

# Variomatrix adaptor for System 45J and Matrix H

## Phase shift circuit allows Variomatrix to decode Matrix H and System 45J

by Michael A. Gerzon, M.A., *Mathematical Institute, Oxford.*

Many hi-fi enthusiasts have Sansui Variomatrix decoders, and the present article describes an adaptor suitable for converting the Variomatrix for decoding signals encoded via the NRDC System 45J or BBC Matrix H systems. While such a decoder cannot by psycho-acoustically optimal, it does permit listening owners to extend the usefulness of their equipment.

The adaptor essentially does the job of converting the 45J or Matrix H signals into a form which the Variomatrix is designed to handle, i.e. into signals which are good approximations of Regular Matrix signals. The optimum method of conversion is slightly different for these two systems, but fortunately involves in both cases the use of a  $58^\circ$  phase-shift network, so that the circuit is kept fairly simple despite its two-fold function.

Essentially, the Matrix H adaptor consists of a  $58^\circ$  phase lead put into the right-channel signal relative to the left channel. The System 45J adaptor adds to this a  $-15\text{dB}$  blend circuit at the outputs of the phase shifters. The six pole phase shifter described gives a  $58^\circ$

shift with  $\pm 4^\circ$  error over the frequency range 44Hz to 17kHz if precision components are used, and so is suitable for use even with a studio-quality Variomatrix. In practice for domestic applications, 5% tolerance components may be used, although the use of 2% resistors will give better results.

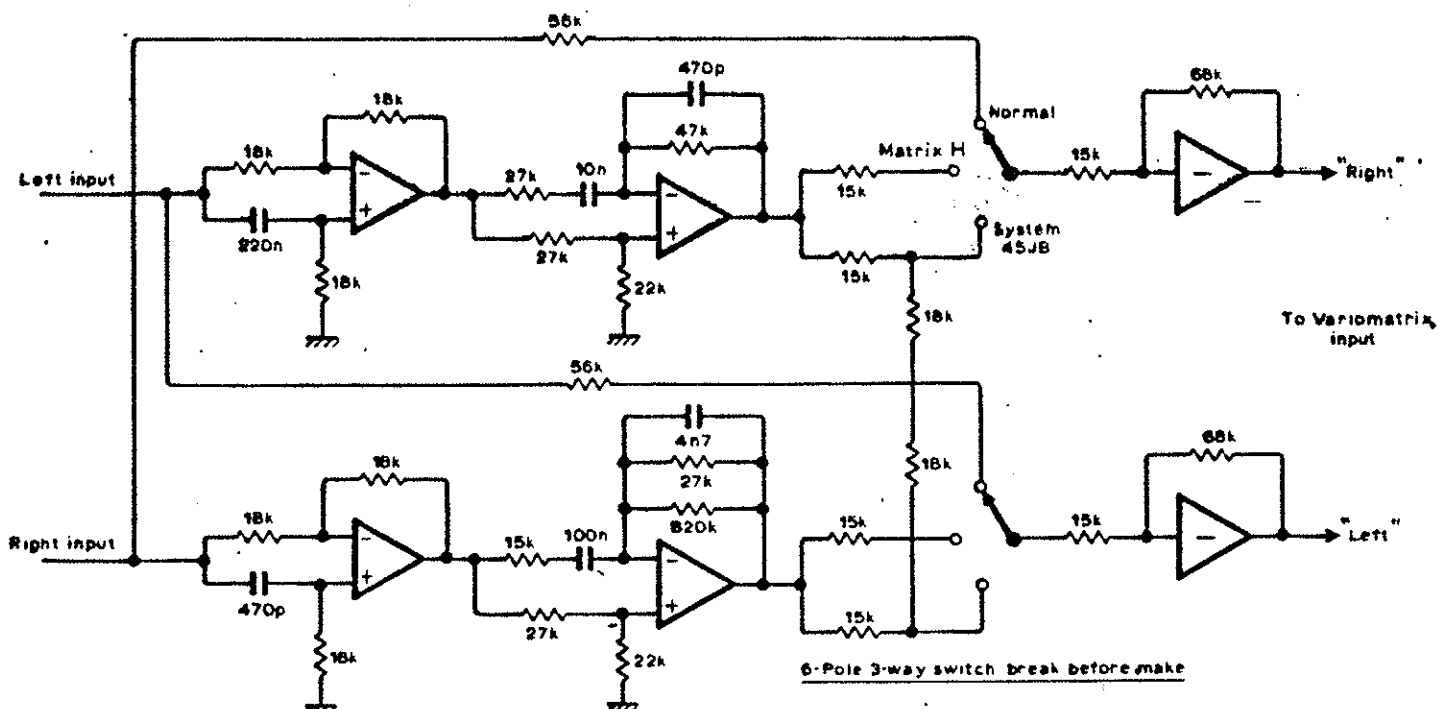
The input circuit of the adaptor is shown in Fig. 1. Depending on the quality desired, the operational amplifiers may be 741 types or special audio types. The circuit is designed to offer a fairly high and resistive input impedance (18k or 14k depending on switch position), and gives approximately unity overall gain in all modes. The mode switch offers three positions: normal (i.e. conventional use for stereo and Regular Matrix), Matrix H, and System 45J.

