TECHNICAL INSTRUCTION
P.4

Stereophonic Broadcasting

AMENDMENT RECORD

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To locate information on BBC equipment, consult the current List of Technical Publications.

CONTENTS

NEW DRAWING SYMBOLS

Certain new drawing symbols used in this Instruction have been introduced to conform to British Standard 3939.

SECTION 1

INTRODUCTION

The purpose of stereophonic sound reproduction is to increase the realism and pleasure of listening to broadcasts and records. It gives the listener some indication of the relative positions and movements of the sources of sound and induces a general effect of spaciousness which is not achieved in monophonic reproduction.

1.1 Directional Location of Sounds

Directional location of the source of a sound is not perfectly understood but for a normal human listener can perhaps be explained as follows.

A sound is transmitted from its source to the ears as a wave pattern in the intervening medium, which is usually air. Because the two ears occupy different positions, the wave pattern can reach them at different times, and with amplitude differences caused by the shadowing effect of the head; also time and amplitude difference variations are produced by slight head movements. These indications together with experience enable the direction of a sound source to be judged.

The predominant factor affecting this judgement is the effect of the difference in the times taken by the wave pattern in reaching the two ears. For simplicity consider a sinewave tone being radiated from a single source. At a particular instant, both ears respond to the sinewave but the difference in path-lengths produces a phase difference in the perceived signals which depends on frequency and on the dimensions of the head. At frequencies around 500 Hz, the angle of phaseshift is such that a rough estimate of the sound-source direction can easily be made, but at higher frequencies, when the difference in path-lengths exceeds half a wavelength, the directional information derived from the resultant phaseshift can be ambiguous. This accounts for the usual difficulty experienced in locating the source of a sinewave tone of a high audio frequency. Natural sounds, however, are almost always complex, with high frequencies contained within a low-frequency envelope, and for location purposes the ears respond to the envelope and rely mainly on the effect of time differences.

Interaural amplitude differences are insignificant below about 700 Hz, because the head is not large enough to produce appreciable changes, and even above this frequency, amplitude differences have only a slight effect.

With a central sound source, for which the interaural time and amplitude differences are zero, the indication is that the source lies in the vertical plane equidistant from both ears, but may be in front of, behind, above or below the listener. Off-centre sources are located in the general areas to the left and right of this plane. The accuracy of location is thus restricted with time and amplitude differences alone and for precise location these must be supplemented by rate-of-change information resulting from head movements.

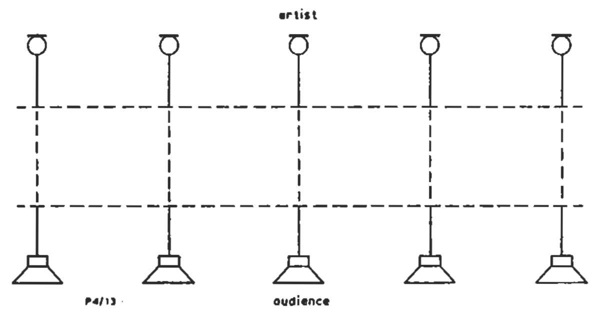


Fig. 1.1 Multiple-channel or 'Wavefront' System

1.2 Stereophonic Reproduction

The aim of stereophonic reproduction is to provide for the listener the additional clarity and possibility of identification which results from sounds appearing to come from positions corresponding to those of the original sources.

Such an effect could be provided by recreating, in the listening room, the wavefront presented to the audience at a live performance. This would require many microphones and loudspeakers and a large number of transmission channels as illustrated in Fig. 1.1. The wavefront idea is thus impracticable for broadcasting purposes although the concept is applied in the cinema where three or more channels are in use.

The system of stereophonic broadcasting used by the BBC is based on a two-loudspeaker formation as illustrated in Fig. 1.2. This system, also employed for stereophonic disk and tape reproduction, requires only two information channels, which should be closely matched in performance; it relies on the fact that two spaced loudspeakers fed with signals differing only in amplitude can produce sounds which combine at the ears to give apparent time differences. A 'sound stage', which occupies the space between and behind the loudspeakers, can thus be created. The best results are obtained when the two loudspeakers are identical but in any installation it is an imperative condition that they are *in phase*, that is, when they are fed with the same signal their cones must move in the same direction at the same time. Fig. 1.3 shows how the signals received by the two ears from the two loudspeakers combine to simulate the time differences required for positional information. Suppose the listener of Fig. 1.2 is at the apex of an isosceles triangle of which the line between the two loudspeakers forms the base, and that the same signal is fed to both loudspeakers. The left ear receives sound from the A loudspeaker slightly earlier than the corresponding sound from the B loudspeaker as indicated in Fig. 1.3(i). The sounds combine to form a resultant signal shown by the dotted curve C. Similarly Fig. 1.3(ii) shows the effect at the right ear. The two resultant signals as shown are coincident in time and the apparent source of sound is directly in front of the listener. If the amplitude of the signal to loudspeaker A is increased, the effect on the combined signals at the left and right ears is indicated by Figs. 1.3(iii) and (iv) respectively.

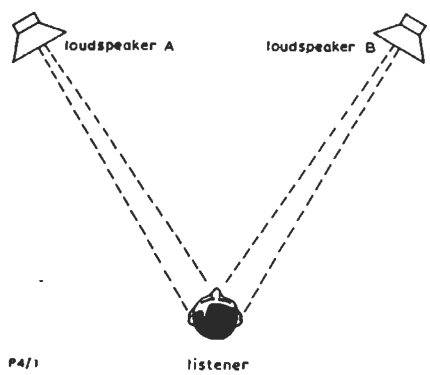


Fig. 1.2 Two-loudspeaker System

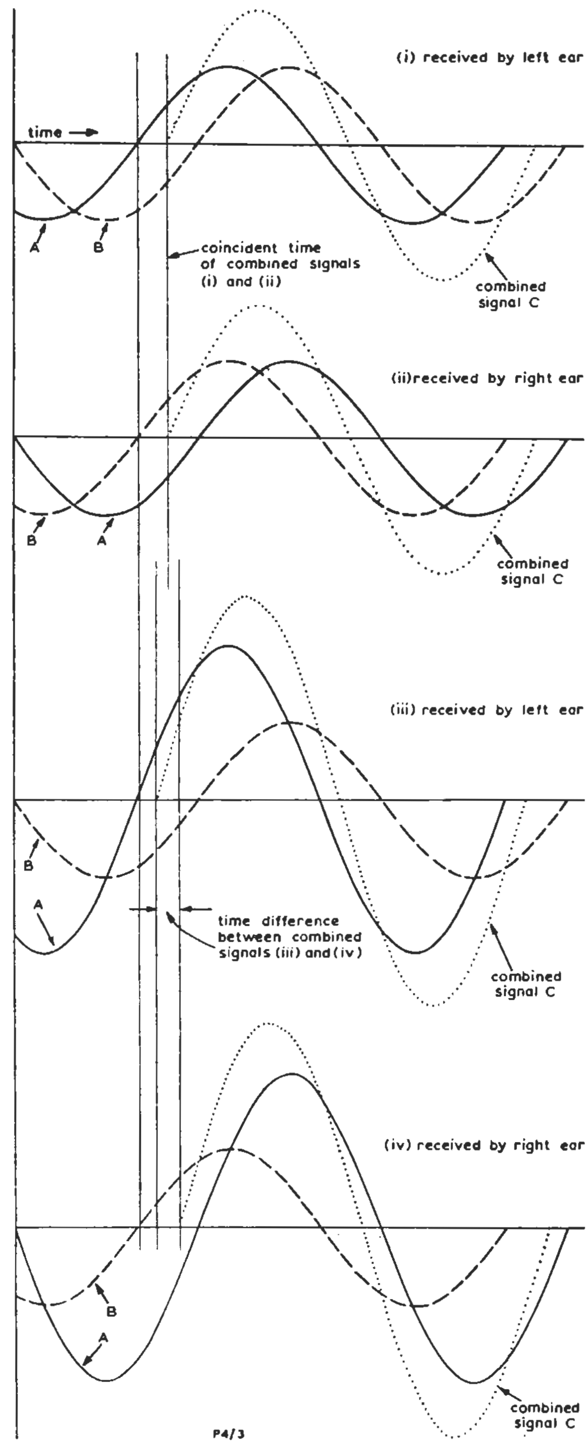


Fig. 1.3 Combination of Sinewaves of Different Phases and Amplitudes to Show Apparent Time Differences

These show that the information is received earlier by the left ear than by the right ear and thus the apparent position (or 'image') of the source moves towards the listener's left.

The movement of the image for a centrally placed listener is roughly proportional to the unbalance, expressed in dB, between the inputs to the loudspeakers. An unbalance of 20 dB is usually sufficient to shift the image from the centre to the extreme edge of the sound stage.

1.3 Microphone Systems

The information to be fed to each loudspeaker is effectively derived from a corresponding microphone in the studio. The microphones can be arranged in either of two basic ways known as the *spaced microphone system* and the *coincident microphone system.*

In the spaced system, the microphones may be arranged as in Fig. 1.4(a); the precise placing is dependent on their polar diagrams. Good stereo-

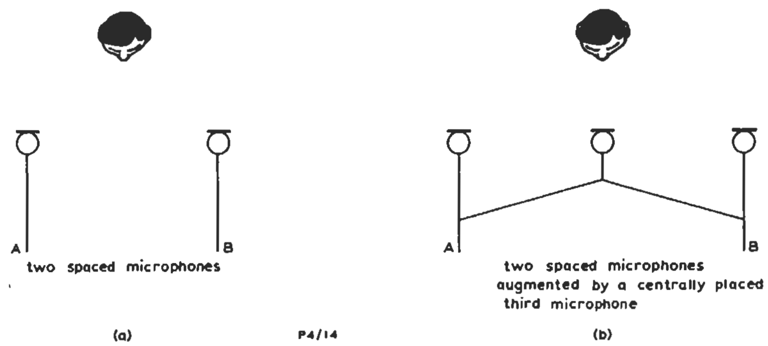


Fig. 1.4 Spaced Microphone Systems

If the listener is off the centre-line between the loudspeakers, the interchannel time differences are modified and the position of the image is different and less sharply defined than for a central listener, but for general listening satisfactory results can be obtained over a fairly wide area. The isosceles triangle formation is essential however for the precise and repeatable assessment of quality and positional information required for good monitoring.

Programme material produced within the BBC for the domestic stereophonic service is intended for reproduction on loudspeakers spaced from 6 to 12 feet apart.

Listening on headphones, in which the separate earpieces are fed with the signals intended for separate loudspeakers, produces a different effect. This is partly because each ear can receive information from one channel only and partly because rate-of-change information resulting from movements of the head with respect to the headphones cannot be derived. These factors combine to create the illusion of the listener being within the scene of action, whereas with loudspeakers the sound stage appears separate from the listener. Headphone monitoring may therefore give an erroneous impression of the programme balance and should be avoided if possible.

phonic results are obtainable with such an arrangement but the system has inherent defects, one of which is known as 'hole-in-the-middle', because the image appears to recede from the listener as the action moves from directly in front of one microphone towards the centre of the stage. Some improvement can be obtained if a third microphone is placed centrally and its output split between the A and B channels as shown in Fig. 1.4(b). Another problem is that the spaced system produces signals between which there are both amplitude and time differences, and is thus difficult to use because programme control systems can control amplitude only.

The coincident system uses two directional microphones mounted in a common housing and with their axes at an angle (usually 90 degrees). Such arrangements are known as *coincident pairs* and they produce good and readily controlled results for a minimum setting-up effort, largely because there is no time difference between their outputs and also because the one housing is easier to handle than a number of separate microphones. Coincident pairs are therefore used whenever possible, and details of several of these arrangements with differing polar diagrams are given in Section 3.

In the foregoing explanations, the letters A and B are used to identify signals and apparatus associated with the separate information channels. In accordance with international standard practice, the signals intended to be reproduced on the listener's left and right are called the *A* and the *B* signal respectively.

the choice of a transmission system.

Listening tests have shown that neither the *A* nor the *B* signal alone is satisfactory, but that a simple addition of the two is usually adequate as a monophonic version of a stereophonic programme. The *A* and *B* signals are therefore added together to

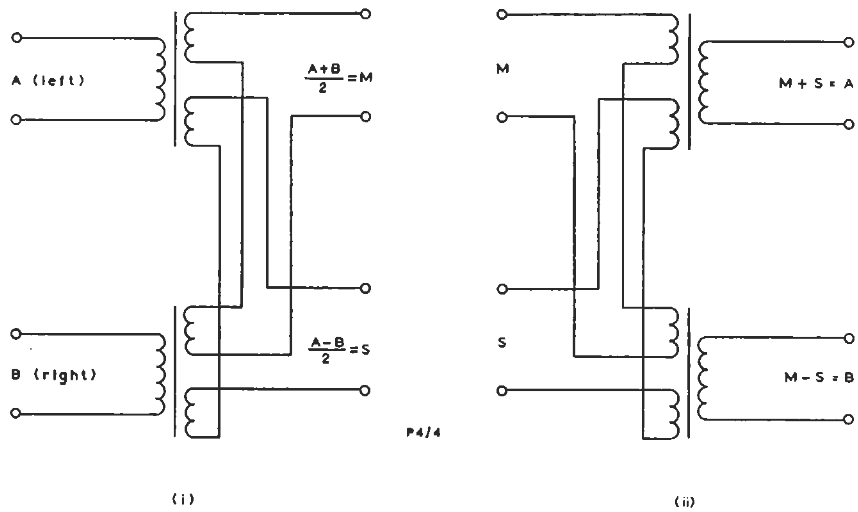


Fig. 1.5 (i) Derivation of M and S from A and B Signals (ii) Retrieval of A and B from M and S Signals

1.4 Transmission Systems

1.4.1 Requirements

The transmission system for stereophonic broadcasting must convey to the receiver the information contained in two separate audio frequency signals *A* and *B*. As there are insufficient broadcasting channels available to provide a separate stereophonic service, the system must be *compatible*; this means that a listener who does not choose to install stereophonic equipment must be able to receive on his existing equipment an acceptable monophonic version of the programmes. The system must also have reverse compatibility, so that stereophonic receivers can handle monophonic programme transmissions.

1.4.2 Compatibility

The compatibility requirement, as defined in the previous paragraph, affects both the methods of producing stereophonic programme material and the means whereby the studio output is transmitted. The two aspects are interdependent and the way in which an acceptable monophonic version of the studio output is derived has a major influence in form a sum signal which is known as the *M* signal*. Thus, if the *M* signal is transmitted in a form which can be accepted by existing monophonic receivers, the compatibility requirement for programme content can be satisfied. Stereophonic receivers require additional information, so that the original *A* and *B* signals can be reproduced. This is given by a difference signal, known as the *S* signal*, which is formed by subtracting the *B* signal from the *A*

* The letters *M* and *S* have become standard abbreviations to identify the sum and difference signals respectively. Their origin stems from an early system of stereophonic reproduction in which two microphones, one facing forwards and one facing sideways, were used. With this arrangement, the terms *M*iddle and *S*ide were adopted to identify the microphones which produced directly signals corresponding to the sum and difference signals of the modern system. (See 3.2.1.)

