.

### 9.7.2. The Voice-frequency Ringer

The Voice-frequency Ringer VFR/1 performs the function of converting the normal ringing current of 17 c/s to 700 c/s and in the reverse direction from 700 c/s to 17 c/s.

Fig. 97 is a simplified diagram, from which it will be seen that in the outgoing direction 17 c/s from a telephone instrument or switch-board is prevented by the 17-c/s stop filter from passing to line, but operates  $C/2$  in conjunction with a capacitor and rectifier, contacts  $C1$ ,  $C2$ , causing 700 c/s to be sent to line for so long as 17 c/s is applied.

For conversion from 700 to 17 c/s, a high-impedance valve circuit is bridged across the line side of the 17-c/s stop filter.

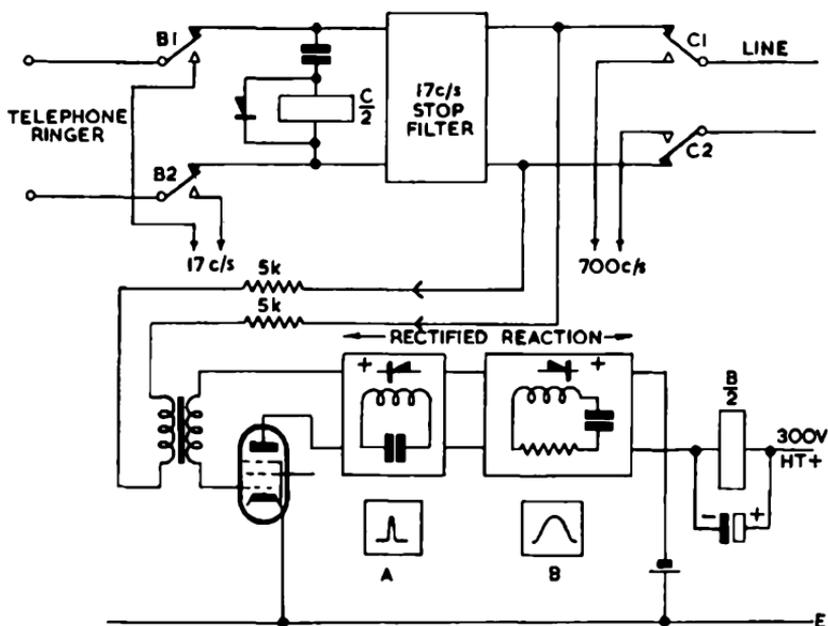


FIG. 97—Voice-frequency ringer: simplified schematic

Received 700 c/s is amplified by the valve which is biased back to pass about 1 mA standing feed and in whose anode circuit are two tuned circuits, *A* and *B*, in series; *A* is sharply tuned to 700 c/s, *B* is very flatly tuned, with a broad peak in the region of 1 kc/s. When a pure tone of 700 c/s is applied, a rectifier circuit associated with *A* causes a positive voltage to be produced in such a direction as to reduce the standing bias and increase the amplification. Thus the anode current rises very quickly and operates *B/2*, *B1* and *B2* sending 17 c/s to the telephone equipment. The electrolytic capacitor across the winding of *B/2* slows down its operating time so that transient 700 c/s components of speech do not operate it. Much greater protection, however, is provided by the negative reaction circuit *B*, which is affected by currents of other than 700 c/s and whose rectifier tends to increase the negative bias on the grid, so reducing the amplification and the current through *B/2*.

The sensitivity of the VFR/1 to 700 c/s is such that it will respond to signals of  $-20$  db to  $-25$  db; it will not respond to 17 c/s.

