Ambisonic Decoders for HDTV

[2TM1.03]
Preprint 3345

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Presented at the 92nd Convention 1992 March 24–27 Vienna

AES

AN AUDIO ENGINEERING SOCIETY PREPRINT

Ambisonic Decoders for HDTV

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Abstract

A new generation of ambisonic decoding technology optimised for use with TV and HDTV is described. Based on B-format encoding plus optional supplementary channels, the new decoders give enhanced frontal stage image stability, so that on-screen sounds are more aligned with on-screen visual images. The signal processing architecture of the new decoders differs from earlier designs, and is discussed in the paper, along with a detailed discussion of the psychoacoustic requirements to ensure optimum subjective localisation quality around a 360° surround sound stage.

1. INTRODUCTION

In the 1970's and early 1980's, a reproduction technology was developed for surround-sound, based on maximising the number of auditory localisation cues produced by loudspeakers, called "Ambisonics". This technology is surveyed in ref. [1] and references therein. However, that technology, although it works extremely well, has some limitations when applied to the special needs of sound with High Definition Television (HDTV). Essentially, the problem is that earlier Ambisonic systems gave frontal-stage images whose apparent position varied with respect to the frontal speakers as the listener moved from one side of the listening area to the other, which results in a mismatch between the sound and associated visual images.

However, Ambisonics has some unique virtues, because it is based on making the maximum possible number of auditory localisation cues consistent with one another. Because of this, it tends to be robust under conditions of "abuse", such as technical misadjustments, speaker misplacements and such things as chairs in front of loudspeakers. It is also capable of convincing illusions of direction around a full 360° azimuthal stage, including side positions. Also, importantly, the mutual consistency of different localisation cues leads to very low listening fatigue, giving a subjectively enhanced quality of sound.

This paper describes a new generation of decoders, based on the same psychoacoustic criteria as earlier Ambisonic decoders, that retain these virtues while giving markedly improved frontal-stage image stability for use with HDTV.

These new decoders use a quite different architecture to that used with earlier decoders. Like earlier Ambisonic decoders, they use a frequency-dependent decoding algorithm to satisfy different auditory localisation mechanisms below and above 700 Hz. Unlike the earlier decoders, these decoders can be used with irregular 5- and 6-speaker layouts (such as those shown in figs. 1 to 4) desired for use with HDTV, where one or two front centre speakers are used in addition to a wider rectangle or trapezium of loudspeakers.

The psychoacoustic requirements of ambisonic decoding can be formulated as a set of equations on loudspeaker feeds, and this paper describes the ambisonic decoding equations and the underlying psychoacoustic theory.

In the past, solutions to these equations were only known for a restricted variety of loudspeaker layouts, including squares, rectangles, regular polygons, and layouts consisting of a number of pairs of loudspeakers diametrically opposed to one another (see [2]). These decoders were based on using a matrix fed by pressure and velocity signals, whereby the pressure and velocity signals were subject to "shelf filters" with differing gains at low and high frequencies to take account of different auditory localisation mechanisms in those frequency ranges. Despite efforts of the first author (MAG) during the last 20 years, he had been unable to find such solutions for the less regular speaker layouts such as those of figs 1 to 4. The second author (GJB) had used empirical means of investigating Ambisonics for cinema auditorium use, and in the course of discussions, provided the novel step that allowed the first author to compute detailed solutions for nonregular loudspeaker layouts.

Although nearly all the detailed mathematical and analytic work reported in this paper is due to the first author, the key insight that made this work possible came from this novel step, which essentially lay in the observation that it was not neccessary for the reproduced pressure signal from an ambisonic decoder to be a shelf-filtered version of a single signal, but that it could be a signal whose directional sensitivity to encoded sounds ("polar diagram") itself varied with frequency.

The new decoders not only satisfy the ambisonic decoding equations for a far greater variety of speaker layouts than did the old, but they also allow considerably improved results, with a far greater range of design options for the tradeoffs and compromises involved in design work. This paper discusses the architecture of the new decoders, outlines the design methodology and tradeoffs, and gives an analysis of the performance of some of these new decoders.

It has to be said that the mathematical analysis and design of such decoders for irregular speaker layouts is very tedious and messy. Not only are the ambisonic decoding equations themselves very nonlinear, which means that the solution involves solving a system of simultaneous nonlinear equations, but it also turns out that the desired solutions lie quite close to "singularities" in these equations, which means that normal methods of numerically solving such equations do not work unless one is very close to the desired solution at the start of the solution. This means that an initial search of the solution space is required.

We have in fact found analytic methods of solving the ambisonic decoding equations, but these result in a solution procedure that requires several pages of mathematics just to describe even in the 5-speaker case , so that we shall not give it in this paper, and even then the decoder optimisation still suffers from the fact that the optimum is very near singularities.

This phenomenon that desirable directional decoders lie near singularities in a solution space appears to be a general one, first encountered by the first author in connection with the computation of optimum "upconversion" decoders for reproducing n_1-speaker stereo via a greater number of stereo speakers [3], and in practice means that solutions take a lot of tedious work to derive. However, once arrived at, the resulting solutions appear to be extremely stable to conditions of mild abuse, such as small alterations of speaker layout shape, which have little effect on the reproduced localisation quality.

In this paper, we shall mainly concentrate on decoders working from B-format, and various enhancements of B-format, described in a companion paper [4] on transmission systems for HDTV sound.

The paper starts with a survey of B-format directional encoding of sound, and its operational aspects, including the "forward dominance" transformation of B-format signals, which turns out to be of key importance in the design of decoders for irregular speaker layouts for reasons we shall describe.

After describing B-format, we shall then summarise the psychoacoustic theory, based on ref. [2], but with a fair amount of interpretative material specific to the design trade-offs in ambisonic decoders. We shall then describe the basic architecture used, and summarise aspects of the design procedure and tradeoffs.

We shall describe the calculated performance of some example solutions for various 5- and 6-speaker layouts, to illustrate the virtues and tradeoffs of the new solutions, which turn out to have markedly improved frontal image stability.

We shall then introduce "enhanced" B-formats, also described in the paper [4] in connection with HDTV transmission systems, and show how these can be used in the new type of ambisonic decoder to provide a yet further improvement in the stability of frontal images.

The outcome of this work is a new generation of ambisonic decoders that, even in their most basic B-format form, are much better suited to use with TV and HDTV than previous decoders, using one or more extra loudspeakers near the TV screen. Moreover, these new decoders are a suitable basis for further enhancement by the addition of extra channels to B-format. These enhancements are closely related to the work reported in [3] and [5-7] on multispeaker stereo, and are also the basis of the transmission hierarchy for HDTV reported in ref. [4].

Reference [8], although dating back to the quadraphonic era, is a useful introduction to the basic design aims of ambisonic systems.

2. B FORMAT

B format is one of a number of signal formats suggested [10] in the early days of Ambisonics to represent surround sound fields, and was intended specifically as a studio production format from which encoded signals for surround-sound transmission could be derived.

We shall adopt the convention shown in figure 5 for sound reproduction in the horizontal plane, with an x-axis pointing in the forward direction and a y-axis pointing in the due left direction, and the azimuth angle of arrival of a sound is measured as an angle θ measured anticlockwise from the x-axis. Thus a due front sound has azimuth θ equal to 0°, a due back sound has azimuth 180°, a due left sound has azimuth $+90^{\circ}$ and a due right sound has azimuth -90°. The left speaker of a standard stereo speaker layout subtending 60° angle at the listener has azimuth $+30^{\circ}$ and the right speaker has azimuth -30°.

B format encodes horizontal sounds into 3 signals W, X and Y, where W is an "omnidirectional" signal encoding sounds from all azimuths with equal gains equal to one, X is a signal encoding sounds from azimuth θ with a gain $2^{\frac{1}{2}}\cos\theta$, giving a forward-facing "figure-of-eight" polar diagram with forward gain $\sqrt{2}$, and Y is a signal encoding sounds from azimuth θ with a gain $2^{\frac{1}{2}}\sin\theta$, giving a leftward-facing "figure--of-eight" polar diagram with leftward gain $\sqrt{2}$. Figure 6 shows the horizontal polar diagrams of the B-format signals. B-format can also be extended to full-sphere directional signals by adding a fourth upward-pointing Z "figure-of-eight" signal with upward gain $\sqrt{2}$, as shown in figure 7, although we do not emphasise such with-height systems in this paper; see [1].

The sound field microphone [13,14,15] is a one-point microphone system capable of encoding a live directional sound field around the microphone directly into B-format signals W, X, Y and Z. Such Sound Field microphones are currently commercially available from AMS as the AMS Mark IV SoundField microphone or the AMS ST250 microphone.

It is also possible to produce horizontal B-format signals by using a B-format panpot, which feeds an input signal directly to the W output with gain 1, and uses a 360° sine/cosine panpot with an additional gain $\sqrt{2}$ to feed the respective Y and X outputs. A unit providing eight channels of B-format panning, capable of being cascaded with additional units to provide more input channels is commercially available from Audio+Design, as is a B-format converter unit which converts the channel strips of conventional stereo mixers with constant power stereo panpots into B-format, using the channel strip panpot plus group selection and an auxiliary feed on each strip to perform B-format panning. This technology is detailed in refs. [9] and [10]; other appropriate

B-format production technologies are described in [11], developed by the IBA, and [12].

Thus B-format production technology exists for creating B-format signals both from live sound fields and from panned monophonic signals. One of the most useful aspects of the B-format representation of surround sound signals is that it allows manipulation of a B-format sound field into a modified B-format sound field by subsequent signal manipulation.

For example, the whole soundfield can be rotated anticlockwise by an angle θ' by producing the new B-format signals:

$$W' = W$$
$$X' = X\cos\theta' - Y\sin\theta'$$
$$Y' = X\sin\theta' + Y\cos\theta' \qquad (1)$$

so that the W' signal still has gain 1, the X' signal has gain $2^{\frac{1}{2}}[\cos\theta\cos\theta' - \sin\theta\sin\theta'] = 2^{\frac{1}{2}}\cos(\theta+\theta')$ and the Y' signal has gain $2^{\frac{1}{2}}[\cos\theta\sin\theta' + \sin\theta\cos\theta'] = 2^{\frac{1}{2}}\sin(\theta+\theta')$ for an original azimuth θ, giving B-format signals with a modified azimuth $\theta+\theta'$. Such rotation controls are implemented both in the Audio+Design B-format panning unit [9] and in the sound field microphone control unit [13].

While such rotation control has many uses, particularly in centering a B-format sub-mix in a desired direction within an overall B-format mix, there is another kind of B-format to B-format manipulation that is more important to the purposes of HDTV production and reproduction, termed <u>forward dominance</u>, which requires a little explanation.

Because $\sin^2\theta + \cos^2\theta = 1$ for all angles θ, one has for B-format signals containing a sound from a direction azimuth θ

$$2W^2 = X^2 + Y^2 \qquad (2)$$

and conversely, any signal gains W, X and Y satisfying equ. (2) are B-format signals for some direction azimuth θ. It is virtually self-evident that rotated B-format signals still satisfy equ. (2); what is less evident is that there are other linear transformations of a highly nontrivial nature of B-format signals that also satisfy equ. (2) after transformation. These transformations, which apply both to horizontal and full-sphere B-format, are technically known as <u>Lorentz transformations</u>. These occur in the theory of special relativity and so have been exhaustively studied, for example see ref. [16].

For our purposes, we only need consider one particular kind of Lorentz transformation of B-format signals, namely the one given by

$$W' = \tfrac{1}{2}(\lambda+\lambda^{-1})\,W + 8^{-\frac{1}{2}}(\lambda-\lambda^{-1})\,X$$
$$X' = \tfrac{1}{2}(\lambda+\lambda^{-1})\,X + 2^{-\frac{1}{2}}(\lambda-\lambda^{-1})\,W$$
$$Y' = Y\,, \qquad (3)$$

where λ is a real parameter having any desired positive value. We leave it as an exercise in elementary algebra for the reader to verify that if equ. (2) is satisfied for B-format signals W, X and Y, then it is also satisfied by the transformed B-format signals W', X' and Y' of equations (3).

