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The following paper is reprinted from the September 1957 issue of the *Proceedings of the Institution of Electrical Engineers,* by kind permission of the Institution. Its concise presentation of the history, theory, and psychoacoustic basis of the stereophonic effect makes this, in the Editor's opinion, an exceedingly valuable reference source.

THE 'STEREOSONIC' RECORDING AND REPRODUCING SYSTEM

A Two-Channel System for Domestic Tape Records.

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SUMMARY

The paper reviews briefly the history of stereophonic reproduction. The principal basic systems with their underlying ideas are described and compared. Some account is given of the supposed mechanism of natural binaural listening from the viewpoint of direction localization.

The principles and practice are discussed of a particular system for domestic use, derived from the early work of A. D. Blumlein, and characterized by the use of spaced loudspeakers driven in phase, to which the name 'stereosonic' has been given. The aims of this system are defined, and the mathematical theory involved in its use is developed. Limitations and sources of error in the results achieved are described.

Equipment used in the making of master recordings and some of the problems of studio technique involved are described. Consideration is given to the form which a domestic stereophonic record should take, and the standards to which such a record should conform, together with the requirements which these impose on the reproducing equipment.

LIST OF PRINCIPAL SYMBOLS

θ = Angle between radius vector of the sound source and the median plane.
θ_t = True angle.
θ_a = Apparent angle.
h = Effective human-ear spacing.
v = Velocity of sound in air.
ϕ = Phase difference of sound at the listener's two ears.
ω = Angular frequency.
ψ = Semi-angle subtended at the listener by a pair of loudspeakers.
μ = Difference of the time of arrival of the sound at the listener's two ears.
L and R = Peak amplitudes at listener's ears due to left and right loudspeakers.
x = Half distance between loudspeakers.
r = Projection of the listener's displacement from the median plane on to the loudspeaker base-line, expressed as a fraction of x.
d = Distance between loudspeaker base-line and the listener.
y_a = Fractional distance of apparent source from central position.
y_t = Fractional distance of true source from central position.
β and γ = Angles subtended by the left- and right-hand loudspeaker with respect to the median plane of the listener.
τ = Time-of-arrival difference of the sound from left- and right-hand loudspeakers at the listener caused by path differences.
μ_L and μ_R = Semi time differences at the listener's ears due to sounds from left and right loudspeakers.
ϕ_L and ϕ_R = Resultant phase of the sound at the left and right ears of the listener relative to the centre position.

(1) THE IMPORTANCE OF STEREOPHONY IN SOUND REPRODUCTION

Normal listening is always binaural, and the two ears are used in conjunction by the brain to interpret the sounds heard. If all the individual sound sources, such as voices and instruments, and the reverberant sounds, are recorded or transmitted by a single channel before being reproduced by one or more loudspeakers, it is not possible to use the binaural sense of location to differentiate between the various sounds as is done in direct listening.

In order to add some spatial realism using only one channel a number of schemes have been tried, such as the use of spaced loudspeakers supplied through frequency-dividing networks, or several loudspeakers fed through delay networks, but none of these is universally successful on all types of programme.

An honest assessment of the sound quality obtainable through a single channel, in which the frequency, harmonic, intermodulation and noise distortions have been reduced to the limits detectable by the human ear, reveals that the reproduction is still far from lifelike. The addition of real binaural listening conditions seems to be the most likely factor remaining.

Many tests have now been made which all show that, if a system is used in which the relative sounds at the two ears are each reproduced as similarly as possible to those heard in direct listening, i.e. with the correct relative amplitude and phase differences for each sound source, the reproduced sound takes on a new quality which is nearer to realism than can be obtained in any other way.

For a given amount of 'ambient' or reverberant sound, it is possible to obtain better definition, i.e. ability to distinguish individual instruments or voices in a chorus. A soloist can be heard clearly through an accompaniment, without having to resort to making the soloist appear unnaturally close to the listener.

The problem of hearing clear bass parts free from the unduly long low-frequency reverberation which is so commonly encountered, is considerably eased, and full bass response without 'boom' can be obtained. It is noticeable that the reproduced sound level can be raised higher than with single-channel reproduction before the average listener becomes irritated. The ability to do this permits a closer reproduction of the original dynamic range of the music in cases where this is great. It also seems possible to obtain a given subjective loudness for less acoustic power than with single-channel operation.

All this is quite aside from any benefits which may accrue from the ability to reproduce illusion of movements and the spatial relationships so essential to drama and opera.

The requirements then seem to be to reproduce, with reasonable economy, in a system suitable for home use, the same sounds at each ear separately which would have been heard by direct listening in an optimum position in front of the performers. This requirement can be limited to sounds arriving from directions covering an angle of, say, 90° in front of the observer for most practical purposes. No method has yet been devised to meet this requirement from all horizontal directions, but neither this nor the need for the indication of vertical angular direction seems to be of major significance.

(2) HISTORY

An explanation of the principles by which the normal human being uses both his ears for the purpose of spatial location has been sought since the classic studies in acoustics of the mid-nineteenth century. Attempts to reproduce this natural ability at a distance were made early in the development of telephony. One of the earliest accounts was given in *L'Électricien* of 1881 by E. Hospitalier, who described the sound system installed in the Paris Opera House.[1] This basically consisted of a pair of microphones spaced on opposite sides of the stage, which supplied at a distance a number of pairs of telephones held to the ears of the listeners, who received an impression of localization.

It was probably not for many years, until sound reproduction for entertainment purposes was receiving more attention, that any serious development work along these lines was undertaken. It was well understood, however, that the use of two similar microphones spaced by a distance equivalent to that between a human being's two ears, operating through two separate channels into a pair of headphones, did, in fact, give a remarkably accurate impression of spatial location. The use of headphones nevertheless robs the observer of the ability to locate sources outside the horizontal plane and prevents discrimination between front and rear, since in these cases it is necessary to move the head relative to the sound field.

In the 1920's it became customary to use loudspeakers for sound-reproducing systems instead of head telephones, and the problem of reproducing sound with directional sense immediately became more complex, for reasons which are discussed later in the paper.

One of the earliest workers in this field was the late A. D. Blumlein. He was convinced at this time that the main contribution towards the sense of spatial location was provided by the difference in time of arrival at the two ears which, at low frequencies, can be interpreted in terms of phase differences.

He invented a system[2] by which the phase difference between the outputs of two similar microphones spaced by a distance small compared with the wavelength, can be converted into two in-phase outputs of different amplitudes driving two loudspeakers spaced by an appreciable distance. If the relative amplitudes corresponding to a given phase difference at the microphones are correctly proportioned, the phase differences at the two ears of the person listening to the two loudspeakers will be a close simulation of the phase differences which the listener would have heard had he been at the microphone position.

Blumlein showed how the correct relative loudspeaker inputs could be derived from two closely spaced pressure microphones by the use of suitable modifying networks which he called 'shuffling' networks. He also showed how velocity microphones could similarly be used, and solved the mathematical relationship between the various parameters in order to obtain the correct apparent position of the source with any given loudspeaker arrangement. He also described a means of recording two channels on a single disc, and, in fact, disc records of this type were pressed and reproduced in 1933. Experimental motion-picture films were made incorporating twin-track optical recordings, and these were reproduced along with large-screen projection in 1935. The industry was not prepared to introduce the wide screen at this date, and therefore further work on the application was abandoned.

Blumlein's method has the advantage of being a 'free field' system, the observer being able to move his head without producing a corresponding movement of the apparent source. It relies on the interaction of sounds from the two loudspeakers at the listener's ears, and cannot utilize headphones.

One of the first successful public demonstrations of stereophonic sound reproduction was given[3] by Bell Laboratories in America in 1934. The system used three microphones at the centre and extremities, respectively, of the sound stage, and they drove, through three identical channels, three loudspeakers similarly disposed on the reproducing stage. Using such a three-channel system, fairly accurate location was obtained in the azimuth sense, and comparatively effective location was possible in depth. In 1939 R.C.A. used a similar arrangement with three channels for the stereophonic recording of a sound film entitled *Fantasia*.

Since the war demonstrations have been given by Philips at Eindhoven of a system using two microphones placed in an artificial head, the outputs of which supply two widely spaced loudspeakers. In such a system the phase differences of the microphone outputs become of no particular significance when reproduced at the two spaced loudspeakers, but the relative amplitudes of the microphone outputs due to masking by the artificial head are reproduced at the loudspeakers, and provide a measure of spatial location to an observer in front of them. Such an effect, of course, can only take place at the upper frequencies, for example those above about 700 c/s, and any directional effect at the low frequencies is neglected.

More recently several American companies have made recordings using two widely-spaced microphones driving spaced loudspeakers via the medium of twin-track magnetic tapes, and one has issued some discs carrying the two channels, one on the outer half and the other on the inner. This has the serious disadvantage of halving the playing time.

The advent of magnetic tape, providing two synchronous but otherwise independent channels, made possible the commercial exploitation of Blumlein's work, resulting in the 'stereosonic' system described in this paper.

This was demonstrated to members of the profession and representatives of the Press in April, 1955. In April, 1956, a full-scale public demonstration at the Royal Festival Hall was given to an audience of 1 800.

(3) COMPARISON OF BASIC SYSTEMS

In the preceding brief historical review the systems employed fall into three basic types. These are described below.

(3.1) The 'Wavefront' System

If an infinite number of microphones, placed in a vertical plane between the source of sound and the listener, were to be connected to an infinite number of loudspeakers in corresponding positions, clearly the radiated wave could be reproduced unaltered and true binaural audition would be preserved.

A line, rather than an area, of microphones and loudspeakers would give perfect location in a horizontal direction, which is adequate for most purposes since location in a vertical plane seems to be impossible except by inclining the head.

An approximation to this condition, with a corresponding limitation in the accuracy of the results, can be obtained using a finite number of microphones and loudspeakers. Contemporary film sound systems use several channels in this manner. The system described in Reference 3 also used this principle.

