Soundfield microphone

Design and development of microphone and control unit

by **Ken Farrar,** Calrec Audio Ltd

Ambisonics and surround sound technology based on psycho-acoustic theory form the nucleus of the design of the soundfield microphone. The complete design combines advanced acoustical, mechanical and electrical precision engineering in a revolutionary way. Recordings made with the microphone and reproduced through a minimum of loudspeakers produce images which are stable and uncoloured, while additional loudspeakers, which need not be full range, allow reproduction of valuable height and reverberant information. The soundfield microphone enables the recording engineer not only to record the total sound field and thus protect his recording from obsolescence, but to compare and dub to conventional forms, adjusting, panning and steering his synthesised, truly coincident "microphones" after the event.

THE DEVELOPMENT OF THE NRDC Ambisonic technology for surround sound recording and reproduction is now well advanced and much has been published by those directly involved. Early attempts to supplement the restrictive conventional stereophonic presentation by hasty additions of extra rear channels in the so-called "quadraphonic" format have proved largely unsuccessful. Their particular inherent weaknesses include difficulties in producing stable images from inter-loudspeaker directions, and the encoding formulae of some systems exacerbates this problem further. It has been clearly shown that using Ambisonic technology, much better use can be made of extra loud speakers and channels, and that if only two channels are available, a decoding system may be employed which gives psycho-acoustic optimisation of the presentation in respect of directionality and freedom from coloration or "phasiness."

Background to microphone design

The theoretical analysis of surround sound psycho-acoustics into the mechanisms of human hearing — by Gerzon[1] — argues that at low frequencies below about 700Hz, where half a wavelength corresponds to the distance between the ears, the information reaching the brain is derived from the sum and difference of the inputs to the two ears. This corresponds at low frequencies respectively to the pressure component of the sound–equivalent to an omni-directional microphone W – and the velocity (pressure gradient) component of the sound – equivalent to a sideways pointing velocity, figure-of-eight microphone Y. As the head may be rotated, a forward pointing velocity, figure-of-eight pick-up is also required to determine direction X, Fig. 1. The vector sum of in-phase forward and sideways velocity (figure-of-eight) signals corresponds to the apparent sound direction according to Makita's theory of sound localization by the ears.

At frequencies between 700Hz and 5kHz sound direction is detected by signal energy and corresponds to an "energy vector" being the addition of vector components pointing at each loudspeaker whose lengths correspond to the energy in that speaker. Above 5kHz the pinnae (flaps) of the ears appear to offer directional information to the brain by differences in coloration they impose on the sound arriving in different directions.

Further it has been found that a listener's ability to localize direction is greatly assisted by moderately reverberant conditions especially where the reverberation is fairly uniformly distributed. To take advantage of this additional ambient directional information, it is necessary to record the reverberation accurately and reproduce it uniformly around the listener. The technique of restricting reverberation to one channel with no directional information does not satisfy the above criteria. Moreover, with current technology, artificial reverberation is also not satisfactory in this respect.

To complete the soundfield symmetry a third velocity component whose axis lies in the vertical is required, corresponding to an upward facing velocity of figure-of-eight microphone, Z.

The above requirements of human directional hearing can be satisfied by

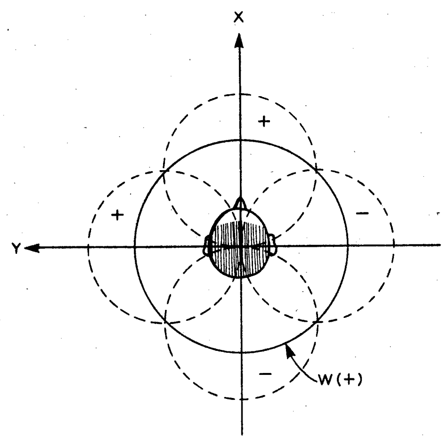
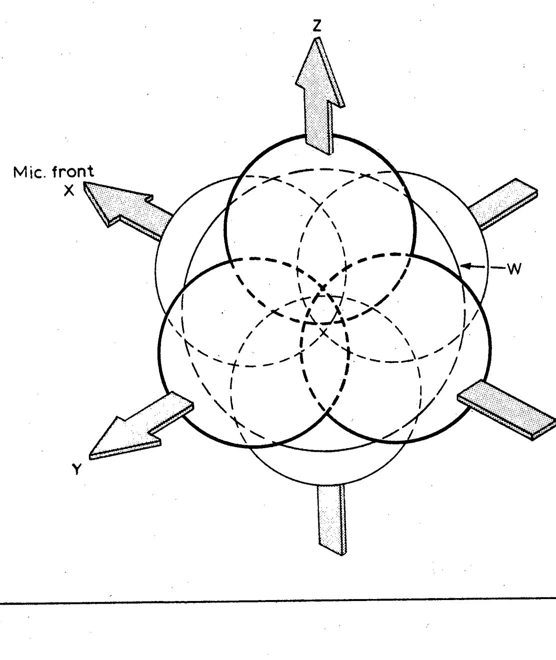


Fig. 1. *At low frequencies, direction is perceived by pressure (w) and velocity (x, y) effects.*

Fig. 2. *B-Format co-ordinates.*



suitable processing of pressure and velocity signal components using an Ambisonic decoder such as that described in references 4 and 7-9. A similar decoder is used in the monitor/output section of the soundfield microphone control unit described later. More detailed aspects of the soundfield microphone principle have been described in references 11 and 13.

Microphone acoustic system

The complete parameters for the design of a microphone to capture the complete soundfield may now be defined as follows. The four signals are known as B-format and soundfield signals should be recorded, stored and generally handled professionally in this form. (Fig. 2).

W–pressure: omni-directional.
X–pressure-gradient (velocity): forward fig.-of-eight.
Y–pressure-gradient (velocity): leftward fig.-of-eight.
Z–pressure-gradient (velocity); upward fig.-of-eight.

The height component Z will probably not be used in reproduction commercially in the immediate future although it is necessary to implement elevation and dominance controls post-session when required and undoubtedly experimental reproduction systems will use it*

The B-format signals are required to be truly coincident and to have good frequency response and well defined polar patterns at all frequencies. It was considered impractical to produce a microphone which generated B-format signals directly; moreover the method chosen has a significant number of advantages over the alternatives. The soundfield microphone uses a unique array of four sub-cardioid capsules mounted as closely as possible in a regular tetrahedron (Fig. 3). They should be imagined as four receivers symmetrically disposed on the surface of a sphere and associated circuits are provided to compensate their practical spacing.