It is easily verified direct from equs. (3) that a due-front B-format sound with W, X, Y gains of 1, $2^{\frac{1}{2}}$ and 0 respectively is transformed into one with a gain λ times larger, whereas a due-rear sound with original gains 1, $-2^{\frac{1}{2}}$ and 0 respectively are transformed into a due rear sound with gain λ^{-1}. Thus the so-called forward dominance transformation of equ. (3) increases front sound gain by a factor λ, whereas it alters rear sound gains by an inverse factor $1/\lambda$, and the relative gain of front to that of back sounds is altered by a factor λ^2, which allows the relative gain of reproduction of rear sounds to be modified to reduce (or increase) their relative contribution.

This use of forward dominance control is important in various applications of B-format to HDTV. In a production application, it can be used to de-emphasise sounds from the rear of a sound field microphone while still giving a true B-format output; such dominance control has been provided in the control units of the AMS SoundField microphone. However, it can also be used in different reproduction modes relying on B-format input signals to de-emphasise rear sounds. In particular, the new B-format Ambisonic surround-sound decoders described in this paper give excessive gain for rear sounds, which can be compensated for by a judicious application of a compensating forward dominance.

Besides altering the front-to-rear level balance, forward dominance also alters the directional distribution and azimuths of sounds (other than those at due front and rear directions). Figure 8 shows the effect of forward dominance with $\lambda = 2^{\frac{1}{2}}$. Without going into the detailed analysis, it can be shown that an original azimuth θ is transformed into a new azimuth θ' given by the equation

$$\cos\theta' = \frac{\mu + \cos\theta}{1 + \mu\cos\theta} , \tag{4a}$$

where

$$\mu = (\lambda^2 - 1)/(\lambda^2 + 1). \tag{4b}$$

If $\lambda > 1$, then all directions are moved towards the front, and if $\lambda < 1$, all directions are moved towards the back by the forward dominance transformation of equ. (3). The width of a narrow stage around due front is multiplied by a factor $1/\lambda$, and of a narrow stage around the back is multiplied by a factor λ, as shown in figure 8, by this transformation, so that forward dominance is a kind of B-format "width control" that narrows the front stage as it widens the rear stage, or vice-versa. The relative front-to-back amplitude gain λ^2, expressed in decibels, is termed the "dominance gain", so that $\lambda = 2^{\frac{1}{2}}$ is said to have a dominance gain of + 6.021 dB. This dominance gain causes images at the sides (azimuths $\pm 90^{\circ}$) to move forward by an angle of $\sin^{-1}\frac{1}{3} = 19.47^{\circ}$ in the B-format sound stage, via equs. (4).

3. DIRECTIONAL PSYCHOACOUSTICS

It is difficult to explain methods of Ambisonic reproduction without some indication of the theoretical methods used to model directional localisation of sounds. The mathematical methods used have been described in several of the author's previous papers. A nonmathematical account was given in [17], and the theory's mathematical details were given in [2], and a good summary of the theoretical framework was recently given in ref. [3], section 5. Since Ambisonic decoders for 360° directional reproduction require a different set of design criteria from the frontal-stage stereo systems described in ref. [3], we repeat some of the theory here in a skeletal outline adapted to our present needs.

The localisation given by signals emerging with different gains g_i from different loudspeakers around a listener can be related to physical quantities measured at the listener location. In particular, it can be shown that localisation given at low frequencies by interaural phase localisation theories below about 700Hz is determined by the vector given by dividing the overall acoustical vector velocity gain of a reproduced sound at the listener by the acoustical pressure gain at the listener. The resulting vector, for natural sound sources, has length one and points at the direction of the sound source. For sounds reproduced from several loudspeakers, the length r_V of this vector should ideally be as close to 1 as possible, especially for sounds intended to be near azimuths $\pm 90°$, and the azimuth direction θ_V of this vector is an indication of the apparent sound direction.

Between about 700 Hz and about 4 kHz (and these figures are merely rather fuzzy indications), and also for noncentral listeners hearing mutually phase-incoherent sound arrivals from different speakers below 700 Hz, localisation is determined by that vector which is the ratio of the vector sound-intensity gain to the acoustical energy gain of a reproduced sound. Again, for natural sound sources, this vector would have length one and point to the sound source. For reproduced sounds, the length r_E of this vector should be as close to one as possible (it can never exceed 1) for maximum stability of the image under listener movement, and its direction azimuth θ_E is an indication of the apparent direction of the image.

These vector quantities can be computed from a knowledge of the gains g_i with which a sound source is fed to each of the loudspeakers, as follows. Suppose one has n loudspeakers all at equal distances from the listening position; let the i'th loudspeaker be at azimuth θ_i and reproduce a sound with gain g_i. (While the theory can be developed for complex gains g_i, see [2 ,3], we here assume that g_i is real for simplicity). The acoustical pressure gain is then simply the sum

$$P = \sum_{i=1}^{n} g_i \qquad (5)$$

of the individual speaker gains. The velocity gain is the vector sum of the n vectors with respective lengths g_i pointing towards azimuth θ_i (i.e. towards the associated loudspeaker), which has respective x-and y-components

$$V_x = \sum_{i=1}^{n} g_i \cos\theta_i \qquad (6x)$$

and

$$V_y = \sum_{i=1}^{n} g_i \sin\theta_i \; . \qquad (6y)$$

By dividing this velocity gain vector by the presure gain P, one obtains a <u>velocity localisation vector</u> of length $r_V \geqslant 0$ pointing in direction azimuth θ_V , where

$$r_V \cos\theta_V = V_x/P \qquad (7x)$$

$$r_V \sin\theta_V = V_y/P \; . \qquad (7y)$$

θ_V is termed the velocity vector localisation azimuth, or <u>Makita</u> localisation azimuth, and is the apparent direction of a sound at low frequencies if one turns one's head to face the apparent direction. r_V is termed the <u>velocity vector magnitude</u> and ideally equals one for single natural sound sources. The two quantities θ_V and r_V are indicative of apparent localisation direction and quality according to low-frequency interaural phase localisation theories, with deviations of r_V from its ideal value of one indicative of image instability under head rotations, and poor imaging quality particularly to the two sides of a listener.

A similar procedure is used according to energy theories of localisation, but with the square g_i^2 of the gain from each speaker replacing the gain g_i. The overall reproduced energy gain is

$$E = \sum_{i=1}^{n} g_i^2 \qquad (8)$$

and the sound-intensity gain is the vector sum of those vectors pointing to the i'th speaker with length g_i^2, which has x- and y-components

$$E_x = \sum_{i=1}^{n} g_i^2 \cos\theta_i \qquad (9x)$$

and

$$E_y = \sum_{i=1}^{n} g_i^2 \sin\theta_i \; . \qquad (9y)$$

By dividing this sound-intensity gain vector by the overall energy gain E, one obtains an <u>energy localisation vector</u> of length $r_E \geqslant 0$ pointing towards the direction azimuth θ_E, where

$$r_E \cos\theta_E = E_x/E \qquad (10x)$$

$$r_E \sin\theta_E = E_y/E \; . \qquad (10y)$$

θ_E is termed the <u>energy vector localisation azimuth</u>, and is broadly indicative of the apparent localisation direction either between 700 Hz and around 4 kHz, or at lower frequencies in the case that the sounds arrive in a mutually incoherent fashion at the listener from the n loudspeakers.

r_E is termed the <u>energy vector magnitude</u> of the localisation, and is indicative of the stability of localisation of images either in the frequency range 700 Hz to 4 kHz or at lower frequencies under conditions of phase-incoherence of sound arrivals. As before with r_V, the the ideal value for a single sound source is equal to 1. Because r_E is the average (with positive coefficients $g_i^2/(\Sigma g_i^2)$) of n vectors of length 1, it is only equal to 1 if all sound comes from a single speaker. Generally r_E is less than 1, and the quantity $1-r_E$ is roughly proportional to the degree of image movement as a listener moves his/her head. Ideally, for on-screen sounds with HDTV, one would like $1-r_E \lesssim 0.02$, but one finds that typically for central stereo images with 2-speaker stereo that $1-r_E = 0.134$, and for surround-sound systems that $1-r_E$ lies between 0.25 and 0.5 .

For frontal stage stereo systems subtending relatively narrow angles (say with stage widths of less than 60°), it is found that the value of r_V is nor critical providing that it lies between say 0.8 and 1.2, but that the value of r_E is an important predictor of image stability. For surround sound systems aiming to produce images at each side of a listener, however, making r_V equal one accurately at low frequencies becomes much more important, since the low-frequency localisation cue is one of the few cues that can be made correct for such side-stage images, and the accuracy of such localisation depends critically on the accuracy of r_V.

It has thus been found that the localisation criteria for front-stage stereo and for surround sound are somewhat different in their practical trade offs.

For all methods of reproduction, it has been found that it is desirable that the two localisation azimuths θ_V and θ_E should be broadly equal, so that any decoding method should ideally be designed to produce speaker feed gains g_i for all localisation azimuths such that

$$\theta_V = \theta_E \qquad (11)$$

at least for frequencies up to around 3½ or 4 kHz. This ensures that different auditory localisation mechanisms give broadly the same apparent reproduced azimuth, especially in those frequency ranges in which more than one mechanism is operative. Equation (11), which is an equation relating the quantities g_i via equs. (5) to (10), can be written in the form

$$E_x V_y = E_y V_x \,, \qquad (12)$$

and is seen in general to be cubic in the gains g_i. If equations (11) or (12) are satisfied, there is a tendency for illusory phantom images to sound more sharp and precise than if θ_V and θ_E differ substantially.

However, sharpness is not the same as image stability, and additional requirements on r_V and r_E are necessary for optimum imaging stability. For surround sound systems, it is highly desirable, under domestic scale listening conditions, that

$$r_V = 1 \qquad (13)$$

for all reproduced azimuths at low frequencies, typically under 400 Hz, at a central listening location. However, above 400 Hz, it is instead desirable that the value of r_E be maximised. With some exceptions, it is generally not possible to design a reproduction system to be such as simultaneously to maximise r_E in all reproduced directions, so that in practice, some design trade off is made between the values of r_E in different reproduced directions. In general, for surround-sound systems, r_E above 400 Hz is designed to be larger across a frontal stage than in side and rear directions, but not to the extent that side and rear sounds become intolerably unstable.

The optimisation of r_E above about 400 Hz is partly a matter of design skill and experience obtained over a period of years, but some of this skill can be codified as informal rules of thumb. It is generally highly undesirable that $1-r_E$ should vary markedly in value for sounds at only slightly different azimuths, since such variations will cause some sounds to be much more unstable than other nearby ones. In general, it is desirable that r_E be maximised at the due front azimuth or across a frontal stage, and it is desirable that the values of r_E in other directions vary smoothly.

A decoder or reproduction system for 360° surround sound is defined to be Ambisonic if, for a central listening position, it is designed such that

(i) the equations (11) or (12) are satisfied at least up to around 4 kHz, such that the reproduced azimuth $\theta_V = \theta_E$ is substantially unchanged with frequency,

(ii) at low frequencies, say below around 400 Hz, equation (13) is substantially satisfied for all reproduced azimuths, and

(iii) at mid/high frequencies, say between around 700 Hz and 4 kHz, the energy vector magnitude r_E is substantially maximised across as large a part of the 360° sound stage as possible.

In large reproduction environments, such as auditoria, it is unlikely that a listener will be within several wavelengths of a central listening seat; under these conditions, the requirement of equ. (13) is desirably not satisfied, although it is still found that satisfying equs. (11) or (12) gives useful improvements in phantom image quality, for reasons too involved to be gone into here.

The Ambisonic decoding equations (11) to (13), plus the requirement for maximising r_E above 400 Hz, are in general a highly nonlinear system of equations. Prior-art solutions to these equations involved the use of loudspeaker layouts with a rather high degree of symmetry, e.g. regular polygons, rectangles, or involving diametrically opposite pairs of louspeakers [2 ,18,1], but the new solutions reported in this paper apply to much less symmetrical speaker layouts.

However, the main novelty of the methods of this paper versus previously known Ambisonic decoders is not in the use of irregular speaker layouts. In all prior art Ambisonic decoders (see [2],[18], [1]), the decoding matrix was frequency dependent so as to ensure that $r_V = 1$ at low frequencies and that r_E was larger at high frequencies, but all such matrices had the special property that the pressure signal P had exactly the same directional gain pattern (as a function of encoded azimuth θ) at low and high frequencies, apart from a simple adjustment of overall gain with frequency. In this paper, by contrast, we for the first time will consider frequency-dependent decoders satisfying the Ambisonic decoding equations where the directional gain pattern of the pressure signal P varies with frequency. Typically, for decoders having better front-stage than back-stage image stability, the back-sound gain divided by front-sound gain for the pressure signal will have a smaller value at low frequencies (for which r_V typically equals 1) than at higher frequencies (for which typically r_E is maximised with a greater value for front stage sounds than for back stage sounds).