An odd feature of the way the adaptor is connected is that (except in normal

mode) all left input signals are fed to the inputs labelled right on the Variomatrix, and vice-versa as shown in Fig. 1. Similarly, all outputs labelled left on the Variomatrix are connected to the corresponding right quadraphonic inputs on the preamplifier, and vice-versa. The reason for the switching shown is to ensure that the left/right interchanging of the Variomatrix inputs and outputs does not occur in the normal switch position, and for this reason, the mode switch is six-pole three-way. Also shown in Fig. 2 is a  $+2\text{dB}$  gain for the back channel outputs in the System 45J mode only; such a  $+2\text{dB}$  gain is necessary for best results. However, constructors may omit these gains from the circuit provided that the front/rear balance control of their system is adjusted to give this  $+2\text{dB}$  rear gain when decoding System 45J.

Owners of Sansui equipment in which the Variomatrix is integrated with the preamplifier and amplifier may not always find it convenient to use the output switching circuitry of Fig. 2, since this would involve breaking into the equipment. For such users, we

Fig. 1. Circuit of pre-Variomatrix adaptor. For best results, components should be 2% tolerance, though 5% should be acceptable.



**Fig 2. Post-Variomatrix circuit,** includes rear channel switched gain compensation. Resisting 5 or 10% tolerance.

suggest that they use the circuit of Fig. 1, for example in the tape monitoring circuit, but with the following modifications.

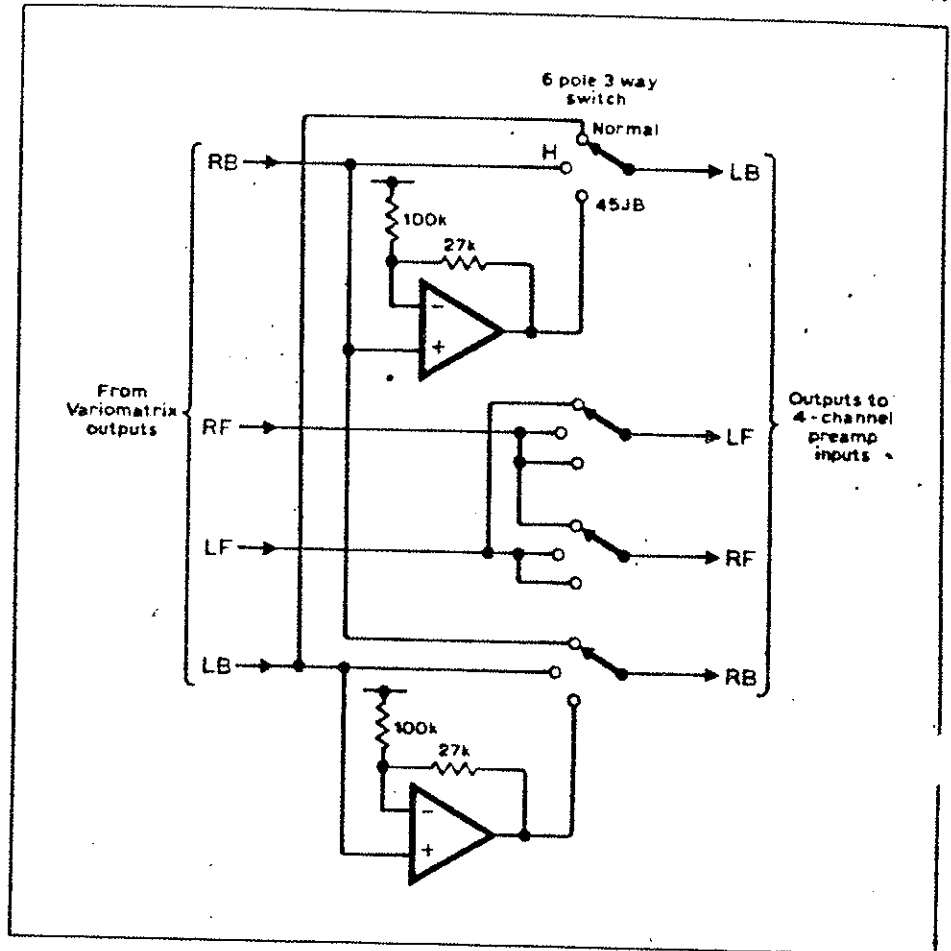
Connect the top output of Fig. 1 to the left Variomatrix input (and not the right), and the bottom output of Fig. 1 to the right Variomatrix input (and not the left), and