Fig. 1 shows the practical arrangement. Detailed tests were

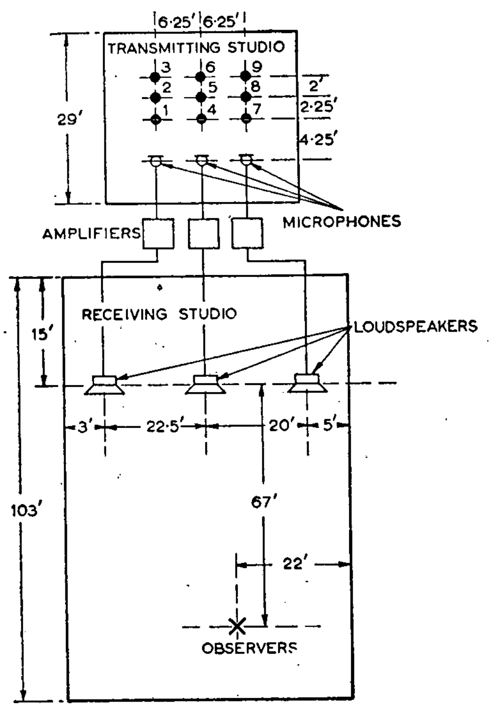

Fig. 1.—Arrangement used by Bell Laboratories in 1934.

made with a loudspeaker located at one of nine positions in the transmitting studio, and the apparent position as estimated by observers in the listening studio corresponded fairly accurately in breadth, and to a reasonable degree in depth, when the observers were not too far from the centre-line. Very-high-fidelity channels were used, and the quality of results obtained by this system was said to be better than anything ever heard prior to that date.

The use of three independent amplifier and loudspeaker channels, although suitable for public use, is quite uneconomical for domestic use. The ultimate simplification of the wavefront system to two channels can, under certain conditions, give pleasing results. For this purpose, two microphones are placed about ten feet apart in front of the sound source, driving two identical amplifying and recording or transmitting systems with two loudspeakers at a similar separation. There is a decided tendency, however, for the sounds to appear to be coming from the two separate speakers, with a gap in between in which the sound is weak.

(3.2) The Reproduction of Correct Sound Pressures at the Ears

A somewhat different approach to that of the wavefront conception is to consider local conditions at the observer's two ears. A system has been devised which relies on a consideration of these differences. The approach is based on the assumption that low-frequency sounds play little or no part in directional localization and that the principal clue to direction is the intensity difference at high frequencies produced by the shadowing effect of the head. It is shown that, if two pressure microphones are used as artificial ears in a dummy head, intensity differences between the outputs occur in accordance with the classic measurements described in Reference 3. It is then demonstrated that, if these outputs are applied to a pair of widely spaced loudspeakers placed symmetrically with respect to an observer, intensity differences do occur at his ears, although they are not as great as the original differences at the microphones. In connection with this system, attempts have been made to show that time differences are less effective than intensity differences in producing impressions of directional localization, and that incorrect time differences can be compensated by modifying the intensity differences. It is then argued that, since the major factor in localization, i.e. intensity differences at high frequencies, is reproduced, natural binaural listening is simulated.

The consideration of local conditions at the ears forms the basis of the 'stereosonic' system, described in detail in Section 4. This springs from the original Blumlein invention. It recognizes that, since each loudspeaker communicates with both ears, differences in magnitude of the sound pressures at the loudspeakers at low frequencies produce phase and not magnitude differences at the ears, since the contributions from the two loudspeakers arrive at slightly differing times. A pair of directional microphones is employed, effectively at a single position, to produce two outputs in phase, differing in amplitude according to the direction of the sound source. These are applied to a pair of spaced loudspeakers so as to produce at the ears a time difference independent of frequency at low frequencies, and an intensity difference using the shadowing effect of the head at high frequencies. It is claimed that this represents the nearest approach yet made to natural listening conditions.

(3.3) Pseudo-Stereophonic Systems

No detailed description will be given of systems in which a single microphone output supplies two or more recording channels at relative levels which are controlled manually in the dubbing stage. Such means are used in the dialogue sequences of wide-screen motion pictures and give an illusion of movement to single voices, etc., but clearly cannot give a simultaneous representation of the direction of a large number of sources as is required for most musical programmes.

(4) THEORY OF THE 'STEREOSONIC' SYSTEM

Before discussing the theory in detail, it will be advisable to review briefly what is known of the mechanism of spatial auditory location in the human being.

(4.1) Mechanism of Angular Localization

Rayleigh[4] in 1896, and Stewart[5] in 1920 both carried out experiments which demonstrated that intensity differences at the ears were insufficient to account for location at the lower frequencies, and that the phase differences had to be taken into account, although above about 1 000 c/s the intensity differences were necessary to avoid ambiguity. In spite of this and the work of others, there has been some reluctance in many quarters to accept the importance of phase differences.

Banister[6] in 1931 and the Medical Research Council[7] in 1932 give prominence to the idea that the time difference rather than phase difference may be the element detected, in which case there is no need for limitation of the effect to the lower frequencies.

In spite of this, and much other and more recent work, no exact explanation can yet be given for the mechanism of sound location. The authors believe that the two principal quantities used by an observer to estimate the angle of arrival of a sound wave are, first, the difference in time of arrival of a wavefront, and secondly, the difference in intensity at the two ears. Of these, by far the more important is the time difference. For sinusoidal waves, a constant time difference at the two ears is equivalent to a relative phase difference proportional to the frequency of the original sound. At low frequencies, where this phase difference is less than, say, π radians, the direction of arrival could be deduced from it. For such a deduction to be possible, the times of passage of each wavefront past both ears must be identifiable to the observer. As the frequency is increased, i.e. the wavefronts follow each other more closely, the point is eventually reached when a wavefront arrives at one ear before the preceding one has reached the other. Since there is nothing to distinguish between the wavefronts, ambiguities arise and it is impossible to interpret the observed time differences uniquely in terms of direction of arrival. These ambiguities start to occur when the ear spacing becomes equal to a half wavelength. At higher frequencies the head becomes an appreciable obstacle and produces an intensity difference, the magnitude of which can be used to assess the direction of arrival. It may also assist in resolving the ambiguities mentioned, and allowing the observed time differences to be interpreted at higher frequencies. Recent work in America[8] shows that the direction of arrival of pure tones can be judged unambiguously with reasonably constant accuracy up to 1 200 c/s. At this frequency the ear spacing approximates to a whole wavelength, suggesting that the first ambiguity due to confusion of wavefronts is overcome, possibly with the assistance of intensity differences. In this case the brain has to choose between two possible directions, widely separated and on opposite sides of the head.

Where the sound waveform is complex, as in the majority of natural cases, the shape of the modulation envelope probably assists in identifying the wavefronts and making possible the interpretation of observed time differences.

With pure tones at high frequencies, when it is necessary to rely on intensity differences to judge direction, accuracy deteriorates, since the amplitude differences at the ears, characteristic of the source position, may be modified by stationary waves and reflections from walls and other obstacles.

(4.1.1) Cochlear Response.

When a sound wave arrives at the ear, the immediate result is the production of an electrical waveform corresponding to that of the instantaneous sound pressure. This is known as the 'cochlear response', and it can be detected by amplifying the potential differences between suitably placed electrodes. The signal, in this form, is unsuitable for analysis by the brain, and its primary purpose seems to be to initiate an electrochemical response in adjacent nerve fibres containing the original information in pulse coded form.

(4.1.2) Action Potential.

This secondary signal is called the 'action potential' and it differs strikingly from the cochlear response. It consists of short pulses, of constant duration and amplitude, apparently occurring at random intervals. The average frequency of these pulses is related to the intensity of the original stimulus rather than its frequency. This latter is determined in some other way, possibly by observing which part of the basilar membrane is responding, although it must be admitted that pitch discrimination is more accurate than can be accounted for by simple resonance. Observations on single nerve fibres[9] show that successive pulses are always separated in time by an integral multiple of the period of the stimulus and that they occur at a particular point on the cochlear response waveform.

When a pulse has been initiated the particular fibre in which it occurs remains inactive for a short interval, known as the 'refractory period', which varies according to the strength of the stimulus. This mechanism limits the pulse rate in a single fibre to a few hundred per second. There are, however, many such nerve fibres associated with each ear, and it is known that at least one fibre will respond at each cycle of the incoming stimulus, up to a frequency of 1 500 c/s, perhaps higher.

Provided, therefore, that the brain can recognize pairs of pulses produced at the two ears by the same sound wave, the original time-difference information is available to it.

(4.1.3) Time-Comparison System.

It is clear that, if the above reasoning is correct, each cycle of the incoming wave gives rise to a pulse of action potential up to frequencies of about 1 500 c/s, and that such pulses occur at a definite point in the cycle. This happens at both ears, and if there were some means of measuring the time differences between their production, the angle of arrival of the sound could be deduced. A theory has been put forward by Jeffress[10] which suggests how this is done.

It must be noted first that transmission of a pulse along a nerve fibre is not simple electric transmission like that along a telephone cable. It is electrochemical in nature; pulses travel without attenuation as they are self-regenerating, and at a relatively slow speed. The rates of transmission as measured do not exceed about 10^4 cm/s, i.e. several times slower than the velocity of sound in air.

The theory then postulates the existence of a nerve combination such that one nerve requires to be stimulated by two others simultaneously before it will respond. Engineers will recognize this combination as being analogous to the 'logical-and gate' ubiquitous in computer circuits. It is suggested that a number of these nerve combinations are spread across the brain and stimulated by pairs of nerve fibres of appropriate length, connected to the two ears. Fig. 2 shows this schematically, in engineering terms, remembering that transmission times are proportional to the physical lengths of the different connections. A response from one of these 'gates' corresponds to a pair of pulses separated by a definite time interval arriving from the two ears. Since the lengths of the nerve fibres are significant, it is clear that the position of the nerve combination in the brain associates it with a definite time interval between stimuli, and thus with a definite direction of arrival of the original sound. In the Figure, a response from the left-hand gate corresponds to a sound wave arriving from the right side of the observer and vice versa.

Some evidence in support of this theory has recently been published. Experiments on cats are described[11] in which clicks.

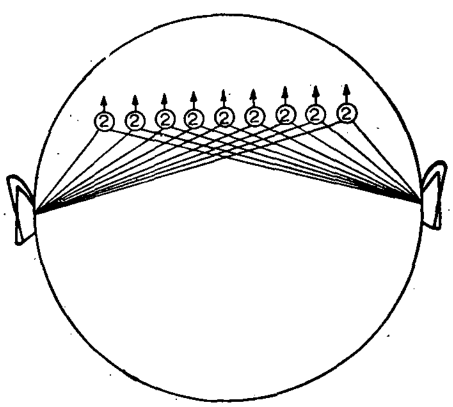

Fig. 2.—Schematic of possible time-comparison system.

separated by a known time interval, were supplied to the two ears independently and the response on the left and right lobes of the brain observed. Maximum response was obtained from the right lobe when the click supplied to the left ear was advanced by a time corresponding to a sound coming from the left of the subject, and vice versa.

(4.1.4) **Application of the Theory.**

The physiology of the brain is insufficiently known to confirm or deny the existence of such a mechanism as that postulated, nevertheless its extreme simplicity makes the idea attractive. The theory forms a useful working basis, and many of the observed facts can be fitted into it. It demonstrates clearly the importance of the first wavefront to arrive at the ears; this has the best chance of triggering a pulse, since the inhibitory mechanism is inoperative. The difficulty of localizing pure tones at high frequency is also shown. Pairs of pulses originating from different cycles of the incoming wave will produce spurious responses from the gates, which the brain will have to discard. It is thought that, under these conditions, intensity differences play an important part. With complex high-frequency waveforms, however, pulses will tend to be grouped round prominent features in the waveform, and this will again enable the time comparison mechanism to operate.