The realisation that the directional gain pattern of the pressure signal P, i.e. for layouts of speakers at identical distances, the sum of the speaker feed signals, can be varied with frequency while still giving solutions of the Ambisonic decoding equations gives an extra degree of freedom in optimising the results of Ambisonic decoders. In particular, at high frequencies, it can be used to make r_V vary substantially with encoding azimuth θ, rather than to be substantially constant with azimuth as it was in earlier prior-art Ambisonic decoders.

Over a decade of experience in designing Ambisonic decoders has suggested two rules-of-thumb in obtaining good high-frequency designs. The first rule of thumb is that when r_E is maximised, it is found that r_E substantially equals r_V, so that the best high frequency designs will tend to be such that r_V varies with encoding azimuth θ according to the same general pattern as r_E, insofar as this is possible. In general r_E varies with direction with higher harmonic components than r_V, so that only rarely can r_E and r_V have exactly the same values for all azimuths, but a fairly close matching of the two quantities generally gives best high frequency results.

The other rule of thumb is that the overall reproduced energy gain E tends to be largest when r_E is smallest, and vice-versa. This is unfortunate insofar as it emphasises the contribution of those directions that are least well localised. Thus, in designing decoders in which r_E is maximised across the frontal stage, it is generally desirable to modify the encoded directional signal by means such as forward dominance of B-format signals to increase the relative gains of frontal sounds to compensate such decoder gain loss. In general, such modifications will alter the reproduced azimuth so that it no longer equals the encoded azimuth θ, but such modifications in practice will not be so large as to introduce gross directional distortions in the reproduced sound image.

It is in general desirable that, even if $\theta_V = \theta_E$ does not precisely equal encoding azimuth θ, one should ensure that the decoded azimuth $\theta_V = \theta_E$ does not vary substantially with frequency (at least up to around 3½ kHz), so as to avoid frequency-dependent image smearing.

4. ENHANCED B-FORMAT AMBISONIC DECODERS

The aim of this section is to describe a new generation of B-format Ambisonic decoders with improved frontal-stage image stability. Figures 1 to 4 show typical speaker layouts that we shall consider for 360° surround-sound reproduction. Figure 1 shows a rectangular speaker layout using left-back L_B, left-front L_F, right-front R_F and right-back R_B speakers at repective azimuths $180^\circ-\phi$, ϕ, $-\phi$ and $-180^\circ+\phi$, supplemented by an extra centre-front C_F loudspeaker. Figure 2 shows a similar 5-speaker layout, except that now the azimuth angles $\pm\phi_F$ of the front pair differs from that $180^\circ \pm \phi_B$ of the rear pair, so that the L_B, L_F, R_F and R_B speakers form a trapezium layout.

Figures 3 and 4 show similar rectangle and trapezium speaker layouts respectively, but this time supplemented by a frontal pair of speakers C_L and C_R at respective azimuths $+\phi_C$ and $-\phi_C$.

There is already a long-known Ambisonic decoder for B-format for the 4-speaker rectangular layout shown in figs. 1 or 3, described in refs. [1], [2], [18] and [12], for which $\theta_V = \theta_E = \theta$ for all encoded azimuths θ. However, this decoder has identical r_E for due front and due back sounds above about 400 Hz, and for square loudspeaker layouts has r_E equal to 0.7071 in all directions, which is not adequate for frontal-stage sounds for use with TV. However, it is possible to show that for the rectangular layouts of figs. 1 and 3, there are other Ambisonic decoders that feed the additional frontal speakers so as to increase r_E for front-stage sounds, at the expense of slightly decreasing r_E at the sides and rear.

Although the speaker layouts of figs. 1 to 4 lack a high degree of symmetry, they are still left/right symmetrical, i.e. symmetrical under reflection about the forward direction. We assume here that we are considering a left/right symmetric speaker layout in which all speakers lie at the same distance from a listener. We seek to find for the various speaker layouts of this kind those real left/right symmetrical linear combinations of the B-format signals W, X and Y such that the equations

$$\theta_V = \theta_E = \theta \tag{14}$$

are satisfied for all encoding azimuths θ in the 360° sound stage. Having found all such solutions, the next step is to find among those solutions ones with $r_V = 1$ for low frequencies and those with maximised r_E at higher frequencies, and to use a frequency-dependent matrix to implement these two matrices in a frequency-dependent manner as an Ambisonic decoder. The method of solution we now describe works for quite general left/right symmetric speaker layouts, although the numerical details of the solution process can be extremely messy in particular cases, requiring the use of powerful computing facilities.

In order to take advantage of left/right symmetry, it is convenient to express the speaker feed signals illustrated in figures 1 to 4 in sum and difference form as follows:

$$L_F = M_F + S_F$$
$$R_F = M_F - S_F$$
$$L_B = M_B + S_B$$
$$R_B = M_B - S_B$$
$$C_L = C_F + S_C$$
$$C_R = C_F - S_C \ . \qquad (15)$$

Because of the left/right symmetry requirement, at any frequency one can write the signals C_F, S_C, M_F, S_F, M_B and S_B in terms of B-format in the following form:

$$S_C = k_C\, Y$$
$$S_F = k_F\, Y$$
$$S_B = k_B\, Y \ ,$$
$$C_F = a_C\, W + b_C\, X$$
$$M_F = a_F\, W + b_F\, X$$
$$M_B = a_B\, W - b_B\, X \ , \qquad (16)$$

where k_C, k_F, k_B, a_C, a_F, a_B, b_C, b_F, b_B are real coefficients (which typically will all be positive, excepting k_C which may be zero).

Figure 9 shows the general architecture of an Ambisonic decoder for the speaker layouts of figs. 1 to 4, based on equs. (15). At the B-format input, there is provided optionally a forward-dominance adjustment according to equs. (3) so that the relative front/back gain balance and directional distribution of sounds can be adjusted. Each of the 3 resulting B-format signals is then passed into a phase compensated band-splitting filter arrangement, such that the phase responses of the two output signals is substantially identical. Typically for domestic listening applications, the crossover frequency of the phase-compensated bandsplitting filters will be around 400 Hz, and the sum of low and high frequency outputs will be equal to the original signal passed through an all-pass network with the same phase response. For example, the low-pass filters in fig. 9 might be the result of cascading two RC or digital first order low-pass filters with low frequency gain 1, and the high-pass filters might be the result of cascading two first order high pass filters with the same time constants, with high frequency gain of minus one; these filters sum to a first order all-pass with the same time constant, and have identical phase responses.

The low-frequency B-format signals resulting are fed to a low-frequency decoding matrix to implement equations (16) for coefficients appropriate below 400 Hz (typically ensuring that $r_V = 1$), and the high-frequency B-format signals are fed to a second high-frequency decoding matrix to implement equations (16) for a second set of coefficients appropriate to the higher frequencies at which r_E is to be maximised. The resulting low and high frequency signals C_F, S_C (where it exists), M_F, S_F, M_B and S_B are then summed together and fed to output sum and difference matrices to provide speaker feed signals suitable for the speaker layouts of figures 1 to 4. In the case of 5-speaker layouts such as those of figs. 1 or 2, or in the case of 6-speaker layouts in the case that $C_L = C_R = C_F$, the S_C signals path and the top sum-and-difference matrix in figure 9 may be omitted.

The use of phase compensation (i.e. phase matching) of the bandsplitting filters in figure 9 is found to be highly desirable for surround sound decoders, since any "phasiness" errors due to relative phase shifts between signal components are magnified by the large 360^o angular distribution of sounds, although in some cases, the use of filters that are not phase matched may prove acceptable. It is also clear that the architecture of fig. 9 can be extended to 3 or more frequency bands by using a three-band phase-splitting arrangement with three decoding matrices, so as to optimise localisation quality separately in three or more bands. Typically a three band decoder might have crossover frequencies at 400 Hz and at or around 5 to 7 kHz so as to optimise localisation in the pinna-colouration frequency region above about 5 kHz [17].

Besides possibly implementing forward dominance and overall gain adjustments, the decoding matrices in figure 9 will, in general, have matrix coefficients that vary with the speaker layout in use, so that a typical Ambisonic decoder implemented as in figure 9 will have a means of causing the matrix ceoefficients to be altered in response to the measured or assumed speaker layout shape and angles shown in figs. 1 to 4. This may be done by a microprocessor software adjustment of coefficients, or by manual adjustment means.

The next section describes how the coefficients of the decoding matrix (16) may be determined for a particular layout shape.

5. SOLVING AMBISONIC DECODING EQUATIONS

We illustrate the method of finding the general left/right symmetric B-format decoder solution to equations (14) with reference to a decoder for the 5-speaker layout of figure 2, assuming speaker feed signals of the form given by equs. (15) with (16). A direct computation using equs. (5), (6), (8) and (9), yields from equs. (15)

$$P = C_F + 2M_F + 2M_B \tag{17a}$$

$$V_x = C_F + 2M_F\cos\phi_F - 2M_B\cos\phi_B \tag{17b}$$

$$V_y = 2S_F\sin\phi_F + 2S_B\sin\phi_B \tag{17c}$$

$$E = C_F{}^2 + 2(M_F{}^2 + S_F{}^2 + M_B{}^2 + S_B{}^2) \tag{18a}$$

$$E_x = C_F{}^2 + 2(M_F{}^2 + S_F{}^2)\cos\phi_F - 2(M_B{}^2 + S_B{}^2)\cos\phi_B \tag{18b}$$

$$E_y = 4M_FS_F\sin\phi_F + 4M_BS_B\sin\phi_B \ , \tag{18c}$$

where, by a slight abuse of notation, we use the same symbols to represent the gains of signals for a given encoding azimuth θ as we do to indicate the signals themselves.

The quantities P, V_x and V_y of equs. (17) are all left/right symmetric real linear combinations of W, X and Y. In particular, from equs. (7), the requirement that $\theta_V = \theta$ as in equ. (14) implies that

$$V_x : V_y = X : Y \ , \tag{19a}$$

so that we may put

$$V_x = 2^{\frac{1}{2}}gX \tag{19b}$$

$$V_y = 2^{\frac{1}{2}}gY \tag{19c}$$

where g is an overall gain factor. In order to simplify the equations following, we may set

$$g = 1 \tag{19d}$$

so as to avoid repeating a lot of factors g in the analysis; however, it will be necessary to multiply the overall decoder coefficients thus obtained in equs. (16) by an overall gain g afterwards, in order to obtain a desired overall gain of reproduction. In particular, it is desirable to match the gains of low and high frequency ambisonic decoding matrices so as to ensure a flat overall frequency response.

The method of solution of the system of equations (17) to (19) with equation (12), which gives the desired signals (16) for the decoder of figure 9 can be summarised as follows, although we shall not give the full details here, since they are very lengthy.

First we substitute equations (17) to (19) into equation (12), and put $S_F = k_FY$ and $S_B = k_BY$ throughout from equs. (16). This gives an equation both sides of which are factored by Y, which factor can then be discarded. The resulting equations can then be expressed entirely in terms of the signals M_F, M_B, X and W by substitutions from equs. (17) and (19), yielding an equation quadratic in these four signals.

By a process of solving quadratic equations in the coefficients, it is then possible to express this quadratic equation in M_F, M_B, X, W as the equality of two quadratic expressions, each of which can be factored into two linear expressions, on one side of the equation these being linear combinations of W and X, and on the other side these being linear combinations of M_F, M_B and X. This factorisation makes use of the equ. (2) to eliminate Y^2 by expressing it in terms of W^2 and X^2. This factorisation process is the most complicated part of the solution process.

These equations are then solved by setting one factor on each side to equal a multiple (which can be chosen at will) of one factor on the other. The other factors on each side are then equal to each other also, which gives a system of two linear equations in M_F, M_B, X and W. These can then be solved for M_F and M_B in terms of X and W, and then the other signals in equs. (16) can be found by substitutions from equs. (17) and (19).

This whole process is quite complicated, and gives solutions in terms of arbitrary predetermined k_F and the multiple C in the factorisation chosen at will. (k_B turns out to be determined by k_F via equs. (17) and (19)).