The advantages of this arrangement are as follows: –

- The four capsules are identical single-diaphragm cardioids of proven design.
- They have individually a very good axial frequency response and the response in other directions is regular when set up as sub-cardioids. This means that the polar patterns are well defined at all frequencies.
- Each of the four capsules contributes an equal component to each of the B-format signals thus allowing effective cancellation of endemic capsule variations from the ideal, particularly when the capsules are well matched as they are.
- Arrangement of separating the pressure and pressure-gradient components into B-format allows each component to be compensated separately for frequency and phase response.
- Tetrahedral array used allows for capsule pairing along discrete axes which greatly facilitates testing and alignment.
- Closeness of the array allows compensations to be applied to produce B-format signal components effectively coincident up to about 10kHz. This contrasts vividly with conventional stereo microphones where capsule spacing restricts coincident signals up to about 1.5kHz.

* The author is presently setting up a reproduction system which reproduces height information, in accordance with system HHJ of the universal HJ surround sound encoding standards for Ambisonic technology.

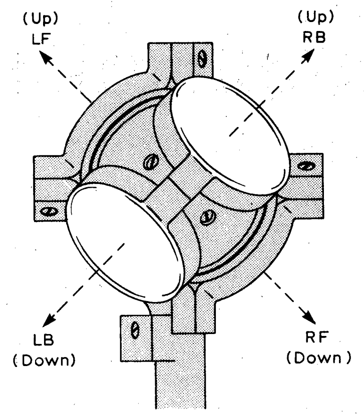


Fig. 3. *The four capsules mounted in tetrahedron.*

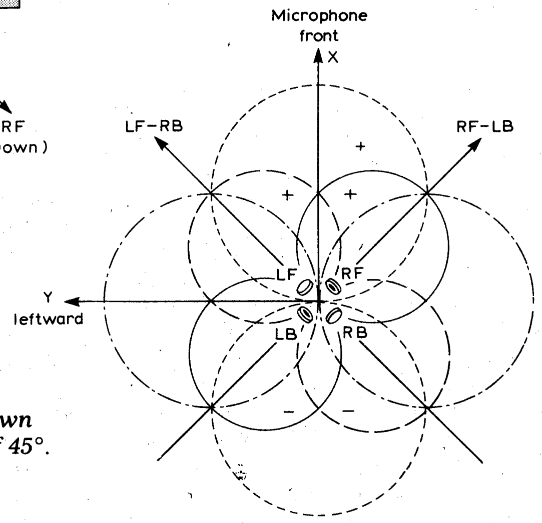


Fig. 4. *Subtracting outputs shown produces horizontal patterns of 45°.*

The capsule signals are known as A-format and correspond to discrete practice except that they are tilted upwards and downwards as shown in Fig. 3, to form the regular tetrahedron. The capsules are paired in the horizontal plane as: left front up and right back up, right front down and left back down. Examination of each of these pairs reveals that they are *symmetrically* tilted from the *vertical* so that if the output signals are subtracted within each pair, the two opposing cardioid patterns produce figure-of-eight patterns whose axes lie along 45° *horizontal* diagonals shown in Fig. 4.

The amplitude of the figure-of-eight patterns thus produced will be reduced from the value obtained if the capsule pairs were back-to-back by cos ϕ, where ϕ is the angle of tilt of each capsule, (35.3°).

If the two diagonal patterns are added, a figure-of-eight pattern facing forward is produced, with an increase in sensitivity of about 3dB (2 cos 45°). This corresponds to

$$X = L_F - R_B + R_F - L_B. \quad (1)$$

Similarly a leftward figure-of-eight pattern is produced by *subtracting* the $R_F - L_B$ figure-of-eight from the $L_F - R_B$ one. This corresponds to

$$Y = L_F - R_B - (R_F - L_B)$$

$$\text{or } Y = L_F - R_B - R_F + L_B. \quad (2)$$

The derivation of an upward figure-of-eight pattern is produced from capsule pairs $L_F - L_B$ and $R_B - R_F$ which produce diagonal figure-of-eight patterns as shown in Fig. 5. This corresponds to

$$Z = L_F - L_B + R_B - R_F. \quad (3)$$

The pressure or omnidirectional component W is produced by adding the

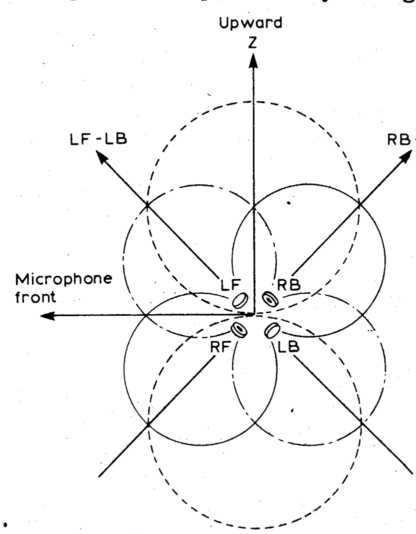


Fig. 5. *Vertical capsule pair outputs.*

four capsule outputs *in-phase* so that

$$W = L_B + L_F + R_F + R_B. \qquad (4)$$

If the microphone is to be used inverted e.g., suspended in a concert hall, it is necessary to reverse the phase of Y and Z only; X still points forwards and W remains omnidirectional. This corresponds to

$$Y_{INV} = -L_F + R_B + R_F - L_B. \qquad (5)$$

$$\text{and } Z_{INV} = -L_F + L_B - R_B + R_F. \qquad (6)$$

The above matrixing and normal/inverted circuit inversion are carried out in the A-B matrix module in the control unit where 16 adjustments are provided allowing for variations in capsule sensitivity. The correct alignment can only be carried out in anechoic conditions where the microphone is rotated in the test field to observe sensitivity and polar patterns of capsules, capsule pairs and finally B-format coordinates. To this end, provision is made for muting each of the capsule A-format signals individually at the input to the control unit. The A/B matrix module carries the serial number of the microphone to which it is adjusted.