Feed the two 56kΩ resistors connected to the normal switch position from the left input for the top switch of Fig. 1, and the right input for the bottom switch of Fig. 1.

When used in this way, no left/right interchanging is used, and the switch need only be two-pole three-way. This method of use does not handle "interior encoded" sounds quite so well, but still generally works. For best results with System 45J with this simplified method of use, the front/rear balance control should be set to give +2dB gain to the rear speakers.

The Matrix H switch position will decode existing BMX discs (e.g. the UD-4 discs of Nippon Columbia) with reasonably accurate results, so that in practice the circuit allows decoding of Regular Matrix, Matrix H, System 45J and BMX.

As the author is connected with the NRDC Ambisonic project, in order to avoid possible misunderstandings it is pointed out that the use of a Variomatrix with the adaptor described will not give proper NRDC Ambisonic decoding



with optimal psychoacoustic results, but is merely a means of enabling Variomatrix owners to use their existing equipment with some of the newer systems.

Also, the method of using the Variomatrix described is solely the author's responsibility, and neither Sansui Elec-

tric Company Ltd nor the BBC would necessarily regard such use as being according to their own recommendations.

The BBC have applied for a patent (34839/74) on the use of a Variomatrix decoder with a prior phase shifting circuit of about 60°. — Ed.

## Surround sound decoders — 6

**Multi-system ambisonic decoder****1—Basic design philosophy**by Michael A. Gerzon, M.A., *Mathematical Institute, Oxford*

This series of articles describes a decoder capable of decoding all major existing and proposed two-channel surround-sound systems, including the Ambisonic System 45J, SQ, Regular Matrix, BMX and BBC Matrix H. For systems other than SQ, the decoder gives full psychoacoustically optimized results using NRDC Ambisonic decoding technology. In addition, Ambisonic playback of mono, stereo and of three-channel studio-format signals is provided. The decoder is suitable for three-amplifier/four-speaker, four-amplifier/four-speaker, and four-amplifier/six-speaker reproduction.

The NRDC Ambisonic project has developed a comprehensive technology for creating, encoding and decoding sound<sup>1-4</sup>. While this ambisonic technology can give of its best only with optimized programme material and encoding (i.e. the System 45J described in reference 1), the methods can be applied to getting improved results from nearly all existing surround-sound systems. Later articles in this series will give the detailed circuit and some constructional details for a decoder for all the above-listed two-channel systems. As this decoder does a great deal more than any previously-published decoder designs, it is necessary to describe its many facilities in some detail, as well as giving some idea of what the circuitry is intended to do.

The aim of any surround-sound decoder is to provide the listener with an illusion of sounds coming from all (horizontal) directions around him. Moreover, if the decoder is well designed, the directions should be those intended by the recording engineer, and should be heard by a listener through the usable listening area. Conventional "quadraphonic" decoder designs give very poor images for sounds in inter-loudspeaker directions, especially at the sides, resulting in a rather gimmicky 'ping-pung-pang-pong' effect at the four corners.

The full theory used to design ambisonic decoders is mathematical, and computing facilities are required to

carry out the extremely complex design calculations involved. It is clearly not possible to give full details here of why the various parts of the decoders have the exact values that they do, but some idea of what is going on can be given.

Two previous articles in *Wireless World*<sup>1,5</sup> have described some, but not all, of the psychoacoustics lying behind ambisonic decoder design. Essentially, the ears use different methods of localizing sounds, not just one or two. The more of these hearing mechanisms that are satisfied the better the sound and accuracy of the result. In particular, if a decoder satisfies several different methods of hearing, the brain has to do far less work to unscramble the complex sound reaching the ears, and the result is particularly "relaxed" listening, with little listener fatigue. While the mathematics of the design is aimed at getting accurate sound localization for all directions, it is undoubtedly true that the biggest musical benefit comes from this consequent low listening fatigue, rather than from any ability to "shoot the pianist".