The coincidence that *m* and *s* are the initial letters of the words 'monophonic' and 'stereophonic' respectively could lead to wrong ideas. In fact, although the *M* signal is virtually indistinguishable from a normal monophonic signal, the *S* signal is not in itself a stereophonic signal. It is part of a stereophonic signal which comprises both the *M* and the *S* signal.

signal. One method of combining the A and B signals to form the M and S signals is shown in Fig. 1.5(i); a similar arrangement can be used to retrieve A and B from M and S as shown by Fig. 1.5(ii). Mathematically, as indicated on the diagrams, the M and S signals are half the sum and half the difference respectively. This factor of two must be taken into account in numerical calculations, but it is otherwise quite common to ignore it, and to talk about the $(A + B)$ and $(A - B)$ signals.

The bandwidth required to transmit two separate a.f. signals prevents stereophonic broadcasting in the long-wave and medium-wave bands, and therefore the transmission system requirements need to be considered only in respect of the Band II f.m. services.

The BBC uses the *pilot-tone system* (also known as the 'Zenith-G.E. System') of stereophonic broadcasting.

SECTION 2

THE PILOT-TONE SYSTEM

2.1 General

The pilot-tone system is used to combine the M and S signals into a single composite signal, known as the *multiplex* signal, which is used to frequency modulate the transmitter. With this arrangement, the S signal is used to produce an amplitude-modulated double-sideband suppressed-carrier signal which is added to the M signal. The frequency of the suppressed carrier is 38 kHz, and this results in the information of the S signal occupying the frequency range from 23 to 53 kHz, which is known as the *sub-carrier channel.* The M signal occupies the normal a.f. range up to 15 kHz and is known as the *baseband channel.* In addition to the sub-carrier and baseband channels, the multiplex signal includes a pilot-tone of 19 kHz which is used by the receiving equipment to reconstitute the suppressed 38-kHz carrier in the correct phase, and hence to extract the S information from the sub-carrier channel. The multiplex signal used to deviate the transmitted carrier thus has a spectrum as shown in Fig. 2.1. The process of producing the multiplex signal from the A and B signals is referred to as *coding*.

design. They need only respond to the baseband channel to reproduce the monophonic signal, and precautions are usually taken to reduce the noise output which would otherwise be present from the 'empty' sub-carrier channel.

2.2 The Multiplex Waveform

An examination of the composition of the waveform of the signal used to modulate the transmitter is useful in understanding the practical techniques of programme control, and of coding and decoding (see Section 2.3).

The modulating waveform is the sum of three components, the audio frequency M signal, the sidebands produced from the S signal by suppressed-carrier amplitude modulation, and the pilot-tone. Fig. 2.2 illustrates the build-up of the waveform starting with the A and B signals. In this instance the A signal is shown as lower in frequency and greater in amplitude than the B signal.

The relationship between the amplitudes of the components of the multiplex waveform is of interest particularly from the programme control aspect.

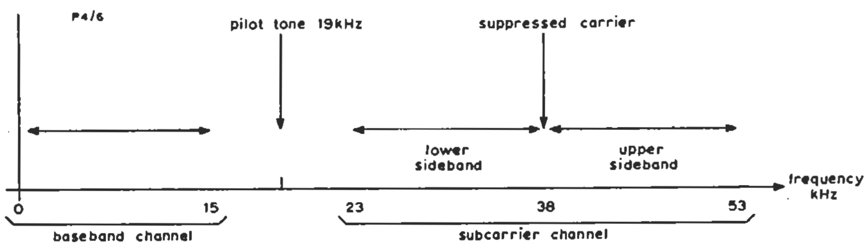


Fig. 2.1 Spectrum of Pilot-tone Multiplex Signal

The frequency-modulated carrier is transmitted in the normal manner and the complete multiplex signal is retrieved at the receiver discriminator. Because of the de-emphasis circuits preceding the a.f. chains in ordinary monophonic receivers, these respond only to the baseband channel and reproduce the compatible M signal, but stereophonic receivers are able to reproduce all the transmitted information.

In monophonic transmissions, the sub-carrier channel is unused and the pilot-tone is absent. The way in which stereophonic receivers reproduce such transmissions varies widely according to

The technique of programme control is explained in Section 4.

It is a feature of the system that the peak value of the multiplex signal (ignoring the effect of the pilot-tone) is determined by the peak value of whichever is the larger of the two signals A and B. This can be seen in Fig. 2.2, which shows that the composite waveform (vii) has the same peak value as (i), the larger of the two waveforms from which it was constructed. Overdeviation of the transmitter can therefore be prevented by controlling the levels of the individual A and B signals to below the level at which either alone would cause overdeviation.

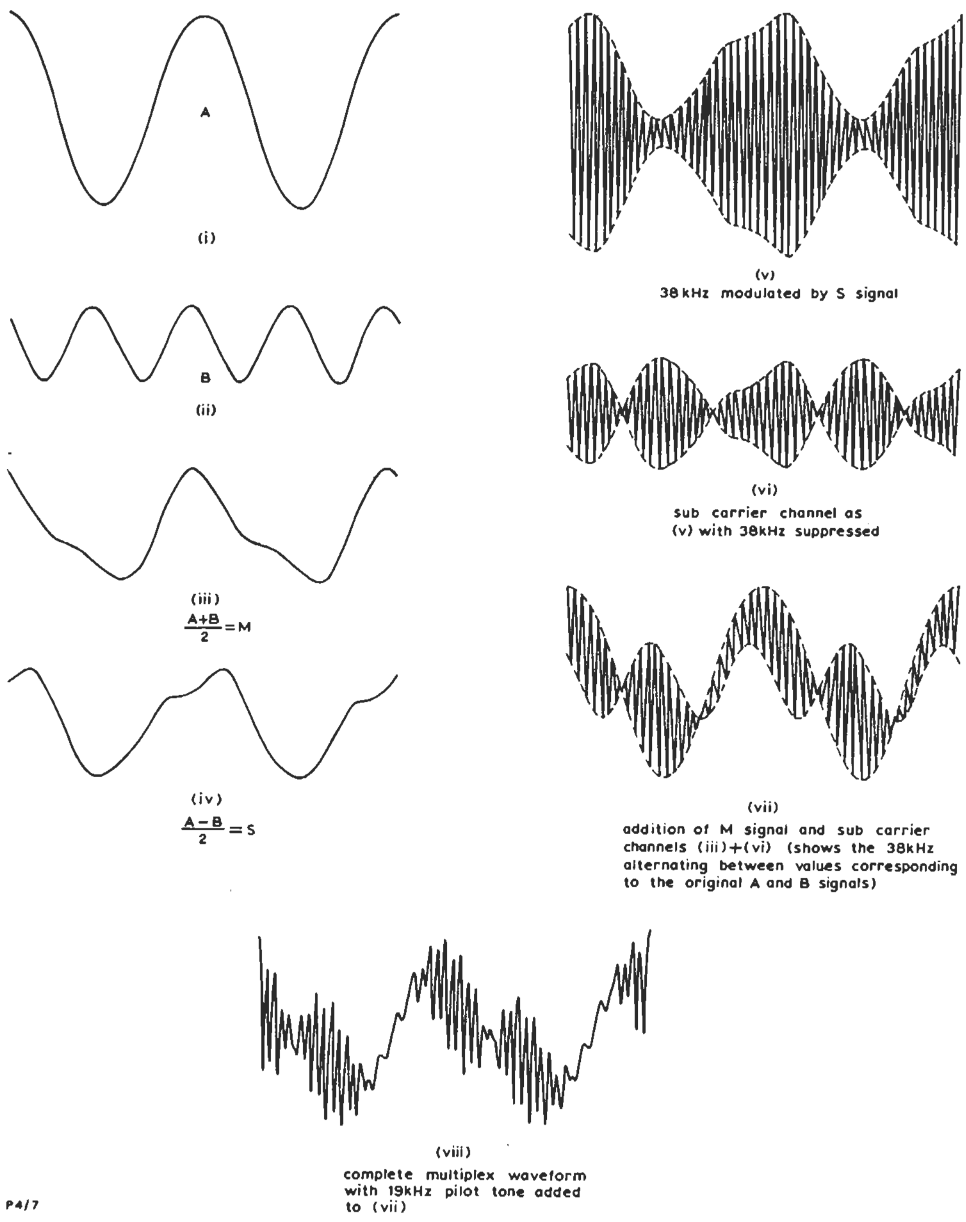


Fig. 2.2 Construction of Pilot-tone Multiplex Signal Waveform

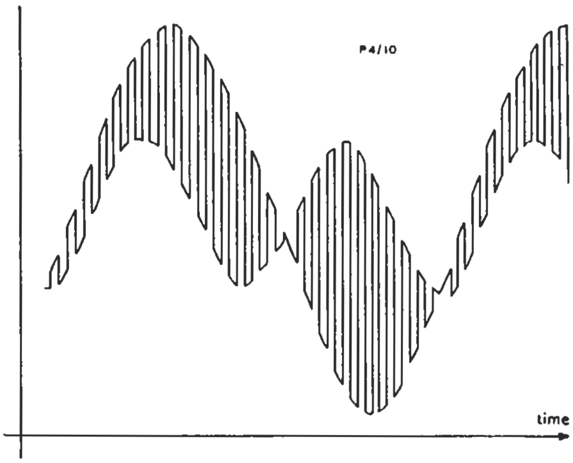


Fig. 2.3 Idealised 'Squared' Waveform of Pilot-tone Multiplex Signal. (Pilot-tone Not Shown)

2.3 Coding and Decoding by Switching Techniques

Fig. 2.2(vii) shows that the waveform alternates at the subcarrier frequency between values corresponding to *A* and *B*. Thus the multiplex signal (without the pilot-tone) can be generated directly from the *A* and *B* signals by applying them to a switch which operates at 38 kHz. Such a process would tend to produce a 'squared' waveform exemplified in Fig. 2.3 and to transmit this a bandwidth covering a number of harmonics of 38 kHz would be required. However, the presence of these harmonics is not essential, and if the signal is passed through a suitable low-pass filter, the harmonic content is removed and the wave becomes identical in form to that produced by the additive system described earlier. The pilot-tone at a specified level and with a specified phase relationship to the sub-carrier is added in the coder to produce the complete information required by the receiver.

The decoding circuits can retrieve the *A* and *B* information from the multiplex signal by the reverse process to that outlined for coding. The pilot-tone is used to control a switch, operating in synchronism with the coder switch at 38 kHz, to produce the *A* and *B* signals directly. The overall process is shown, much simplified, in Fig. 2.4.

2.4 Signal-to-noise Ratio

The use of the pilot-tone system for stereophonic transmission reduces the signal-to-noise ratio of the received programme below that obtainable for monophonic transmission in the Band-II f.m. services. The theoretical degradation, with compatible monophonic reception, is about 4 dB, but adjustments to the line-up at the transmitter are made to reduce the practical degradation to about 2 dB. This is a relatively insignificant change from normal and satisfies the compatibility requirement. For stereophonic reception the degradation is about 20 dB, but because of the excellent performance of the monophonic f.m. transmissions (which have potentially about 80 dB signal-to-noise ratio) this does not cause difficulties except at the fringe of the service area.

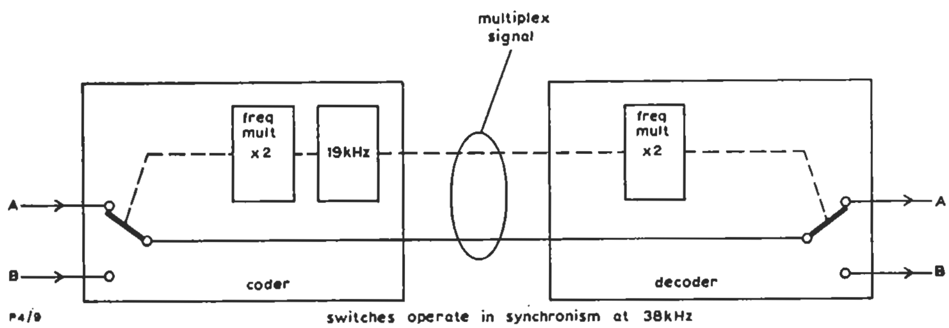


Fig. 2.4 Coding and Decoding Using Switches: Simplified Diagram

SECTION 3

MICROPHONE AND STUDIO TECHNIQUES

3.1 Introduction

The difficulties inherent in the use of spaced microphone systems have led to a preference for coincident microphones in BBC studio practice. In the following descriptions, therefore, emphasis is placed on coincident microphone systems. These may be supplemented in some instances by additional microphones to achieve the desired overall stereophonic effect, although in particular situations other arrangements are sometimes used.

3.2 Coincident Microphones

3.2.1 General

Two identical directional microphones mounted in the same housing and with their axes at an angle are said to form a *coincident pair*. In the examples which follow, the polar diagrams of the separate microphones indicate the responses from which the *A* and *B* signals are derived, and by addition and subtraction of these polar diagrams it is shown that in each instance the effective polar response which leads to the *M* signal is equivalent to that of a forwards-facing microphone, whereas the *S* signal is equivalent to the output of a sideways-facing figure-of-eight microphone.*

The compatible version of the output of a coincident pair is thus the same as the output of a single microphone the polar response of which is the same as that of the relevant *M* characteristic. The *S* characteristic usually leads to a high indirect-to-direct sound pick-up ratio, with the result that the stereophonic version of the programme contains a higher proportion of reverberation than the monophonic version.

The *M* and *S* responses of a coincident pair depend on the polar diagrams of the individual microphones and on the angle between them. It can be shown mathematically† that cross-mixing of the *A* and *B* outputs (or, alternatively, independent adjustment of the derived *M* and *S* signals) is equivalent to physical adjustment of the angle between the microphones. For practical convenience, therefore, most coincident pairs are set with 90 degrees between the individual units and adjustment of the characteristics is made as required by selection of polar diagrams and by electrical means.

The term 'useful angle', employed in the following examples, refers to the angle of the arc from which sounds can be accepted without introducing either peculiar positional effects in the reproduced programme or out-of-phase signals in the separate microphones of the coincident pair. Out-of-phase signals produce unpleasant results for the listener, because their source cannot be located at any point in space. It is important also to realise that although the useful angle can range up to 360 degrees, the reproduced sound stage is limited by the positions of the listener and loudspeakers to an arc of about 60 degrees. When the balance is planned, allowance must therefore be made for the possible resultant width compression.

The following details are indicative of commonly used polar-diagram configurations, but some microphones have a selection of responses with intermediate shapes.

A practical line-up procedure for a typical coincident pair is given in Appendix B.

3.2.2 Microphones with Crossed Figure-of-eight Characteristics

In Fig. 3.1, the polar diagrams of two figure-of-eight microphones which produce the *A* and *B* signals are shown in full and broken lines respectively. Sound from point X is reproduced only by the left loudspeaker and sound from point Y only by that on the right. Sound from point Z is reproduced equally by both and therefore appears to originate from the centre of the sound stage. Because of the symmetry of this arrangement, sounds from the rear of the combination are treated similarly, but there is a left-to-right inversion.