Altogether, the best chance of a clear direction seems to be at those lower frequencies where the action-potential pulses can only be paired in one way to give a time difference corresponding to a possible angle of arrival. This does not mean that the most accurate indication will be obtained at the lowest frequencies. The slower rate at which the waveform crosses the zero-pressure axis and the higher threshold as shown in the Fletcher–Munson curves combine to produce a loss of resolution that appears as an error in estimated direction. Nevertheless the sense of direction is very strong at low frequencies, and the authors do not subscribe to the view that low frequencies play no part in angular localization.

(4.2) The Aim of the 'Stereosonic' System

The aim of the 'stereosonic' system is to reproduce at the ears of the listener, in as large an area as possible in front of a pair of loudspeakers, the same vector sound pressures as he would have experienced by direct listening in a corresponding position in front of the sound stage. In other words, although the spacing between the loudspeakers must, in general, be less than the width of the original sound source (except for some solo instruments and unaccompanied solo voices), if the apparent angular width of the reproduced sound is the same as that subtended at the optimum position for direct listening, the listener will imagine that he has been conveyed to the optimum position for direct listening in the recording studio. So far as is known, the ratio of reverberant to direct sound provides the only clue to distance of the source, except for any possible change in quality due to absorption of high frequencies at a distance. In general, the reverberation of the average domestic room is small compared with that of concert halls used for large musical combinations, and so the sense of distance thus conveyed is not substantially altered.

It is impossible to achieve this aim with perfection, but an attempt is made in the system to reproduce the actual relative phases and thus the inter-aural time differences at the lower frequencies and the relative intensities at the upper frequencies.

It should also be clearly understood that the system is intended to operate only with spaced loudspeakers and thus transmit the spatial effect when listening in free space. The system will not function correctly if the two channels are connected to left and right headphones.

(4.3) Mathematical Theory

(4.3.1) **Vector Pressures at Ears in Direct Listening.**

Fig. 3 shows an actual source of sound S before two ears E_L and E_R, spaced by a distance h in such a position that its

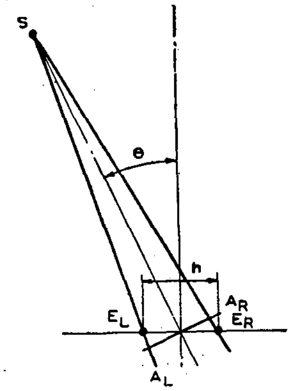

Fig. 3.—Time differential of sound at two ears from oblique source.

direction is at an angle θ to the face-on position. The sound to E_R, however, will travel a distance E_RA_R further than, and that to E_L a distance E_LA_L less than, the average. If v is the velocity of sound in air, the sound will take a time $(h \sin \theta)/2v$ to travel from E_L to A_L. Thus the time interval between the arrivals of sound at the two ears will be $(h \sin \theta)/v$. Hence, if h is small compared with the distance from the source, the magnitude at each ear will be the same, but there will be a phase difference

$$\phi = \frac{\omega h \sin \theta}{v} \quad . \quad . \quad . \quad . \quad . \quad . \quad (1)$$

where ω is 2π times the frequency of the sound wave.

At higher frequencies, when the wavelength is short compared with h, the phase angle will be large and ambiguous but the time delay will be the same. Owing to the masking effect of the head, the magnitude is not subject to accurate calculation but has been determined experimentally.[12, 13]

If these phase differences and magnitudes can be simulated in reproduction, the received sound will appear to come from the same angle θ.

(4.3.2) **Reproducing System.**

For domestic purposes it is generally permissible to use two loudspeakers only, and they must be operated in such a way as to reproduce as nearly as possible the required vector pressures at the ears. Since the bulk of acoustic energy in music and speech occurs below about 700 c/s, it is important that the system should operate effectively at low frequencies.

In Fig. 4, W_L and W_R represent two similar loudspeakers

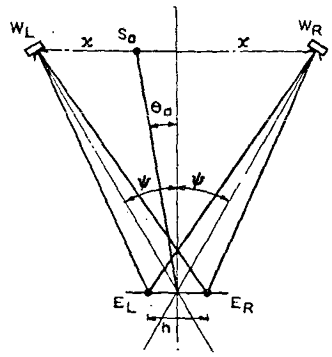

Fig. 4.—Apparent source deduced from the time differential of sound produced at two ears by two loudspeakers operating at different sound levels.

supplied with inputs which are in phase but of different magnitudes. The loudspeakers subtend an angle 2ψ at a centrally placed listener whose ears E_L and E_R are separated by a distance h, which is assumed to be small compared with the loudspeaker distance. From eqn. (1) the phase difference between E_L and E_R for sound coming from either speaker will be $2\omega\mu = (\omega h \sin\psi)/v$, where 2μ is the difference in time of arrival at E_L and E_R.

Let the average instantaneous sound pressure at both ears be $L \sin \omega t$ from W_L and $R \sin \omega t$ from W_R. The sound pressures at each ear can be calculated as follows:

	Average	*At* E_L	*At* E_R
From W_L	$L \sin \omega t$	$L \sin \omega(t + \mu)$	$L \sin \omega(t - \mu)$
From W_R	$R \sin \omega t$	$R \sin \omega(t - \mu)$	$R \sin \omega(t + \mu)$

Total pressure at $E_L = L \sin \omega(t + \mu) + R \sin \omega(t - \mu)$

$$=\sqrt{(L^2+R^2+2LR\cos 2\omega\mu)}\sin\left[\omega t + \arctan\left(\frac{L-R}{L+R}\tan\omega\mu\right)\right]$$

Similarly, the total pressure at E_R is.

$$\sqrt{(L^2+R^2+2LR\cos 2\omega\mu)}\sin\left[\omega t - \arctan\left(\frac{L-R}{L+R}\tan\omega\mu\right)\right]$$

Hence the phase difference between E_L and E_R is ϕ_2,

$$\text{where } \phi_2 = 2\arctan\left(\frac{L-R}{L+R}\tan\omega\mu\right)$$

$$= 2\arctan\left(\frac{L-R}{L+R}\tan\frac{\omega h\sin\psi}{2v}\right)$$

When $\omega\mu$ and $\phi_2 \leqslant \pi/2$

$$\phi_2 = \frac{L-R}{L+R}\frac{\omega h \sin\psi}{v} \quad . \quad . \quad . \quad . \quad (2)$$

Thus, if the loudspeakers are supplied in phase with correct relative amplitudes, the phase difference ϕ_2 at the ears can be made to be the same as that from a sound source at any angle within $\pm\psi$. In addition, if L or R is permitted to become negative, i.e. to have a phase reversal, the apparent direction lies outside the limit of $\pm\psi$ at frequencies where this analysis is valid. This phenomenon is not readily observable in practice possibly because it is accompanied by a certain degree of amplitude cancellation. It has, however, been observed under laboratory conditions.[8]

(4.3.3) **Microphone Systems.**

When Blumlein's first experiments were carried out in 1929–30 the only readily available microphones were pressure-operated and had substantially circular polar diagrams. In order to obtain the required inputs to the loudspeakers the first arrangement to be used consisted of two such pressure microphones separated by about 8 in (i.e. typical distance between ears). The outputs were therefore the same in magnitude, for any source angle, but there was a phase difference given by eqn. (1). To convert this into the required amplitude difference demanded by eqn. (2), an ingenious circuit was used in which the two microphone outputs were first summed and differenced to produce two new voltages in quadrature. The difference voltage was integrated, i.e. multiplied by a factor proportional to $1/\omega$ and rotated by 90°. The sum voltage was uniformly attenuated by a suitable amount. These two modified voltages were again summed and differenced, giving two final voltages, in phase but of different amplitudes, which it can be shown are of such values that, when applied to two loudspeakers, will, according to eqn. (2), produce at the ears the same phase angles as those at the microphones.

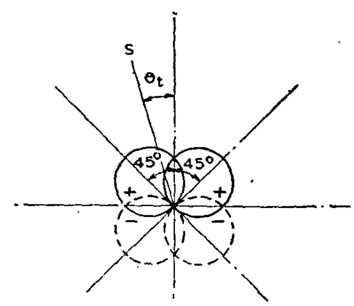

Fig. 5.—Polar characteristics of a pair of velocity microphones at 90°.

Fig. 5 represents two velocity microphones, e.g. ribbon type, with their axes of maximum response at 90°. Such microphones have a response which is proportional to the sine of the angle between the source and the plane of the ribbon. A source S_t is assumed to be at an angle θ_t from the median axis of the microphones. The outputs from the microphones will be in phase at all values of θ_t if the microphones are placed on the same vertical axis but their outputs L and R are given by

$$L \propto \sin(45° + \theta_t)$$
$$R \propto \sin(45° - \theta_t)$$

whence it can be shown that

$$\frac{L-R}{L+R} = \frac{\sin\theta_t}{\cos\theta_t} = \tan\theta_t \quad . \quad . \quad . \quad . \quad (3)$$

(4.3.4) **Performance of Complete System at Low Frequencies.**

If the outputs of the microphones are connected (after suitable amplification) to the loudspeakers, the resultant phase difference between the sounds at the ears will be

$$\phi_2 = \frac{L-R}{L+R}\frac{\omega h \sin\psi}{v} \quad . \quad . \quad . \quad . \quad (2)$$

$$= \tan\theta_t \frac{\omega h \sin\psi}{v} \quad \text{from eqn. (3)}$$

The ears will interpret this as an apparent sound source at an angle θ_a, where $\sin\theta_a = \phi_2 v/\omega h$, from eqn. (1);

i.e.
$$\sin\theta_a = \tan\theta_t \sin\psi \quad . \quad . \quad . \quad . \quad . \quad (4)$$

Fig. 6 shows the apparent angle plotted against the true angle for various values of ψ. It will be seen that, when the listener

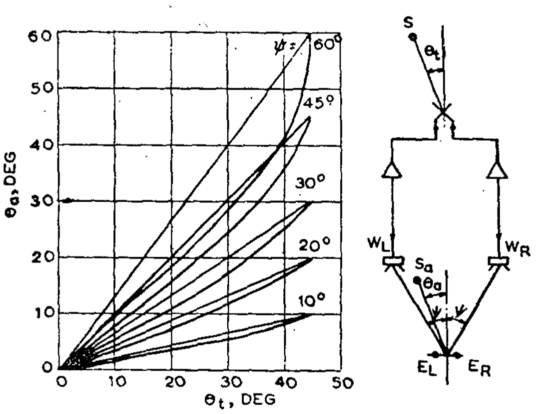

Fig. 6.—Apparent angle plotted against true angle for velocity microphones at 90°.

is at such a distance that the loudspeakers subtend an angle of 120°, the apparent angle is very close to the true angle up to ±35°. When the listener is at other distances from the loudspeakers, but still on the centre line, the apparent source remains at the same fraction of the total loudspeaker spacing, thus showing that a correctly proportioned sound picture is presented although the angular scale has been altered. The angular distortion is such that the apparent source appears to be somewhat nearer the centre than it should be over most of the range.