### 9.7.3. Selective Signalling

Other frequencies are used in order to provide the facility of selective signalling—i.e., to allow sharing of the circuit by two or more users . . . a “party line”. Referring to Fig. 98, if a telephone is

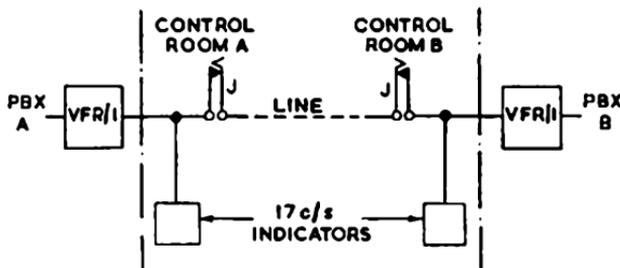


FIG. 98—Selective signalling

plugged into jack *J* and a 17-c/s ring is sent, it will not reach PBX, because a 17-c/s stop filter is incorporated in the VFR/1; but provided the circuit is a suitable one and not a carrier, it will be received at the other end on an indicator, connected as shown to the inner springs of a similar jack. When the other end answers by plugging a telephone into the jack, the two PBXs are cut off. Such a system is in use over some circuits, the jacks being in control rooms (which therefore have priority), with the circuits extended to the PBX at each end; but it can work with satisfaction only when the traffic originated by both users is small. A 700-c/s ring has no effect on the 17-c/s indicators; 17 c/s does not affect the 700-c/s receiver.

#### 9.7.4. *D.C. Signalling*

An economical method of signalling over short four-wire circuits is by d.c. converted to and from the local-end 17 c/s signal by means of the equipment shown in Fig. 99, and generally referred to as A.C./D.C. ringing.

Station A rings with 17 c/s into the shunted relay circuit  $C-A/1-MR$ , via relay contacts  $B1$  and  $B2$ .  $A/1$  operates  $A1$ , connecting a 50-volt d.c. supply to the phantom of one of the lines (normally the return line). At the receiving end  $B/2$  operates, sending a local 17 c/s ring.

Speech transmission takes place as described in section 9.5.2. The unit can, of course, be coupled to any other circuit employing 17-c/s signalling.

#### 9.7.5. *Recall Unit*

In place of the complicated Post Office supervisory system the BBC requires only a means of recalling an intermediate PBX operator. When this facility is not incorporated in the PBX switchboard, it is provided by the Recall Unit  $RCL/1$ , which comes into operation when a PBX cord circuit is connected. In effect it enables the distant PBX to attract the attention of the intermediate PBX by sending a series of rings, so "flashing" the supervisory indicator. One such unit is associated with each ringer panel.

#### 9.7.6. *Code Ringing Panel*

It sometimes happens that three fairly adjacent stations have a community of interests—e.g., control rooms at A, B and C, where low-loss circuits connect A-B and B-C. Any one station may want to call only one of the other two. Selective calling is accomplished in this case by connecting both circuits together through a code ringing panel at the intermediate station. If station A wants to call B, he gives one average-length ring, which calls B but does not pass through to C. In replying, B enters the break-jack looking towards A, and speaks. If A wants C, he gives two rings separated by not less than 40 milliseconds or more than 3 seconds; the second ring prevents B being called and relays a ring to C. The circuit provides for the case where A and B are engaged and C attempts to ring A; the ring is not lost but calls B. B can, of course, call either A or C by sending one ring.

The system is less expensive than the voice-frequency method, but cannot be used by PBX since it requires the use of a code; it is, therefore, confined to use on engineering circuits.