A similar, and even more complicated, solution process involving factorisations of quadratic expressions in signals, has been found for speaker layouts with 6 or more speakers, and an explicit solution algorithm written for the 5- and 6-speaker cases. In the 6 speaker case, the solution involves 3 predetermined variables rather than the 2 of the 5-speaker case (involving a choice of k_C as well), and 5 and 6 predetermined variables for the respective 7- and 8-speaker case.

It turns out that, arising from the various factorisations and quadratic equations in the solution process, there is more than one family of solutions - four different families in the 5-speaker case. Of these families, one has markedly better-performing decoders (measured by largeness of r_E) than the others. However, the complications of the solution process do not end with the selection of the appropriate signs and factorisations to give the best family of solutions.

The values of the predetermined parameters k_F and C (in the 5-speaker case) need to be chosen to yield the "best" low and high frequency solutions. The "best" low frequency solution is that that gives $r_V = 1$ in all directions Θ, i.e. that solution for which P = 2W. The only problem here is that it turns out that, even within the "best" family of solutions, for any given speaker layout angles ϕ_F and ϕ_B (see fig. 2), there are either no values of the parameters k_F and C for which a P = 2W solution exists (in which case no fully ambisonic decoder at low frequencies is possible), or else there are <u>two</u> such values of the predetermined parameters for which P = 2W.

In the former case (no $r_V = 1$ solution), one can only find that solution with the highest possible value of r_V for low frequency use. In the latter case, it turns out that one of the two $r_V = 1$ solutions has a markedly better r_E performance than the other, and care must be taken that this solution is found. The process of finding the solution is complicated by the fact that all the good solutions lie within a very tiny range of values of the parameter k_F, all very close to a value of k_F that leads to singular (non-real) behaviour in the solutions. Thus, to locate values of the parameters k_F and C that yield good solutions, one first of all has to locate the "singular" values of k_F and then search nearby.

The finding of high-frequency solutions is complicated by the fact that they involve a tradeoff of r_E among different directions, so that it is impossible completely to justify one tradeoff as being superior to another. After extended investigations, we have provisionally decided on those solutions having the following two properties for high frequency use:

(i) a solution such that $P = 8^{\frac{1}{2}}W + b_P X$ for some real constant b_P, and

(ii) such that an azimuth $\theta = 0^o$ (front-centre) sound gives zero output from the L_B and R_B speakers.

Condition (i) ensures that $r_V = 0.7071$ for azimuth $\pm 90^o$ sounds, the same value as for the optimum high-frequency square-layout 4-speaker B-format decoder [2], [18]. Condition (ii) helps ensure that the important front-centre "dialogue" position does not suffer from rear speaker crosstalk, which helps ensure good results even in a large auditorium situation.

Even in the high frequency case, one finds that there are two sets of values of the predetermined parameters k_F and C that satisfy conditions (i) and (ii), and care must be taken to choose the one giving the largest value of r_E.

As the reader must have realised, finding the optimum low and high frequency solutions to a given speaker layout, although involving no mathematics more advanced than algebra and the solution of quadratic equations, is a quite lengthy and tedious process, even with computer aids, because of the multiple families of solutions and the need to search in the parameter space for the best solutions. Methods of finding the solutions for given arbitrary values of the speaker angles ϕ_F and ϕ_B suitable for use in consumer decoders need to be relatively fast and automatic, and methods involving interpolation from solutions for known layouts may be used. We have computed a "grid" of solutions for use in such an interpolation procedures.

Once the low and high-frequency solutions satisfying $\theta = \theta_V = \theta_E$ for B-format have been determined, one then needs to match their reproduced energy gains by adjusting the constant g mentioned in connection with equs. (19), so as to ensure a flat frequency response. One also needs to check the localisation parameters r_V, r_E, θ_V and θ_E in the crossover region between the low and high frequency bands to ensure that this is still satisfactory. In practice, the localisation azimuths θ_V andθ_E in the crossover region differ from θ only by a small fraction of a degree and the gain varies only by a small fraction of a dB, so that the crossover region causes no problems in performance, but this should be checked anew for every new design in case an exception to this good behaviour arises.

In table 1, we list the parameters of one particular solution, that for a 5-speaker layout for which $\phi_F = 45^o$, $\phi_B = 50^o$, with values of the "psychoacoustic parameters" r_V, r_E and gain of reproduction in dB. It will be seen that r_E is largest at the front and smallest at the back. By way of comparison, a conventional 4-speaker B-format ambisonic decoder has $r_V = 1$ and $r_E = 0.6667$ at low frequencies in all directions, and $r_V = r_E = 0.7071$ at high frequencies in all directions. It will be seen that the decoder of table 1 gives significantly enhanced frontal-satge image stability (i.e. larger r_E) than the 4-speaker decoder, and also that r_V and r_E roughly track in value at high frequencies as azimuth varies, which we have found to be highly desirable if the value of r_E is to be maximised.

An unfortunate side-effect in this decoder, however, is that the poorly localised rear-stage sounds are reproduced almost 4 dB louder than the more important and better-localised frontal stage sounds. By preceeding the decoder with a forward dominance transformation (3), or by incorporating such a forward dominance transformation within the decoding matrix coefficients a_C, a_F, a_B, b_C, b_F, b_B, k_C, k_F, k_B, one can compensate for this front/back gain imbalance.

6. DECODER PERFORMANCE

As with previous ambisonic decoders, the actual subjective performance of the new decoders varies considerably according to the shape of the speaker layout used, with "long thin" layouts giving better front and back localisation but poorer side localisation stability, and "wide" speaker layouts giving better side but poorer front and back localisation stability.

In this section we show the results of studies of decoders for various 5-speaker (and one 6-speaker) layouts, to show what the localisation parameter tradeoffs are. All the decoders listed have been adjusted so that (i) low and high-frequency reproduced energy gains are matched for a flat frequency response, and (ii) forward dominance, which alters reproduced azimuths of sounds, has been applied to equalise front and rear stage reproduced gains.

Table 2 lists a range of low- and high-frequency designs for 9 different 5-speaker layouts computed by the methods of sections 6 and 7, including forward dominance to compensate for front/rear gain variations and a gain adjustment of the high frequency decoder to ensure that it has the same gain at reproduced azimuths $\pm$ 45^{o} as the low frequency decoder. The cases ϕ_F = 35^{o}, 45^{o} and 55^{o} and $\phi_B - \phi_F$ = 5^{o}, 0^{o} and -10^{o} are listed. Because both the low and high frequency decoder matrices are chosen according to "objective" criteria, it is possible to use quadratic interpolation to derive 5-speaker Ambisonic decoders for other intermediate values of ϕ_F and ϕ_B - ϕ_F .

It has been found, however, that low-frequency r_V = 1 solutions do not exist for all angles ϕ_F, ϕ_B. For example, for the values ϕ_F = 35^{o}, ϕ_B = 55^{o}, there is no r_V = 1 solution. In general, such low frequency solutions are found to exist for $\phi_B \leqslant \phi_F$, but in general, ϕ_B cannot be more than about 10% or 15% larger than ϕ_F before an r_V = 1 solution can no longer be found. In cases where the r_V = 1 solution does not exist, one should seek to use a low frequency solution having as large a value of r_V as is possible if this still gives a greater r_V than at high frequencies.

The attainable value of r_E at high frequencies for front stage sound is clearly enhanced by the use of the 5-speaker decoder, as seen in table 2, as compared to similar 4-speaker rectangle decoders, thanks to the significant output from the C_F speakers, with r_E typically being increased from 0.7071 for a square layout to around 0,835 when a C_F speaker is added. This almost halves the desgree of image movement for front stage sounds. It will be seen that the value of r_E at the sides and back is not drastically reduced, although the average value for r_E over the whole 360^{o} stage is not increased, and in fact is slightly reduced.

Thus, although the value of r_E at the front is not brought up to ideal values very close to 1, the use of a 5-speaker Ambisonic decoder provides an improved image stability, as compared to previous designs [1], without giving an unacceptable loss of the rest of the surround sound 360° stage. Thus a 5-speaker Ambisonic decoder designed as in this paper matches TV use a great deal better than earlier decoders, and makes good use of just three transmission channels, although there is still a need for enhancing front-stage results by adding extra transmision channel signals.

The use of six speakers gives a further improvement of r_E at the front, improving image stability further, while still giving reasonable values of r_E (typically around 0.6) around the rest of the sound stage, as the Ambisonic decoder solution listed in table 3 illustrate. The degree of frontal stage image movement of the 6-speaker decoder is typically only 40% of that encountered with 4-speaker decoders. Where possible, the use of six speakers is preferable to five in terms of frontal image stability.

7 . SUPPLEMENTARY AMBISONIC CHANNELS

While the 5- and 6-speaker B-format decoders give greatly improved frontal-stage image stability without much loss of image stability around the rest of the sound stage, the further improvements in frontal stage image stability required for HDTV require the use of more than three signals, and the question arises as to how additional signals can be added to B-format so as to give improved r_E across the frontal stage, preferably using the speakers already used for 5- or 6-speaker decoding. Moreover, these additional signals should continue to work well as the shape of speaker layout is altered to fit into different room shapes.

One approach would simply be to suplement the B-format signals by separate signals conveying 3 or 4-speaker stereo to just the frontal speakers, using the methods discussed in refs. [3, 5-7] for frontal stage stereo. Unfortunately, this would imply the use of six transmission channels to feed the 5-speaker layouts of figs. 1 and 2 or seven channels to feed the layouts of figs. 3 or 4 - i.e. more channels than loudspeakers.

This problem arises because of the need to adapt the way signals are decoded according to the speaker layout used, and the way that 360° surround-sound Ambisonic decoding varies with layout is very different from the way that, say, 3-speaker stereo signals are varied (if at all). Thus, while the 5-speaker feeds for a particular speaker layout might suppress one channel's worth of a six-channel B-format plus 3-channel stereo transmission's information, the information being suppressed varies according to the speaker layout, making the transmission of the extra channel's worth of information non-redundant.

We have sought a compromise approach, whereby the Ambisonic ability to adapt to give optimal decoded results via a variety of speaker layouts is retained, but where most of the advantages of having additional frontal stereo channels are retained in terms of improved frontal image stability. The aim is to use the minimum possible number of additional channels to obtain as much of the improved stability as possible without the necessity of transmitting three additional channels. It turns out that the new 5- and 6-speaker decoders described above provide a far better basis for such a structured frontal-stage enhancement of frontal stage image stability by adding extra audio channels than did prior-art 4-speaker Ambisonic decoders for B-format.

At the very simplest, one can add one additional channel signal, denoted by E which incorporates a feed signal, with gain one, intended to be fed only to a centre-front loudspeaker. Such an isolated centre front signal has been found to be important in film and HDTV applications in that typically dialogue and other sounds from the centre of the screen area are more important than any other directions, and experiments in using Ambisonics plus a front-centre speaker feed by one author (GJB)(unpublished) have confirmed that such a method works well in cinema applications. However, having only a single sound position that is highly stable is rather inflexible and unsubtle for many applications.

Nevertheless, such an added E channel, in combination with B-format signals can yield useful benefits. A second added channel F can be used largely to cancel front-to-rear stage crosstalk (which is largely due to the Y signal) and to widen the frontal stage, thereby giving, in combination with E and the 3 B-format signals, a frontal stage reproduction closely approximating 3-channel frontal stereo. Any sounds assigned to such a high-stability frontal stage should also be encoded conventionally into the 3 B-format signals so that users discarding the E and F signals will still get B-format reproduction incorporating those sounds.

These considerations make us propose an <u>enhanced</u> <u>B-format</u>, comprising (up to) five signals W, X, Y, E and F for studio production applications in horizontal surround-sound with enhanced frontal image stability. As shown in figs. 6 and 10, and discussed also in ref. [4], this encodes signals from azimuth θ into the five channels with respective gains:

W with gain 1

X with gain $2^{\frac{1}{2}}\cos\theta$

Y with gain $2^{\frac{1}{2}}\sin\theta$

E with gain $k_e(1 - 3.25(1 - \cos\theta))$ for $|\theta| \leq \theta_S$
and gain 0 for $|\theta| > \theta_S$

F with gain $2^{\frac{1}{2}}k_f \sin\theta$ for $|\theta| \leq \theta_S$,
and gain $-2^{\frac{1}{2}}k_g \sin\theta$ for $|180^\circ-\theta| \leq \theta_B$, and gain 0 otherwise,

where θ_S is a frontal stage half width, typically between 60° and 70°, θ_B is a rear stage half width, typically around 70°, and the gains k_e, k_f and k_g may be chosen between zero (for pure B-format) and one (for reproduction effect purely from the front or rear stages). The coefficient 3.25 may be subjected to slight changes in value between 3 and $3\frac{1}{2}$; the value suggested is a provisional proposal.