There are thus two useful angles each of 90 degrees and they are bounded by the positions of the dead axes of the two microphones. Sources outside the useful angles produce out-of-phase signals

* Some organisations use microphones to produce the *M* and *S* signals directly and derive the *A* and *B* signals from these. The *M* and *S* signals produced by this method show considerable similarities to those produced by coincident pairs, but problems of matching the frequency responses of microphones with dissimilar polar diagrams are introduced. These problems usually outweigh the advantages offered, and the direct *M* and *S* system is not used in BBC practice. (See also the footnote to page 1.4.)

† BBC Engineering Division Monograph No. 38, Appendix IV.

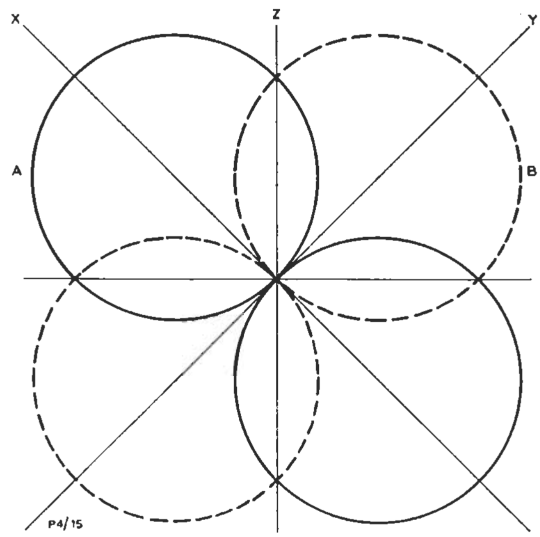


Fig. 3.1 Crossed Figure-of-eight Coincident Microphones: Polar Diagram

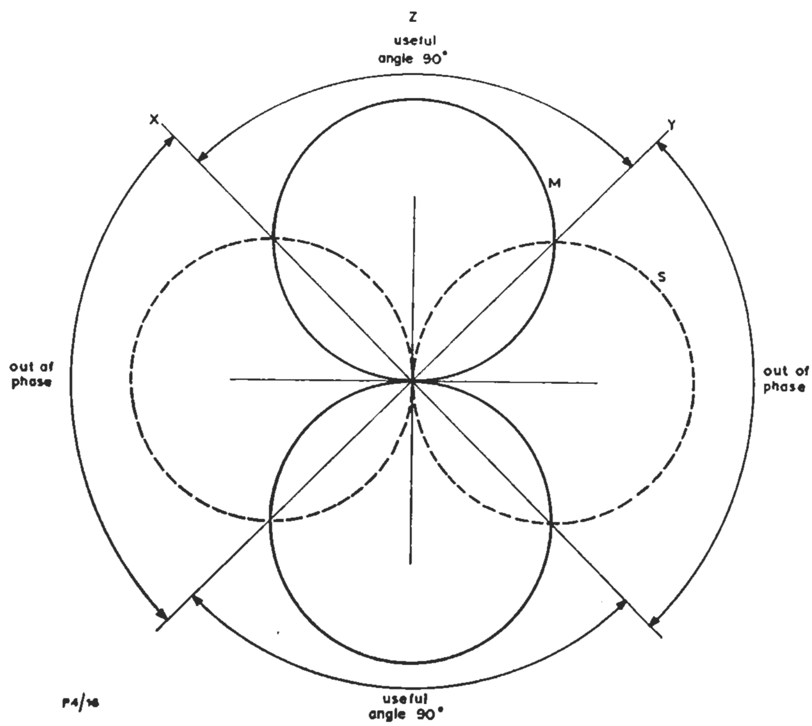


Fig. 3.2 Crossed Figure-of-eight Coincident Microphones: M and S Characteristics

because they are picked up by the front of one microphone and the back of the other.

Combining the microphone polar ordinates to obtain $(A + B)$ and $(A - B)$ gives the result shown in Fig. 3.2. The full line shows the polar diagram for the M signal and the broken line shows that for the S signal. The M output (which would be obtained from a monophonic receiver) is theoretically the same as that from a single forward-facing figure-of-eight microphone.

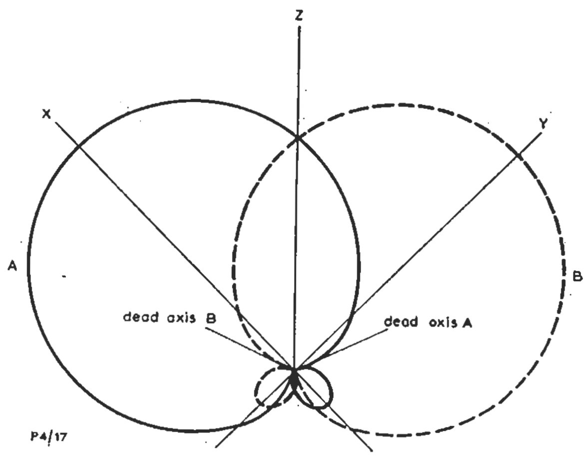


Fig. 3.3 Crossed Cottage-loaf Coincident Microphones: Polar Diagram

3.2.3 Microphones with Crossed Cottage-loaf (Hypercardioid) Characteristics

This is a very common arrangement in coincident microphone technique. The individual polar diagrams shown in Fig. 3.3 combine to produce the M and S characteristics indicated by Fig. 3.4. As in the previous example, the limits of the useful angle are determined by the positions of the dead axes. For microphones with cottage-loaf characteristics, the dead axes are about 110 degrees from the live axis, and with the two units at 90 degrees this produces a forward-facing useful angle of about 130 degrees as shown in Fig. 3.4. The small in-phase lobe at the rear of the combination is of little practical value.

3.2.4 Microphones with Crossed Cardioid Characteristics

The polar diagrams of the individual microphones are shown in Fig. 3.5 and the M and S polar diagrams are shown in Fig. 3.6. The useful angle of this coincident pair is about 180 degrees, although

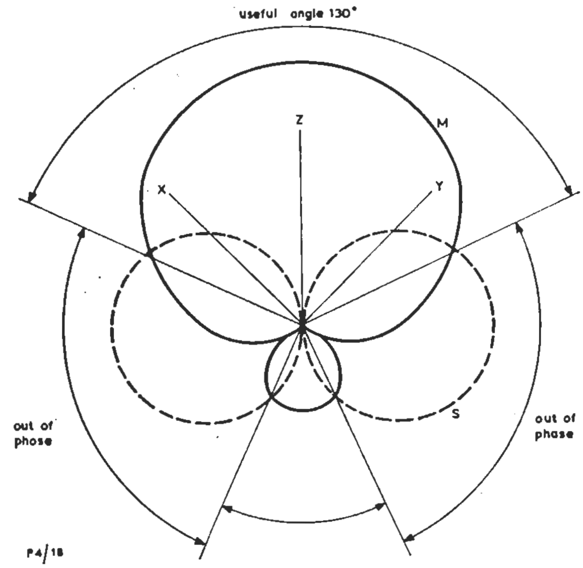


Fig. 3.4 Crossed Cottage-loaf Coincident Microphones: M and S Characteristics

sounds from an extra 45 degrees each side can be accepted and are reproduced at the extremes of the stage. The effect of sources in the remaining 90-degree arc is in practice unpredictable, because the response in this region is determined by the normally unimportant rear sensitivities of the separate microphones.

A coincident pair of cardioid microphones does not have an out-of-phase region, except possibly at the extremes of the frequency range where differences between the polar responses of the individual microphones may occur.

*3.2.5 Cardioid Microphones Back-to-back**

This formation is an exception to the usual 90-degree angle between the microphone axes. As shown by Fig. 3.7, the arrangement gives an omnidirectional characteristic for the M signal and the usual sideways-facing figure-of-eight for the S signal.

The characteristics of this arrangement give an advantage over the coincident pairs described earlier, in that the impression of direct and indirect

* Microphones should not be mounted physically back-to-back, because reflections may occur between the rear faces of the separate units, and in practice either a side-by-side or one-above-the-other arrangement is preferable.

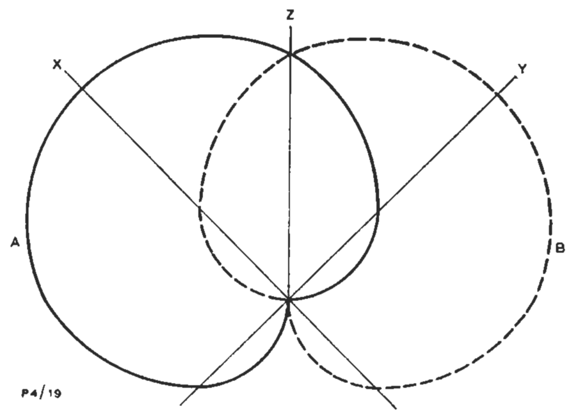


Fig. 3.5 Crossed Cardioid Coincident Microphones: Polar Diagram

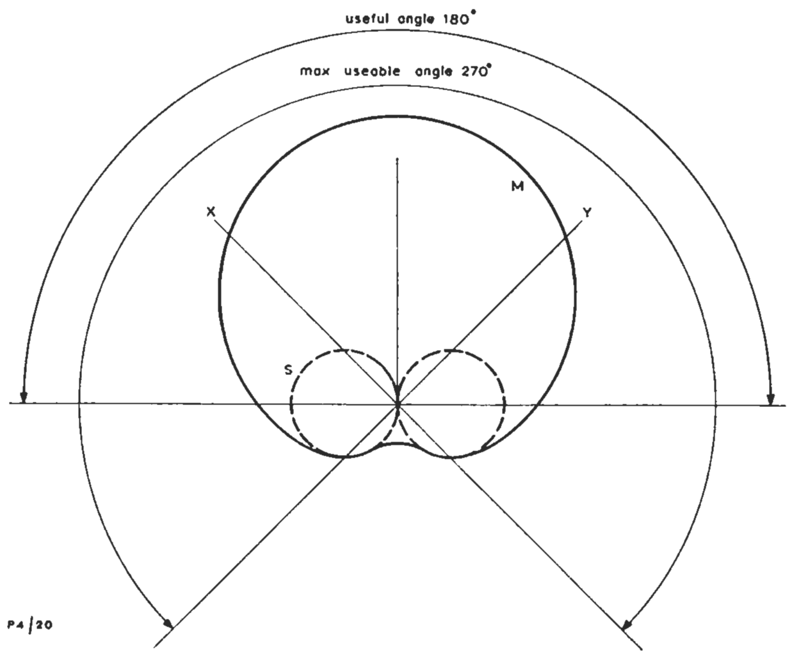


Fig. 3.6 Crossed Cardioid Coincident Microphones: M and S Characteristics

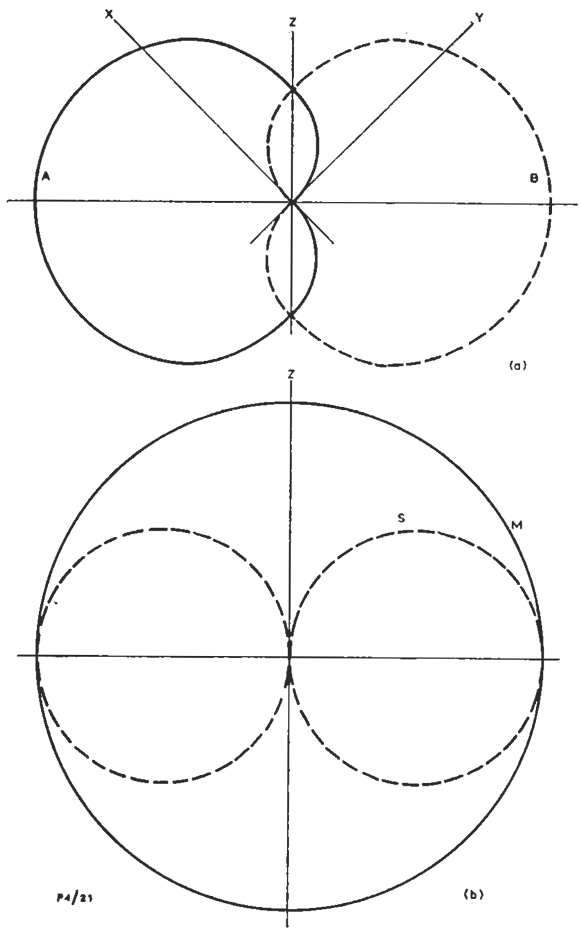


Fig. 3.7 Back-to-back Cardioid Microphones: (a) Polar Diagram and (b) M and S Characteristics

sound is roughly the same for listeners to both the stereophonic and the monophonic versions of the programme. However, careful disposal of the artists with respect to the microphones is necessary to avoid peculiar positional effects on stereophonic reproduction.

Some microphones have responses which are more directional at high frequencies than at low frequencies. This may give poor high-frequency performance at the centre of the stage, with consequential degradation of quality for all listeners and dispersion (blurring) of off-centre stereophonic images.

3.3 Balance

3.3.1 General

In the following account, some of the problems met in providing a correctly-balanced stereophonic programme are indicated, and examples of techniques to overcome these problems are outlined.

When a programme is being transmitted monophonically, the aims are to reproduce the internal balance of the performance, and to maintain the correct perspective, given by impressions of space, presence, clarity and so on. These aims can be achieved by:

(a) choosing a microphone with particular directional characteristics,
(b) tilting the microphone to modify the effect of its characteristics,
(c) adjusting the height and distance of the microphone from the performers.

In stereophony the scale-of-width is an additional factor. The sound stage (the distance between the loudspeakers) is between about 6 and 12 feet wide. A full orchestra is usually required to almost fill this stage as shown in Fig. 3.8. The reverberation must be such that the orchestra appears to be sufficiently far behind the vertical plane through the loudspeakers to be realistic in scale.

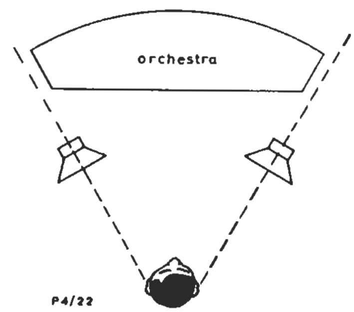


Fig. 3.8 Reproduction of Orchestra in Correct Scale-of-width

Suppose a crossed figure-of-eight coincident pair is to be used and the foregoing requirements must be met. The forward useful angle is 90 degrees and the orchestra must subtend this angle at the microphone to fill the stage. The coincident pair is then as close as it can be to the orchestra without introducing out-of-phase effects, but either the balance or the perspective or both may be wrong.

There may be excessive reverberation. In monophony the microphone could be moved closer to the orchestra but, as has just been explained, this would cause out-of-phase effects in stereophony. A solution is to use a coincident pair with crossed cottage-loaf characteristics. The wider angle allows the pair to be moved closer to the orchestra and at the same time the reduction of sensitivity at the rear of the microphones results in less reverberation being picked up.

If there is too little reverberation, the microphone can be moved away from the orchestra, which then appears to be narrower. This is not necessarily bad, because reverberation still fills the whole of the sound stage. Reverberation is often added to the balance of a coincident pair by using the output of an additional pair of microphones (not necessarily coincident) which are placed at the rear of or high up in the auditorium.