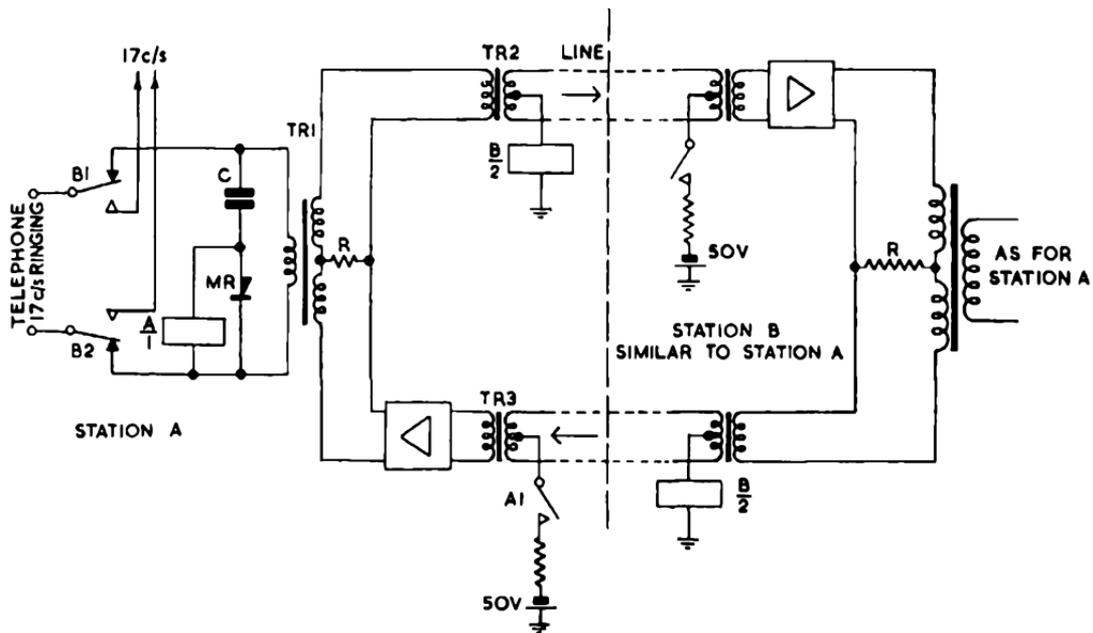


FIG. 99—D.C. signalling



CREED TELEPRINTER NO. 7B WITH N/3 KEYBOARD AND PERFORATING  
ATTACHMENT

(With acknowledgments to Messrs. Creed and Co. Ltd.)

*PLATE 8*



CREED AUTOMATIC TELEPRINTER TRANSMITTER NO. 6S/4  
(With acknowledgments to Messrs. Creed and Co. Ltd.)

## 9.8 TELEGRAPHY

### 9.8.1. *Manual and Mechanical Telegraphy*

For manual telegraphy one of the simplest codes is Morse, which makes use of two unequal-length signals, one three times the other, and spaces between letters and words. The introduction of mechanical senders and receivers led to the development of machines operated from a keyboard like a typewriter. By using such machines, the sending operator does not have to encode each letter himself, nor has the receiving operator to decode it; furthermore, a printed message can be obtained at the other end of a circuit. Such a system reduces the need for operating skill, speeds up the service and increases accuracy and reliability.

### 9.8.2. *Telegraph Codes*

The code most widely used for land-line telegraphy is the five-unit code. That is, a group of five elements (each of which may take up one of two conditions) can be arranged so that 32 different characters can be transmitted. The actual code is shown in Fig. 100; the last six are "functional" signals, i.e., they control movements of the mechanism but do not print. By international agreement each element except the last (which is the stop signal) is of 20 milliseconds duration. The condition when no signals are being sent is known as "marking"; when signals are being sent, the marking condition is punctuated by pulses of "spacing". A "start" signal precedes every character, and is of spacing polarity; the "stop" signal follows every character and is of marking polarity. Thus every character consists of seven elements—one start, five code, and one stop—lasting 150 milliseconds; and there is no pause between characters.

### 9.8.3. *Start and Stop Signals*

The start and stop signals are essential to preserve synchronism between sender and receiver; absolute synchronism is unnecessary and would involve additional costly equipment. A sufficiently close approximation is obtained by fitting a governor to keep machines within 0.5 per cent of the standard speed, and starting the receiver mechanism afresh for each character by a pulse from the sender, so ensuring that at least the beginning of each character is correctly timed; even a 5 per cent speed difference has negligible effect after a 150-millisecond interval. The receiving mechanism is brought to rest at the end of each character and the synchronism re-established at the start of the next character. The method is known as *start-stop* or *arhythmic*.

STUDIO ENGINEERING FOR SOUND BROADCASTING

START	CODE					STOP	LETTERS CASE	FIGURES CASE
	1	2	3	4	5			
							A	-
							B	?
							C	:
							D	WHO ARE YOU?
							E	3
							F	%
							G	@
							H	£
							I	8
							J	BELL
							K	(
							L	)
							M	.
							N	,
							O	9
							P	0
							Q	
							R	4
							S	'
							T	5
							U	7
							V	=
							W	2
							X	/
							Y	6
							Z	+
							CARRIAGE RETURN	
							LINE FEED	
							LETTERS	
							FIGURES	
							SPACE	
							ALL SPACING	

FIG. 100—The teleprinter telegraph code

### 9.8.4. Telegraph Speed

Speed has been internationally fixed on the basis of the 20-milli-second elementary signal referred to above. The unit of speed in telegraph transmission is the *baud* which represents one code element per second. Since the elementary signal is 20 milliseconds, the speed is said to be 50 bauds. (In terms of words per minute, since each character occupies 150 milliseconds, 6.6 characters are transmitted each second; assuming plain language to average six characters per word, the transmission speed is 66 words per minute.)