There is considerable difference between the sensitivity of the pressure-gradient (velocity) components X, Y and Z and that of the pressure component W due to the following reasons.
1. Capsule signals are added to produce W but subtracted to produce X, Y and Z.
2. The capsules are sub-cardioids, with polar response 2+ cos θ approximately, not cardioids (1+ cos θ), which increases the pressure, W component.
3. The tilt of the capsule pairs reduces diagonal velocity components and hence X, Y and Z signals.
4. The three directional components X, Y and Z of the pressure-gradient require an additional 3dB to conform with the standardized B-format levels. (This sets the energy levels in X, Y and Z similar to that in W for average programme.)

The pressure-gradient components X, Y and Z require a total boost of about 13dB so that the B-format signals match correctly at frequencies where the wavelength is long compared to capsule and array dimensions e.g. 500Hz. At very low frequencies, W requires some boost since it is made up of signals from velocity type capsules which characteristically do not have an extended l.f. response.

At frequencies where capsule spacing compares with wavelength, equalisation circuits take effect to maintain apparent coincidence in B-format signals to about 10kHz. The overall microphone performance is extremely good to 20kHz beyond which the output is rolled off at 12dB/octave.

All these equalization circuits are contained in the A-B matrix module.

Fig. 6. *Amplifiers in body of unit.*

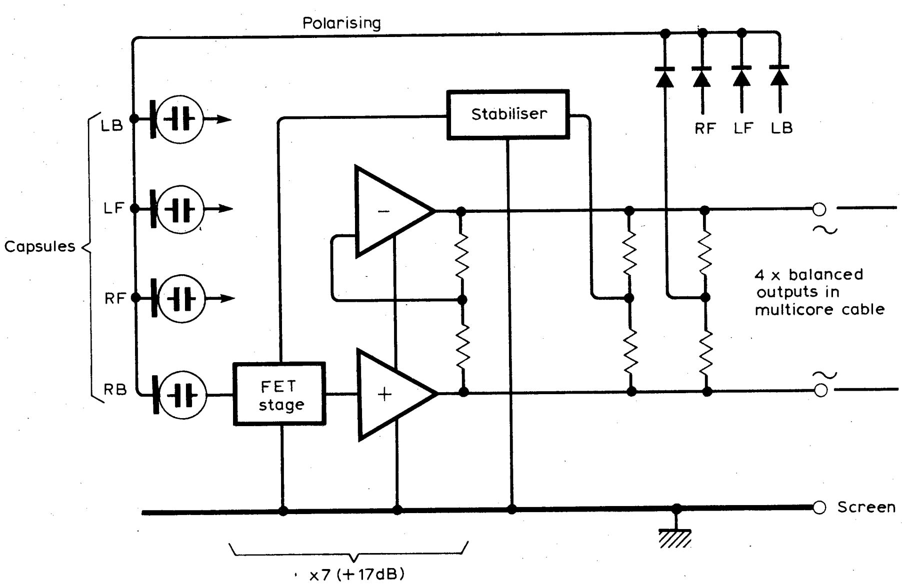
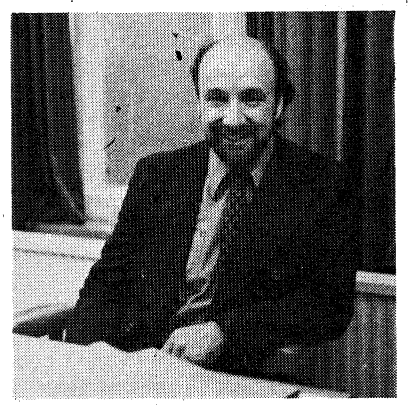


Fig. 7. *One of four identical amplifiers.*



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Microphone amplifiers

The microphone body contains four identical head amplifiers mounted on two similar printed circuit cards, Fig. 6. Each amplifier consists of a field effect transistor low-noise pre-amplifier with a gain of +11dB. The f.e.t. stage drives two operational amplifiers in an electronic balanced configuration effectively adding a further +6dB to make +17dB (×7) overall (Fig. 7) Each pre-amplifier is phantom-powered along its output balanced lines from the control unit, each supply being separately stabilised within the microphone. Each circuit contributes to the polarising of all four capsules so that any or all circuits polarise all capsules. This arrangement together with the stabilised supplies allows signal levels equivalent to 138dB s.p.l. at 1kHz to be handled before clipping occurs. The capacitance of a full length (150 metres) of cable restricts the output to 134dB s.p.l. at 10kHz but this allows an adequate margin over normal loud programme which rarely exceeds 110dB s.p.l. (130dB s.p.l. corresponds to a *very* loud sound). At 138dB s.p.l. the microphone outputs are about 8 volts r.m.s. (+20dBm approx). The microphone signal output level is, in fact, about 5mV/microbar. □

To be continued.

References

The references will appear in Part 2 of the article.

Soundfield microphone — 2

Detailed functioning of control unit

by Ken Farrar, Calrec Audio Ltd

Ambisonics and surround sound technology based on psychoacoustic theory form the nucleus of the design of the soundfield microphone (News, Aug. 1978). The design combines advanced acoustical, mechanical and electrical precision engineering in a new way. Recordings made with the microphone and reproduced through a minimum of loudspeakers produce images which are stable and uncoloured, while additional loudspeakers, which need not be full range, allow reproduction of valuable height and reverberant information. The soundfield microphone enables the recording engineer not only to record the total field sound and thus protect his recording from obsolescence, but to compare and dub to conventional forms, adjusting, panning and steering his synthesized, truly coincident "microphones" after the event.

THE MICROPHONE INPUTS to the Soundfield control unit are electrically balanced having a common mode rejection to interfering signals better than −60dB, 20Hz to 20kHz. The input preamplifiers have a gain of +14dB but may be preceeded by a −20dB attenuator (A−20) if the microphone is used in very loud conditions.

Following the AB matrix, the B-format signals are controlled by a four-gang rotary fader and additional gain of +6dB, +14dB and +30dB (fader max.) may be added to allow a maximum microphone sensitivity of 68dB s.p.l. for 0dBm levels at the recording outputs. These amplifiers are designed to withstand overloading to +24dBm.

The recording output level may be monitored by a peak programme meter which may be switched to X, W, Y, or Z and having facility to increase meter sensitivity by 20dB.

The recording signal and the replay signal from the four-track tape recorder may be monitored, (thus allowing checking of recording quality) by a chain of amplifiers contained in a series of modules in the same manner as the input circuits (Fig. 9).

The first two modules in the output (monitor) chain are soundfield control modules. Soundfield-1 provides azimuth and elevation adjustments and soundfield-2 provides dominance to the sound sphere either up/down or front/back. These controls are operative on replay or dubbing unless the outpts are used for recording.