By comparison with an orchestra, a small group, such as a string quartet, gives rise to slightly different scale-of-width considerations. Fig. 3.9 shows two ways in which such a group could be made to appear to the listener; (a) is used to give the impression that the quartet is in a concert hall, whereas (b) gives some impression that the quartet is in the listening room.

Whatever the nature of the broadcast, however, it is always important that some thought should be given to the relationship between width of image and apparent distance of image from listener. There are three methods of controlling image width:

1. by varying the microphone polar diagram,
2. by varying the microphone position, and
3. by the use of electrical systems described in Section 4.

With methods 1 and 3, action to narrow the image reduces the accompanying reverberation, whereas with method 2 such action increases reverberation.

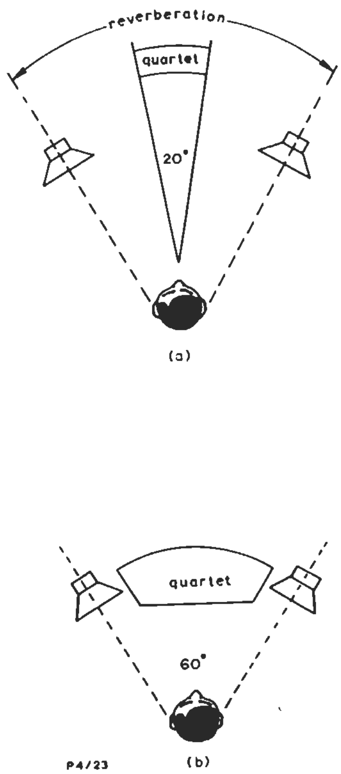


Fig. 3.9 Two Methods of Presenting a Quartet

3.3.2 Close-microphone Techniques

It is not always possible to obtain a satisfactory balance using a single coincident pair, even with sources which are apparently balanced within themselves. Close-microphone (or 'spotting') techniques can be combined with either the spaced or the coincident system to produce good results, but great care is needed to ensure that conflicting positional information is not produced by the various microphones.

A common technique is to employ a single coincident pair to establish the overall positional picture and then to augment its outputs with those from a number of single spotting microphones. The outputs of these microphones are electrically adjusted to give positional agreement with the overall picture.

When a spotting microphone is used in this way, any reverberation it picks up is not spread across the reproduced sound stage, but appears from the same place as the direct sound. This leads to a 'tunnel' effect which can be avoided by using very close techniques and adding any necessary reverberation by other methods.

Suppose that a single spotting microphone is to be used for a soloist who is standing off-centre. The output of this monophonic microphone, if divided equally into the A and B channels, would conflict with the outputs of the coincident pair and tend to make the soloist appear to be at the centre of the stage. The output of the single microphone must therefore be split in the correct proportions for feeding the two channels in agreement with the outputs of the coincident pair. This is known as 'steering' or 'panning'.

The problem is worse if two or more spotting microphones are needed, because they can effectively form one or more stereophonic pairs and produce positional information which usually disagrees with that from the main microphones.

In addition, adjustment of the levels of any of the spotting microphones alters the spurious positional information and makes electrical compensation impossible. The spotting microphones must therefore be as close as possible to the source they are intended to augment and, by careful positioning, screening and choice of polar diagram, they must be arranged to pick up the minimum possible sound from other sources.

A multiple source, such as a group, which has a definite width but requires support, cannot be handled by a single spotting microphone, because the output would be too narrow to fit the overall picture. Instead, a spaced or coincident pair must be used and its outputs must be arranged to simulate the correct width with the individual sources in their correct positions.

In general, the output of any spotting microphones used must agree with and reinforce the output of the main microphones unless some special effect is desired.

3.4 Deliberate Positional Changes

The limitations on the shape of the sound stage imposed by the two-loudspeaker formation sometimes require the establishment of positional information which does not agree with the physical layout of studio performances. It may also be desirable to introduce deliberate positional changes, because listeners often expect a particular arrangement of performers which cannot be achieved in the studio.

Suppose a choir and orchestra were positioned as shown in Fig. 3.10(a), using a single coincident microphone pair. Such an arrangement would give unsatisfactory results, as indicated in Fig. 3.10(b), and it might be better to simulate the more usual arrangement, shown in Fig. 3.10(c). This result could be achieved as follows:—

Sound from the orchestra is picked up by a coincident pair in the usual way as shown in Fig. 3.10(d). Additional microphones for the choir are placed at P and Q and the output of P, which inevitably contains pick-up from the left-hand side of the orchestra, is steered towards the left of the sound stage where it agrees with the outputs of the coincident microphones. The output of the microphone at Q, which picks up less of the orchestra, is steered towards the right of the stage and thus the overall effect of Fig. 3.10(e) is obtained.

It is generally desirable to establish positional information by use of a coincident pair, but there are some exceptions. For example, at a recording session with a modern dance orchestra, the musicians may have to be placed in a layout which, because of acoustic considerations and limitations of space, disagrees with their normal layout. In this instance, a very close microphone technique could be employed, and both a balance and the impression of a particular layout could be achieved by electrical means.

In introducing positional changes, care must be exercised to avoid making the sound stage appear too 'flat', that is, with too little depth from front to back. With a choir and orchestra such as considered earlier, for example, it would be wrong to make the choir and the orchestra appear to occupy the same depth plane.

3.5 Artificial Reverberation

In making a stereophonic balance, it is usually necessary to add reverberation, especially with close microphone techniques. The reverberation must be in stereophonic form, and as mentioned previously, is often obtained from a pair of microphones in the concert hall. Artificial means such as echo rooms, plates and springs can be employed as in monophonic productions, springs being particularly successful, but in all instances the output must be stereophonic.

3.6 Artistic Compatibility

Most of the audience is likely to be listening to the monophonic version ($A + B$) of a transmitted stereophonic programme and it is important to achieve artistic compatibility as well as technical compatibility between the two ways of listening to the same material. This makes careful consideration necessary to several features which produce effects that are either deliberate or unavoidable. Monitoring of both versions of the programme is therefore advisable to ensure that each is satisfactory. The main characteristics for which a noticeable difference exists are

(a) loudness and reverberation,
(b) aural selectivity, and
(c) extraneous noises.

(*a*) *Loudness and Reverberation*

A stereophonic receiver reproduces all the information from the M and S signals, whereas a monophonic installation is responsive only to the M signal. The M and S responses of the crossed figure-of-eight coincident pair (Fig. 3.2) illustrate the division of information from an off-centre source at X or Y into the M and S channels.

A source at the centre of the stage does not

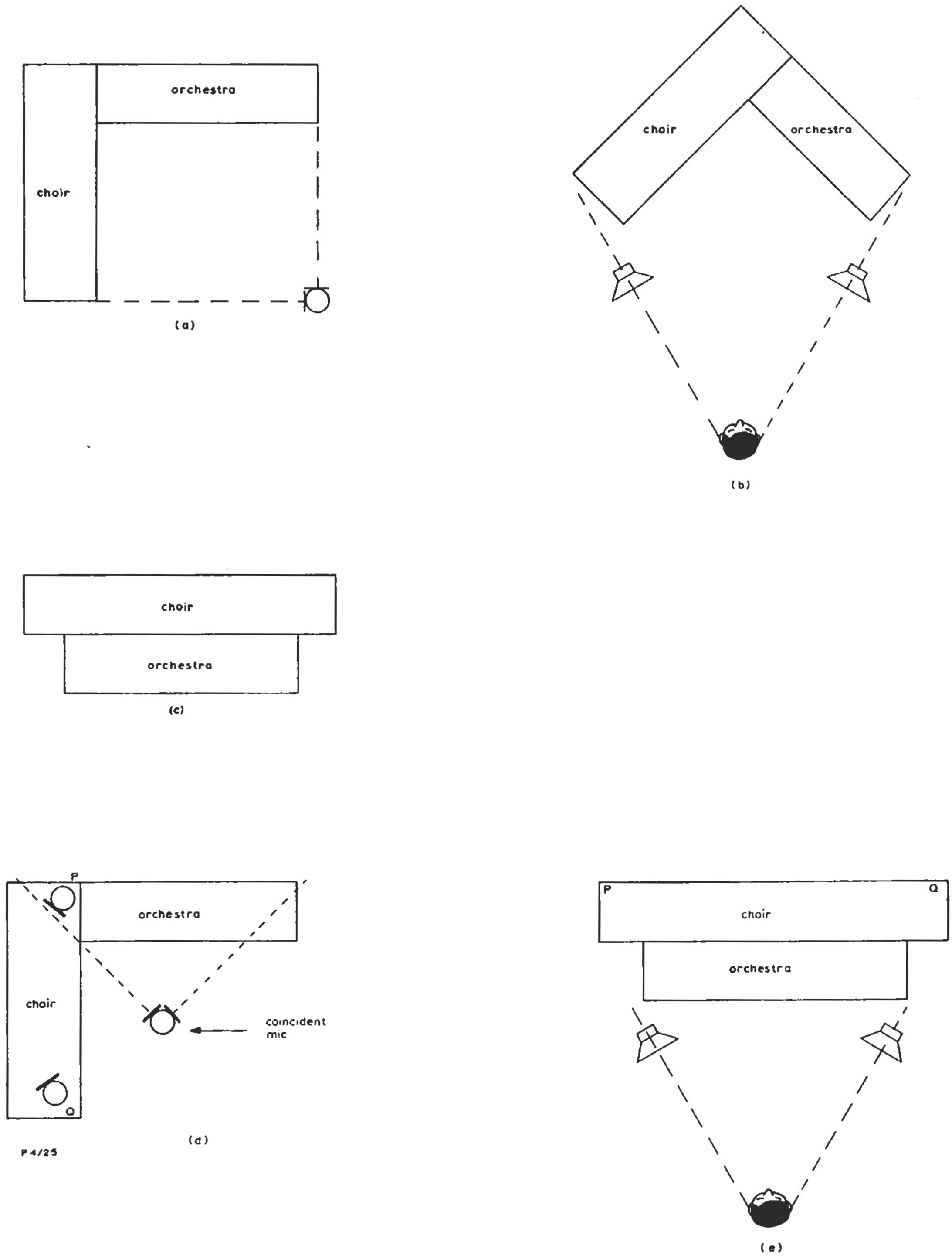


Fig. 3.10 Illustration of Deliberate Positional Changes Produced by Microphone Placing

contribute to the S channel and its total effect is reproduced equally in both versions of the programme. If the source moves off-centre, its output is divided between the M and S channels and is reproduced completely by stereophonic equipment, whereas in monophonic reproduction the S channel is lost and central sources can thus appear over-prominent. With spaced microphone systems, there is an additional complication, because only central sources produce synchronous signals in the A and B channels. The resultant phase differences with off-centre sources can lead to peculiar effects because of partial additions and cancellations at different frequencies.

The S signal usually carries much of the reverberance information and this is not therefore available in the monophonic version of a transmission which can thus appear to be closer (or 'drier.') than the stereophonic version. As a partial solution to this problem, it may be possible to find a compromise balance and an improvement may perhaps also be obtained by introducing additional central echo; sometimes, back-to-back cardioid microphones can be used, if off-stage noises are acceptable and problems of poor high-frequency response on the central axis are not introduced.

(*b*) *Aural Selectivity*

With stereophonic reproduction many discrete sounds can exist simultaneously, as in ordinary surroundings, and a listener can exert a degree of selectivity by concentrating on a sound from a particular direction. This facility of directional selectivity is often called the 'cocktail party effect'. With monophonic reproduction, however, all the sounds come from one place and the facility is therefore unusable. For this reason, important sounds must sometimes be deliberately accentuated for the benefit of listeners to the monophonic version, and monophonic (as well as stereophonic) monitoring is essential to ensure compatability.

The listener to the stereophonic version can be confused by the effect of positional changes which may pass unnoticed not only in monophonic monitoring but also in stereophonic monitoring where there is visual contact with the performance. For example, a character in a dramatic production may cross from one extreme of the stage to the other between successive speeches. In monophony this would not create any difficulty, and the stereophonic sound stage would also agree with the visual information imparted at the monitoring point. However, the sudden transposition apparent to the remote listener to the stereophonic version might be disturbing and it would be better to convey the impression of the character's movements by some means such as audible footsteps during the interval between his speeches.

(*c*) *Extraneous Noises*

The effect of extraneous noises is potentially much more annoying to listeners with a stereophonic installation because such noises can be clearly defined spatially and may distract attention from the area of action. This can be very noticeable when a stage performance is broadcast and theatre conditions require microphones near floor level. In particular the foot noises of a performer standing close to a coincident pair can, because the two microphones are mounted one above the other, produce time differences in the outputs. This leads to positional shift in the reproduced sound stage with the artist's foot apparently situated to one side of his mouth.

SECTION 4

PROGRAMME CONTROL

4.1 General

The apparatus and methods used in studios for the control and monitoring of stereophonic programmes are in most respects similar to those employed for monophony but with the addition of facilities peculiar to stereophonic working. These additional facilities include provision for control of the width of the reproduced sound stage, the positioning of particular sources and the derivation of artificial reverberation in stereophonic form.

Control desks for stereophonic programmes usually contain both stereophonic channels and monophonic channels, and pairs of the monophonic channels can often be used together as additional stereophonic channels. The following descriptions exemplify arrangements which may be met in practice although none may correspond exactly to a particular desk in use.

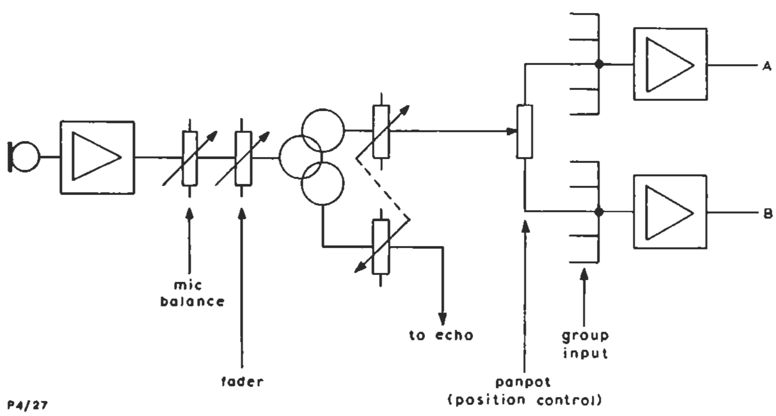


Fig. 4.1 Monophonic Channel Adapted for Stereophonic Working: Simplified Diagram

4.2 Monophonic Input Channels

A monophonic channel of a stereophonic control desk accepts the output of a single microphone and feeds it, after channel control, to the *A* and *B* group inputs in the proportions required to place the reproduced image correctly on the sound stage.

Such a channel is shown in Fig. 4.1. It is operated like a conventional monophonic channel up to the input of the differential fader, or *panpot* (an abbreviation for 'panoramic potentiometer'). One form of this control comprises ganged variable attenuators which operate in opposition as shown in Fig. 4.2. The input signal is applied to both attenuators, and when the control is centred these feed the signal equally into the *A* and *B* channels. This results in the reproduced image being positioned at stage centre. Movement of the control to off-centre alters the proportions of the signal fed to the *A* and *B* channels and the image can thus be positioned as required. A monophonic channel of this type is used for the spotting microphone technique described in Section 3.