(4.3.5) Performance at High Frequencies.

If the velocity microphones have a constant polar characteristic at all frequencies, the ratio of voltages applied to the loudspeakers is independent of frequency. At frequencies above about 700 c/s, however, the foregoing analysis will not hold, since the phase angle between the pressures at E_L and E_R will be ambiguous. At these frequencies the masking effect of the head, however, causes the left ear to be more affected by the left loudspeaker than by that on the right and vice versa, and it can be easily demonstrated experimentally that directional information can be conveyed at the higher frequencies.

Subjective tests were made with a number of observers using two loudspeakers supplied with known relative voltages from a source of recorded music. First a filter was inserted passing all frequencies up to 700 c/s from a variety of sources of sound including male and female speech, solo, orchestral and brass-band music. The experiment was repeated using all frequencies above 600 c/s. Quite definite location within about ±2° was obtained in each case, but whereas at low frequencies the angle was in agreement with that predicted from eqns. (1) and (2) for a given loudspeaker ratio, that obtained at high frequencies was greater. The relationship obtained is in agreement with that published by other workers who rely primarily on intensity differences.[15] By introducing a factor of approximately 0·7 into the ratio $(L - R)/(L + R)$ above 700 c/s, the results for high and low frequencies can be brought into line, except for extreme positions of the source.

(4.4) Basis of Recording System

To meet the requirements set out in Section 4.2, therefore, the microphone system can consist of a pair of velocity microphones with mutually perpendicular axes of maximum response. The outputs from these microphones can be summed and differenced and the difference channel subjected to a loss of 3 dB at all frequencies[14] above 700 c/s. These voltages are then again summed and differenced and used to supply two identical loudspeaker channels. If the loudspeakers are equidistant from the observer and subtend an angle of approximately 90° at him, he will receive substantially correct directional information at all frequencies within the spectrum. At greater distances from the loudspeakers the listener will observe the same apparent position of the source, relative to the loudspeakers.

An important feature of a system using two coincident velocity microphones at 90° is that the r.m.s. sum of the outputs of the loudspeakers is constant for a source at a constant distance from the microphones, regardless of its direction. It is this feature which enables the system to reproduce a uniform sound field between the loudspeakers without the tendency to leave a 'hole' in the centre. This latter effect can be very pronounced with the spaced microphone system.

(4.5) Sources of Error

Although the two amplifying channels and the microphones and loudspeakers have been assumed to be distortionless in the above explanation, it is clear that they cannot be perfectly matched in amplitude and phase, particularly the transducers. The effect of the small inevitable differences in frequency response can be calculated from the preceding theory, but such transducers will, in general, introduce phase shifts before amplitude differences become very pronounced.

Calculations show that a phase shift in one channel produces no change in the central position, i.e. when the two channels are at the same level. A phase shift in one channel of as much as 90° can introduce error factors of 2 at small values of θ_t, 1·5 at $\theta_t = 26°$ and 1·2 at $\theta_t = 39°$.

(4.5.1) Effect of Asymmetric Listening Position.

The preceding calculations have assumed an observer symmetrically placed with regard to the loudspeakers and facing the centre line. If the observer turns his head the apparent source will appear to remain in the same position, as in natural listening, but if he moves to either side of the centre line the apparent position will move in the same direction, e.g. from S_a to S'_a in Fig. 7.

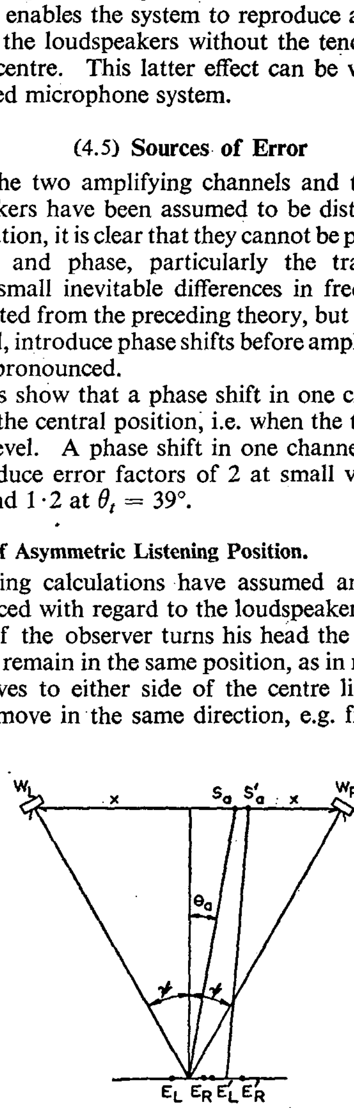

Fig. 7.—Change of position of apparent source with displaced listening position.

The reasons for this are, first, that the loudness from W_R has increased and that from W_L decreased, and their relative phases are altered owing to the difference in the path lengths. Secondly, the angles subtended by the loudspeakers at the observer have also become unequal. The effect of these inequalities makes the calculation of apparent angle exceedingly laborious, but it is possible, by making simplifying assumptions, to appreciate their general effect. Fig. 8 shows the position of the apparent source of Fig. 7 for various listening positions plotted against the positions of the real source with respect to the microphones. In Fig. 8, r is the projection of the observer's

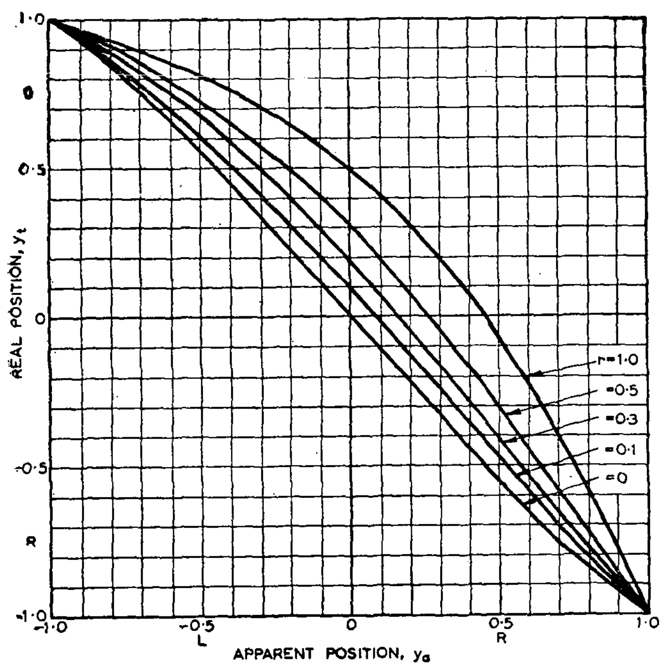

Fig. 8.—Apparent position plotted against true position for asymmetric listening position.

displacement on the loudspeaker base-line, expressed as a fraction of x. The calculations ignore the effect of the path difference on the relative phases at the listener of the sound from W_L and W_R; i.e. it applies where such a path difference is small compared with a wavelength. The curves bear out the statement that the apparent source moves in the direction of the observer.

In order to take account of the effect of path differences it is necessary to fix the scale of the layout shown in Fig. 7, so that a definite phase shift can be associated with any given listening position at a particular frequency. For this purpose let d equal 10 ft to represent a typical domestic arrangement. A listening position is chosen corresponding to $r = 0 \cdot 5$, and the apparent position is plotted against frequency in Fig. 9 for various values of the true angle θ_t. It will be seen that, as the frequency increases from zero, deviations from Fig. 8 are initially small, but they increase rapidly above 100 c/s, reaching a maximum in the region of 200 c/s, beyond which the function approaches its initial value once more before reaching a second maximum at about 600 c/s and again returning to its original value.

The calculations have not been made for frequencies above which the theory given in Section 4.3.2 is valid.

The function may reach a finite maximum, which in all cases is seen to lie outside the loudspeaker base-line, or it may become imaginary. In this case, the phase difference at the ears is so large that it cannot be interpreted in terms of any real angle of arrival. These regions are the ones in which the path difference approximates to an odd number of half wavelengths, the sound from the two loudspeakers arriving substantially out of phase at the two ears.

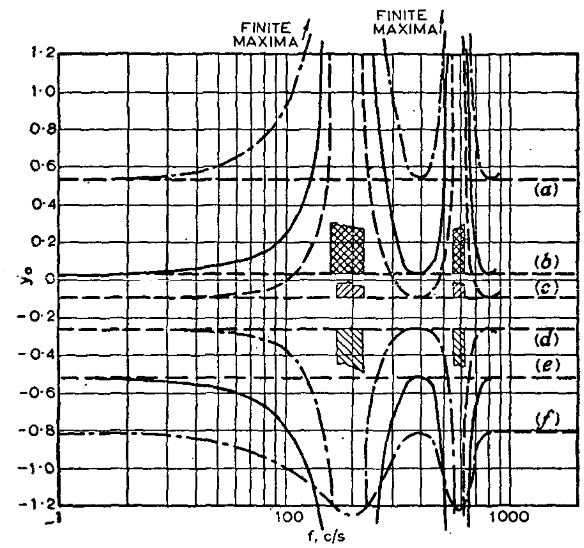

Fig. 9.—Apparent position as a function of frequency for asymmetric listening position.

(a) $\theta_t = 36 \cdot 9°$.
(b) $\theta_t = 18 \cdot 4°$.
(c) $\theta_t = 11 \cdot 3°$.
(d) $\theta_t = 0$.
(e) $\theta_t = -18 \cdot 4°$.
(f) $\theta_t = -36 \cdot 9°$.

Fortunately, at these critical frequencies the resultant magnitude becomes small, thus allowing the remaining sound, which has small phase errors, to predominate. The importance of scale is obvious from the curves; halving all dimensions will allow the system to operate to twice the frequency before the anomalous regions are approached.

The effect of the offset listening position is worst for true angles near the centre, when L and R have about the same amplitude. (The extreme positions, $\theta_t = \pm 45°$, are unaffected, since only one loudspeaker is operating.)

(5) RECORDING EQUIPMENT

(5.1) Microphone Combinations

(5.1.1) Pressure Microphones.