### 9.8.5. The Telegraph Waveform

Bearing in mind the speed of 50 elements per second, it can be seen that the signal produced by a series of elements which are alternately marking and spacing (positive and negative) is a square wave of fundamental frequency 25 c/s, with harmonics extending to infinity. Over v.f. circuits, marking is represented by presence of tone and spacing by its absence.

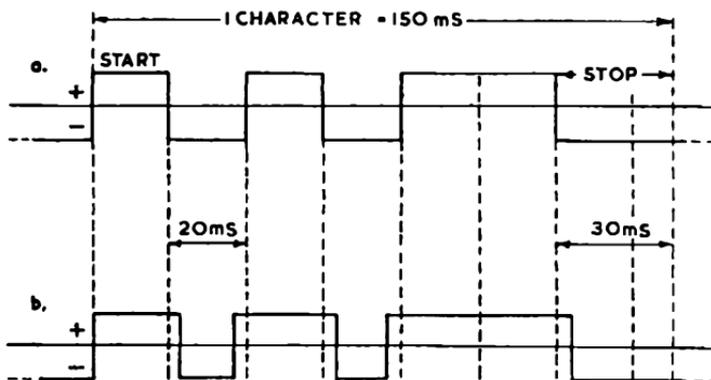


FIG. 101—D.C. teleprinter signal letter (S), a = perfect signal, b = distorted signal.

Fig. 101 shows the letter S (lower case) or its relative in the upper case: after the start signal (20 milliseconds spacing) which invariably precedes every character, the five code elements are (i) Mark, (ii) Space, (iii) Mark, (iv) Space, (v) Space—followed by the invariable stop signal (30 milliseconds marking).

There is no break between elements of the same sign, and theoretically the change-over between elements of opposite sign is instantaneous. In practice, however, neither this nor the square waveform, nor the precise elemental length of 20 milliseconds is perfect, but the receiving equipment can tolerate surprisingly large departures from the ideal before producing the wrong answer.

### 9.8.6. Telegraph Distortion

Distortion on telegraph circuits takes several forms; the most common is the lengthening of all space elements and the corresponding shortening of the others, or vice versa (bias distortion). Speed differences cause another form of distortion, and random distortion is present in most systems. All can be measured, traced and corrected.

### 9.8.7. Nature of Generated Telegraph Signals

The usual signal generated by telegraphic senders is d.c., interrupted or reversed when signalling.

#### 9.8.7.1. D.C. Operation

The telegraph current is, therefore, irregularly cut up into 20-millisecond code elements, either positive or negative, requiring a band-width of about 80 c/s for efficient transmission, and subject to distortion during transmission. The signal must undergo moderation before it can be accepted for transmission.

#### 9.8.7.2. Voice-frequency Operation

We have seen that a frequency band of 200 c/s (approximately 2,500 c/s to 2,700 c/s) is available above the speech frequencies in each BBC communication channel. Consider a tone of frequency 2,600 c/s, modulated with telegraph signals; there will be produced from the static modulator MR/4 a double-sideband with frequencies

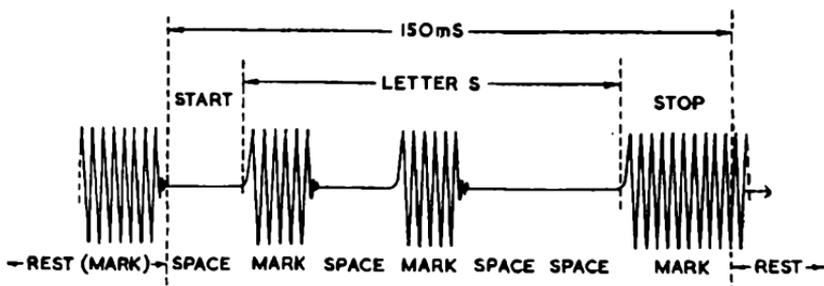


FIG. 102—Form of v.f. teleprinter signals (letter S).

extended during signalling from about 2,500 c/s to 2,700 c/s. From this point onwards the signal is in the form of pulses of 2,600 c/s (Fig. 102).