Two such monophonic channels can be associated as shown in Fig. 4.3 to produce a stereophonic channel; their inputs must then be the *A* and *B* outputs of either a spaced pair or a coincident pair of microphones. To prevent unbalance during operation, the two individual channel faders must be operated together, and physically-adjacent channels with the faders mechanically connected are normally used. Clips are provided to connect the control knobs of quadrant faders.

The panpots provide control of both position and width. With both panpots at the same setting, the image has no width but can be at any position on the stage dependent on the particular setting. Full

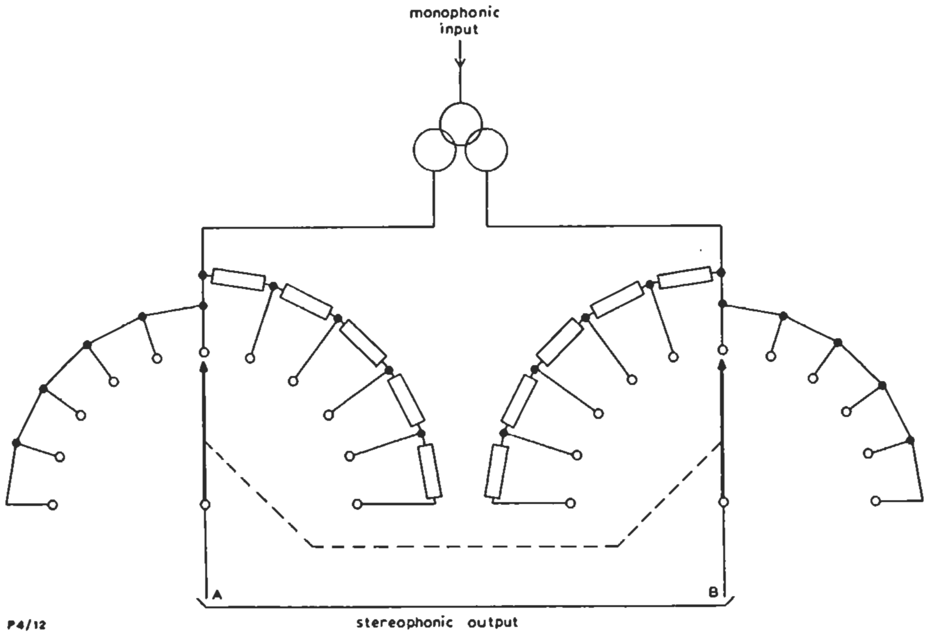


Fig. 4.2 Simplified Diagram of a Typical Panpot

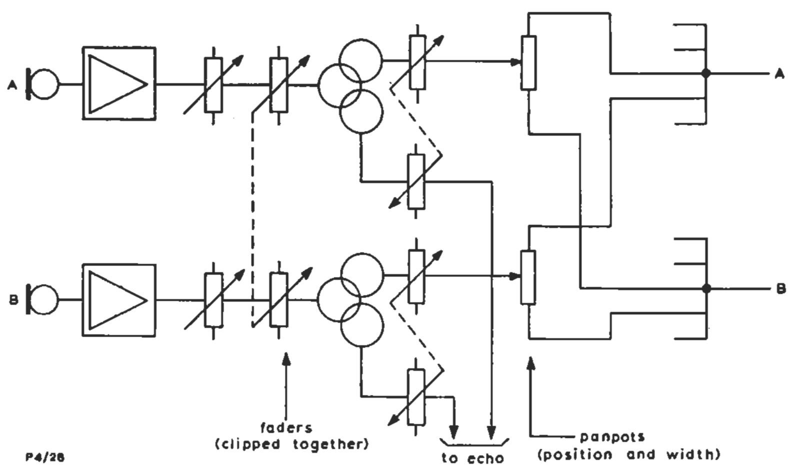


Fig. 4.3 Two Monophonic Channels used together as a Stereophonic Channel

width is achieved by steering the two signals entirely to their respective channels; that is by setting the panpots to opposite extremes of their travels. This arrangement is used on some desks as the exclusive means of providing stereophonic channels.

(d) *Channel Position Offset Control.* These reverse-ganged attenuators allow a change of balance to be introduced which shifts the reproduced images across the stage. The usual limit is an unbalance of about 9 dB. The degree of offset

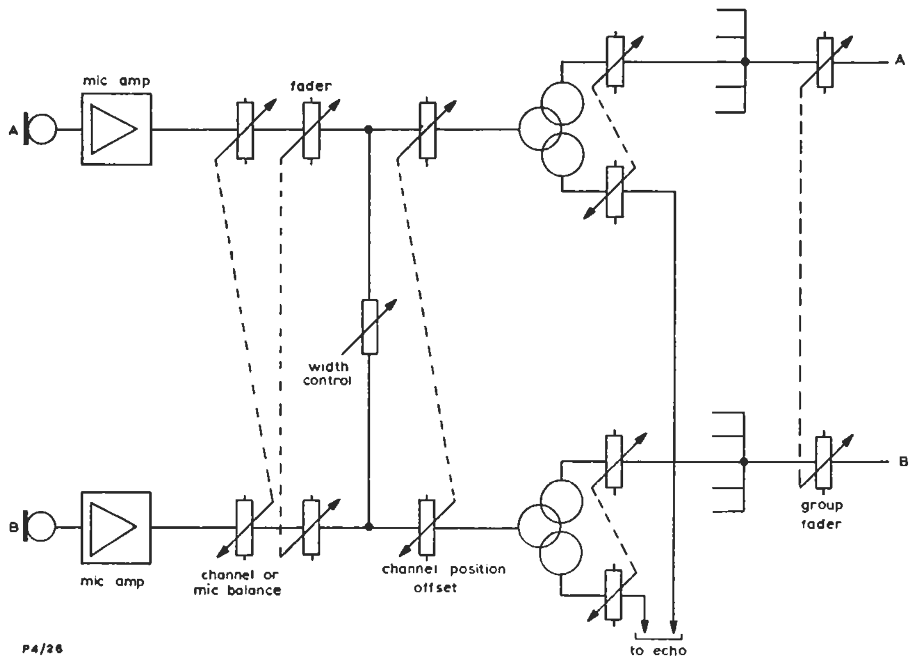


Fig. 4.4 Stereophonic Channel

4.3 Stereophonic Input Channels

The arrangement shown in Fig. 4.4 is typical of an exclusively stereophonic channel. Its main features are:

(a) *Microphone or Channel Balance Control.* Used in technical line-up of the equipment and must not be adjusted subsequently.

(b) *Channel Fader.* A permanently-ganged pair of faders which operate on both signals simultaneously.

(c) *Width Control.* In this example a reduction of attenuation between the *A* and *B* channels introduces cross-mixing which results in a narrowing of the reproduced sound stage. Widening of the image is not possible in this instance but could be effected to some extent by introducing a phase-reversal into the cross-connection between the signals.

is dependent also on the position of the width control.

(e) *Echo Arrangements.* The echo is taken separately from the *A* and *B* channels in this example and returns to the group inputs in the normal way.

(f) *Group Fader.* Two or more groups may be provided and channels may be selected to these in the conventional manner.

In addition to the limitations on the control of width, mentioned in (c), the arrangement suffers from a disadvantage resulting from the extreme difficulty of ensuring that the wipers of a ganged pair of stud faders both move between studs at exactly the same time. Any discrepancies tend to cause the reproduced image to jump from side to side during a fade, an effect known as 'flicker' or 'fader wiggle'.

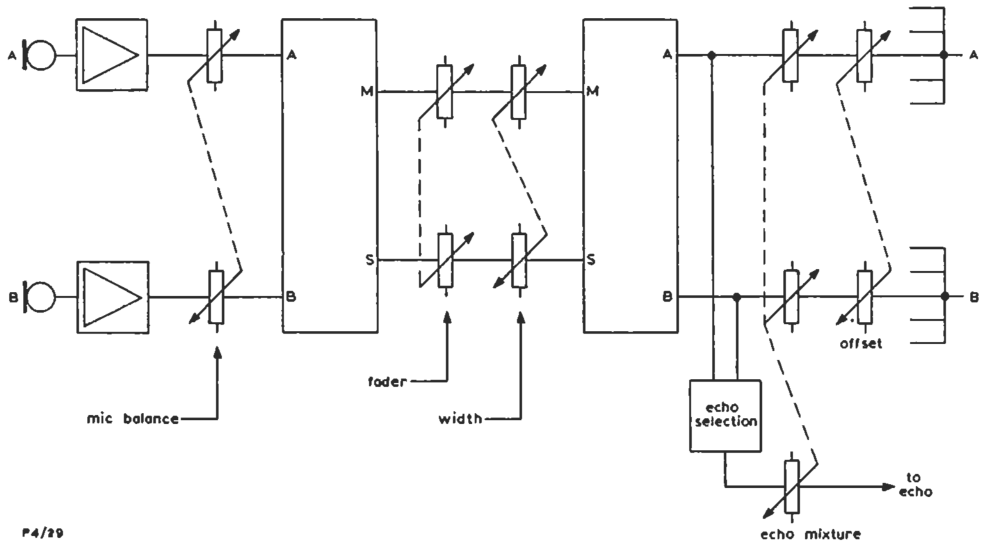


Fig. 4.5 Stereophonic Channel with Control on the M and S Signals

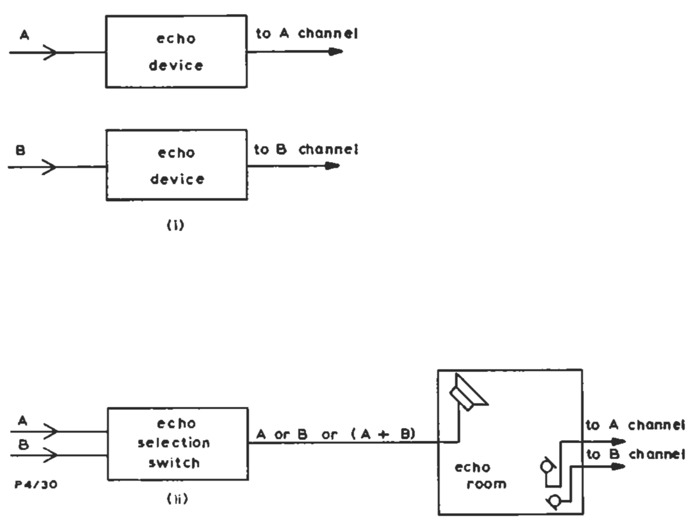


Fig. 4.6 Two Methods of Deriving Stereophonic Artificial Reverberation

As an alternative to the arrangement shown, fading and width control can be exercised on the *M* and *S* signals; this overcomes the disadvantage mentioned above. Fig. 4.5 typifies such an arrangement. The image can be narrowed by reducing the ratio of *S* to *M* signals and some widening can be achieved by increasing this ratio. Fader wiggle does not occur with this arrangement but the effects of fader imperfections appear as slight variations in width which are less objectionable than fader wiggle.

4.4 Artificial Reverberation

There are many possible methods of deriving artificial reverberation from the stereophonic signals; the choice depends largely on the programme material and available facilities. As mentioned in Section 3.5, the reverberation must be presented in stereophonic form as separate signals to the *A* and *B* channels.

Fig. 4.6 exemplifies two of the methods used. In (i) the separate *A* and *B* signals are processed independently; this arrangement might be used with the type of desk circuit shown in Fig. 4.4. Method (ii) could be used with the desk arrangement of Fig. 4.5; in this instance the 'echo go' signal can be *A*, or *B*, or ($A + B$), and is determined by the position of the echo selection switch. The usual arrangement for serious music is to use an ($A + B$) input to a single device with independent outputs.

4.5 Output Arrangements

Stereophonic control desks provide simultaneous outputs in both stereophonic and monophonic form. The stereophonic output comprises the separate *A* and *B* signals; the monophonic output is the ($A + B$) signal, which is fed to areas incapable of handling stereophony. Fig. 4.7 is a simplified diagram of a typical output arrangement which gives stereophonic and monophonic outputs, both with and without announcements.

Visual monitoring facilities vary between desks. Normally at least two P.P.M.s are provided, one with a double movement showing the *A* and *B* outputs independently, and the other with a standard single movement showing the ($A + B$) output. The ($A + B$) P.P.M. does not necessarily measure across the ($A + B$) monophonic output, but may derive its input from the separate *A* and *B* signals in the monitoring circuits. Sometimes monitor selection arrangements are provided, either to switch various monitoring points to the main P.P.M.s mentioned above, or to switch these points to a third, subsidiary, meter. These provisions may allow the monitoring of clean feed, main output, tape replay and so on. In some instances the facility to monitor the *S* signal, ($A - B$), is available also.

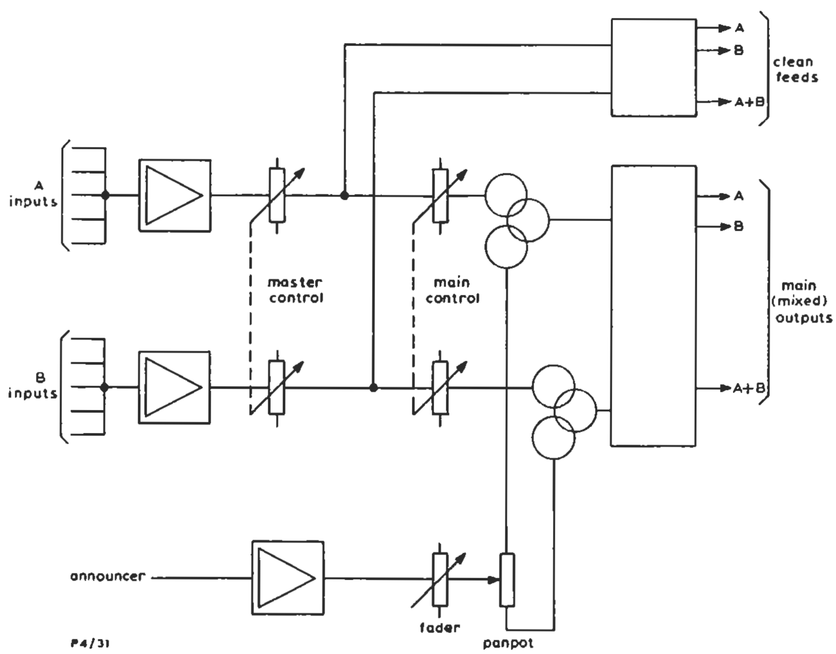


Fig. 4.7 Control Desk Output Arrangements

SECTION 5

STUDIO LINE-UP AND PROGRAMME LEVELS

The existence of two versions of a stereophonically-produced programme can lead to difficulties in monitoring because at some places, where the full signal is available, the A and B signals can be monitored individually whereas at others the $(A + B)$ signal* only is available. The problems arise because the A and B signals can be either identical or non-identical. When the A and B signals are identical, so that $A \equiv B$, the peak level of the $(A + B)$ signal (whether tone or programme) is greater by 6 dB than that of either A or B alone. When the two signals are not identical, the peak level of the $(A + B)$ signal can be from 0 to 6 dB greater than that of either A or B alone, depending upon the degree of correlation. For practical purposes the average level of $(A + B)$ is taken to be 3 dB above that of A or B. In the special instances where the source is at an extreme of the stereophonic stage, only the A or the B signal is present and the $(A + B)$ signal therefore is at the same level as that signal. Thus there can be differences of up to 6 dB between the levels indicated at the stereophonic and monophonic monitoring points.