Enhanced B-format thus allows, by variations of the gains k_e and k_f (which should preferably be roughly equal), the assignation of frontal stage sounds anywhere between pure B format and pure frontal stage postioning. As a production format, it allows reproduction in a large variety of different modes, as is shown in ref. [4].

For Ambisonic reproduction via 5 or 6 loudspeakers, figure 11 shows a typical architecture for decoding enhanced B-format signals, incorporating an Ambisonic decoding algorithm as described earlier for pure B-format signals. Across the frontal stage for $k_e = k_f = 1$, it will be seen that $F = Y$ and it will further be seen that for front centre sounds, $W = E$ and $X = 2^{\frac{1}{2}}E$. Thus the signals

$$W-E$$

$$X - 2^{\frac{1}{2}}E$$

and

$$Y - F \tag{7-1}$$

equal zero and cancel for due front sounds, and Y - F continues to cancel across the rest of the frontal stage, whereas, as θ increases towards 60° and the gain of E falls to zero and then becomes negative, the other two of the signals of equ. (7-1) become large, but in a manner that causes very little output from rear speakers.

Thus the first step in an Enhanced B-format Ambisonic decoder is to derive the "cancelled" signals of equ. (7-1) to feed a conventional 5- or 6- (or greater) speaker B-format Ambisonic decoder, and to take the E signal and to feed it with an appropriately chosen gain via a phase-compensating all-pass network (to match the filter networks in the Ambisonic decoder) to feed centre front loudspeakers directly.

For azimuth zero sounds with $k_e = 1$, this gives ideal localisation of centre front sounds. For sounds at other azimuths, the change of the sign of E's gain towards the edges of the frontal stage mean that the directly fed E signal now tends to cancel the centre-front speaker feeds deriving from the output of the B-format Ambisonic decoder, leaving largely just the front left and right speaker feeds. if one then provides the frontal speakers with a multiple of the F signal (passed though another phase-compensating all-pass network) as a left/right difference signal, the width of this largely frontal stage reproduction can be given a desired degree of left/right separation.

By this means, the architecture of figure 11, with intial "cancellation" of the enhancement channnels E and F from the B-format signals before these are Ambisonically decoded, and the provision of direct speaker feed signals, via phase compensation networks, from E and F, can provide substantially conventional 3-speaker stereo from frontal-stage sounds with $k_e = k_f = 1$, with relatively low crosstalk onto rear speakers, provided that the Ambisonic decoder design is a type having additional frontal stage speakers of the kind described above such as in figures 6 to 10. The cancellation by E of a central speaker feed for encoded azimuths near $\pm 60^o$ can be adjusted for a given decoder design by a careful choice of the direct spaeker feed gains of the E signal. In particular, while figure 11 shows the E and F signals as being simply fed forward and mixed into the C_F, S_C and S_F signal paths in a manner that is (apart from phase compensation) independent of frequency, in a practical design, a judicious feed of a small amount of E signal to the M_F and M_B signal paths, and of the F signal to the S_B signal path in small amounts, possibly with a frequency dependence in the gain, can yield a small but useful improvement in the overall performance of front-stage stereo sounds.

It will thus be seen that the diagram of figure 11 illustrates the structure of an enhanced B-format decoder only in its most basic form, and that slightly more complex direct feeds of the E and F signals, with the dominant components feeding respectively C_F and S_C and S_F may be used to optimise front-stage performance, possibly using gains that vary somewhat with frequency.

In typical 5-speaker decoders, it is found that the gain of the E signal fed to C_F is typically around $g_E = 2$, and the gain g_F of the F signal fed to S_F is typically around 1 to ensure broadly "discrete" frontal 3-speaker stereo. These figures vary somewhat with the Ambisonic decoder design and speaker layout.

The function of the E signal is to increase the "separateness" of the frontal speaker feeds, especially that of centre-front, whereas the F signal has the effect of cancelling out the left/right difference signal from the rear speakers and increasing it at the front, thereby conveting signals from true Ambisonic surround-sound signals to ones dominantly reproduced from a frontal stage.

The gains k_e and k_f that give predominantly discrete speaker feeds at the front are around 1, and if one wishes to keep rear speaker levels low for frontal stage sounds, it is desirable to put $k_f = 1$. However, in general, an improved localisation quality of phantom front-stage images is typically achieved not with $k_e = 1$, but with k_e having a value near 0.4 or 0.5, as is shown by computed values of Θ_V and Θ_E for decoders of the form of figure 11.

The design of the best direct gains for the E and F signals for each B-format Ambisonic decoding design, for each speaker layout, is a matter of subjective tradeoffs of different aspects of frontal-stage localisation quality by the designer, and does not form a strict part of the system standards for enhanced B-format, but rather a decoding option that may be varied within quite wide limits. It is, of course, necessary to ensure that reasonable results <u>can</u> be obtained, and the basic architecture of fig. 11 based on the 5- or 6-speaker Ambisonic B-format decoders described in sections 4 and 5 of this paper, or its minor modifications suggested above, does broadly achieve the desired result of enhanced frontal-stage image stabilty very similar to the use of separate frontal-stage stereo transmission channels, while still incorporating full B-format surround sound signals in an economical manner.

While the above has explained how the decoder of fig. 11 uses the additional channels E and F to provide an essentially "discrete" frontal stage for sounds with azimuth θ of magnitude less than θ_S, the same method also reduces rear-to-front stage crosstalk for rear stage azimuths θ such that $|180^\circ-\theta| \leq \theta_B$ when k_b has a value near 1. This is because subtraction of F from Y as in equs. (7-1) in such a rear stage has the effect of doubling the amplitude gain of Y, and thereby increasing the magnitude of the difference signals S_C, S_F and S_B in figure 11 by a factor 2, rather than cancelling them out. The addition of F to the front stage difference signal S_F thereby largely cancels out the contribution of F and Y to the frontal stage, while increasing its contribution to the rear stage, thereby reducing rear-to-front crosstalk.

Thus the overall effect of a five- or 6-speaker ambisonic decoder supplied with the enhancement channels E and F is to allow effectively discrete front and rear stages to be achieved, with enhanced stability of front-centre sounds. By reducing k_e, k_f and k_b to near zero, the decoder automatically reverts to true B-format decoding, so that one can freely mix 3-channel B-format material in with effectively "discrete" stages.

In ref. [4], the use of these enhanced B-formats to provide a complete system hierarchy for transmission of HDTV surround sound is discussed, and it was there shown that a method of transmission compatible with existing "3:2" stereo systems was possible, which also supported ambisonic formats. In total, the 5-channel transmission hierarchy of ref. [4] supports 11 directional encoding modes in a fully mutually compatible fashion, including 4 "ambisonic" azimuthal encoding modes and 7 conventional "front stage" plus "rear stage" stereo modes. This hierarchy would not have been possible to formulate without the new 5- and 6-speaker ambisonic B-format decoders described in this paper, because the use of the supplementary E and F channels only gives sensible results when added to a basic 5- or 6-speaker B-format ambisonic decoder. The enhancement does not work correctly with previously-known 4-speaker decoders.

8. FURTHER DEVELOPMENTS

At the time of writing of this paper, a prototype B-format decoder of the kind described in this paper has been designed and built, using analog technology with digitally-controlled decoder coefficients, but no extended series of listening tests has yet been completed.

Experience over nearly two decades has given a high degree of confidence in the interpretation of the psychoacoustic localisation theory used in this paper, but fine details of design may still be influenced by listening trials.

The design methods of this paper can also be applied to directional encoding systems other than B-format, most notably to 2-channel surround sound encoding systems such as stereo and UHJ to enhance frontal image stability, although such decoders are expected to perform less well than those operating from B-format. At present, however, there is no detailed experience with such 2-channel decoders, and design work on such decoders has not yet begun.

However, it is expected that in due course, once the formidable computational design work has been done (and this will be aided by the availability of advanced symbolic manipulation software packages such as Mathematica or Maple to reduce the drudgery of the mathematics), such decoders may lead to significant enhancements in performance from the large libraries of UHJ [1] ambisonic recordings now in record company catalogues, thereby extending the life of older ambisonic formats.

There is still a need to provide quick algorithms to compute coefficients in the ambisonic decoding matrices as a function of speaker layout shape. Each decoder described in table 2 of this paper took about 5 hours of design work using computer aids, due to the need to do manual searches of singularities and optimisations, whereas a consumer decoder will need its"layout controls" to compute the optimum coefficients in not more than a few seconds. Fast layout control algorithms are currently under development, and we are confident of solving this problem, at least for 5 and 6- speaker B-format decoders, in the near future by refining methods we have already devised.

There is also a need to comprehensively survey possible 6-speaker decoder solutions. While the general solution algorithm has been found, the amount of work required to survey all reasonable 6-speaker layouts is large, and again, more efficient algorithms (and better mathematical software) is expected to expidite this task. Our initial survey of a few 6-speaker solutions has indicated that they offer a significant improvement in performance for TV applications as compared to 5-speaker decoders, giving significantly better frontal image stability.

Although these decoders are still a little way off from being realisable as consumer products, mainly for the reasons just described, we are already in a position to evaluate the performance of the new generation of decoders in HDTV applications, and expect to be able to move on to the consumer design phase in the forseeable future, certainly in time for the mass marketing of HDTV technology.

The future is also expected to see the detailed investigations of decoders using 7, 8 and 9 speakers for large auditorium and cinema use, and of periphonic (full sphere) decoders for with height reproduction with improved frontal image stability. The new decoding technology should not only improve domestic results in TV applications, but also allow better ambisonic results in large auditorium applications than the prior technology.

9. CONCLUSIONS

This paper has described a new generation of ambisonic decoders using 5 or 6 speakers, with the additional speakers being placed close to the front of the sound stage to provide improved frontal image stability. Working from the B-format 3-channel azimuthal directional encoding, such decoders provide significantly improved results in connection with TV sound, particularly in providing improved diretional matching of sounds with associated visual images, as comapred to older B-format decoding technology via 4 loudspeakers.

These new decoders depart from the "shelf filtering" architecture of older ambisonic decoders, since in general, no decoder of the old architecture exists which can be applied to the 5- and 6-speaker layouts appropriate to HDTV. The new architecture involves decoders whose reproduced pressure signal has a polar pattern to encoded azimuthal sounds that varies with frequency, unlike older decoders.

The new decoders not only involve a novel architecture, but also make use of the forward dominance transformation of B-format signals in order to restore what would otherwise be an imbalance in the gains of front and rear stage reproduced sounds. This leads to a slight narrowing of the reproduction of the frontal B-format sound stage, but in a manner that helps improve artistic compatibility with other reproduction modes for the same signal.

The use of "enhancement" channels added to B-format to improve front and rear stage stability even further has been described, based on the new decoders with supplementary signal feeds. Such enhancement channels also allow the development of a comprehensive hierarchical transmission system (described in ref. [4]) for HDTV sound which incorporates all the main known formats up to 3:2 stereo as well as ambisonic formats in a mutually compatible way.

The design method for the new decoders has been outlined, with all the fundamental theory given, but the detailed mathematical derivations have only been summarised, without details, because of their great complexity. However, the main design tradeoffs and procedures have been described, with a detailed tabulation of the likely subjective performance of ambisonic B-format decoders via a variety of speaker layouts.

The improved results of the new decoders, based on a novel suggestion by the second author (GJB),make ambisonics, with all its subjective advantages over non-psychoacoustic approaches to surround sound, an attractive option both for sound only and for use with future HDTV transmissions. In particular, the existance of a 3-channel surround-sound option greatly enhances the operational flexibility of HDTV sound systems, as well as offering a more satisfying sound quality.

Patent note

Some of the methods described in this paper are the subject of patent applications.