Azimuth control

Consider the horizontal directional components of the B-format signals X and Y, Fig. 10. Suppose that they are passed through a circuit such that X′ (output) = −Y and Y′ (output) = X. This gives results equivalent to the microphone being turned to face C_R.

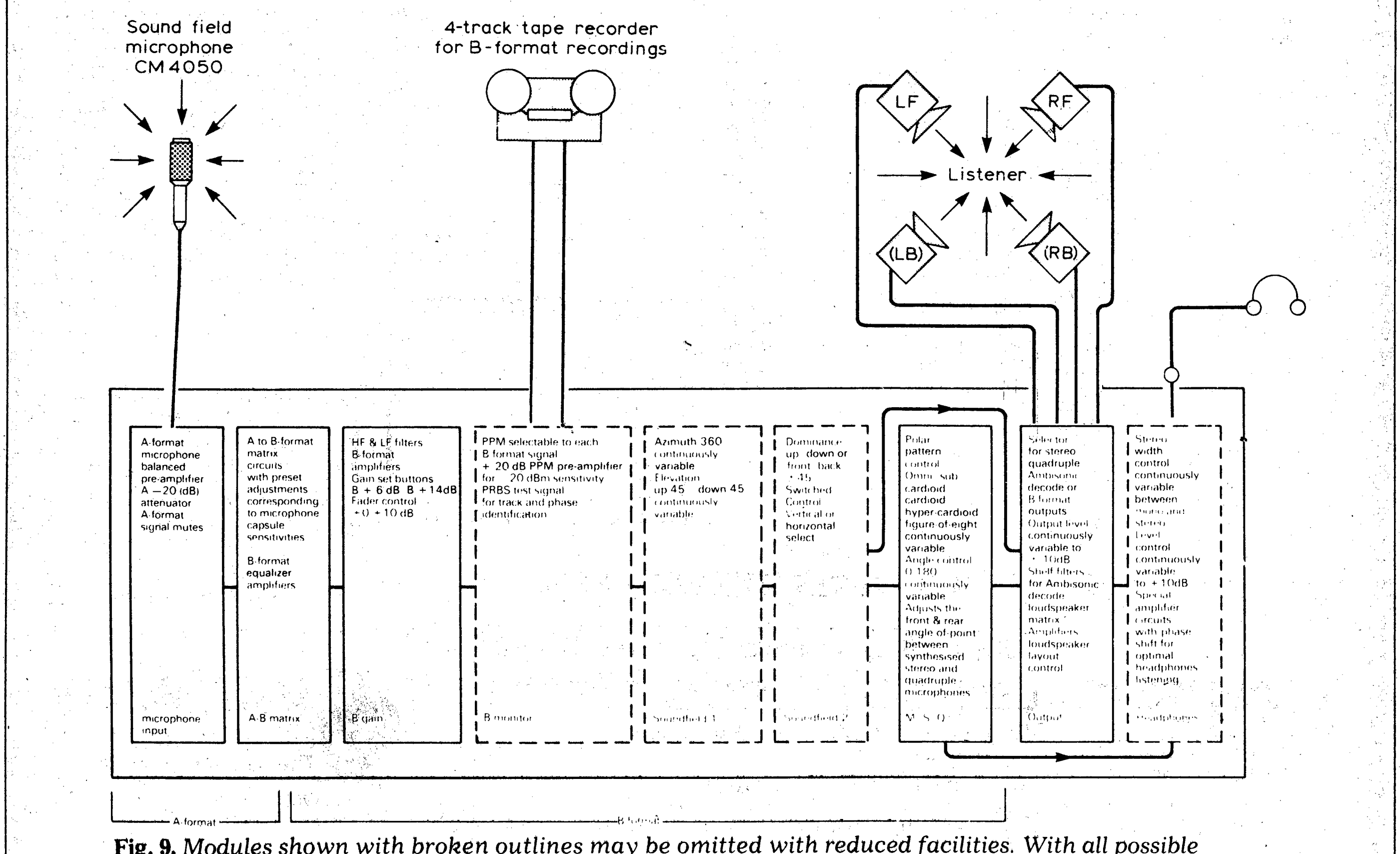


Fig. 9. *Modules shown with broken outlines may be omitted with reduced facilities. With all possible modules left out equipment operates as a "Super Stereo" microphone with steerable facilities. These controls may still be operated post session on B-format material.*

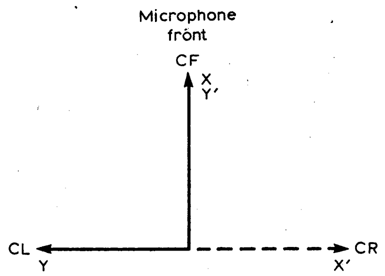
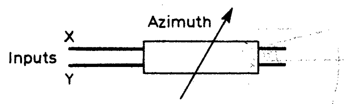


Fig. 10. *Principle of azimuth control.*



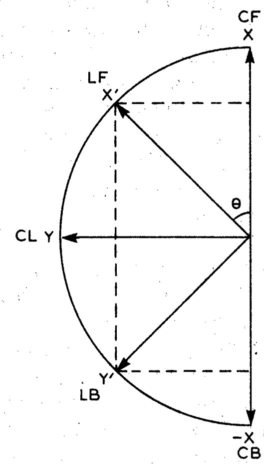
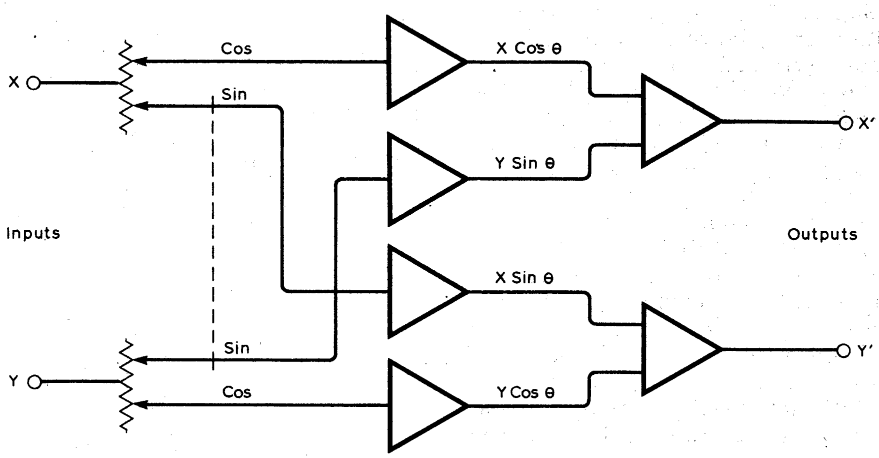