A method adopted to minimise the difficulties is to adjust the gain of the outputs so that for identical signals the $(A + B)$ level is 3 dB greater than the level of either A or B alone. This means that for most of the time, with normal programme signals, stereophonic and monophonic monitoring points have substantially the same indications and the total potential discrepancy of 6 dB is thus divided about equally between the stereophonic and monophonic signals. Normal line-up procedure is to use identical signals comprising line-up tone at −3 dB in phase on both channels and this results in the $(A + B)$ output, after adjustment, being at zero level so that observers in non-stereophonic areas need not be aware that the signal is derived from a stereophonic source.

Table 5.1 gives a summary of the levels for various conditions. Two examples of identical signals are tone applied to the input of both channels in parallel, and exactly central speech. Normal stereophonic material, however, is in general of non-identical form.

* It was mentioned in 1.4.2 that, although the M signal is theoretically $\frac{1}{2}(A + B)$, the division by two is often ignored. This is especially so in programme control areas where the levels of all the signals can usually be adjusted independently and it is therefore convenient to regard the sum signal as a simple addition of A and B.

TABLE 5.1
LEVELS MEASURED WITH 3-dB GAIN DIFFERENCE BETWEEN STEREOPHONIC AND MONOPHONIC OUTPUTS

Nature of Signals	*Level Measured on P.P.M.*		
	A	B	$A + B$
Identical	zero −3 dB +5 dB	zero −3 dB +5 dB	+3 dB zero +8 dB
Non-identical	+8 dB +11 dB *nil*	+8 dB *nil* +11 dB	+8* dB +8 dB +8 dB

* This level is +8 dB on average, but can vary from +5 to +11 dB.

Monitoring should, in general, be mainly with respect to the $(A + B)$ signal because it is important to avoid changes in loudness for listeners to the monophonic version of the programme. This gives rise to some difficulty because, as can be seen from Table 5.1, a source entirely at one side of the stage can peak to +11 dB without any indication of excess level on the $(A + B)$ P.P.M. It was shown in Section 2.2 that the peak value of the multiplex signal which modulates the transmitter is determined by the peak value of whichever is the larger of the two signals A and B. A source predominantly to one side may therefore cause operation of limiters which precede the coder and it is thus necessary, at the programme control point, to pay particular attention to the A and B P.P.M. indications when such conditions are likely to occur.

The $(A - B)$ signal has little practical value in monitoring of programme material but is frequently used as a check of channel balance. Because identical signals should not produce any $(A - B)$ output, the channels can be balanced by adjusting for a null whilst listening to the $(A - B)$ signal. This gives a more accurate balance than adjusting for equality by direct measurement.

SECTION 6

DISK AND TAPE RECORDINGS

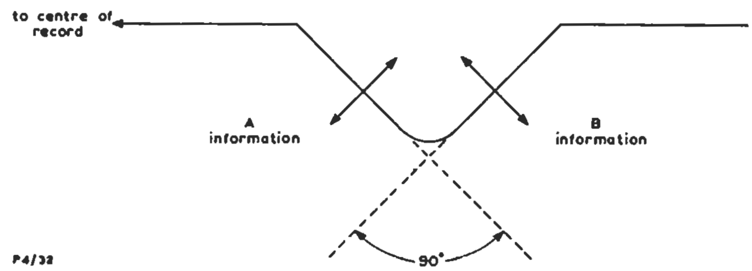


Fig. 6.1 Directions of Modulation in Stereophonic Disk Recording

6.1 Disk Recording and Reproduction

6.1.1 General

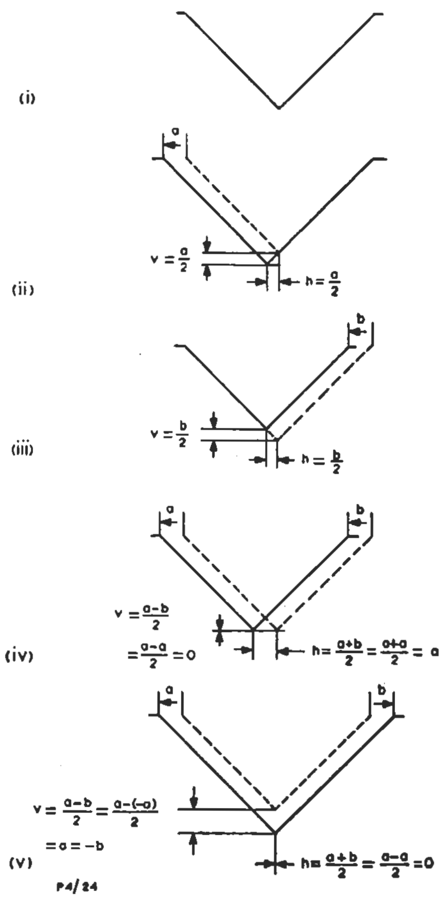
A monophonic recording has a V-shaped groove which is laterally modulated to correspond to a single audio-frequency signal. A stereophonic recording is required to carry information corresponding to two audio-frequency signals, *A* and *B*, and the two sets of information are cut independently into the two walls of the groove as indicated in Fig. 6.1. The effect of this double modulation in a stereophonic recording is to vary the groove both horizontally and vertically, whereas in a monophonic recording the groove has a constant depth if second-order effects are ignored. The simplified diagrams* given in Fig. 6.2 show that the horizontal and vertical groove displacements in stereophonic recording are analogous to the *M*, or $(A + B)$, and the *S*, or $(A - B)$, signal respectively.

A pickup used for stereophonic reproduction must be capable of converting the simultaneous horizontal and vertical stylus motions into separate electrical outputs, whereas a pickup used for monophonic reproduction should only produce an electrical output when the stylus moves horizontally. The extraction of the stereophonic information requires a stylus tip of smaller radius than is necessary for monophonic reproduction. Because the intensity of pressure exerted on the record is inversely proportional to the square of the tip radius, the maximum playing weight permissible for stereophonic records is smaller than for monophonic ones.

* In these very simplified diagrams, *a* and *b* represent the horizontal components of the modulation on the left and right walls of the groove.

Typical figures are 2-3 gm playing weight and 0·0125 mm (0·5 mil) tip radius for stereophonic equipment compared with 8 gm and 0·025 mm (1 mil) for monophonic equipment. For use with both monophonic and stereophonic records a so-called elliptical stylus is often used; it is of approximately elliptical cross-section made by grinding major and minor radii of curvature of about 0·0175 mm (0·7 mil) and 0·0075 mm (0·3 mil) and mounted with the major axis at right angles to the groove. The smaller radius at the point of contact with the groove wall permits tracking of high frequencies at high modulation levels while the larger radius across the groove prevents the tip of the stylus from riding too low in the groove. However there are many problems inherent in the manufacture and mounting of elliptical stylii and the BBC usually uses a 0·0175-mm radius spherical tip as a compromise.

Stereophonic records must be played only on equipment specially designed for this purpose, otherwise the grooves can be so damaged that virtually all stereophonic information is permanently destroyed. Stereophonic pickups are fitted to most monophonic equipment used by the BBC, although suitable monophonic pickups are commercially available and these extract the monophonic version $(A + B)$ of the stereophonic information from the record. Second-order effects (mostly pinch-effect) cause some vertical movement of the stylus when monophonic records are played. This results in out-of-phase outputs from a stereophonic pickup and it is usual to parallel the outputs so that the out-of-phase signals cancel.



unmodulated groove

modulation of A wall only

$$\left[\; M = \frac{A+B}{2} = \frac{A}{2} \qquad S = \frac{A-B}{2} = \frac{A}{2} \;\right]$$

modulation of B wall only

$$\left[\; M = \frac{A+B}{2} = \frac{B}{2} \qquad S = \frac{A-B}{2} = \frac{-B}{2} \;\right]$$

identical modulation of both walls: $A = B$

$$\left[\begin{array}{l} M = \dfrac{A+B}{2} = \dfrac{A+A}{2} = A \\[2ex] S = \dfrac{A-B}{2} = \dfrac{A-A}{2} = 0 \end{array}\right]$$

modulation of both walls with equal amplitudes but opposite directions: $A = -B$

$$\left[\begin{array}{l} M = \dfrac{A+B}{2} = \dfrac{A+(-A)}{2} = 0 \\[2ex] S = \dfrac{A-B}{2} = \dfrac{A-(-A)}{2} = A \end{array}\right]$$

Fig. 6.2 Analogy Between M and S Signals and Vertical and Horizontal Modulation Components in Stereophonic Recording

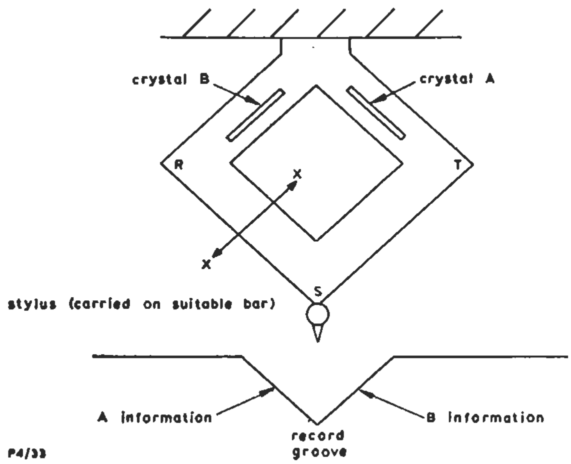


Fig. 6.3 Action of Stereophonic Crystal Pickup

6.1.2 Stereophonic Pickups

(a) Crystal Pickup

Fig. 6.3 illustrates the basic configuration of a typical crystal pickup in which two separate crystals provide the *A* and *B* outputs. The crystals are mechanically connected to a flexible diamond-shape structure. With modulation of the *A* wall of the groove only, the stylus moves parallel to the line XX and the limb RS of the diamond flexes sideways with little effect on crystal *B*. The stylus movement is transmitted along the arm ST with resultant distortion of crystal *A* which produces an electrical output. Modulation of the *B* wall of the groove produces a similar effect on crystal *B*. Modulation of both walls is resolved into left and right components by the diamond structure. In practice the degree of separation is about 15 dB at mid frequencies.

(b) Moving-coil Pickup

The action of a moving-coil stereophonic pickup is illustrated by Fig. 6.4. The two coils *A* and *B* which provide the stereophonic output signals are at right angles to each other and are able to move in the magnetic field, as a result of stylus movement, about an effective pivot at their common centre. The diagram shows the coils positioned in their normal (unmodulated) state, each with the plane of its circumference parallel to the appropriate wall of the record. In this position neither of the coils embraces any of the lines of magnetic force.

Suppose that only the *A* wall of the groove is modulated. The stylus then moves in a direction parallel to the *B* wall and the motion is transferred to the coils in such a way that coil A 'rocks' about the pivot point and is disturbed from its normal plane but coil B 'rolls' within its plane. The rocking motion makes coil *A* cut the lines of force and produce an output, whereas the rolling motion of coil *B* takes place without cutting the magnetic field. Similarly,

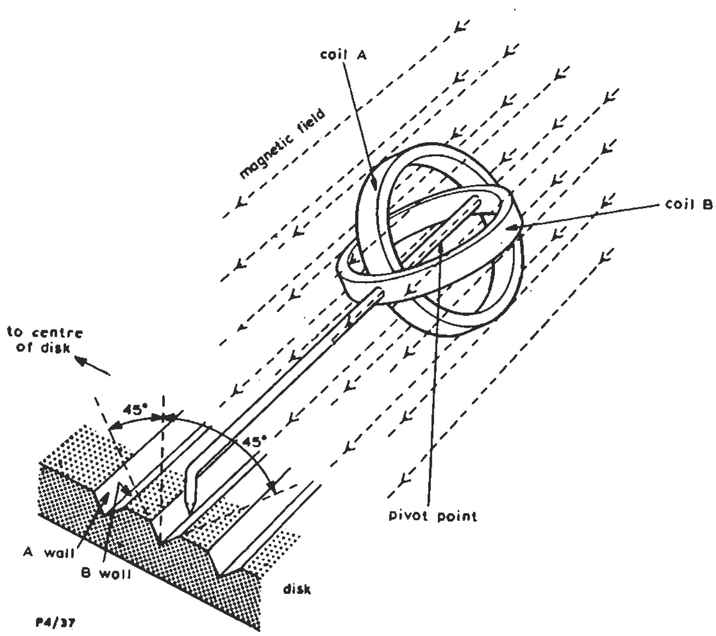


Fig. 6.4 Action of Stereophonic Moving-coil Pickup

modulation of the *B* wall alone produces an output from coil *B* but not from coil *A*. When both walls of the groove carry modulation, each coil produces the appropriate electrical output. The degree of separation achieved by a moving-coil pickup is about 25 dB at mid frequencies.

6.2 Tape Recording and Reproduction

6.2.1 Standards

Stereophonic tape recordings are made by using twin tracks on conventional 6.25-mm (¼-in) tape. The two tracks, upper and lower for the A and B signals respectively, are separated by a guard track which remains unmodulated.

twin-track work where a different programme may be recorded on each track.

The presence of the guard-track reduces the overall active tape-width and leads to a poorer signal-to-noise ratio and also to low reproduced level when the tape is played back on a full-track head. If 1-kHz tone is recorded at a given short-circuit flux* by a full-track head and also on both tracks simultaneously of narrow guard-track and wide guard-track heads, the relative respective levels from a full-track reproducing head will be 0, −1 dB and −3·5 dB approximately. Conversely if a wide guard-track repro head is used it will never scan the centre section on the tape so its output will depend on the short-circuit flux regardless

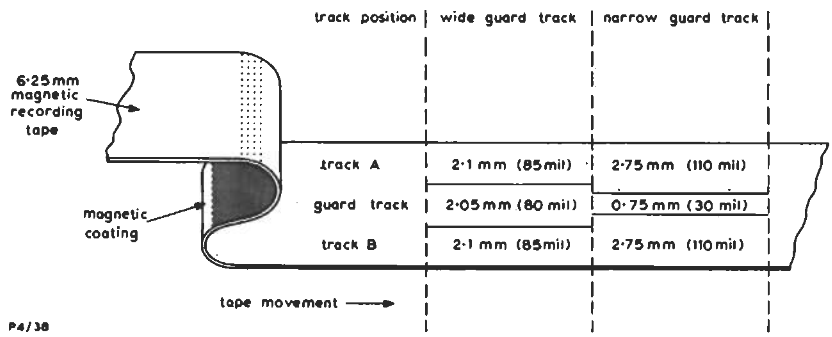


Fig. 6.5 Twin-track Stereophonic Tape Recording Standards

Two standards are in general use and are known as *wide guard track* and *narrow guard track;* the dimensions of the tracks for both standards are given in Fig. 6.5. These are not at present internationally agreed standards but are typical figures taken from the specifications of head manufacturers. The only definite specification is that the guard track shall be at least 0·75 mm (BS 1568: Part 1: 1970). Many European countries use recording machines in which the heads face towards the rear of the machine, with the magnetic coating towards the operator; on these machines the upper track is B and the lower is A, so that the recorded tape is directly interchangeable with the BBC standards.