Some aspects of sound that ambisonic decoders are designed to optimize are now described. At low frequencies, below 500 or 700Hz, there are three important aspects of sound localization: the "Makita" direction of a sound (the direction one turns to face the apparent sound direction), the "velocity magnitude" (the degree to which the sound stays in its correct localization as one turns to other directions), and the "phasiness" (the degree to which unwanted components of sound not in phase with the desired sound are heard). It turns out that for all systems other than SQ, it is possible to design a decoder matrix below 700Hz to get the Makita localization correct for all encoded sound directions. In addition, a careful adjustment of the gain of the various signal components in the decoder permits the velocity magnitude to be made correct also. Thus, at low frequencies, a listener will hear all directions correctly reproduced in direction, no matter which way one faces.

Phasiness is more of a problem with two-channel systems, as it is not

possible to design decoders that get rid of it altogether. The effect of phasiness is not only to blur the sound image, but also to create an unpleasant sensation often described as "pressure on the ears" that actually makes some people feel sick, although others don't seem to notice it much. Studies by the BBC<sup>6</sup> and NHK<sup>7</sup> have given a good indication of how much phasiness can be tolerated. In addition, it is found in practice that phasiness is more acceptable for sounds behind the listener than for sounds in front. Two-channel ambisonic decoders are therefore designed for very low phasiness in the front sector of sounds, while giving rather higher phasiness in the rear.

There is another reason why phase shifts cause poor directional reproduction that comes into action below 300Hz. In a real-world listening room, the loudspeakers are at a finite distance from the listener (often about 2.5 metres for British listening rooms), which means that the sound wave from each loudspeaker arrives as a curved wave-front at the listener, rather than as a plane wave. This curvature can be shown to cause the "phasiness" components of the reproduced directional sound to be converted into rotations of sound images around the listener at low frequencies. However, it is possible to remove these low-frequency errors by means of two high-pass filters in the "velocity signal" paths in the decoder. These speaker-distance compensation filters are RC types with -3dB points at about 20Hz for 2.7m speaker-to-listener distance, this may seem too low to worry about, but listening tests here confirmed the design theory and show that image displacements of as much as 15 to 30° can occur for instruments such as double basses when situated behind the listener unless distance compensation is used. Distance compensation does not turn a bad decoder into a good one, but it does give a "tighter" and more well-defined sound to an already good decoder design.

The ears use different methods of locating sounds at higher frequencies say from 700Hz to 5kHz. However, a rather magical result emerges from the

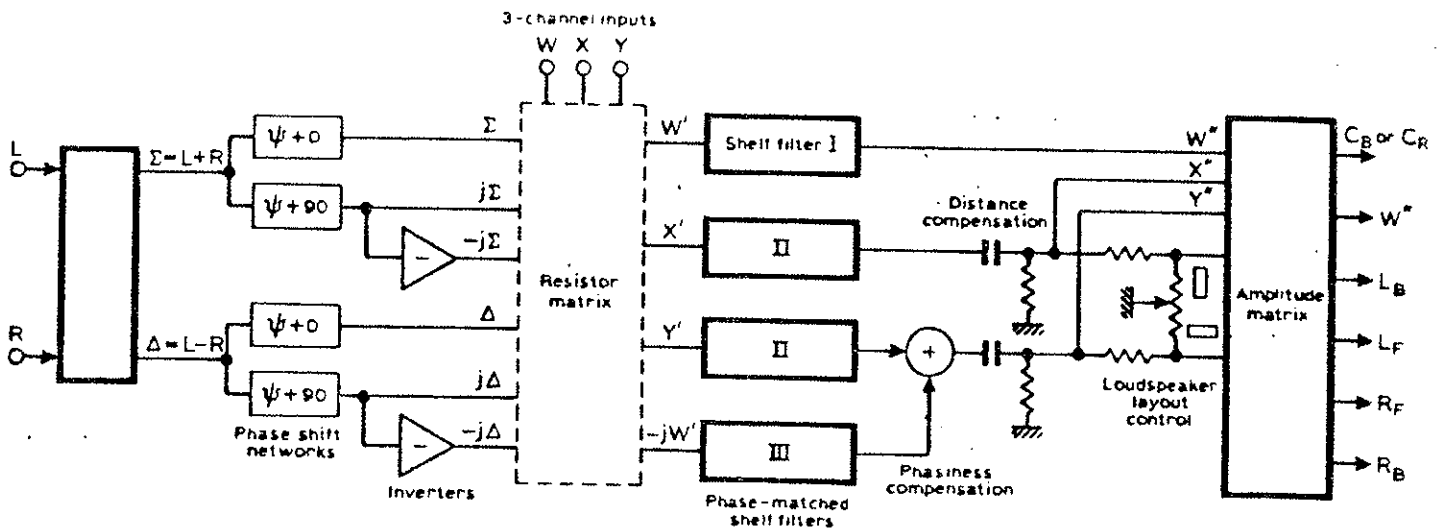


Fig. 1. Block diagram of multi-system ambisonic decoder, switching arrangements omitted. Shelf filters, inoperative for SQ decoding, depend on system being decoded, as does the resistor matrix. Also not shown is switching for  $C_B$  or  $C_R$  output.