A similar demodulation process could be employed at the receiving end, but it is altogether simpler and more reliable to rectify the 2,600 c/s pulses and in this way control an electro-mechanical relay which produces the positive and negative d.c. from a local source to operate the receiving mechanism.

## 9.8.8. Telegraph Circuits

Telegraph circuits can be provided in a number of ways, e.g., metallic conductor with earth return; metallic pair; phantom with earth return; phantom GO and RETURN; derived on a telephone circuit as a tone below, above, or in the middle of speech frequencies by suitable filtering; or devoting a whole telephone channel to telegraph circuits. (Up to 24 such narrow-band channels can be derived on one telephone channel.)

## 9.8.8.1. Simplex and Duplex

There are two commonly used methods of operating telegraph circuits—simplex and duplex. Broadly, by simplex we mean transmission in one direction at once. Fig. 103 shows in simple form a

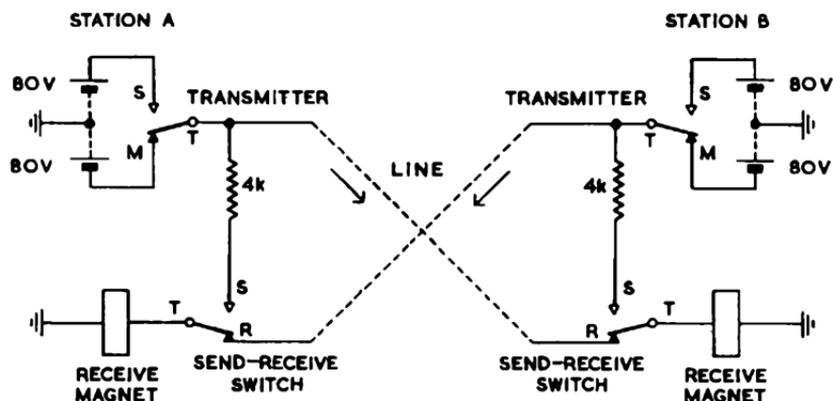


FIG. 103—Double-current simplex circuit

double-current simplex circuit. The transmitter armature normally rests on *M* and sends a “mark” to line; during signalling it moves backwards and forwards between mark and space contacts, sending negative or positive currents via the line to the distant station; they then pass through the receiving magnet via the send-receive switch. The latter switches the local receiver across the send line for the duration of each character in order to provide a local record. Note the cross-over in the line and that the system uses a separate wire for sending in each direction, with common earth return.

## 9.8.8.2. Omnibus Working

We have mentioned in section 9.4.3 a method of working in which several teleprinters are connected to one circuit so that any one station can send to all the rest. This is comparatively simple but requires slightly unusual equipment because BBC stations are in one

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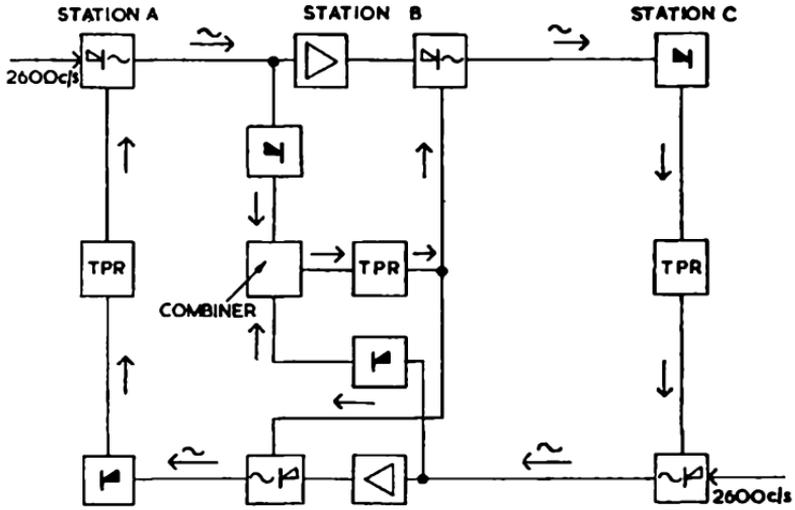


FIG. 104—Omnibus working

long chain and the connecting circuits use different methods of transmission. The terminal stations originate the telegraph carrier tone which is then modulated by one station at a time, in both directions; it will be seen that this requires two modulators controlled by the teleprinter, and two receiving units. The method adopted to combine the four units at an intermediate station is to feed them all into a bridging unit as shown in Fig. 104.