Fig. 11. *Azimuth co-ordinate components. (Right).*

Fig. 12. *Azimuth circuit.*



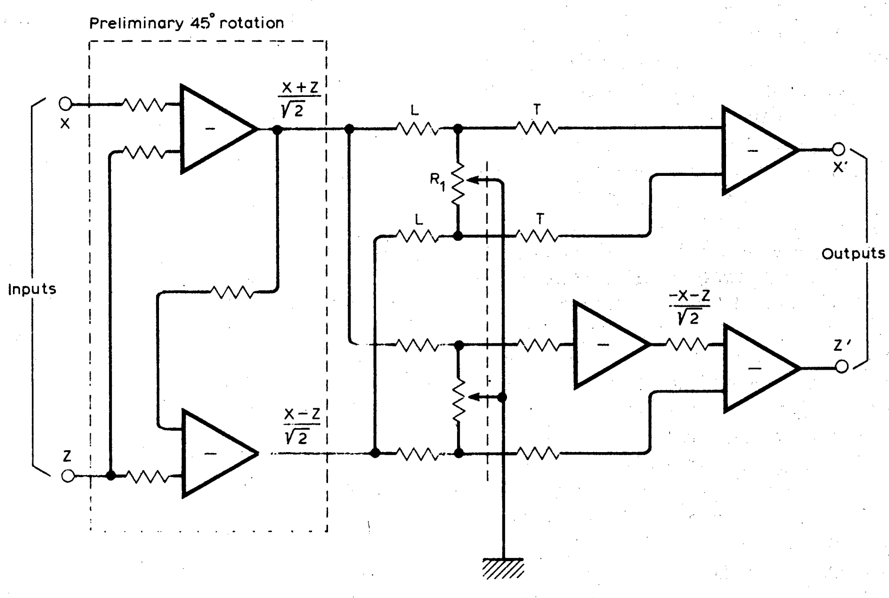


Fig. 13. *Elevation circuit. Amplifiers indicated schematically are used in the virtual earth mixing mode.*

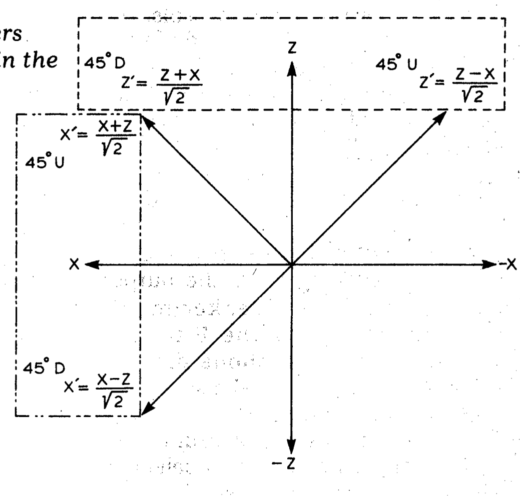


Fig. 14. *Dominance control. Sound directions are moved around the sphere in the manner illustrated for a forward or upward dominance angle. Simultaneously microphone sensitivity (W and X or Z) is increased in the direction of the dominance (front or up in the example) and reduced in the opposite direction (back or down in the example).*

Similarly if $X' = Y$ and $Y' = -X$ the microphone would behave as if it faced the C_L direction. $Y' = -Y$, $X' = -X$ corresponds to the microphone facing C_B and so forth.

If the microphone is required to face, say, L_F (Fig. 11) then X' needs to be composed of components X and Y but with the same overall sensitivity as X (or Y). The peak sensitivity of X' (and Y') is required to remain constant and the orthogonal components must satisfy the following sine/cosine relationship

$$X' = X\cos\theta + Y\sin\theta$$
$$\text{and } Y' = Y\cos\theta - X\sin\theta$$

In the example given

$$X' = X\cos 45° + Y\sin 45°$$

$$\text{so that } X' = \frac{X}{\sqrt{2}} + \frac{Y}{\sqrt{2}} \qquad (7)$$

$$\text{and similarly } Y' = \frac{Y}{\sqrt{2}} - \frac{X}{\sqrt{2}} \qquad (8)$$

A continuously variable azimuth control requires the use of a twin-gang sine/cosine potentiometer in the circuit of Fig. 12.

Elevation control

If rotation of the microphone is required only over a restricted range of ±45° such as the elevation control, a less sophisticated circuit may be used, Fig. 13. This control needs to act only on the X and Z co-ordinates since it is required to rotate the microphone forward and backward about the Y-axis.

The circuit firstly produces sum and difference signals $(X+Z)\sqrt{2}$ and $(X-Z)\sqrt{2}$ corresponding to 45° vertical rotation in either direction. A control R_1 varies the mix to X' (output) so that at the extreme positions 45°U and 45°D each of the two signals is passed respectively. These corresponde to X' as shown in Fig. 14. In the centre, 0°, position $X' = X$.

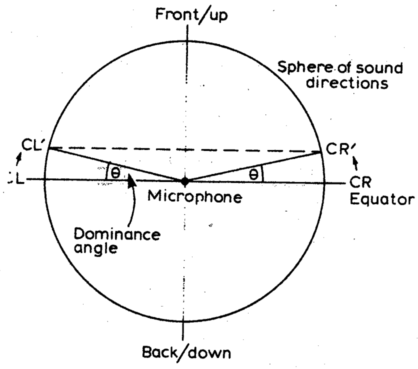
L, T and R values are chosen so that over the range of the control, the modulus of X' remains constant, the components following a sine/cosine law as with the azimuth control.

Dominance control

The soundfield-2 module introduces an effect called dominance. To see its effect, imagine the incoming sounds as arriving from different points on the surface of a large sphere centred at the microphone. The dominance control modifies the effective directions of arrival of sounds, and also their loudness. In the case of vertical dominance, the control effectively displaces all sound on the sphere upward or downward, also making the microphone more sensitive to sounds in the direction (up or down) toward which the sounds are displaced.