REFERENCES

[1] M.A. Gerzon, "Ambisonics in Multichannel Broadcasting and Video", J. Audio Eng. Soc., vol. 33 no. 11, pp. 859-871 (1985 Nov.)
[2] M.A. Gerzon, "General Metatheory of Auditory Localisation", Preprinted at the 92nd Audio Engineering Convention, Vienna (1992 Mar.) - republication of privately circulated 1977 paper.
[3] M.A. Gerzon, "Optimum Reproduction Matrices for Multispeaker Stereo", Preprint 3180, 91st Audio Engineering Society Convention, New York (1991 Oct.)
[4] M.A. Gerzon, "Hierarchical System of Surround Sound Transmission for HDTV", to be presented at the 92nd Audio Engineering Society Convention, Vienna (1992 March)
[5] M.A. Gerzon, "Hierarchical Transmission System for Multispeaker Stereo", Preprint 3199, 91st Audio Engineering Society Convention, New York (1991 Oct.)
[6] M. Gerzon, "Production Approaches to Multispeaker Stereo", Studio Sound (1992 April/May) to be published
[7] M.A. Gerzon, "Problems of Upward and Downward Compatibility in Multichannel Stereo Systems", to be published

[8] M.A. Gerzon, "Criteria for Evaluating Surround-Sound Systems", J. Audio Eng. Soc., vol. 25 no. 6, pp. 400-408 (1977 June)
[9] M.A. Gerzon & G.J. Barton, "Ambisonic Surround-Sound Mixing for Multitrack Studios", Conference paper C1009, 2nd AES International Conference "The Art and Technology of Recording", Anaheim, California (1984 May 11-14)
[10] R. Elen, "Ambisonic Mixing - An Introduction", Studio Sound, vol. 25 no. 9, pp. 40-2,44,46 (1983 Sept.)
[11] C.P. Daubney, "Surround Sound: An Operational Insight", IBA Tech. Rev. no. 14 pp. 52-56 (1981 June) reprinted in Studio Sound, vol. 24 no. 8, pp. 52-4, 56, 58 (1982 Aug.)
[12] M. Gerzon, "Ambisonics Part Two: Studio Techniques", Studio Sound, vol. 17 no. 8, pp. 24, 26, 28-30 (1975 Aug.)
[13] K. Farrar, "Soundfield Microphonone", Wireless World, vol. 85, pp. 48-50 (1979 Oct.) and pp. 99-103 (1979 Nov.)
[14] J.H. Smith,

[15] M.A. Gerzon, "The Design of Precisely Coincident Microphone Arrays for Stereo and Surround Sound", Proc. 50th Audio Engineering Society Convention, London (1975 March)
[16] I.M. Gelfand, R.A. Minlos & Z. Ya Shapiro, "Representations of the Rotation and Lorentz Groups and Their Applications", The Macmillan Company, New York, 1963
[17] M.A. Gerzon, "Surround Sound Psycoacoustics", Wireless World, vol. 80, pp. 483-486 (1974 Dec.)
[18] M.A. Gerzon, British Patents 1,494,751 , 1,494,752 , 1,550,627 , and 2.073,556. Corresponding U.S. Patents: 3,997,725 , 4,086,433 , 4,081,606 , and 4,414,430 .

TABLES

Low frequency decoder design 5-speakers, $\phi_F = 45^o$, $\phi_B = 50^o$

$k_F = 0.50527$, $C = 1.13949$

$C_F = 0.34190\ W + 0.23322\ X$, $M_F = 0.26813\ W + 0.38191\ X$, $S_F = 0.50527\ Y$

$M_B = 0.56092\ W - 0.49852\ X$, $S_B = 0.45666\ Y$

decoder performance:

$\theta=\theta_V=\theta_E$	r_V	dB	r_E	L_B	L_F	C_F	R_F	R_B
0	1.0000	2.551	0.7494	-0.1441	0.8082	0.6717	0.8082	-0.1441
15	1.0000	2.641	0.7396	0.0471	0.9748	0.6605	0.6049	-0.2872
45	1.0000	3.246	0.6807	0.5191	1.1553	0.5751	0.1448	-0.3943
60	1.0000	3.645	0.6481	0.7677	1.1570	0.5068	-0.0806	-0.3509
90	1,0000	4.386	0.6017	1.2067	0.9827	0.3419	-0.4464	-0.0849
135	1.0000	5.065	0.5815	1.5161	0.3915	0.1087	-0.6190	0.6028
180	1.0000	5.255	0.5832	1.2659	-0.2720	0.0121	-0.2720	1.2659

High frequency decoder design 5 speakers, $\phi_F = 45^o$, $\phi_B = 50^o$

$k_F = 0.54094$, $C = 0.93050$

$C_F = 0.38324\ W + 0.37228\ X$, $M_F = 0.44022\ W + 0.23386\ X$, $S_F = 0.54094\ Y$

$M_B = 0.78238\ W - 0.55322\ X$, $S_B = 0.42374\ Y$

decoder performance:

$\theta=\theta_V=\theta_E$	r_V	dB	r_E	L_B	L_F	C_F	R_F	R_B
0	0.8158	3.046	0.8273	0	0.7709	0.9097	0.7709	0
15	0.8115	3.175	0.8148	0.1818	0.9577	0.8918	0.5617	-0.1284
45	0.7806	4.029	0.7431	0.6529	1.2150	0.7555	0.1331	-0.1946
60	0.7576	4.585	0.7059	0.9102	1.2681	0.6465	-0.0569	-0.1278
90	0.7071	5.620	0.6550	1.3816	1.2052	0.3832	-0.3248	0.1831
135	0.6462	6.625	0.6306	1.7593	0.7473	0.0110	-0.3346	0.9919
180	0.6240	6.938	0.6294	1.5648	0.1095	-0.1432	0.1095	1.5648

Table 1 Example of 5-speaker Ambisonic decoder design according to the methods of section 5, for the speaker layout of figure 2 with $\phi_F = 45^o$ and $\phi_B = 50^o$, including values of psychoacoustic localisation parameters, overall energy gain in dB and speaker feed gains. High frequency front/back gain imbalance can be compensated by 3.893 dB forward dominance before decoding, and high frequency decoder can be matched in gain to low frequency decoder at azimuth $\theta_V = \theta_E = 45^o$ by a 0.784 dB gain reduction of the high frequency decoder.

Low frequencies $\phi_F = 35^o$, $\phi_B = 25^o$

$k_F = 0.73675$, $C = 0.90593$, forward dominance = 3.8636 dB, gain = 0 dB

High frequencies $\phi_F = 35^o$, $\phi_B = 25^o$

$k_F = 0.74762$, $C = 0.80803$, forward dominance = 3.8636 dB, gain = -0.4217 dB

psychoacoustic analysis

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.9009	3.820	0.8952	0.9120	3.868
15	12.03	1.0000	0.8319	4.130	0.8900	0.8512	4.162
45	36.69	1.0000	0.5424	5.705	0.8504	0.5912	5.699
60	49.61	1.0000	0.4399	6.379	0.8186	0.4988	6.388
90	77.36	1.0000	0.3447	6.837	0.7412	0.4190	6.983
135	125.29	1.0000	0.4405	4.820	0.6306	0.5192	5.699
180	180.00	1.0000	0.8384	1.586	0.5843	0.7567	3.868

Table 2a 5-speaker Ambisonic decoder design for $\phi_F = 35^o$, $\phi_B = 25^o$.

Low frequencies $\phi_F = 35^o$, $\phi_B = 35^o$

$k_F = 0.63701$, $C = 1.00206$, forward dominance = 4.1113 dB, gain = 0 dB

High frequencies $\phi_F = 35^o$, $\phi_B = 35^o$

$k_F = 0.65584$, $C = 0.83708$, forward dominance = 4.1113 dB, gain = -0.5188 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.8686	3.928	0.8853	0.8915	3.865
15	11.86	1.0000	0.8251	4.118	0.8806	0.8527	4.057
45	36.21	1.0000	0.6122	5.155	0.8443	0.6633	5.127
60	49.00	1.0000	0.5227	5.626	0.8147	0.5847	5.642
90	76.56	1.0000	0.4317	5.899	0.7418	0.5102	6.103
135	124.62	1.0000	0.5225	4.175	0.6345	0.5916	5.127
180	180.00	1.0000	0.8087	1.860	0.5886	0.7561	3.865

Table 2b 5-speaker Ambisonic decoder design for $\phi_F = 35^o$, $\phi_B = 35^o$.

Low frequencies $\phi_F = 35^o$, $\phi_B = 40^o$

k_F = 0.59675, C = 1.20337, forward dominance = 4.2769 dB, gain = 0 dB

High frequencies $\phi_F = 35^o$, $\phi_B = 40^o$

k_F = 0.62175, C = 0.86543, forward dominance = 4.2769 dB, gain = -0.5464 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.8471	4.153	0.8803	0.8812	3.949
15	11.75	1.0000	0.8150	4.283	0.8758	0.8511	4.098
45	35.89	1.0000	0.6431	5.017	0.8412	0.6946	4.956
60	48.58	1.0000	0.5619	5.358	0.8129	0.6244	5.384
90	76.03	1.0000	0.4681	5.519	0.7424	0.5523	5.773
135	124.17	1.0000	0.5185	4.060	0.6367	0.6136	4.956
180	180.00	1.0000	0.6908	2.339	0.5908	0.7368	3.949

Table 2c 5-speaker Ambisonic decoder design for $\phi_F = 35^o$, $\phi_B = 40^o$.

Low frequencies $\phi_F = 45^o$, $\phi_B = 35^o$

k_F = 0.58727, C = 0.91902, forward dominance = 3.1169 dB, gain = 0 dB

High frequencies $\phi_F = 45^o$, $\phi_B = 35^o$

k_F = 0.60353, C = 0.85140, forward dominance = 3.1169 dB, gain = -0.7244 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.8214	3.834	0.8284	0.8535	3.844
15	12.56	1.0000	0.7946	3.953	0.8250	0.8306	3.957
45	38.19	1.0000	0.6495	4.627	0.7991	0.7045	4.619
60	51.52	1,0000	0.5809	4.946	0.7780	0.6439	4.960
90	79.77	1.0000	0.5077	5.108	0.7260	0.5785	5.277
135	127.27	1.0000	0.5872	3.817	0.6495	0.6300	4.619
180	180.00	1.0000	0.7678	2.354	0.6168	0.7273	3.844

Table 2d 5-speaker Ambisonic decoder design for $\phi_F = 45^o$, $\phi_B = 35^o$

<u>Low frequencies</u> $\phi_F = \phi_B = \phi = 45^o$

k_F = 0.52936, C = 1.00589, forward dominance = 3.5692 dB, gain = 0 dB

<u>High frequencies</u> $\phi_F = \phi_B = \phi = 45^o$

k_F = 0.55707, C = 0.89149, forward dominance = 3.5692 dB, gain = -0.7857 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.7771	4.169	0.8194	0.8349	4.027
15	12.24	1.0000	0.7645	4.213	0.8165	0.8226	4.083
45	37.28	1.0000	0.6869	4.468	0.7937	0.7472	4.426
60	50.36	1.0000	0.6432	4.579	0.7749	0.7046	4.612
90	78.31	1.0000	0.5860	4.535	0.7273	0.6457	4.791
135	126.08	1.0000	0.6113	3.637	0.6543	0.6435	4.426
180	180.00	1.0000	0.6810	2.839	0.6219	0.6742	4.027

<u>Table 2e</u> 5-speaker Ambisonic decoder design for $\phi_F = \phi_B = \phi = 45^o$.

<u>Low frequencies</u> $\phi_F = 45^o$, $\phi_B = 50^o$

k_F = 0.50527, C = 1.13949, forward dominance = 3.8929 dB, gain = 0 dB

<u>High frequencies</u> $\phi_F = 45^o$, $\phi_B = 50^o$

k_F = 0.54094, C = 0.93050, forward dominance = 3.8929 dB, gain = -0.7838 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.7494	4.497	0.8158	0.8273	4.208
15	12.01	1.0000	0.7430	4.502	0.8131	0.8192	4.239
45	36.63	1.0000	0.6996	4.513	0.7918	0.7655	4.429
60	49.54	1.0000	0.6705	4.489	0.7740	0.7313	4.536
90	77.27	1.0000	0.6177	4.309	0.7285	0.6724	4.640
135	125.21	1.0000	0.5824	3.710	0.6567	0.6325	4.429
180	180.00	1.0000	0.5832	3.308	0.6240	0.6294	4.208

<u>Table 2f</u> 5-speaker Ambisonic decoder design for $\phi_F = 45^o$, $\phi_B = 50^o$.