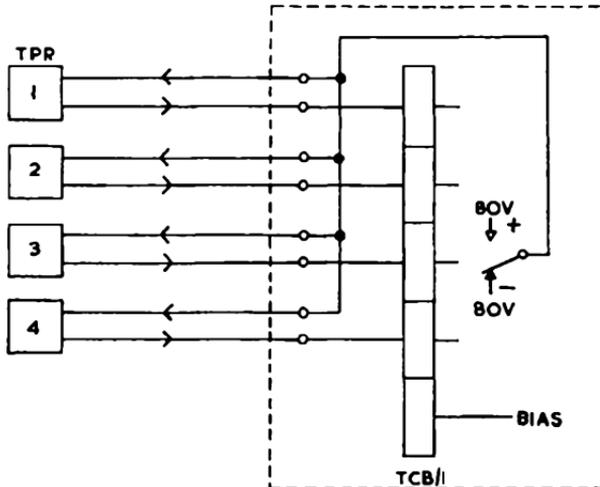


FIG. 105—Principles of the telegraph combiner

### 9.8.8.3. The Combiner

A simple means of connecting up to four teleprinter circuits together is frequently used. The usual relay has four working and two bias windings, so by connecting each incoming circuit to one of the working windings and applying the correct amount of bias-adjusting current to the bias windings, the relay tongue will respond to any incoming signal. Then by connecting the tongue (Fig. 105) to all four outgoing circuits, each machine will receive what any other sends. This apparatus is called a combiner. With some circuits of long delay it may be that reflections will cause trouble, but this can be cured by fitting suppressor relays in the offending circuits.

## 9.9 TELEGRAPH INSTRUMENTS

The standard instrument used for normal work is the Creed Teleprinter, No. 7B; in stations where traffic is heavier or circuit arrangements are more complex, the Creed Automatic Transmitter and Reperforator have been installed, and connected to a small switchboard.

### 9.9.1. *The Teleprinter*

Teleprinter No. 7B is a combined manual keyboard sender and printing receiver, both sent and received messages being typed either on a paper tape or on a continuous web of paper  $8\frac{1}{2}$  in wide, the latter being adopted in the BBC Plate 7 (page 192) shows a 7B teleprinter with dust-cover, N/3 keyboard, perforating attachment on the extreme left, and associated tape reel holder on the right; the latter two items are required only when an automatic transmitter is also installed.

Unit construction has been successfully applied to these instruments so that it is easy to replace a whole complicated section and the teleprinter will function immediately without further adjustment.

#### 9.9.1.1. Keyboards

Machines now in use are fitted with Type M or Type 7N/3 keyboards which have three rows of keys and a space bar, arranged as in Plate 7; two characters are allocated to most keys—a lower and an upper case, or letter and figure case. If the “letters” key is depressed, all following characters are in the lower case; depression of the “figures” key changes letters to upper case; thus the full range of characters usually required is available with 31 keys. Certain locking devices are incorporated to ensure that (a) any key once depressed is not released until transmission of the start, code and stop impulses has been completed, and (b) no other key can be depressed in that same period. Consequently, the maximum practicable speed falls somewhat short of the theoretical 66 w.p.m.

### 9.9.1.2. Receiver

The received current passes through the coils of the receiving magnet, whose armature controls the receiving mechanism; the start pulse connects the receiving mechanism to the already-running motor, the five-unit code elements cause the selecting train to place the typehead in the correct position for typing the wanted letter, and the stop element disconnects the mechanism and brings it to rest. The automatic start device, operated by the first space impulse, switches the motor on; if no signal is received for  $1\frac{1}{2}$  minutes, the device switches the motor off again. Assuming that the machine is in order and complete with paper and typewriter ribbon, reception can take place even when no operator is in attendance.