As stated in Section 4.3.3, the earliest experiments were necessarily carried out with pressure microphones. These had substantially spherical polar diagrams—at least, at low frequencies. The problem of producing magnitude differences was solved by giving them a small but definite spacing, summing and differencing their outputs, and integrating the difference vector before recombining to drive the loudspeakers. The significant parameter in such a case is the actual spacing employed. If this exceeds about a quarter wavelength of the highest working frequency, the resultant outputs to the loudspeakers become of opposite polarity and the system breaks down. Small spacing is desirable on this score. However, if the spacing is too small, the difference vector will diminish until it is comparable with the normal random differences in the microphones. Moreover, the amplification in the difference channel has to be increased to compensate, in addition to having a characteristic inversely proportional to frequency as already mentioned. Low-frequency electrical noise introduced by the amplifiers thus becomes the factor limiting the reduction in microphone spacing. A way out of this difficulty was sought by using more than one pair of microphones to cover the required frequency band, but, as can be imagined, this arrangement was cumbersome in practice and complicated theoretically. Pressure microphones were finally abandoned after a composite microphone system had been tried, using two pairs of crystal elements, one pair being spaced 8 cm apart and operating up to 1 000 c/s and the other being spaced 1 cm apart and operating from 1 000 c/s upwards.

(5.1.2) **Velocity Microphones.**

By this time, ribbon microphones were becoming available. As is well known, these have a cosine-law polar characteristic in the horizontal plane. A pair of these can be used in two ways to produce the required outputs for driving a pair of loudspeakers. For example, they may be used with one microphone having its axis directed to the centre of the sound stage and the other as close as possible but with its axis at right angles. In this condition the outputs from the microphones will be in phase and invariant with frequency (in so far as their characteristics are flat), and will correspond to the sum and difference vectors described in Section 5.1.1 after manipulation but before recombination to drive the loudspeakers.

Alternatively, they may have their axes equally inclined to the centre of the sound stage. In this condition their outputs are suitable for direct amplification and reproduction by a pair of spaced loudspeakers. Such outputs may be considered to be of 'left and right' type rather than 'sum and difference', as in the previous instance.

In either case it will be noted that the operation of integration that had to be performed in the case of pressure microphones is unnecessary. Consequently, although ribbon microphones are generally less sensitive than pressure types, the overall signal/noise ratio of the system is improved.

(5.1.3) **Other Types of Microphones.**

It will be apparent that any pair of microphones having a polar characteristic other than circular can be used in this way to produce 'left and right' type signals of different amplitudes at a pair of loudspeakers. Whether these combinations are of value depends on the uniformity with which their polar characteristics can be maintained with frequency, as well as the degree within which the microphones can be matched in amplitude and phase.

Combinations of microphones having dissimilar polar characteristics may also be employed to produce 'sum and difference' type outputs, provided that their amplitude and phase characteristics are well matched. Combinations of polar diagrams which appear to show promise are those of circular with cosine, which would give the same directional response as the pressure microphones used in the early experiments, and cardioid with cosine, which would give substantially a one-sided version of the same thing.

(5.1.4) **Current Practice.**

The combination of two cosine microphones has so much to recommend it compared with other known arrangements that the principal effort has been concentrated on its improvement. If a pair of ribbon microphones is used, or even a specially-designed double-element ribbon type, the difficulties encountered are still quite serious. At low frequencies the fundamental resonances of the two ribbons cause relative phase shifts unless these are adjusted to occur at the same frequency and to have the same degree of damping, whilst at high frequencies the ribbon microphone having the theoretical cosine-law polar diagram has yet to be designed. Departure from constancy of polar diagram with frequency will upset the requirement that the amplitude differences for any given angle of arrival shall be independent of frequency. Nevertheless, with all their limitations, some very acceptable recordings have been made using ribbon types.

Within recent years the condenser microphone has undergone considerable development, enabling directional characteristics to be obtained from a double-diaphragm element of small size. Since these elements are substantially free from mechanical resonances within the audible spectrum, their amplitude and phase characteristics show great uniformity. This is fortunate as the polar characteristics show some variation from sample to sample, particularly at low frequencies. Selection has thus to be principally on the basis of polar response. Individual adjustments to match their sensitivities can easily be made. Pairs of these microphones, mounted with their elements as close together as possible in a single cylindrical holder, have been used with some success. Like all arrangements with one element above the other it is necessary to ensure that the common axis is perpendicular to the plane in which the various sounds lie, otherwise the vector relationships are upset.

(5.2) Studio Techniques

The problems encountered in making a 'stereosonic' record are not all of an engineering nature. The finished record must have a pleasing tonal balance, and the apparent spatial distribution of the instruments, if not of prime importance, must not sound unnatural. Using a single pair of microphones, it is often extremely difficult to satisfy the required conditions. In single-channel recording, good results have sometimes been achieved using only one microphone, but the occasions when this is possible are the exception rather than the rule. Generally, in order to achieve a proper balance, multi-microphone techniques have to be adopted. For obvious reasons this cannot be done when the direction of the various sources is significant. It is therefore necessary to rely on correct disposition of the performers in the recording studio to achieve the required tonal balance. In this connection, it may be remarked that the double-cosine combination has two working arcs each 90° wide on opposite sides of the common axis. This has been found useful in practice, particularly for large choral works, in which the orchestra has been arranged on the one side, with choir and soloists on the other. The adoption of such unusual layouts, however, often makes the task of the conductor more difficult, since he may not be able to see all the performers without turning round. Considerations of this sort, which may be termed the technicalities of musical performance, often preclude the use of the purely engineering solution and make the task of the recording engineer even more complicated.

Under difficult circumstances it is possible to envisage the need for more than one microphone pair, but it is realized that the use of additional microphones may introduce as many problems as it solves.

(5.3) Microphone Amplifiers and Equalizers

Amplifier design follows conventional lines. It is, of course, essential that pairs of amplifiers used in this type of recording have accurately matched frequency and phase characteristics, and that their gain be constant over long periods. In general, however, modern designs employing negative feedback have no difficulty in achieving the required consistency. The same requirement for accurate matching applies to any microphone equalizers that are used, e.g. for bass correction when working in close proximity to an artist.

(5.4) Mixers

As stated in Section 5.2, the use of a single microphone pair for stereophonic recording is almost mandatory. It is nevertheless possible to foresee circumstances in which it may be necessary to use a second pair or to introduce extraneous effects, Some form of mixer is therefore necessary, although this will generally be a simple device compared with those used in conventional recording, which may have to cater for relatively large numbers of microphone channels.

A stereophonic mixer in current use does permit the use of two microphone pairs with independent level controls. It also includes means for providing the correction at high frequencies referred to in Section 4.4. This requires that the signals, if not of the sum-and-difference type, shall have been subjected to a sum-and-difference operation.

A further facility is provided which gives an additional degree of flexibility in the recording studios. This consists of a differential control permitting the relative amplification of the 'sum and difference' channels to be altered over a limited range. This control allows the apparent angle of the total reproduced sound to be modified and can be used to good effect in several ways. The microphones may have to be withdrawn from the artists in order to get more 'ambience' into the recording. This will reduce the apparent stage width, but the full width can be restored by suitably attenuating the sum channel relative to the difference. Similarly, if the need for great 'presence' calls for a very close microphone position, the reproduction may cause a solo instrument to sound much too large, and this can be corrected by attenuating the difference channel relative to the sum.

The signals are finally subjected to a second sum-and-difference operation to convert them to 'left and right' type for application to the input of a twin-track tape recorder. The replay amplifiers of the machine drive two matched loudspeakers suitably adapted for stereophonic monitoring.

A simplified diagram showing the operations performed in a typical recording channel is given in Fig. 10.

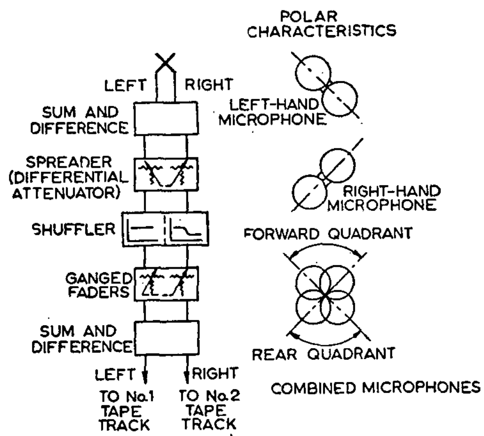

Fig. 10.—Functional schematic of recording system.

An interesting feature of the control system is the peak-level indicator. The use of two such meters, one on each channel, is unsatisfactory owing to the difficulty of observing them simultaneously. In this equipment a peak-level indicator circuit,[16] with the ability to measure the true level of short transients, is connected to each recording channel, and the meter on the mixer panel can be switched to either channel at will. Alternatively, for general use, a circuit is provided whereby the meter is electronically switched to whichever channel has the higher level. The operator then uses his main gain control in the ordinary way to avoid overloading either channel on the tape.

(5.5) Magnetic-Tape Recording Machines

In normal record production it is the practice, irrespective of the final form of the commercial article, to make master recordings on magnetic tape for convenience of play-back, editing, etc. These advantages apply equally to stereophonic recording. The system described employs professional machines of high quality, modified by the addition of twin-track heads and twin-head amplifier channels.

A standard tape speed of 15 in/sec has been chosen for two reasons:

(*a*) Satisfactory standard of master quality can be obtained at this speed.

(*b*) The majority of single musical works can be recorded within the full 11 in-diameter tape spools and thus simplify the copying process.

The record heads are provided with bias from a common oscillator to avoid heterodyne beats due to residual crosstalk.

(6) RECORDING AND REPRODUCING METHODS

(6.1) Choice of Media

It is essential that the recording of the two channels should be made on a medium which allows the two tracks to be permanently linked together so that the correct phase relation between the signals is maintained throughout any copying process and replay operation.

In 1931 when Blumlein conceived the basic idea of his system, the only recording medium capable of giving reasonable results was the wax-cut disc with its electrolytic copying process. He therefore proposed applying the two signals from the microphone system to the complex transducer so arranged as to cut a single groove. If the axes of movement of the transducer armatures carrying left and right signals are inclined at 45° to the surface of the wax and are thus at 90° to each other, the resulting lateral cut can be arranged to represent the sum of the two channels and the 'hill and dale' to represent the difference. Such a disc could be played on a normal gramophone to give the equivalent reproduction of a single-channel recording. The early experimental samples of the complex-cut disc suffered from considerable interaction at the higher frequencies and from excessive background noise. The fine-groove technique may simplify the problem, but to produce a complex-cut disc to the standards demanded by the modern record is very difficult, and the problem has not yet been solved satisfactorily.