The extent of the displacement is marked on the control as an angle which is the extent by which the "equator" of the sound sphere is displaced above (or below) its normal horizontal position, Fig. 15. The control provides nine selected positions, four either side of normal, 0°, the maximums being ±45°: the control can be used either for up/down (vertical) or front/back (horizontal) dominance.

Increasing dominance in the circuit progressively changes the pressure (omni-directional) component W into a sub-cardioid by adding an increasing amount of corresponding pressure-gradient component. That is Z for up, —Z for down, X for front and —X for back. At 45°, W is still not quite a cardioid. Simultaneously Z (for up/down) or X (for front/back) has an increasing component of W added to it or subtracted from it to convert Z or X from figure-of-eight to hyper-cardioid. At 45° Z or X is a cardioid. Fig. 16 shows the condition for +30° up. Compensation is simultaneously provided in each case such that the ratio of the energy in the velocity signals to the energy in the pressure signal remains unchanged although a use of the control is to emphasize particular directions and/or de-emphasize others so a change in programme level is usually heard. For example a typical use is to set for UP dominance so as to reduce the sensitivity of the microphone to audience noise. Alternatively the microphone may be apparently moved closer to the sound stage by the use of front dominance. What in fact happens is that the sensitivity to front or direct sounds is increased whilst that to back or reverberant sound is reduced. The circuit to achieve this is shown schematically in Fig. 17.



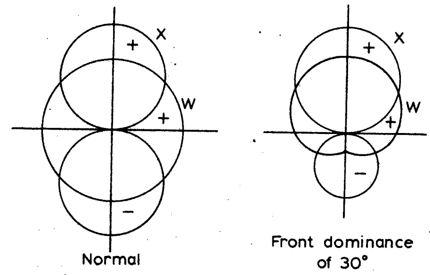


Fig. 16. *Effect of dominance control on B-format components.*

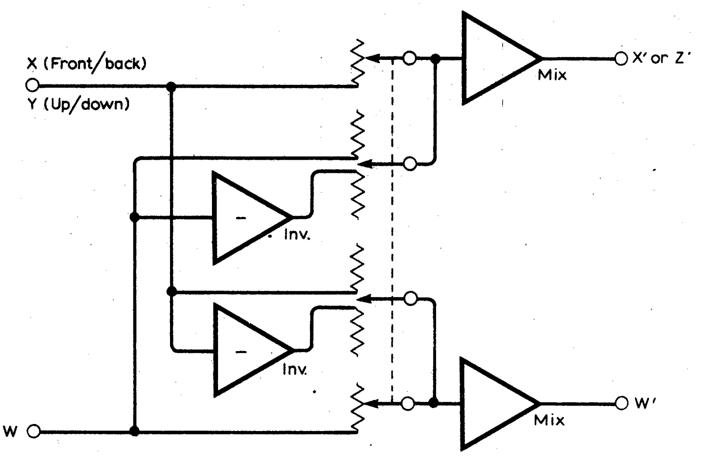


Fig. 17. *Dominance circuit schematic.*

Output controls

Following the soundfield controls, the B-format signals may be passed via the gain control to the output sockets if "B" output is selected, at 0dBm level. This condition would be used for dubbing using the soundfield controls to make adjustments or during recording if it was felt necessary to use the controls in this condition.

A more likely arrangement is to select Ambisonic decode and use the output sockets for four-loudspeaker monitoring thus recording the B-format signals from the microphone directly and allowing experimental use of the soundfield and other controls in the monitor chain during the recording.

When Ambisonic decode is selected, the B-format signals are passed from the gain control into shelf filters in the output module. The shelf filters are all-pass circuits with identical phase shifts of 90° at 400Hz. The effect of the shelf filters is to boost the gain of W relative to that of X or Y by 3dB at high frequencies. The subsequent loudspeaker matrices thus produce 120° hyper-cardioids at low frequencies at the four corner positions and 135° hyper-cardioids above about 1kHz. This produces optimum psychoacoustic performance in accordance with the theory of references 1, 4, 7 to 10.

If the listening loudspeakers cannot be placed in a regular format, a loudspeaker layout control shown in Fig. 18: allows variation of the X:Y aspect ratio from 1:2 to 2:1 to compensate. Fixed

Table 1: Effective matrix equations for Ambisonic B-format decoder with square loudspeaker layout.

Low frequencies	Mid and high frequencies
$L_B=W-X+Y$	$L_B=W-\frac{X}{\sqrt{2}}+\frac{Y}{\sqrt{2}}$
$L_F=W+X+Y$	$L_F=W+\frac{X}{\sqrt{2}}+\frac{Y}{\sqrt{2}}$
$R_F=W+X-Y$	$R_F=W+\frac{X}{\sqrt{2}}+\frac{Y}{\sqrt{2}}$
$R_B=W-X-Y$	$R_B=W-\frac{X}{\sqrt{2}}+\frac{Y}{\sqrt{2}}$

distance compensation in the form of RC high-pass filters of pressure gradient components X and Y is provided for typical monitor loudspeaker distances of 2 to 3 metres from the listeners to compensate the increase in velocity components at very low frequencies due to sound wavefront curvature.

The loudspeaker matrix formulae at low frequencies and at mid-band and above for a square loudspeaker layout are given in Table 1.

If the outputs are selected to quadruple or stereo/mono the shelf filters and loudspeaker layout control are bypassed and the output matrix is set for corner cardioids i.e.,

$$L_B=W-\frac{X}{2}+\frac{Y}{2}$$

$$L_F=W+\frac{X}{2}+\frac{Y}{2}$$

$$R_F=W+\frac{X}{2}+\frac{Y}{2}$$

$$R_B=W-\frac{X}{2}+\frac{Y}{2}$$

On stereo/mono and L_B and R_B are switched off.