Immediately the transmitting mechanism starts, the receiving magnet is switched automatically from the receiving line and connected between the sending line and earth, and only returns to the receiving line at the end of each character. This function, performed by the send-receive switch, enables a local copy or local record to be printed at the sending end, while still enabling the distant station to "break in".

The paper at both ends of the circuit is moved along letter by letter, or upwards line by line, or returned to the beginning of the line, entirely by the same transmitted signals, so that the local record is an exact replica of the distant copy—given that the machines and circuit are in order.

### 9.9.1.3. Signalling Power Supplies

The source of signalling power must be capable of supplying at least 40 mA at 80 V alternatively positive or negative in relation to earth potential, and have good smoothing and regulation. The actual current for the receiving magnet is standardised at 20 mA.

The sending and receiving mechanisms are operated by the same fractional h.p. electric motor, running at a governed speed of  $3,000 \pm 15$  r.p.m. from the 50 c/s mains and geared down to the required working speed.

### 9.9.2. *Automatic Working*

Automatic working is being more widely adopted to enable maximum use to be made of line time, especially on systems which require distribution to several circuits or where circuits are poor, or if repetitions are required. The messages are stored by means of a parchment tape in which holes representing messages are punched in accordance with the five-unit code, and are later transmitted by an automatic transmitter at the maximum speed of 66 w.p.m. continuously. No special equipment is required at the receiving teleprinter.

The tape must be prepared on a perforator, either a separate instrument, an attachment to the printer, or a reperfector operated electrically from another printer. (See sections 9.9.2.2 and 9.9.2.3.)

The advantages of automatic working are: (i) greatly increased speed; (ii) economy of circuit use; (iii) storage of messages for subsequent transmission; (iv) possibility of many repetitions without further manual operation, and (v) regeneration of signals. The last advantage makes it possible to relay traffic over say two circuits whose individual distortion is within limits, but whose combined distortion makes it impossible to send over both in tandem.

### 9.9.2.1. The Transmitter

The automatic transmitter used by the BBC is designed to convert messages stored as punched tape into a train of electrical impulses on the line in accordance with the five-unit code, and to insert automatically the start and stop elements. (This is illustrated in Plate 8, page 193.)

On the section of tape illustrated in Fig. 106, the word PARIS appears twice separated by four spaces; each character occupies one vertical line, reading from top to bottom (ignoring the continuous

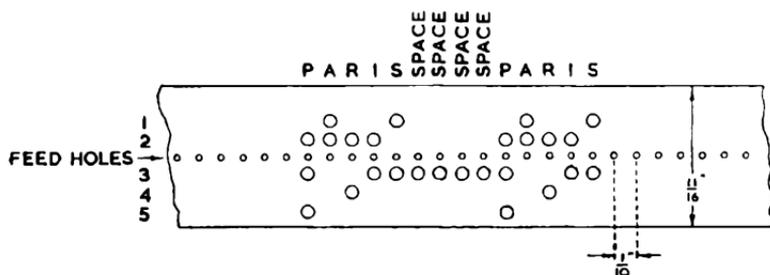


FIG. 106—Teleprinter tape: formation of characters

row of small holes which merely serve to move the tape forward), the distance between the vertical lines being one-tenth inch. As the tape is fed through the transmitter, five needle-like “peckers” explore the tape one after another at 20-millisecond intervals. Where a hole is punched, the corresponding pecker passes through, and the transmitter sends a mark; if there is no hole, the pecker cannot move, and a space is sent. A tape-control lever is fitted to ensure that if for any reason the tape tightens up transmission is stopped, and re-started when the tension is released. The transmitter is driven by a small motor kept to  $1,500 \pm 7$  r.p.m. by a governor. A send-receive switch is incorporated so that if required the distant station can break in.

## 9.9.2.2. The Perforator

The punched tape (Fig. 106) is prepared by a perforating attachment fixed to the teleprinter (see Plate 7); with this, a printed record is prepared while the tape is being punched, and errors can be seen and corrected at once. If not needed, the attachment can be disconnected, the teleprinter then functioning in the normal way. As mentioned under 9.3.3, typing speed can average only about 25 w.p.m. on a long-term basis, so that two perforators and therefore two operators are needed to feed one automatic transmitter.