The separation of the two active tracks by the guard track, and the consequent spacing of the two parts of the reproducing head, minimises the amount of crosstalk which appears on one channel as a result of modulation of the other channel.

Crosstalk figures of the order of −38 dB at 1 kHz rising to −30 dB at 50 Hz and 10 kHz are regarded as acceptable for stereo but are not satisfactory for

of the format of the recording head. To minimise errors of compatibility the BBC has adopted a hybrid standard of narrow guard-track recording, wide guard-track reproduction; thus the maximum error caused by guard-track variations is only 1 dB whereas the crosstalk between A and B tracks is less serious than if narrow guard-track repro were used. Some recorder manufacturers are only able to supply wide guard-track record heads and so recordings made on these machines will reproduce about 3·5 dB low on a full-track machine but as the proportion of stereo reproducing machines in service increases this error becomes less serious.

6.2.2 Recording Levels

Recorded levels on the tape are specified in terms of short-circuit flux and there are several units in common use. The preferred S.I. unit is the nanoweber per metre (nWb/m) but because track widths are commonly measured in millimetres a more

* See the first paragraph of Section 6.2.2.

convenient unit is the picoweber per millimetre of recorded track width (pWb/mm). Since 1 nWb equals 1000 pWb and 1 m equals 1000 mm, 1 pWb/mm equals 1 nWb/m. Another unit in common use outside the BBC is the millimaxwell per millimetre (mM/mm); the conversion for this is 1 mM/mm equals 10 pWb/mm or 10 nWb/m.

To compensate for the degradation of signal-to-noise ratio caused by the narrow active track-width a higher short-circuit flux is used on stereo recording machines than on mono machines. At present (1972) this level difference is 4 dB, set up as follows:

A standard BBC line-up tape (1968 level) which has a short-circuit flux of 160 pWb/mm, corresponding to 160 nWb/m or 16 mM/mm, is used to align the repro amplifiers but the repro gain controls are set to give an output level of –4 dB (3 on the PPM) on each track instead of zero level. If the record gain controls are now adjusted so that zero level into each input produces zero level out of the corresponding output, the required 4-dB higher short-circuit flux (peak level 640 pWb/mm or 64 mM/mm) will be recorded. The tapes used for stereo, normally B.A.S.F. LR56 but occasionally Agfa PER 555 or others, will handle this peak flux but require higher bias current and different equalisation from the BBC Type 100 used for mono. Another advantage of these tapes is that, being matt-backed, they wind much more evenly onto the spools than Type 100 and so are less easily damaged when boxed; slight edge damage affects the tracks of a stereo tape differentially and produces disturbing jumps in the position of the sound image.

As described in Section 5, a zero level A + B signal will be produced with tone at –3 dB on both A and B chains (in phase); the line-up tone recorded on a stereo tape is at –3 dB on each track, i.e., 180 pWb/mm.

If a stereo machine is used for recording monophonic programme controlled to zero volume (peaks to +8 dB) this should be applied at –3 dB volume (line-up tone –3 dB, peaks to +5 dB) to both tracks of the recorder in parallel. This will then reproduce normally on a stereo machine (zero volume on A + B) and, assuming a narrow guard-track record head, at approximately normal level on a full-track repro machine, for the following reasons:

(a) The loss caused by the unmodulated narrow guard track is about 1 dB.
(b) The short-circuit flux is 4 dB higher than for a normal mono machine.
(c) The programme is applied at –3 dB to each track.

If the record head has a wide guard track the level from a wide guard-track repro head remains unchanged but in (a) above the guard-track loss becomes about 3·5 dB so that the level from the full-track repro head is about –2·5 dB.

Table 6.1 shows how many variables need to be considered in measuring levels in the stereo recording and reproducing processes and also the impossibility of making mono and stereo recordings precisely interchangeable in level under all conditions of programme. The table gives the levels indicated by correctly aligned A,B and A+B peak programme meters, with the 3-dB gain difference (See Section 5 and Table 5.1).

With only a few exceptions the narrow guard-track (N.G.T.) record, wide guard-track (W.G.T.) repro configuration is used throughout the BBC so only the figures in heavy type need be considered for most purposes.

The values of short-circuit flux in the table involve an over-simplification as this is a frequency-dependent parameter but the figures given are correct for signals at approximately 1 kHz. Short-circuit flux figures are included here because they are a fundamental measurement of recorded level and many organisations outside the BBC refer their peak recorded levels to specified short-circuit flux levels. For example the meter on many Dolby A Processors has calibrations corresponding to 185 pWb/mm and 320 pWb/mm (actually marked 18·5 and 32 mM/mm). BBC stereo line-up tone, nominally at 180 pWb/mm, is within normal measuring accuracy (actually 0·25 dB) the same as the 18·5-mM/mm Dolby level (also referred to as NAB level).

TABLE 6.1

Signal	*Short Circuit Flux (pWb/mm)*	*Record Head*	*Repro Levels (dB)*			
			Full-track	*W.G.T. Stereo*		
				A + B	*A*	*B*
Tone (mono line-up)	**160**	**F.T.**	**0**	**−1**	**−4**	**−4**
Tone (stereo line-up)	**180**	**N.G.T.**	**0**	**0**	**−3**	**−3**
	180	W.G.T.	−2·5	0	−3	−3
Mono peak level	**400**	**F.T.**	+8	+8	+5	+5
	400	**N.G.T.**	+8	+8	+5	+5
	400	W.G.T.	+5·5	+8	+5	+5
Average stereo (random phase) peak level	**640**	N.G.T.	+8	+8	+8	+8
	640	W.G.T.	+5·5	+8	+8	+8
Peak level A only	**640**	N.G.T.	+5	+5	+8	nil
	640	W.G.T.	+2·5	+5	+8	nil
Tone at peak level on A and B simultaneously in phase	**640**	N.G.T.	+11	+11	+8	+8
	640	W.G.T.	+8·5	+11	+8	+8

AJ 10/68
Revised DGMS (X) 9/72

SECTION 7

CONTINUITY AND CONTROL ROOM ARRANGEMENTS

7.1 General

Stereophonic programme outputs from studios or O.B. sources are passed to B.H. London control room whence they can be routed to a continuity suite. At present (December 1968) stereophonic broadcasting is limited to a part-time service, on the Radio 3 network only, and facilities for this have been provided by modifying the existing monophonic equipment in and associated with Continuity Suite D.

The modified continuity suite (Fig. 7.1) retains all the normal facilities for monophonic working. The principal additions for stereophony are:

(a) A selection system for stereophonic sources.
(b) Six stereophonic input channels (also usable monophonically, with the same signal on both programme circuits).
(c) Three stereophonic tape channels.
(d) A stereophonic gram-desk.
(e) Full aural and visual monitoring.
(f) Remote control of coder operating mode.

Except for the stereo gram-desk, the studio facilities are monophonic. The monophonic equipment feeds identical signals to the stereophonic *A* and *B* channels, so that central images result.

7.2 Source Selection

The main source-selection system cannot handle stereophonic-source outputs and an auxiliary system has been added for this purpose. Selection of a stereophonic source, using the two systems in combination, requires a particular sequence of operations to be performed.

The first operation is identical with the normal method of selecting a monophonic source and when completed the programme line, carrying the ($A + B$) output of the stereophonic source, is switched to the channel input. At the same time, the usual cue and control circuits are provided. Additionally, an extra switching circuit is connected to allow operation of the otherwise independent stereo-selection system which is carrying the separate *A* and *B* signals from the source.

The stereo-selection system comprises a relay matrix and allows the outputs from any of 10 sources to be connected to any of four channels in the continuity suite. When the switching circuit is made ready at the end of the mono-selection operation, the appropriate relay can be energised by operating the channel *Stereo Select* pushbutton. The *A* and *B* signals are then available to the channel but are not connected until the channel *Mono/Stereo* selection button is pressed. This button is a push-on/push-off type with an internal lamp which lights when the *Stereo* condition is selected. In the *Mono* condition, the input to both halves of the channel is obtained from the ($A + B$) line routed via the mono-selection system.

7.3 Stereophonic Channels

All the stereophonic channels in the continuity suite are of simple form as shown in Fig. 7.1. Four of the six channels can be fed from the main or the stereo source-selection system as described previously. The other two channels are intended to be used with sources not available on the stereo source-selection system, and must be fed via cord circuits.

7.4 Output Arrangements

Two independent output chains are provided and each has separate *A*, *B* and *Mono or* ($A + B$) channels. In each instance the *Mono or* ($A + B$) signal is obtained by direct addition of the *A* and *B* channel signals in a resistive network.

When the continuity is working in the *Mono* condition, a 3-dB pad is switched into the *Mono or* ($A + B$) output, because the *A* and *B* channels are carrying identical signals. This helps to maintain equality of output levels in the *Mono* and ($A + B$) conditions, as explained in Section 5.

The two outputs, *Main* and *Single Transmitter*, take the usual form which allows continuity-studio-originated signals to be fed to the single-transmitter chain while the main chain takes a feed from the input channels.

7.5 Coder Remote Control

The programme coder for the pilot-tone system is situated at Wrotham transmitting station. During stereophonic service periods, the coder accepts the *A* and *B* outputs from the continuity and delivers a multiplex signal to modulate the transmitter. During monophonic periods, however, the coder is not required to function in this way, and its mode of

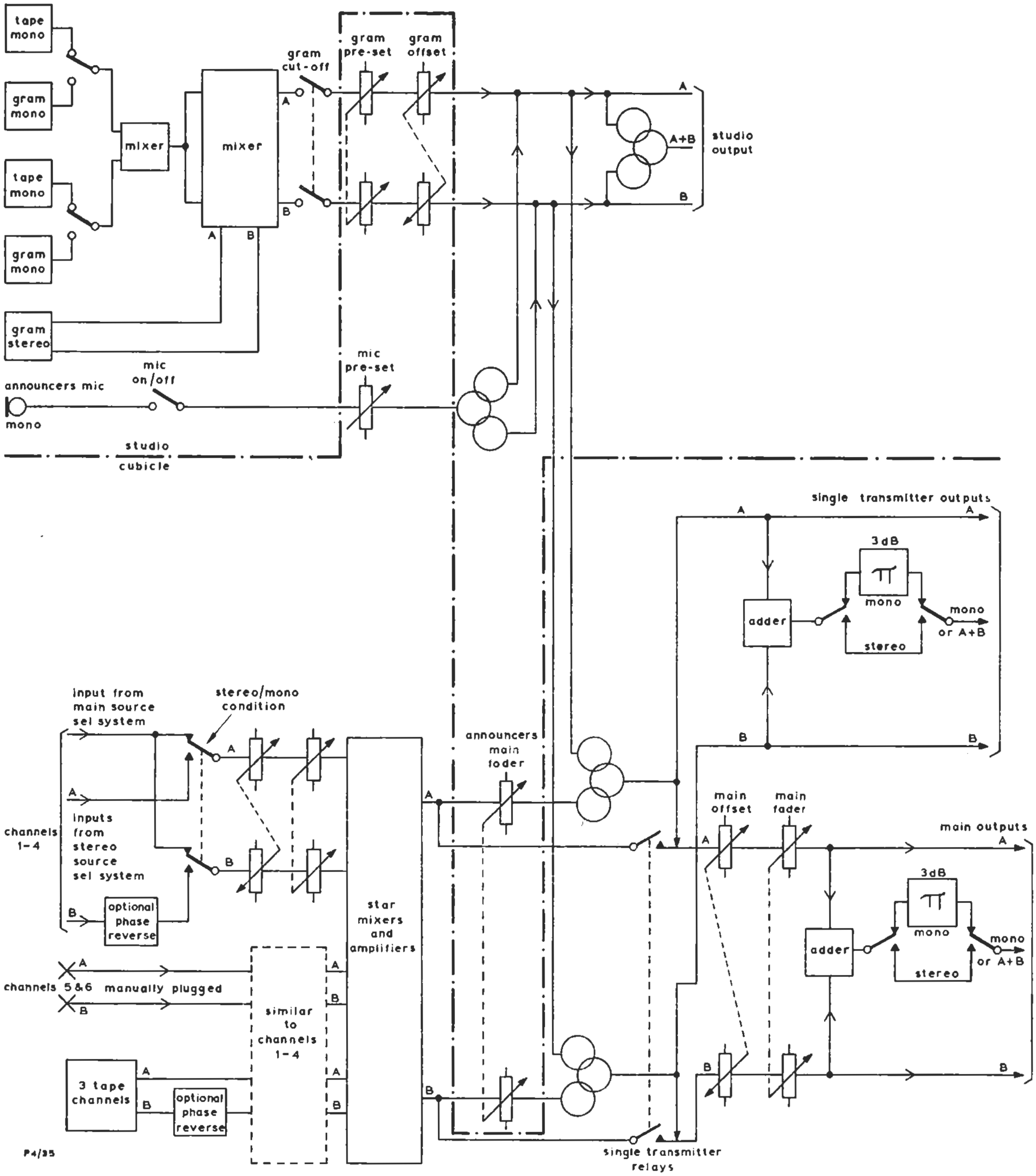


Fig. 7.1. Modified Continuity Suite: Simplified Diagram

operation is then altered by means of relays. These relays are controlled via phantom circuits on the programme lines, either from B.H. control room or from the continuity cubicle. Similar control can be exercised to change over between the main and standby equipment at the transmitting station. Details of the present arrangement are given in Section 8.

7.6 Monitoring

Two commercial bookcase-type loudspeakers are provided for stereophonic monitoring. They can also be used for monophonic monitoring, although for this an LS5/1A is available. The two stereo-monitoring loudspeakers can either be fed individually with the appropriate *A* or *B* signal, or they can both be fed with the *M* or the *S* signal. Either speaker can be muted in any of the three conditions. The monitoring signals can be derived from prefade points, from the position output, or (except for the *S* signal) from a check receiver.

The main and prefade monitoring circuits each include two double P.P.M.s, one of which shows the *A* and *B* and the other the *M* and *S* signals.

SECTION 8

DISTRIBUTION OF STEREOPHONIC SIGNALS

8.1 Introduction

The distribution of stereophonic material requires two information channels, which carry the A and B or the M and S signals, and should be closely matched as mentioned in Section 1.2. Any difference in the gain or the transmission time of the two paths between studio and coding point can cause imperfections in the reproduced sound stage.

For simplicity, suppose that the two channels have different gains but are otherwise identical, and that the A and B signals are sent along them. The stereophonic sound stage at the receiving end will be compressed and biased toward the side of the channel with higher gain, and also the 'compatible' sum signal, M, will be unlike an $(A + B)$ signal derived at the sending end. The latter objection could be overcome by deriving the M and S signals at the sending end and passing them along the two channels. However, this system introduces cross-talk between the derived A and B signals as shown below.