The controls of polar pattern and angle are now operative and work as follows. At 0° angle Y is reduced to zero and X enhanced 3dB to maintain proper levels. Similarly, at 180° X is removed

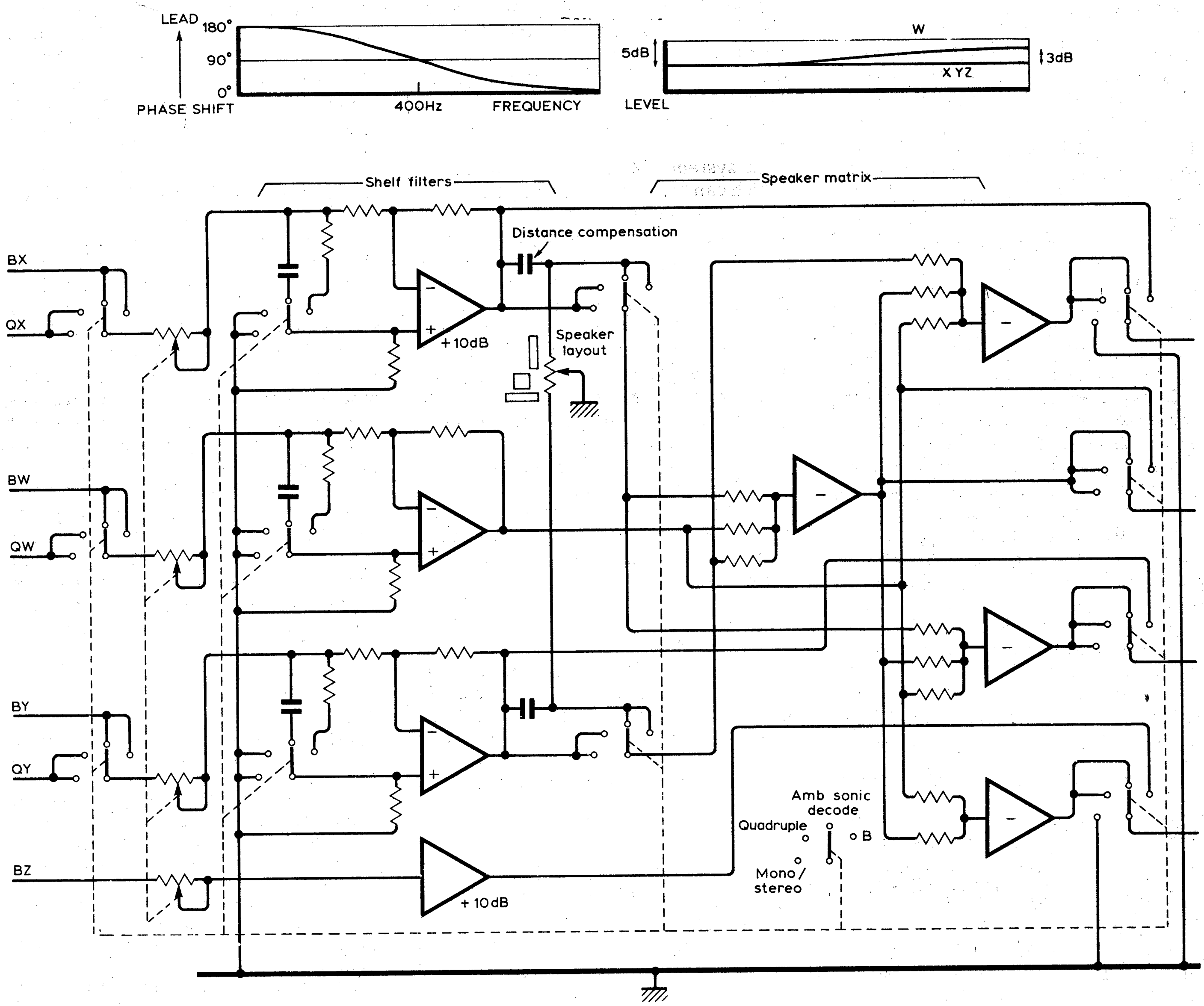


Fig. 18. *Circuit of output module showing shelf filters, low frequency distance compensation, louspeaker layout control which follows a sine/ cosine gain law and loudspeaker matrix.*

and Y increased 3dB. The control follows a sine/cosine law similar to the elevation and layout controls, Fig. 19. The polar pattern control provides unity gain to X, Y and W at the cardioid position. For omni, W is increased 6dB and X and Y turned off. Similarly, at figure-of-eight X and Y are increased 6dB and W turned off. Intermediate settings provide continuously adjustable patterns such as sub-cardioids and hyper-cardioids.

Typical microphone conditions can be set up as follows.

- Angle θ, output stereo/mono. This corresponds to mono and the synthesized single microphone can be set for any pattern from omni-directional through cardioid to figure-of-eight. It can of course be panned and tilted as previously described, using the soundfield 1 controls, Fig. 20(a).

- Angle 90°, output stereo/mono.

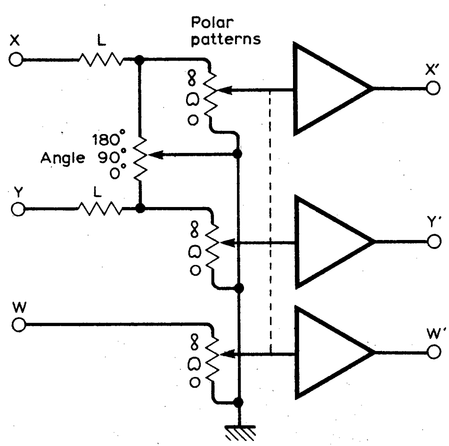


Fig. 19. *Microphone angle and polar pattern circuit.*

This corresponds to a truly coincident stereo pair whose patterns may be varied as above, set in the familiar 45°/45° configuration. They may as a pair be tilted down or up and turned as required, Fig. 20(b).

- Variation of the angle control now adjusts the angle between the pair of microphones which can be as much as 180° if desired. This together with the polar pattern control, azimuth and elevation gives infinite stereo flexibility, Fig. 20(b).

- With the output set to quadruple, four such microphones may be synthesized Fig. 21. This configuration may be changed as shown.

The angle control now changes the angle between the front and rear pairs of "microphones" simultaneously.

All polar patterns may be varied together and identically, The whole array may be panned and tilted as before.

The left and right front stereo output signals may be monitored continuously using stereo headphones irrespective of the output mode selected. Thus what is heard in the headphones is subject to the angle and polar patterns controls all the time. These controls may be set for a particular output condition and in this event, there is a separate headphones stereo width control which may be set down to mono if required. The headphones employ a unique circuit with a phase advance applied to the Y signal (this is equivalent to $S(=L-R)$ in the M/S format). The result is a more natural, sharper stereo presentation.

Conclusions

Clearly the soundfield microphone and master recording of the four B-format signals would appear to fulfil the need for a new and versatile standard but is necessary to consider the inclusion of pan-potted material. In some situations emphasis needs to be given to an individual or section. Single microphones may, for this purpose, be panned into a B-format presentation using special, but fairly simple, circuit techniques although it should be stressed that such enhancement microphones should be used with discretion so as not to confuse and distort the "acoustic hologram" of the soundfield microphone.