## 9.9.2.3. The Receiving Perforator

Received signals can be made to prepare tape by means of a receiving perforator, usually shortened to reperforator, Incoming signals actuate the standard electro-magnet which controls motor-driven selecting and punching units and an auto-start device, which starts the motor with the first signal and switches it off after about 1 minute if no further signal arrives.

9.9.3. *Electrical Circuits*

The electrical circuitry of these instruments is simple and fairly flexible. Fig. 107 shows the connections of the 7B printer; different circuit arrangements can be obtained by changing leads and straps on connection strips fixed to the underside.

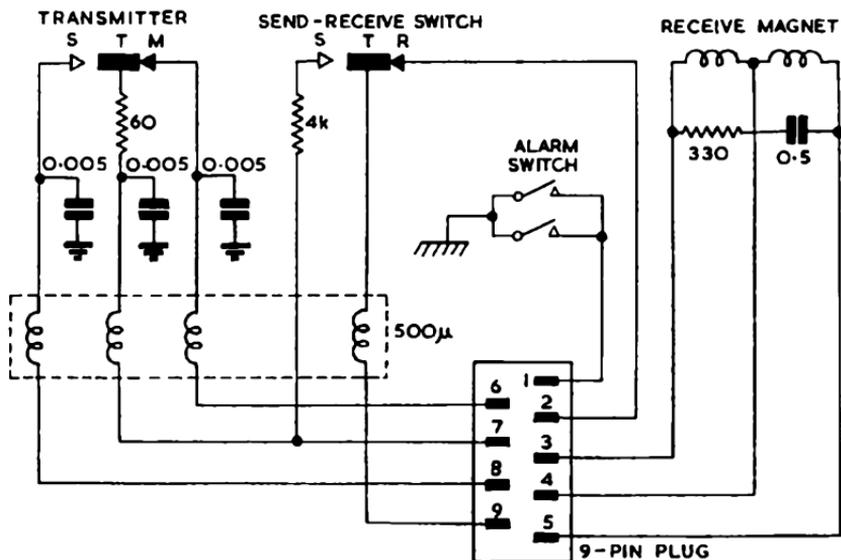


FIG. 107—Teleprinter: electrical circuit

9.9.3.1. Spark Quenching

As usual when circuits are made and broken, sparking occurs at the switch contacts. Spark-quenching measures are therefore essential to absorb and dissipate the inductive surge when the circuit

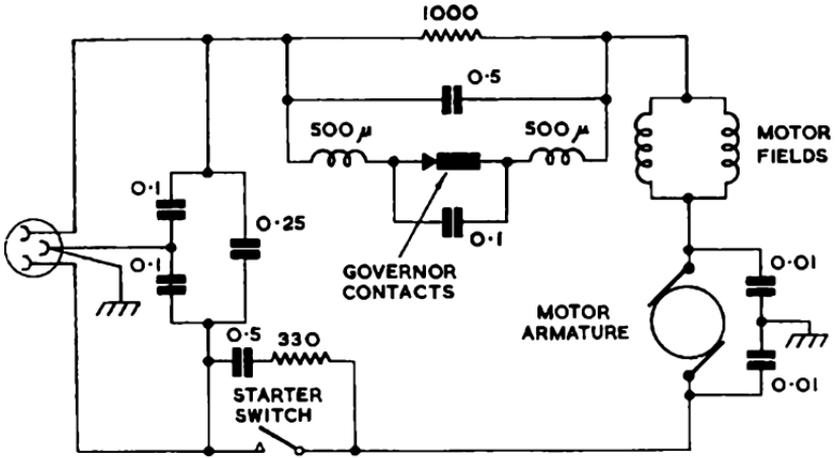


FIG. 108—Power circuit for teleprinter motor

is broken. They are shown in Fig. 108, which also gives the complete power circuits to the motor driving the operating mechanism.

Precautions are also taken to minimise r.f. interference; these are shown in Figs. 107 and 108.

The equipment used in the BBC telegraph service is described fully in a technical instruction on the carrier communication system (Instruction No. L.1).

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