In the United States stereophonic disc records have been marketed in which the lateral-cut tracks are disposed in two concentric bands. This record is replayed with two pick-up heads mounted on a common arm. The operation of locating both pick-up styli in their correct grooves is a delicate one. The frequency response and distortion characteristic of the two tracks will differ quite appreciably when there is a large difference between the track diameters.

There have been various suggestions for modifying the frequency band of the signals of one of the channels, so that both channels can be recorded on one groove by means of a single transducer. One such method proposed by Livy[17] suggests that one set of signals is arranged to modulate a carrier, then to select the lower sideband and apply this, together with the normal signal band of the other channel, to a wide-range cutter head. Reproduction is effected by using a wide-range pick-up, separating the two frequency bands and applying the upper band to modulate a carrier of the same frequency as that used for recording. The lower sideband produced by this modulator contains the same frequency components as the original signal. Livy also proposed to record the carrier frequency on the disc so that it would be used during reproducing to control the frequency of the local oscillator. This double-bandwidth system demands a very high performance in terms of frequency response, and thus if a frequency range up to 10 kc/s be required, the full bandwidth of the system will be at least 20 kc/s. This wide bandwidth is very difficult to attain, particularly at the inner groove diameters.

If recordings are made side by side on a continuous film, the synchronization problems are largely overcome, and while it is recognized that optical methods have been used, the recent development of the magnetic-tape process makes it a more attractive system.

(6.1.1) **Crosstalk.**

The overall crosstalk between channels should be better than 30 dB at all frequencies. In the production of a commercial tape record there are at least four stages where crosstalk can take place:

Master recording.
Master replay.
Copy recording, commercial tape record.
Copy replay.

If copy masters are used two more stages of crosstalk are introduced. In practice, it has been found that the same amount of crosstalk is introduced during the recording process as during the reproducing process.

In well-designed heads a crosstalk of -50 dB can be attained, and if the crosstalk at the various copying stages can be added arithmetically the overall figure for a commercial tape record will be -38 dB, or if a copy master is used, -34 dB. If it is required to replay either twin-track or single-channel half-track tape records from the same head system, the crosstalk of this replay head must be better than -55 dB, and preferably better than -60 dB, in the 1 000–3 000 c/s region.

Crosstalk can take place owing to the mutual inductance of the heads and owing to leakage fields from one track to the head on the other track. The mutual inductance can be reduced by placing magnetic screens between the heads and separated from them by brass or aluminium spacers. These screens should be large enough to shield the magnetic circuit and the windings. The usual back gap should be eliminated, and the head dimension should be kept as small as possible.

A useful reduction of crosstalk can be achieved by either series or parallel cross-connection of the windings, so that a small signal from one head is injected in anti-phase to the leakage signal on the other head. If carefully adjusted the series cross-connection can reduce the residual crosstalk by at least 10 dB.

Appreciable crosstalk may occur if the front edge of the outer magnetic screening shield is too close to the magnetic tape. The flux from one track enters the shield and then passes to the magnetic yoke of the other head operating on the other track. A clearance between the tape and the shield of 0·2 in is sufficient to avoid this effect.

(6.2) Track Standards

The early experimental stereophonic magnetic-tape recordings were made with the heads displaced by $2\frac{11}{16}$ in. This enabled the conventional half-track heads to be used on the existing recording and replay machines, and the separation between the heads was sufficient to make crosstalk negligible. When copy tapes were made for issue as commercial records it was soon obvious that the use of in-line heads would make for easier acceptance as an international standard for domestic and commercial tape records. It was also decided to change to in-line heads on the master recordings, so that the process of editing was simplified. Furthermore, the use of in-line heads eliminates phase shift between the signals on the two tracks owing to small variations of elasticity of the tape along its length.

Standardization has been effected on the designation of the tracks, which is as follows:

If the tape moves from left to right and with the active side facing away from the observer the top track shall be designated *No. 1 track* and shall carry the recording for the left-hand channel as viewed from the audience. The bottom track shall be designated *No. 2 track* and shall carry the recording of the right-hand channel.

The replay response characteristic (100 microsec) and the track dimensions are in accordance with Amendment No. 1 to B.S. 1568: 1953, relating to tape speeds of $7\frac{1}{2}$ in/sec. The track dimensions are as shown in Fig. 11.

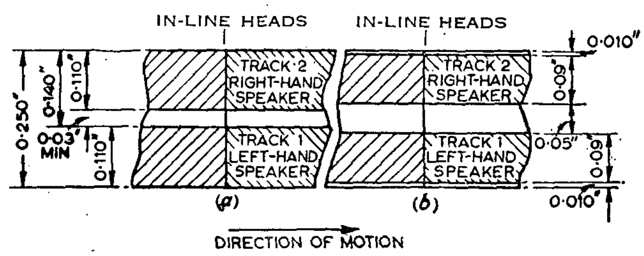

Fig. 11.—Track standards.
(*a*) Record-head track.
(*b*) Replay-head track.
Views looking on coated side of tape.

(6.3) General Requirements for Domestic Operation

The tape transport mechanism may be of the conventional type, but the associated twin-track magnetic replay head must conform to the standards laid down above. Since the signals from the two tracks on the tape are left and right, it is essential that the gains of the two replay channels should be closely matched, and for this purpose preset controls should be provided for initial adjustment. It is a further requirement that the equality of gain should be maintained for all settings of the main coupled gain control. Large relative phase shifts in the two channels should be avoided, particularly at low frequencies.

The underlying principle of the system demands that the loudspeakers should possess, as far as possible, uniform polar response characteristics in the horizontal plane at all frequencies. It has been suggested that departure from uniformity can be an advantage in maintaining the apparent position of the source for a wide range of listening positions. Such non-uniform polar characteristics cannot reasonably be attained except at high frequencies. Even if it were possible to extend the directional characteristics down to the low frequencies, the beneficial results claimed would not be realized on account of the path differences, as outlined in Section 4.5.1. At the high frequencies non-uniform polar characteristics have the disadvantage that the overall tonal balance will vary with the listener's position. Furthermore, the directional response will emphasize any background hiss and will tend to identify the two loudspeakers as separate sound sources and thus interfere with the illusion. In addition, the domestic user might have difficulty in positioning the loudspeakers with sufficient accuracy.

If cone loudspeakers are used the rear radiation should be suppressed, and this can be conveniently done by using a closed box baffle. Such baffles can have quite small volume, and they can be designed to occupy a very small floor area. Any loss of bass response due to the small enclosed volume can be compensated electrically in the power amplifiers.

(6.3.1) **A Commercial Model.**

One form of domestic reproducer has been described by Smith and Martin,[18] but for the sake of completeness a brief general description may be of interest. The complete machine consists of two consoles each containing a loudspeaker group and a power amplifier. One of these consoles also contains a tape deck with its associated head amplifiers, tone and gain controls. Fig. 12 is a schematic. Bass and treble ganged tone

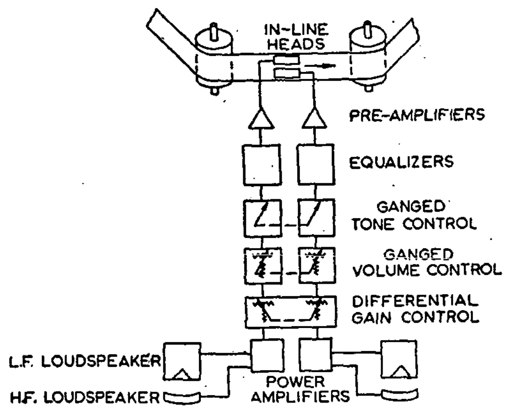

Fig. 12.—Functional schematic of reproducer.

controls are provided, in order to allow some adjustment for the different acoustic conditions which may be met.

In order to maintain uniformity between the two channels the controls are of the stepped-switch type. The main gain control consists of two ganged switches with 12 steps of 3dB each. There is a differential gain control on the panel to take up any slight drift in amplifier gains and to allow for asymmetry in the listening room. A preset continuous control on the amplifier chassis balances the gain during factory adjustment. The power amplifiers are rated at a peak output level of 10 watts.

In order to give the best horizontal polar distribution the elliptical cone loudspeakers are mounted in closed rigid box baffles of $3\frac{1}{2}$ ft^3 capacity, the major axis of the cone ellipse being vertical. The frequency range of this speaker is limited at the upper end to about 5000c/s, in which region the electrostatic speaker takes over and continues beyond 15kc/s. The electrostatic speaker consists of a curved metal back plate 24in long by $1\frac{1}{4}$in wide, over which is laid a membrane metallized on the side away from the back plate. This speaker is driven from a separate amplifier and is supplied with a bias potential of 300 volts.

(6.3.2) Domestic Listening Conditions.

In general, 'stereosonic' tape records are balanced with regard to tonal quality and perspective, for reproduction in the average home lounge in which the reverberation time is of the order of 0·5sec or less. This reverberation period is short compared with that in the studio or concert hall in which the recordings will have been made; it is not likely to interfere with the listener's impression of the original studio conveyed to him by the stereophonic effect; and there will be no confusion due to the two distinct reverberations which might blur the detail. Provided that the walls and the floor are reasonably absorbent it is found that, in spite of errors due to asymmetry, one can move about over a large floor area in front of the loudspeakers without losing the major benefits of this type of reproduction. As stated earlier in the paper, the optimum position for listening is at the apex of an equilateral triangle with the base-line formed by the line joining the two loudspeakers; this is the arrangement used for monitoring during recording. In the average room it is satisfactory to set the loudspeakers 10 or 12ft apart, but in a restricted space a very fair performance can be obtained down to distances of as little as 4 or 5ft.

(6.4) Operation of the System in Large Halls

Special difficulties attend the reproduction of 'stereosonic' records on a large scale, apart from the general fact that, like all other types of record, they are balanced primarily for domestic conditions.

The most serious errors are caused by increased path differences. These occur, as outlined in Section 4.5, down to lower frequencies and over a much greater proportion of the listening area than is the case when operating at the scale for which the system was designed.

In a small room reflections from the walls, etc., decay rapidly and follow each other at such short intervals that the resulting sounds appear to the listener to coalesce. In large halls the relatively undamped primary reflections from the surroundings arrive at the listeners' ears at intervals sufficiently great to cause an appreciable distortion of the sound picture.

Ideally the system requires that the loudspeakers shall be uniformly radiating point sources. In the domestic case a reasonable approximation to this can be achieved because single small units can be constructed to handle the required power. In large-scale reproduction, multiple units or single units of appreciable size are necessary; this departure from the ideal can cause a depreciation of the stereophonic definition.

(7) ACKNOWLEDGMENTS

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(9) APPENDIX

(9.1) Change of Position of Apparent Source under Asymmetric Listening Conditions

In the asymmetric case there are three principal reasons for departure from the simple theory applicable to central listening. These are as follows:

(*a*) Operation of the inverse-square law to alter the magnitudes of the sound pressures from the loudspeakers at the listener.
(*b*) The phase difference due to the unequal path lengths from the loudspeakers to the listener.
(*c*) The unequal angles subtended by the loudspeakers at the listener.