It is necessary to have encoding standards for public use which preserve as far as possible the qualities of the system. They must allow proper mono and stereo compatibility and, because of the existing commercial outlets, include the possibility of encoding the surround sound effect into just two audio channels. Whatever system is chosen should, however, be adaptable to three of four-track systems in such a way that additional features such as height may be included.

The term for a system of this type is C-format (consumer format) and there are a number of proposals in existence, the Ambisonic proposal being System UHJ (12) in which the basic two-channel is HJ (or BHJ). This matrix specification is the now-accepted two-channel standard superceding the earlier NRDC system 45J and BBC System H.

In system THJ a third channel (T) which can be band limited may be added to improve directionality. In the case when the third channel is band-limited the system is termed a 2½-channel system. The UK Independent Broadcasting Authority is undertaking experimental 2½-channel HJ broadcasts. When available a fourth channel may be used for either to emphasize loudspeaker positions in a square layout; QHJ, or to present a full soundfield with-height directional effects, HHJ.

The last-mentioned specifications are kernel not matrix specifications and as such may be applied to multi-track as well as B-format or other four-track recorded material. See references for further details.

Acknowledgements

The tetrahedral array of capsules is based on an application of the mathematical theory of sampling on the surface of a sphere developed by Michael Gerzon at the Mathematical Institute of Oxford who devised much of the basic design architecture described. Ambisonics technology was developed by Professor Peter Fellgett of the University of Reading, John Wright of IMF Electronics and Michael Gerzon of the Mathematical Institute of Oxford under the auspices of the National Research Development Corporation. Thanks to Geoffrey Barton of the University of Reading for invaluable design assistance, notably his computer simulations of the acoustical performance and the resulting design of spaced-to-coincident conversion filters. I am indebted to co-director Clem Beaumont without whose devoted expertise in the production of superb capacitor capsules and associated acoustical design work this project would not have been possible.

The Calrec Soundfield microphone and NRDC Ambisonic technology are the subject of the United Kingdom patent No. 1494751 and US patent Nos 3997725 and 4042779 together with all corresponding patents in other countries and all other patents pending.

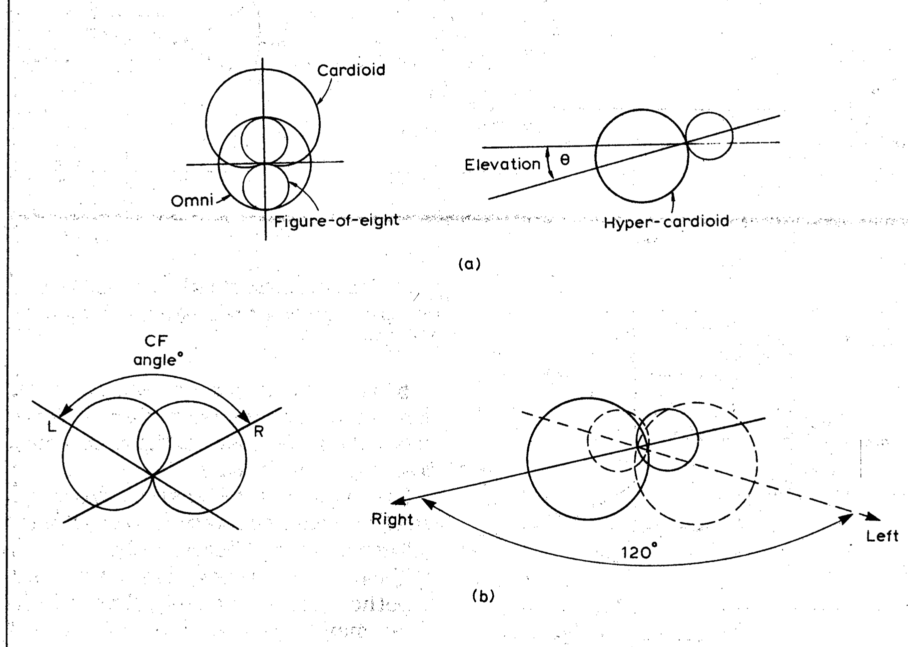


Fig. 20. *Two polar patterns (a) above, and two stereo configurations (b), below. Synthesized cardioid microphones are shown left and a stereo pair of synthesized microphones set hypercardioids at an angle of 120° and tilt downward seen from direction C_F are shown right.*

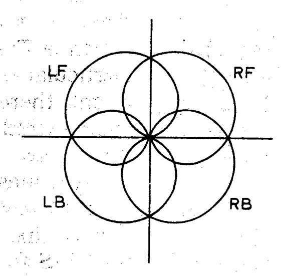


Fig. 21. *Typical quadruple configuration.*

References

1. Gerzon, M. A., Surround-sound psychoacoustics, *Wireless World*, 1974 vol. 80, pp.483-66.
2. Fellgett, P. B., Perspectives for surround-sound, *Hi-Fi Sound Annual*, 1974.
3. Fellgett, P. B., Ambisonics – part one: general system description. *Studio Sound*, August 1975.
4. Gerzon, M. A., Ambisonics – part two: studio techniques, *Studio Sound*, August 1975, pp. 24, 26, 28, 30. Corrections October 1975, p.60.
5. Gerzon, M. A., Periphony: with height sound reproduction, *J. Audio Eng, Soc*, vol. 21, 1973, pp. 2-10.
6. Gerzon, M. A., Criteria for evaluating surround sound systems, *J. Audio Eng. Soc.* vol. 25, 1977, pp. 400-8.
7. Gerzon, M. A., NRDC Surround-sound system, *Wireless World*, April 1977, vol. 83, pp.36-8.
8. Gerzon, M. A., Design of ambisonic decorders for multispeaker surround sound, AES New York (reprint) November 1977.
9. Gerzon, M. A., Multi system ambisonic decoder, *Wireless World*, July 1978, vol. 84, pp. 43-7.
10. Gerzon, M. A., Optimum choice of surround-sound encoding specification, 50th AES convention Paris, March 1977.
11. Raven, P. and Gerzon, M. A., US Patent 4,042,779, 16 August 1977.
12. NRDC. Encoding standards for NRDC Universal HJ Surround Sound Encoding System.
13. Gerzon, M. A., Design of precisely coincident microphone arrays for stereo and surround sound, 50th AES convention, London, March 1975. □

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