<u>Low frequencies</u> $\phi_F = 55^o$, $\phi_B = 45^o$

$k_F = 0.50933$, $C = 0.92862$, forward dominance = 2.2870 dB, gain = 0 dB

<u>High frequencies</u> $\phi_F = 55^o$, $\phi_B = 45^o$

$k_F = 0.53280$, $C = 0.89997$, forward dominance = 2.2870 dB, gain = -1.0511 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.7185	4.036	0.7509	0.7882	3.961
15	13.17	1.0000	0.7146	4.044	0.7497	0.7834	3.976
45	39.91	1.0000	0.6889	4.087	0.7402	0.7507	4.071
60	53.69	1.0000	0.6724	4.095	0.7324	0.7286	4.125
90	82.48	1.0000	0.6463	4.021	0.7125	0.6881	4.178
135	129.42	1.0000	0.6412	3.675	0.6819	0.6564	4.071
180	180.00	1.0000	0.6516	3.435	0.6682	0.6515	3.961

<u>Table 2g</u> 5-speaker Ambisonic decoder design for $\phi_F = 55^o$, $\phi_B = 45^o$

<u>Low frequencies</u> $\phi_F = \phi_B = \phi = 55^o$

$k_F = 0.47329$, $C = 1.01391$, forward dominance = 3.0674 dB, gain = 0 dB

<u>High frequencies</u> $\phi_F = \phi_B = \phi = 55^o$

$k_F = 0.51350$, $C = 0.94696$, forward dominance = 3.0674 dB, gain = -1.0292 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.6613	4.702	0.7455	0.7770	4.399
15	12.59	1.0000	0.6658	4.649	0.7445	0.7775	4.374
45	38.29	1.0000	0.6954	4.281	0.7369	0.7763	4.208
60	51.64	1.0000	0.7112	4.035	0.7304	0.7684	4.109
90	79.94	1.0000	0.7080	3.675	0.7135	0.7227	4.007
135	127.40	1.0000	0.5943	3.841	0.6857	0.6040	4.208
180	180.00	1.0000	0.5225	4.129	0.6725	0.5439	4.399

<u>Table 2h</u> 5-speaker Ambisonic decoder design for $\phi_F = \phi_B = \phi = 55^o$

<u>Low frequencies</u> $\phi_F = 55^o$, $\phi_B = 60^o$

$k_F = 0.45808$, $C = 1.13695$, forward dominance = 3.6033 dB, gain = 0 dB

<u>High frequencies</u> $\phi_F = 55^o$, $\phi_B = 60^o$

$k_F = 0.50892$, $C = 0.99169$, forward dominance = 3.6033 dB, gain = -0.9719 dB

psychoacoustic analysis:

		low frequencies			high frequencies		
θ	$\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	0.00	1.0000	0.6259	5.227	0.7441	0.7742	4.732
15	12.21	1.0000	0.6337	5.142	0.7433	0.7766	4.688
45	37.21	1.0000	0.6890	4.531	0.7363	0.7872	4.392
60	50.27	1.0000	0.7231	4.116	0.7303	0.7843	4.212
90	78.20	1.0000	0.7300	3.550	0.7144	0.7295	4.024
135	125.99	1.0000	0.5327	4.093	0.6870	0.5581	4.392
180	180.00	1.0000	0.4251	4.694	0.6736	0.4745	4.732

<u>Table 2i</u> 5-speaker Ambisonic decoder design for $\phi_F = 55^o$, $\phi_B = 60^o$

<u>Low frequencies</u> $\phi = 45^o$, $\phi_C = 15^o$

$k = 0.08476$, $C' = 0.53306$, $k_C = 0$

<u>High frequencies</u> $\phi = 45^o$, $\phi_C = 15^o$

$k = 0.2$, $C' = 1.04567$, $k_C = 0$

<u>psychoacoustic analysis</u>

	low frequencies			high frequencies		
$\theta=\theta_V=\theta_E$	r_V	r_E	dB	r_V	r_E	dB
0	1.0000	0.8084	0.954	0.8540	0.8708	1.449
15	1.0000	0.7730	1.219	0.8431	0.8400	1.763
45	1.0000	0.6171	2.697	0.7687	0.7047	3.562
60	1.0000	0.5629	3.458	0.7180	0.6551	4.560
90	1.0000	0.5297	4.489	0.6194	0.6068	6.197
135	1.0000	0.6087	4.782	0.5187	0.6003	7.602
180	1.0000	0.6877	4.574	0.4860	0.6070	8.012

<u>Table 3</u> example of 6-speaker Ambisonic decoder design with $\phi = 45^o$, $\phi_C = 15^o$ and $k_C = 0$, without forward dominance or gain adjustments.

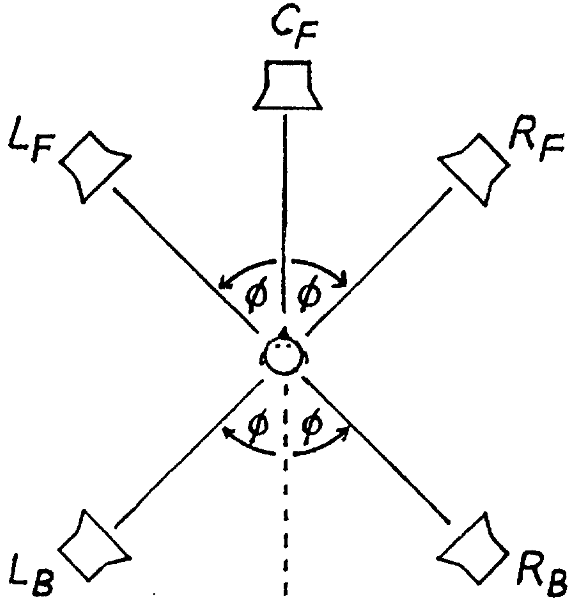


Figure 1. 5-speaker rectangle plus centre-front speaker layout for ambisonic decoding.

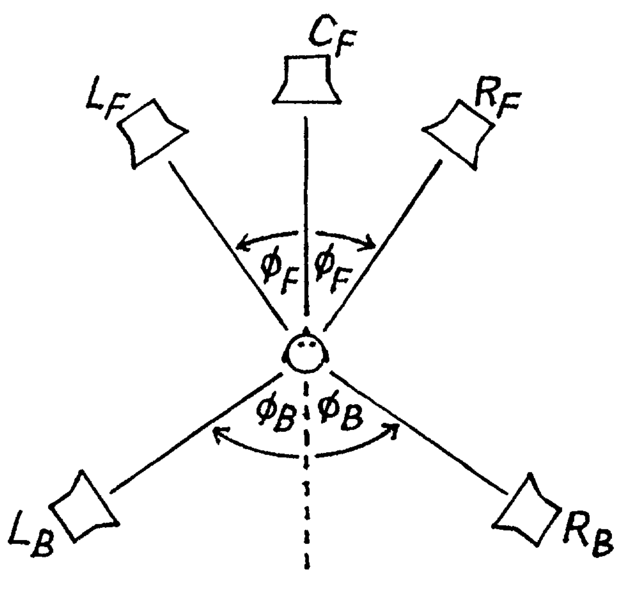


Figure 2. 5-speaker trapezium plus centre-front speaker layout for ambisonic decoding.

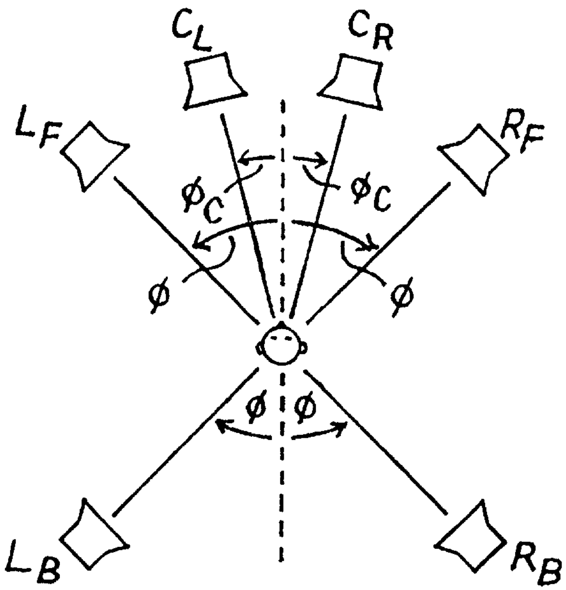


Figure 3. 6-speaker rectangle plus frontal centre pair layout for ambisonic decoding.

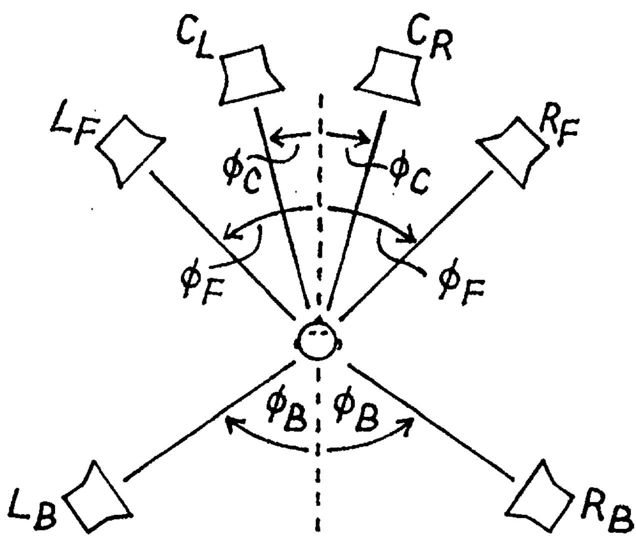


Figure 4. 6 speaker trapezium plus frontal centre pair layout for ambisonic decoding.

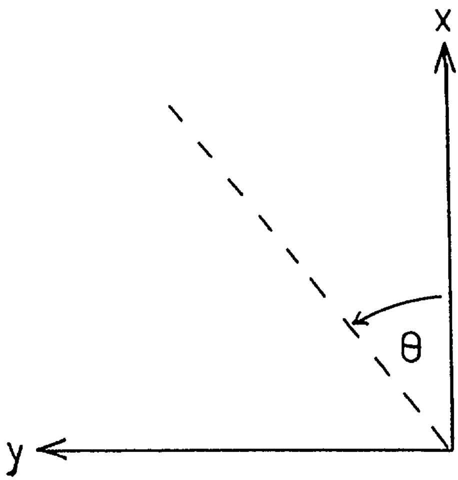


Figure 5. x- and y- rectangular coordinate conventions and azimuth angle convention used in this paper.

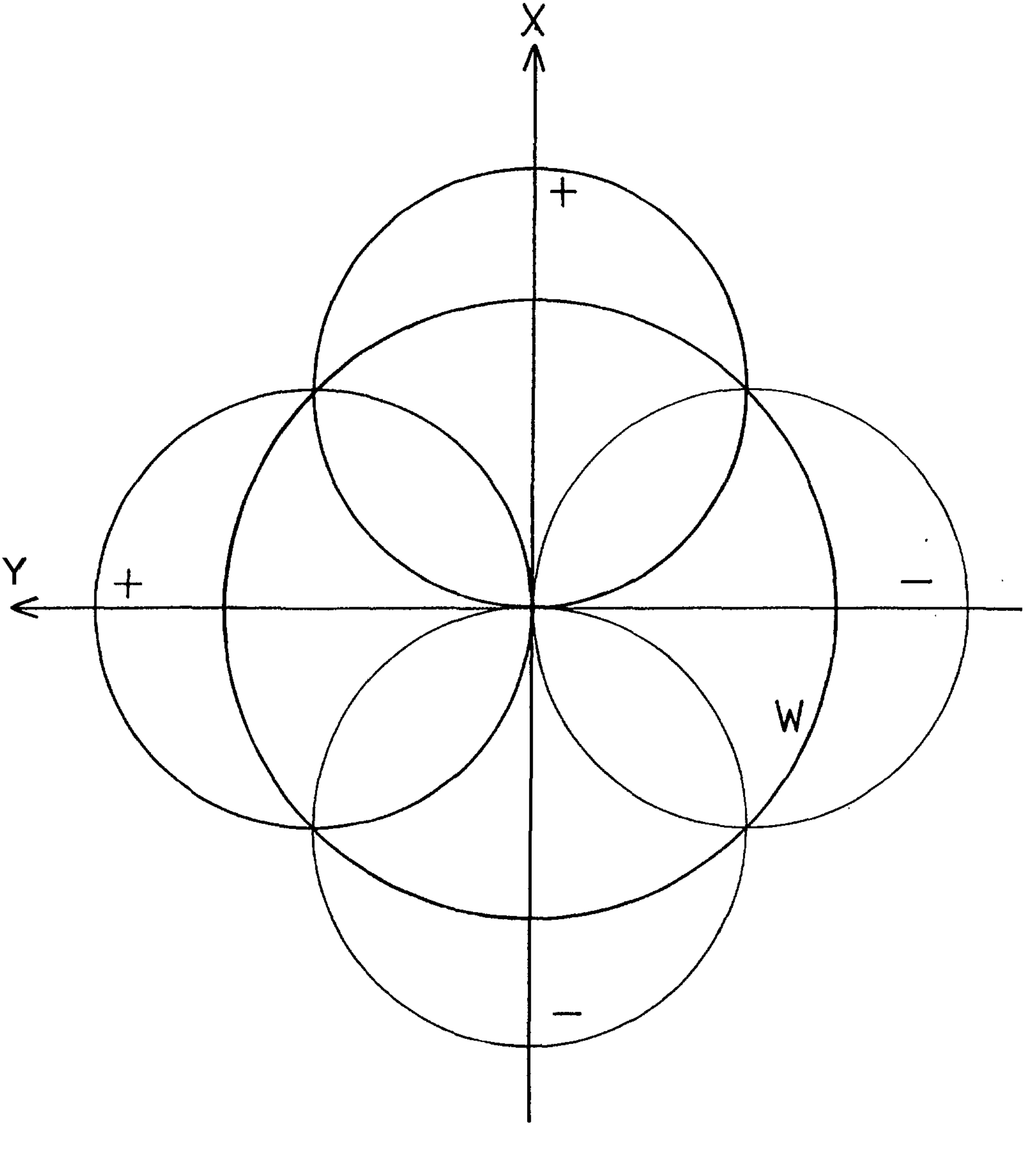


Figure 6. Directional patterns for horizontal B-format signals W, X and Y.