The method of calculation of apparent position is shown below. Fig. 13 shows diagrammatically a source S on a sound stage PQ, towards which a microphone pair M is directed. The microphone outputs are reproduced on a similar stage by a pair

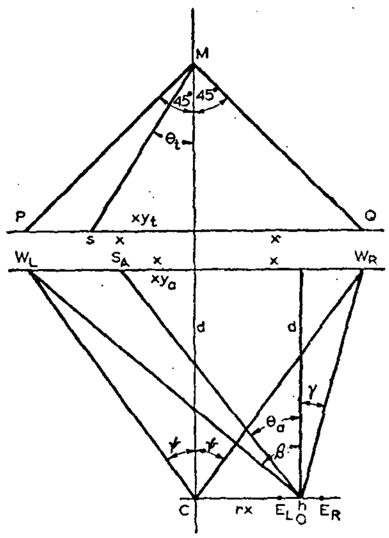

Fig. 13.—Position of apparent source in the asymmetric case.

of loudspeakers, W_L, W_R, assumed to have uniform polar characteristics over the frequency range considered. The virtual image S_a thus produced is observed by a listener O displaced from the central position C.

Then let

$2x$ = Distance between loudspeakers W_L and W_R.
d = Distance between listener O and base-line.
rx = Displacement of listener parallel to base-line.
y_a = Fractional distance of apparent source S from central position.
y_t = Fractional distance of real source from central position.
L, R = Magnitudes of sound pressures due to W_L and W_R at C.
β, γ = Angles subtended at listener by W_L and W_R.
τ = Difference in time of arrival of sounds from W_L and W_R at O due to path difference.
μ_L, μ_R = Time differences at ears E_L and E_R relative to O of sounds from W_L and W_R.
ϕ_L, ϕ_R = Resultant phase relative to O of sound at left and right ears.
θ_a = Apparent angle of arrival of sound at O.
θ_t = True angle of source with respect to microphones for magnitudes L and R from the loudspeakers.

Then:

$$OW_L = d\sqrt{[(1+r)^2 \tan^2\psi + 1]}$$

$$OW_R = d\sqrt{[(1-r)^2 \tan^2\psi + 1]}$$

$$\tau = \frac{OW_L - OW_R}{v}$$

$$= \frac{d}{v}\{\sqrt{[(1+r)^2 \tan^2\psi + 1]} - \sqrt{[(1-r)^2 \tan^2\psi + 1]}\}$$

$$\mu_L = \frac{h \sin\beta}{2v}$$

$$\mu_R = \frac{h \sin\gamma}{2v}$$

Relative pressures at the listener's ears are as follows:

	At E_L	At E_R
From W_L	$\dfrac{L \sin\omega(t - \tau + \mu_L)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}}$	$\dfrac{L \sin\omega\,(t - \tau - \mu_L)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}}$
From W_R	$\dfrac{R \sin\omega(t - \mu_R)}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}$	$\dfrac{R \sin\omega(t + \mu_R)}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}$

The resultant phases at the ears relative to O, derived from the above, are:

$$\phi_L = \arctan\left\{\frac{\dfrac{L \sin\omega(\mu_L - \tau)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}} - \dfrac{R \sin\omega\mu_R}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}{\dfrac{L \cos\omega(\mu_L - \tau)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}} + \dfrac{R \cos\omega\mu_R}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}\right\}$$

$$\phi_R = -\arctan\left\{\frac{\dfrac{L \sin\omega(\mu_L + \tau)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}} - \dfrac{R \sin\omega\mu_R}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}{\dfrac{L \cos\omega(\mu_L + \tau)}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}} + \dfrac{R \cos\omega\mu_R}{\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}\right\}$$

whence it can be shown that the total resultant phase difference at the ears is:

$$(\phi_L - \phi_R) = \arctan\left\{\frac{\dfrac{L^2 \sin 2\omega\mu_L}{(1+r)^2 \tan^2\psi + 1} - \dfrac{R^2 \sin 2\omega\mu_R}{(1-r)^2 \tan^2\psi + 1} + \dfrac{LR2 \sin\omega(\mu_L - \mu_R) \cos\omega\tau}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}{\dfrac{L^2 \cos 2\omega\mu_L}{(1+r)^2 \tan^2\psi + 1} + \dfrac{R^2 \cos 2\omega\mu_R}{(1-r)^2 \tan^2\psi + 1} + \dfrac{LR2 \cos\omega(\mu_L - \mu_R) \cos\omega\tau}{\sqrt{[(1+r)^2 \tan^2\psi + 1]}\sqrt{[(1-r)^2 \tan^2\psi + 1]}}}\right\}$$

Making the approximation, on which the system is based, that $\omega\mu_L$ and $\omega\mu_R$ are small, thus implying also that ϕ_L and ϕ_R are small:

$$(\phi_L - \phi_R) \to \tan(\phi_L - \phi_R) \to \left\{ \frac{\dfrac{L^2 2\omega\mu_L}{(1+r)^2\tan^2\psi+1} - \dfrac{R^2 2\omega\mu_R}{(1-r)^2\tan^2\psi+1} + \dfrac{LR2\omega(\mu_L-\mu_R)\cos\omega\tau}{\sqrt{[(1+r)^2\tan^2\psi+1]}\sqrt{[(1-r)^2\tan^2\psi+1]}}}{\dfrac{L^2}{(1+r)^2\tan^2\psi+1} + \dfrac{R^2}{(1-r)^2\tan^2\psi+1} + \dfrac{LR2\cos\omega\tau}{\sqrt{[(1+r)^2\tan^2\psi+1]}\sqrt{[(1-r)^2\tan^2\psi+1]}}} \right\}$$

But

$$(\phi_L - \phi_R) = \frac{\omega h \sin\theta_a}{v}$$

Therefore

$$\theta_a = \arcsin\left[\frac{v}{\omega h}(\phi_L - \phi_R)\right]$$

Substituting for μ_L and μ_R in the expression for $(\phi_L - \phi_R)$:

$$\theta_a = \arcsin\left\{ \frac{\dfrac{(L/R)\sin\beta}{(1+r)^2\tan^2\psi+1} - \dfrac{(R/L)\sin\gamma}{(1-r)^2\tan^2\psi+1} + \dfrac{(\sin\beta - \sin\gamma)\cos\omega\tau}{\sqrt{[(1+r)^2\tan^2\psi+1]}\sqrt{[(1-r)^2\tan^2\psi+1]}}}{\dfrac{L/R}{(1+r)^2\tan^2\psi+1} + \dfrac{R/L}{(1-r)^2\tan^2\psi+1} + \dfrac{2\cos\omega\tau}{\sqrt{[(1+r)^2\tan^2\psi+1]}\sqrt{[(1-r)^2\tan^2\psi+1]}}} \right\}$$

The fractional distance of the apparent source from the central position is then given by

$$y_a = \frac{d\tan\theta_a}{x} - r = \frac{\tan\theta_a}{\tan\psi} - r$$

Similarly, the fractional distance of the real source from the central position before the microphones is given by

$$y_t = \frac{\tan\theta_t}{\tan 45^\circ} = \tan\theta_t$$

45° being half the working angle of the microphones.

DISCUSSION BEFORE THE RADIO AND TELECOMMUNICATION SECTION, 20TH FEBRUARY, 1957

Mr. J. Moir: I am professionally interested in the reproduction of sound films in which three-channel stereophonic reproducer systems are common, but I have an interest in domestic stereophonic systems as an enthusiastic amateur.

I have been playing these stereosonic magnetic-tape recordings for some months, and my only comment on their performance is that the authors have understated the advantages. The spatial distribution and the impression of size that a good stereophonic system gives to an orchestra is most effective and most important. The individual instruments and sections of the orchestra are clearly separated, which gives a degree of clarity and definition to the performance that cannot be obtained from a monaural system, however much the amplitude distortions are reduced. A monaural system with 0·5% distortion does not sound so 'clean' as a stereophonic system with 5% distortion.

It has been appreciated for the last 25 years that the public will not accept a reproduction of an orchestra with the full original frequency range or volume range. Many reasons have been advanced, all with some degree of truth, but there is no doubt that the use of monaural reproducer systems is a major factor in this preference for range restriction.

It has often been said that the advantages of a stereophonic system are too subtle for the ordinary members of the public to appreciate, but my experience tends to the opposite point of view. Fifteen or twenty of my friends have heard these stereophonic tapes run on my domestic equipment and have had the opportunity of making an immediate comparison with a long-playing recording of the same work run monaurally. After about ten seconds of monaural reproduction the invariable comment is on the flatness.

Though I am fairly enthusiastic about these stereophonic recordings I am not so enthusiastic about the authors' explanation of how the system works. They suggest that their two-channel system gives stereophonic results because it allows the brain to compare the amplitude of the signals at the two ears, the signal at each ear being the vector sum of signals reaching the ear from both loudspeakers. They support this suggestion with an elegant mathematical analysis in which I can find no questionable step, but in spite of this, I do not believe that the stereophonic results can be explained in this way or so simply.

A source emitting a single sound pulse in the studio will result in two pulses reaching each ear, but I would suggest that only the first pulse at each ear reaches the brain. The nervous system transmits amplitude indications as a pulse code, pulse-repetition rate being related to sound intensity. However, the maximum pulse rate cannot exceed 1 000 pulses/sec, while 300–500 pulses/sec is the more usual maximum. This pulse-rate limitation is imposed by the maximum rate at which the transmitting-end cells can be chemically recharged by the blood stream. Thus, after a discharge, the nerve cannot transmit a second signal for a time which is always greater than 1 millisec. However, the head dimensions are such that the second acoustic pulse will appear from the remote loudspeaker with a time interval which is always less than 0·6 millisec. At first sight, it would appear that no stereophonic effects could be produced if the sound source emitted short pulses, though in fact the stereophonic effect is most marked.

Our own work indicates that there are clues to source position distributed over the whole of the audio-frequency range though the majority of the information is contained in the frequency range above 500 c/s.

Dr. E. C. Cherry: During the course of a recent informal meeting* I expressed the opinion that stereophony, using two-loudspeaker systems, is impossible. In view of the authors' demonstration, it seems behoven upon me to explain myself further.