If the M and S signals are subjected to amplifications of p and q times respectively, the received signals can be denoted by Mp and Sq. Addition and subtraction of these now gives A' and B' instead of A and B. Thus:

$$\begin{aligned} A' &= Mp + Sq \\ &= \tfrac{1}{2}(A + B)p + \tfrac{1}{2}(A - B)q \\ &= \tfrac{1}{2}A(p + q) + \tfrac{1}{2}B(p - q). \end{aligned}$$

Similarly,

$$B' = \tfrac{1}{2}A(p - q) + \tfrac{1}{2}B(p + q).$$

Thus, if p and q are unequal, each of the reconstituted signals contains a component from the other channel.

The simple inequalities considered could be easily overcome by introducing adjustable gain into each channel, but practical programme distribution circuits usually have nonlinear amplitude/frequency and phase/frequency characteristics*, and this makes it difficult to provide two long-distance circuits which are sufficiently alike to be used for stereophony. Because of the stringent requirements, the need for matched lines is kept to a minimum by coding the A and B signals at an early point in the distribution network. Subsequent destinations are then fed with the pilot-tone multiplex signal over wideband circuits, and except for monitoring and test purposes, the A and B signals are retrieved only at the listeners' decoders.

The remainder of this Section is concerned with the distribution arrangements provided for the present stereophonic broadcasting service, which is carried by a limited number of Radio 3 transmitters. Any extension of stereophonic broadcasting, either geographically or into additional programme networks, is likely to use similar principles.

*The effect of these characteristics is explained in BBC Engineering Monograph No. 56.

8.2 Distribution of A and B Signals to the Coding Point

All Radio 3 programmes are fed to Wrotham transmitting station via two matched P.O. lines. During stereophonic service periods the lines carry the A and B signals from B.H. control room, but during monophonic periods each line is fed with the same signal so that one acts as reserve for the other. The stereophonic A line is used as the normal monophonic circuit, with the B line in reserve. As indicated by Fig. 8.1, duplicate sets of programme input equipment including coders are provided so that main and standby facilities are available at all times. The diagram is much simplified but shows the effect of *Mono Normal*, *Mono Reserve* and *Stereo* switching which is achieved in practice by relays controlled either locally or via phantom circuits from B.H.

With the switching system at *Stereo*, the A and B signals are fed to the main and standby programme input equipment chains simultaneously. In each chain the two signals pass via individual pre-emphasis networks into ganged limiters which are arranged to operate together whenever either signal reaches the preset limiting level. By this means the necessary identical treatment of the A and B signals is maintained when either signal approaches the level which would cause over-

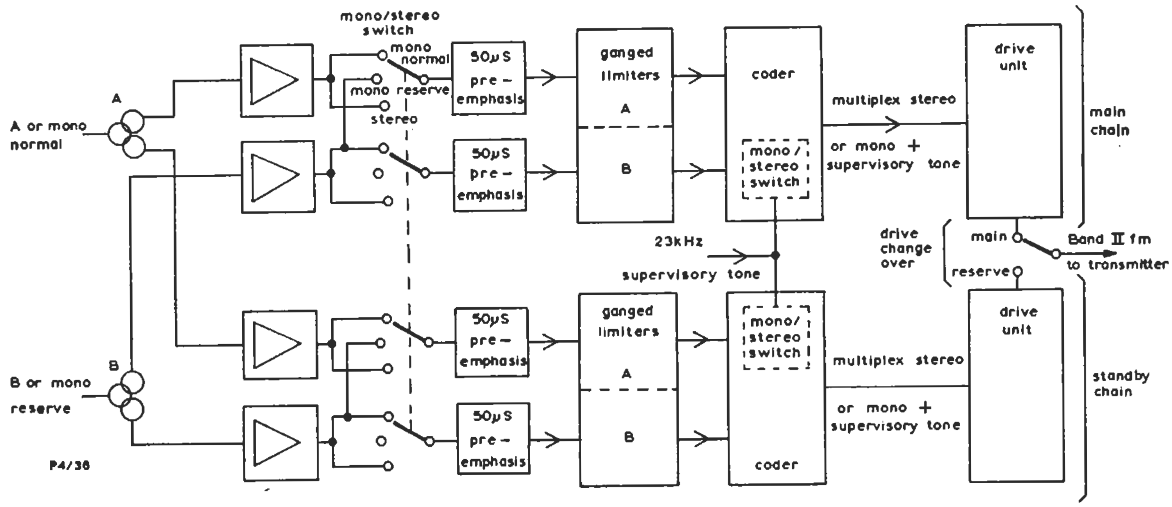


Fig. 8.1. Arrangement of Programme Input Equipment, Coders and Drive Equipment at Wrotham

deviation of the transmitter. The pre-emphasised and controlled signals are passed to the coder, which produces from them the multiplex signal to frequency modulate the oscillator in the transmitter drive unit.

The *Mono Normal* position of the switching system connects the lines to the programme input equipment in the same way as the *Stereo* position, but in the *Mono Reserve* position the outputs of the *B*-line hybrid transformer are passed to the *A* pre-emphasis networks. In both *Mono* positions there is also extra switching, which alters the operation of the coder. The multiplexing action is then suppressed and instead the coder is used as a combining stage to add an externally derived 23-kHz supervisory tone to the output of the *A* limiter. This combined signal is the monophonic service modulating signal to the transmitter drive unit. The supervisory tone is used for automatic switching and monitoring purposes during monophonic service, but during stereophonic periods these functions are performed by the pilot-tone.

8.3 Distribution Between Transmitting Stations

8.3.1 Main Stations

The multiplex signal (or mono-plus-supervisory-tone signal) is sent to transmitting stations remote from the coding point by a combination of rebroadcast and s.h.f.-link techniques. Fig. 8.2 shows the distribution to Sutton Coldfield and Holme Moss from Wrotham.

The s.h.f.-link station at Whipsnade picks up the signal radiated by Wrotham and, using a double-superheterodyne receiver, changes the carrier to an intermediate frequency of 1 MHz. This i.f. carries the multiplex information as f.m., and is used as the modulation input of the s.h.f. transmitter. The signal is then passed, via the non-demodulating s.h.f. repeater station at Whichford, to Sutton Coldfield where the 1-MHz frequency-modulated i.f. is retrieved.

The 1-MHz i.f. is applied to a discriminator to recover the multiplex signal, and this is used to modulate the broadcast transmitter at Sutton Coldfield. The multiplex input for the Holme Moss transmitter is obtained in a similar way via the s.h.f.-link station at Macclesfield Forest where the broadcast signal from Sutton Coldfield is received. On this particular route the distance can be covered without a repeater station.

It is of interest to note that the practical application of the technique outlined uses an otherwise conventional double-superheterodyne receiver in which the discriminator is physically separated from the second frequency changer by many miles. The gap is effectively closed by including the s.h.f. link in the second i.f. chain.

8.3.2 Relay Stations

Some relay stations use a translator to rebroadcast the output of a main transmitter and these carry the stereophonic service whenever the parent transmitter does so. It does not follow, however, that the

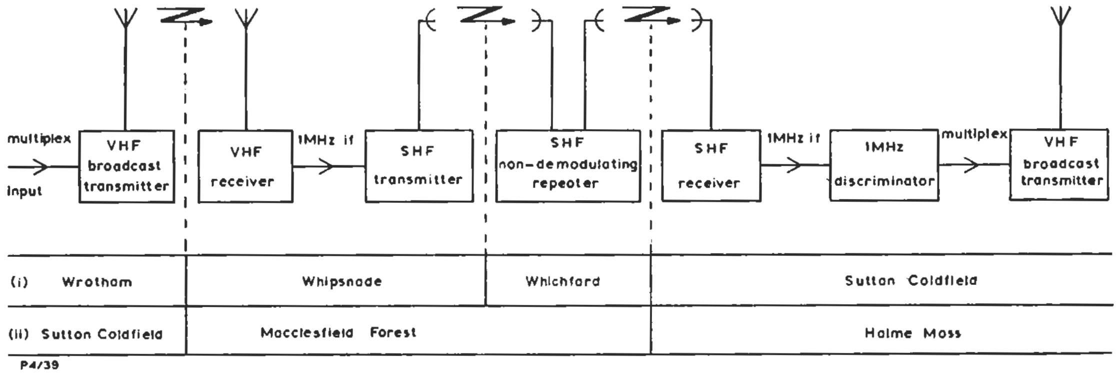


Fig. 8.2. Distribution between Wrotham and Holme Moss

relayed stereophonic service is automatically satisfactory. This depends on local conditions and in particular on the signal strength at the input to the translator.

Other relay stations use rebroadcast receivers to derive their modulation input. At these, modifications to the receiver and to the transmitter drive unit are required to enable the wideband multiplex signal to be handled. Unmodified relay stations of this type transmit the compatible monophonic version of the programme during stereophonic service periods.

SECTION 9

TRANSMITTING STATION TECHNIQUES

9.1 General

The introduction of stereophonic broadcasting into an existing monophonic service has two major effects at transmitting stations. These effects concern (a) the transmitters and their drive units, and (b) the programme input equipment and line-up procedure. Other matters, such as (c) monitoring facilities and (d) reserve programme feeds, also need consideration, but the arrangements for these are usually peculiar to individual stations. For (a) and (b) the main need is to avoid distortion and maintain identity of treatment of the two information channels (baseband and subcarrier) of the multiplex signal.

9.2 Transmitter and Drive Unit

Theoretically, frequency modulation of a carrier by a wideband multiplex signal results in a deviated carrier signal which requires wider bandwidth r.f. circuits than those needed for a monophonic signal*. In practice, however, it is not usually necessary to modify the transmitter, but some changes are required in the 'audio-frequency' and modulator circuits of the drive unit so that the wideband multiplex signal can be handled. Some types of drive unit cannot be modified and have to be replaced by new units designed for wideband application.

9.3 P.I.E. and Line-up

9.3.1 General

Identical treatment of the two information channels of the multiplex signal can be maintained only if the apparatus used has a flat response over a bandwidth greater than the 30 Hz to 53 kHz spectrum. The techniques employed to pass the multiplex signal from the derivation point to the modulator input are therefore more like those used for video signals than the normal sound techniques. The only processing of the multiplex signal in this path is either linear amplification or attenuation to provide the correct deviation of the transmitted carrier.

*See Designs Department Technical Memorandum No. 5.45(68), Appendix I.

9.3.2 Transmitter Line-up with either Pilot-tone or Supervisory Tone

The relationships between the components of the coder output signal (either stereo-multiplex or mono-plus-supervisory-tone) are fixed at the coding point to agree with the specification of the particular system. In each instance there is one component the level of which does not depend on the a.f. content of the modulating signal and which can be used as a line-up signal. This component is either (a) the 19-kHz pilot-tone of the stereo-multiplex signal, or (b) the 23-kHz supervisory tone present with the monophonic signal.

When there is no a.f. input to the coder, the deviation of the transmitter can be set up as follows:

1. Connect a *wideband* deviation meter to a suitable test point.
2. Check which tone is incoming, either 19 kHz or 23 kHz.
3. Adjust the input level to the modulator to give the appropriate reading on the deviation meter. This is:
   (a) 6·75kHz deviation with 19-kHz pilot-tone, or
   (b) 4·5 kHz deviation with 23-kHz supervisory tone.

9.4 Monitoring and Test Arrangements

Automatic monitoring, including executive switching to bring in standby equipment, is accomplished by using the pilot-tone (or the mono supervisory tone) in a conventional system based on principles outlined in Instruction T.10.

Provision is made also for stereophonic aural monitoring, using a decoder which can be fed with a multiplex signal derived from various sources in the distribution and transmitting chain. These sources include the final radiated carrier, from which the multiplex signal can be extracted either by a receiver or by a wideband deviation meter.

The decoder can also be used for routine measurements, but specialised equipment is necessary for a thorough performance check either of the complete chain or of individual items.

Useful investigations can be made by employing a locally controlled coder in conjunction with a decoder and standard test equipment.

AJ 12/68

APPENDIX A

REFERENCE INFORMATION

Colour Code

The colour code given in Table A.1 has been adopted in studio and O.B. practice to identify the A, B, $(A + B)$ and $(A - B)$ channels. It is employed on individual input and output positions on jack-fields and on interconnecting cords and cables. Programme meter needles are also coloured and in some instances, with meters which can be switched alternatively to two channels, the needles are appropriately two-coloured.

TABLE A.1

Channel	*Colour*
A	RED
B	GREEN
$(A + B)$	WHITE (in O.B. practice this is $(A + B)$ Clean Feed)
$(A + B)$ with Announcements	BLUE in O.B. practice only
$(A - B)$	YELLOW

$(A + B)$ may be denoted by M or *Sum*.
$(A - B)$ may be denoted by S or *Difference*.

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APPENDIX B

LINE-UP PROCEDURE FOR COINCIDENT MICROPHONES

General

The suggested procedure given here requires two operators, one in the studio to speak into and make adjustments to the microphones and the other to carry out checks and make adjustments at the control point. It includes checks of all the important points related to the use of a coincident pair and has the advantage that it can be performed before the control equipment is finally adjusted for stereophonic balance. Although the procedure is given for a particular type of coincident pair, the A.K.G. Type C.24, the sequence has general application and can be adapted as necessary to suit other microphones.

The A.K.G. Type C.24 consists basically of two Type-C.12 microphones, one above the other in a common housing. One microphone is fixed relative to the housing but the other can be twisted to provide the required angle between the axes. The C.24 can be used either suspended or fixed on a stand. In each instance a check must be made to ensure that the individual outputs are fed to the *A* and *B* channels in accordance with an accepted standard; that is, the *A* channel input is always derived from the *lower* microphone of a coincident pair.

Procedure

1. Set the polar-diagram switches of both microphones to the same characteristics, preferably the one to be used for the programme if this is known; otherwise use the figure-of-eight characteristic.

	Operator X at the Microphones	*Operator Y at the Control Point*
2.	Adjust the angle between the capsules to 90 degrees and speak into each in turn to identify them appropriately as left and right.	Put the control and monitoring equipment to the stereo condition and check for correct left and right indications from operator X.
3.	Adjust the angle between the capsules to 0 degrees. (Make sure that the microphones are not back-to-back; the front faces are either grey or silver-coloured and the rear faces are black.)	Introduce a phase-reversal into one channel and put the loudspeaker selector switch to ($A + B$).
4.	Speak into the front of the microphones.	(a) Adjust the *Mic. Bal.* or *Channel Bal.* controls for minimum output from the loudspeakers. (b) Remove the phase-reversal and restore the loudspeaker switch to stereo. Check that operator X's speech is heard from about midway between the loudspeakers.
5.	Walk slowly around the microphones while speaking towards them.	Listen and check that the speech remains approximately central throughout. (Slight variations of position when operator X is near the dead axes of the microphone can be ignored.) The loudness should vary according to the polar characteristic in use, but a substantial positional shift indicates that the two microphones are operating with different polar characteristics and the procedure must be restarted when the fault has been cleared.
6.	Restore the angle between the capsules to 90 degrees and speak from left, centre and right in turn.	Check for correct positional sense.