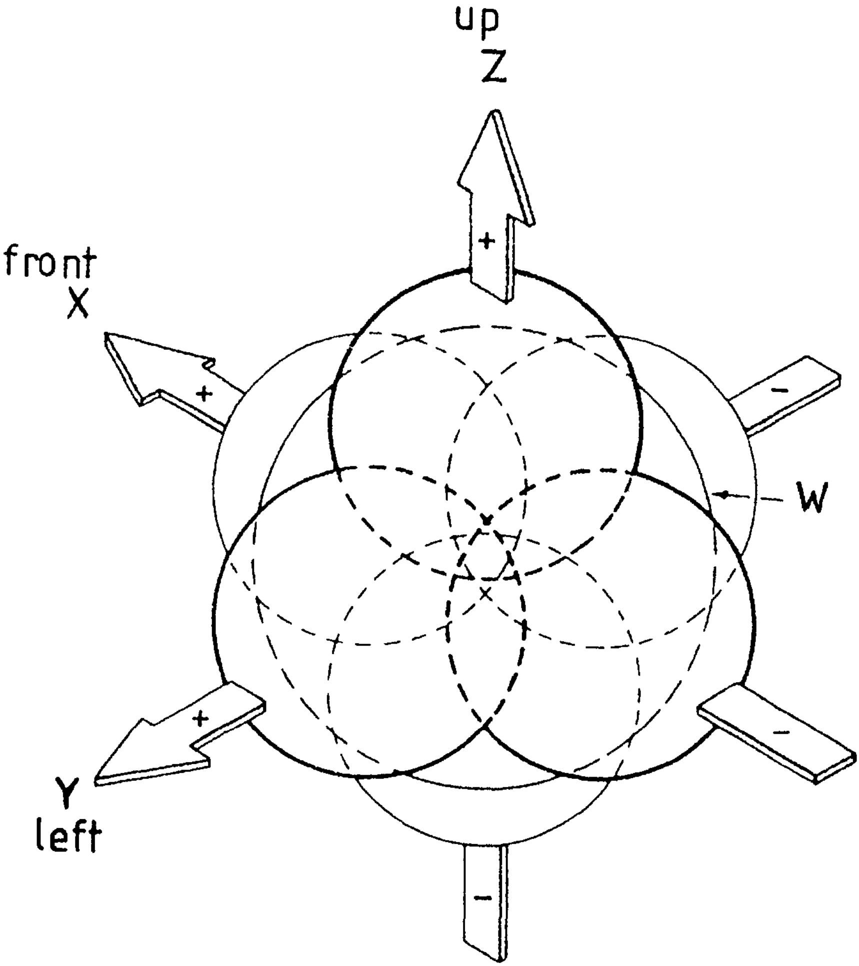


Figure 7. Directional polar patterns for full-sphere B-format signals W, X, Y, Z .

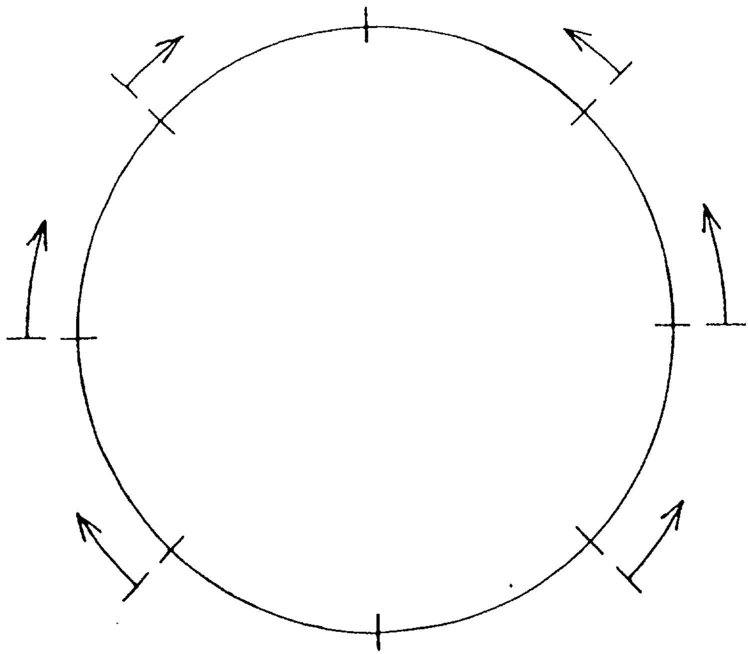


Figure 8. Alterations of reproduced azimuth caused by 6 dB forward dominance.

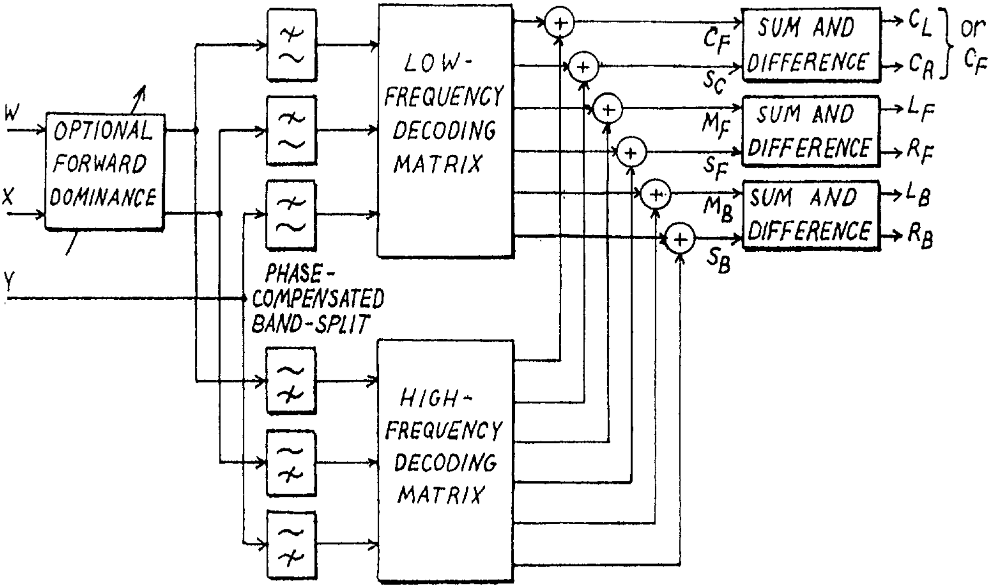
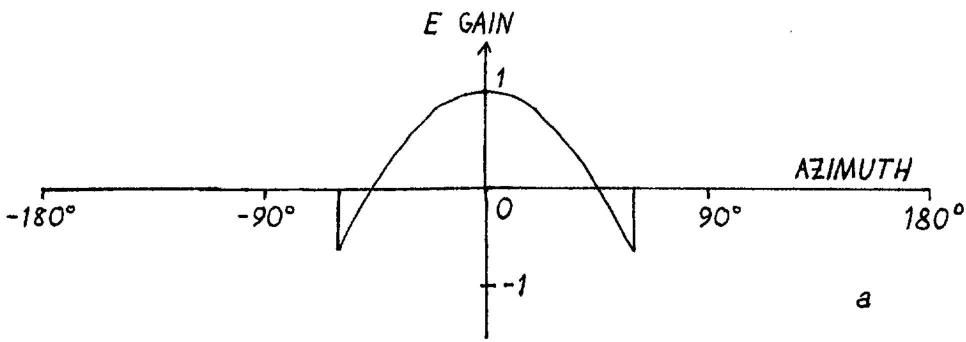


Figure 9 . Architecture of B-format ambisonic decoder for the speaker layouts of figs. 1 to 4.



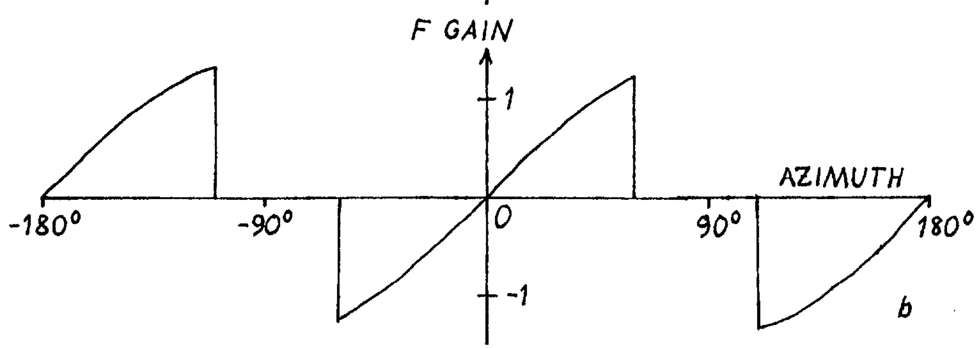


Figure 10. Gain as a function of direction azimuth θ for the signals E (fig.10a) and F (fig.10b) for $\theta_S = 60^o$, $\theta_B = 70^o$ and $k_e = k_f = k_b = 1$.

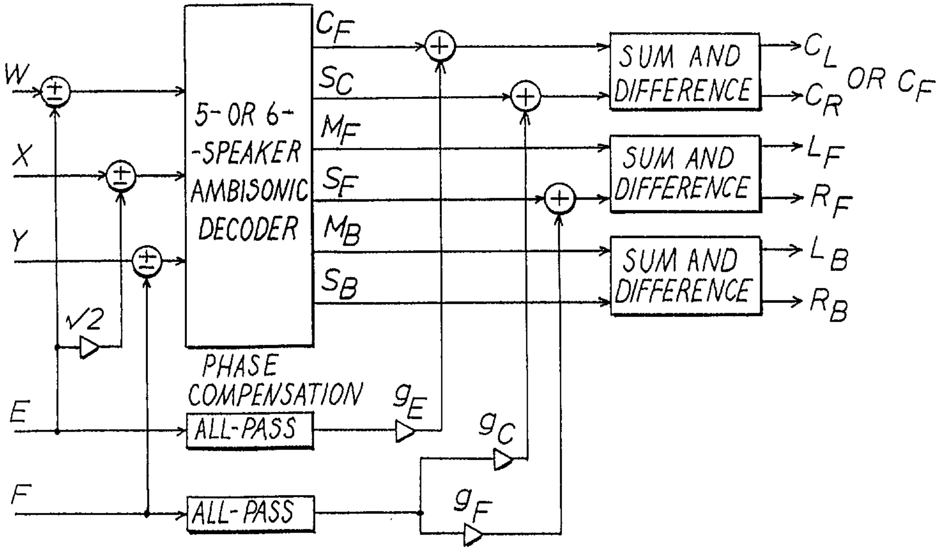


Figure 11. Decoder for enhanced Ambisonic B-format signals, showing input "cancellation" followed by direct feeds to the outputs. Simplified schematic.