I repeat that it is impossible, but the practice of engineering continually involves the solution of impossible tasks; it is a regular search for compromise. In the present context, the theoretical impossibility arises from several reasons, not the least important being that the acoustic environments in the recording studio and in the home or concert hall are simply not reciprocal. Why, it may be pertinent to ask, did the demonstration take place in the Great Hall of Northampton Polytechnic and not, more conventionally, in the Lecture Theatre? Again, the acoustic conditions may be adjusted to some extent, but they cannot be suitable for all listening situations; neither can the listener be free to move far from a centre-line or other given contour without the apparent directive qualities vanishing.

The binaural directive location of a sound, in real-life environments, is not yet understood; more fundamentally, the psychological 'projection' of sounds to lie 'outside' our heads is not understood, nor are the physical controlling factors. Binaural phase (time) difference and amplitude difference, by themselves, do not account for such projection and angular discrimination.

The brain makes great use of the differences between the signals reaching the two ear portals, and there are several differences other than time and amplitude (e.g. effects of room acoustics and the listener's experience of typical environment effects; head-turning effects; head diffraction; the varied angle of arrival of wavefronts, and impedance mismatch at the ear portals). I agree with Mr. Moir that Fourier-type analysis, and theory such as that given in the paper, are best laid aside when trying to understand stereophonic hearing. Statistical analysis rather than Fourier analysis may be the brain's function.

My remarks do not purport to be disparaging in the least, but I wish merely to stress the need for much more psycho-physical study, not only of binaural hearing, but of other sensory functions in communication. Perhaps then we may understand results such as those which were so well demonstrated by the authors.

Mr. G. Millington: In the early days of electrical recording I did some work under Mr. P. W. Willans on high-fidelity recording of the piano. We found that, with a single microphone in a fixed position, it was extremely difficult to avoid an uneven response in which some notes had a dully wooden sound while others were unnaturally brilliant. When one note was cured by moving the position of the microphone, it was usually at the expense of another in a different part of the register.

I should like to ask the authors whether the placing of the microphone system is similarly critical when using two microphones at the same position but orientated in different directions. As far as I could tell, the recording given was perfect in this respect, and was as fine an example of piano recording as I have heard.

Mr. A. M. Thornton: I assume that the system is intended to have a great future in high-quality and more realistic recording, and that ultimately we shall have this sort of stereophonic quality in our homes.

In the case of the single instrument—the piano—the quality and fidelity were the most satisfying that I have experienced, but when it came to the orchestral item 'Peter and the Wolf' I was very disconcerted. The notes alternated from one side of the stage to the other exactly like the sound of the ping-pong ball in the table-tennis game previously demonstrated. I realized then that that is how the conductor must hear the music, i.e. quite differently from the vast majority of the audience who are sitting some distance away. This raises the interesting point: Does the conductor endeavour to produce a good effect at, say, the middle of the hall, or is he the only person really hearing the music as it should be heard?

In using the system for producing recordings in the home, should the listener be virtually transferred from a position in the auditorium to the conductor's rostrum, or should the aim be to 'seat' him effectively in a position, say, centrally five rows from the front?

Major W. V. G. Fuge: In the demonstration with two people speaking at the same time and recorded stereophonically, it seemed possible to listen at will to either conversation by mentally concentrating without moving the head, so that a mental process could switch the attention from one voice to the other.

Is that the advantage of the stereophonic recording, i.e. that different people listening to the same music can concentrate on any particular instrument they wish from time to time and switch their attention to what pleases them?

Mr. F. Oakes: Mr. Thornton asked whether the stereosonic recording should reproduce what the conductor hears from the rostrum. The answer is in the negative, because the balance of instruments is adjusted during rehearsal to provide the correct sound in the auditorium. Depending on the acoustic properties of the concert hall, this may well mean that what the conductor hears is far from acceptable, whilst correct placement of the microphones further from the orchestra will produce a satisfactory balance, such as would be enjoyed from a good seat in the auditorium.

Mr. J. K. Webb: To anyone who has been privileged to hear a demonstration of the authors' stereosonic reproducing system in an average-size living room, 'revelation' is about the only word adequately to describe it. This scheme is surely a mutation which is bound to set a completely new standard and inevitably expose the deficiencies of 'monosonic' recording.

It must always be borne in mind, when considering domestic sound reproduction, that one cannot contemplate the enormous range of sound levels with which the authors have battered our eardrums in their demonstration, however commendable this may be from the technical angle. Let it be said in praise of music in the home, as it was of Cordelia, 'Her voice was ever soft, gentle and low—an excellent thing in woman'. To circumvent the Fletcher–Munson effect and otherwise wrest the maximum satisfaction from a limited range of loudness is a problem which perhaps merits more attention than it has so far been given.

Mr. R. Vermeulen (*Netherlands: communicated*): There can be no doubt about the benefit, nay the necessity, of stereophony, if one really wants a life-like reproduction and not just 'hi-fi'. By its very nature, even the most perfect loudspeaker cannot possibly be any better than a hole of the same dimensions in the wall of the concert hall. It is still something of a miracle that this hole can be enlarged into a wide window frame by means of only two loudspeakers, but we have to accept this as a fact.

Such a window seat would not be considered quite satisfactory in the concert hall. What we really want is a seat right in the middle of the hall. Therefore, though I agree that at present and for most practical purposes 'the reproduction can be limited to sounds arriving from directions covering an angle of, say, 90° in front of the observer', I do not consider this to be the final solution of perfect music reproduction, but expect a still further development.

It is true that we can imitate the orchestra itself by stereophonic methods, but this imitation orchestra, like the real one, will only sound well in a hall with good acoustics and not in a room with poor acoustics or in the home. In order to have a completely life-like reproduction it is necessary also to simulate the sound waves reflected from the ceiling and from the walls of the concert

* 'The Psychology of Communication', *Journal I.E.E.*, 1957, 3, (New Series), p. 209.

hall. These are not limited to an angle of 90°, but they come from all sides, and it is essential that they do so. The reverberant sound must not only have the right reverberation time; it must also have the right diffuseness.

It is possible to simulate such a diffuse reverberation artificially by placing loudspeakers all round the auditorium and feeding them with suitably delayed music, using several different time delays. By adding such a stereo-reverberation to the stereophonic reproduction, a completely life-like reproduction can be obtained.

I am impressed by the elegant and convincing way in which the authors have reduced the intensity differences of the loudspeakers to phase differences at the ear of the listener, but I still have some doubt whether binaural hearing can be explained by only one single principle. Phase differences with pure tones are not equivalent to differences in time of arrival of clicks, and I cannot see, at present, how the precedence effect, for instance, can be fitted to the authors' theory. Moreover, there is some experimental evidence that differences of intensity at the ear of the listener cannot altogether be neglected. I am not yet completely convinced that it will be wise to base our techniques on a single effect only.

THE AUTHORS' REPLY TO THE ABOVE DISCUSSION

Mr. H. A. M. Clark, Dr. G. F. Dutton, and **Mr. P. B. Vanderlyn** (*in reply*): As Mr. Moir states, it is agreed that directional clues of various kinds occur at all audible frequencies, and their relative importance must vary to some extent with the spectral content of the sounds with which a system is concerned. In the present case, where the reproduction of music is the chief aim, it is contended that low frequencies play at least as important a part as high frequencies. In the design of a stereophonic system, due regard must be paid to the types of clue capable of reproduction within the physical and economic framework, and if any are found to be mutually exclusive, a choice must be made as to which to reject. In the case of the stereosonic system, it was decided to exclude the 'precedence' clue, since to reproduce it would necessitate specification of the precise listening position to be adopted, from which the observer would not be allowed to move. For this reason, the system will not deal correctly with the case cited by Mr. Moir, i.e. a single short sound pulse. It is fair to say, however, that naturally occurring single sound pulses are infrequent, and analysis of most sounds of short duration reveals them to be composed of many cycles of pressure variation. For localization, the system, as Mr. Moir states, allows a comparison of aural amplitudes, but this occurs at high frequencies only; at low frequencies the mechanism of vector summation described in the paper permits a phase comparison.

Many workers in this field of endeavour will at times be inclined to agree with Dr. Cherry's statement that two-channel stereophony with loudspeakers is impossible. Much depends on the precise meaning attached to the terms 'stereophony' and 'impossible'. He does not deny, however, that reproduced sounds having a pronounced spatial character have been demonstrated, at least to those listeners in the more favoured seats. Perhaps these qualifications may help to illuminate his statement.

In the engineering basis of the system, nothing is assumed about the way in which the human 'black box' operates, nor is any such assumption necessary, since all that is attempted is a reconstruction of the acoustic conditions that would obtain at an observer's ears were he actually present in the recording studio. One requirement for satisfactory reproduction is that the listening room should not add appreciably to the total reverberation, and it should be stressed that the system is designed for domestic use where this is normally the case and where the listener can occupy the optimum position. It is not intended for large-scale demonstration.

Before finally rejecting any explanation, based on Fourier analysis, of the effects produced, it must be remembered that the first operation performed on a sound when it reaches the human cochlea is a Fourier analysis.

In reply to Mr. Millington, the use of a double microphone driving two separate loudspeaker channels does largely avoid the unevenness in tonal quality in some parts of the register caused by standing waves in the studio.

Mr. Thornton dislikes the alternation of sounds between one side of the stage and the other. Although this is certainly what the conductor hears there was no intention to give a reproduced performance from this viewpoint. The arrangement of the demonstration was intended to convey the impression of an orchestra occupying the full area of the stage. It is contended that such an impression would be realistic, but if a domestic user feels that the reproduction is on too large a scale dimensionally, it is always open to him to reduce the distance between his loudspeakers.

Major Fuge is correct in supposing that one of the advantages claimed for the system is an increased freedom for selective listening to a particular part of the recorded sound.

Mr. Webb raises the difficult question of domestic sound levels. It may well be that a sound can be made to appear louder than it really is by controlled juggling with the frequency response of the reproducing channel. The operating range of such a control is likely to be limited, however, and many will think that nothing can recreate the sensation of a really loud sound but reproduction at a level comparable with the original. In this respect he is, to some extent, at the mercy of the recording engineer, since, if modern recordings are not permitted to reach reasonably high levels during reproduction, the quiet passages will be lost in background noise.

Mr. Vermeulen mentions the further application of stereophony to include reverberation from the back of a hall. Many of us who have not had the good fortune to hear the results of his experiments have, nevertheless, read about them with interest, and it may well be that where such means as he describes are applicable and economically feasible, they will add considerably to the subjective enjoyment of a reproduced performance.

Finally, in reply to him and to other speakers, it is not our aim to produce a complete theory of binaural hearing, least of all to ascribe it to a single mechanism. The stereosonic system came about as the result of considering what naturally occurring clues exist, and of attempting to recreate a few of them. Others, such as the 'precedence' clue, have had to be rejected. If the explanations offered to fit the observed facts are not looked upon with favour, it can only be remarked that they are not put forward in any spirit of pontification, but that, at least, they have the merit of some experimental foundation.