design theory that states that, in effect, the basic sound localization of a decoder will automatically be the same at low and higher frequencies provided that the loudspeaker outputs of the decoder are derived via a particular type of amplitude matrix, the matrix involved depending only on the loudspeaker layout being used (by the listener).

Besides getting the basic high frequency localization correct, it is necessary to minimize phasiness in this frequency range also, and to ensure that the sound image does not move around as the listener faces other directions. To get this last requirement right, it turns out that the best decoder design at higher frequencies involves different relative signal gains from those apt at low frequencies, so that the decoder has to be made frequency-dependent. The

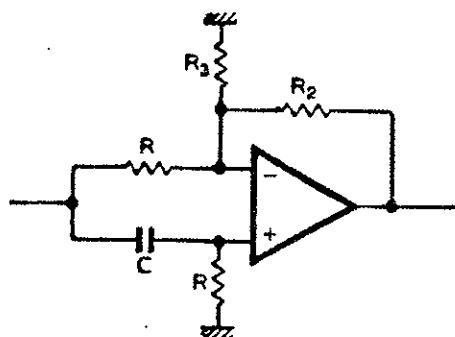


Fig. 2. Phase-compensated shelf-filter circuits allow frequency-dependent decoding. Conventional RC shelf filters would cause unwanted phase differences between signal paths.

effect of minimizing the image movement as the listener rotates his head is to avoid an unpleasant "in-the-head" sensation often, but incorrectly, described as "closeness" of sound by other authors<sup>9</sup>.

There are numerous other detailed aspects of decoder design, particularly those involving the way reverberation is reproduced (where a careful choice of encoding system such as 45J<sup>1</sup> can help), the effect heard by listeners seated away from the centre of the listening area, and the tone quality of the sound. It may seem strange that absolutely flat frequency response reproduction can sound coloured when reproduced through several speakers, the coloration depending on the precise speaker feeds used in the decoder. Many simple "matrix quadraphonic" decoders suffer from a tubby bass or harsh treble due to these effects. However, it is possible to account for most of these effects by the psychoacoustic design theory, and to minimize them in the decoder design. In practice, sounds encoded at the back are allowed to sound a little more colored than frontal sounds in ambisonic designs, because the ears appear to be more tolerant of marginal faults at the back, although one should assume that they are infinitely tolerant!

All decoder designs for two-channel encoding systems are a compromise between conflicting factors, and no design can achieve perfect performance in all ways. In this respect, the design of decoders is akin to loudspeaker design in being in the final analysis an art based on experience and listening. However, the science (i.e. the comprehensive psychoacoustic theory) is a very necessary part of reducing the almost infinitely complex design problem to a point where the designer can be sure of achieving his particular compromise as well as possible. The compromises inherent in these designs are based on the requirements:

- for front-stage material, the surround-sound should be subjectively superior to stereo for musical listening

(few existing designs meet this minimal requirement!),

- good results for listeners facing non-frontal directions and in non-central listening positions, especially behind-centre, and

- "musicality" of effect on both "ambient" and "surround" programme material, leading to low listening fatigue.

To some extent these requirements conflict with those based solely on the localization of direct sounds, such as in the experimental results quoted in ref. 10, where image sharpness for a forward-facing central listener has been achieved in a simple matrix decoder at the expense of "in-the-head" sound and severe image mislocation for non-forward-facing and non-central listeners. Good single-sound localization for most directions and listening positions can be achieved using a signal-actuated variable matrix decoder, but such decoders give a high level of listening fatigue on music due to the constant variation of signal parameters. Such decoders may be useful for surround-drama, where accuracy of localization becomes more important than "musicality" or low listening fatigue, and a fully-fledged ambisonic 'variable matrix' design is under development for such specialist applications. However, there is no doubt that a non-variable decoder is going to remain the preferred method of serious listening to music despite its superficially less "impressive" performance.

The basic diagram of the ambisonic decoder to be described in detail in later parts is shown in Fig. 1. Left and right signal inputs are fed to a sum-and-difference matrix to derive the sum  $\Sigma = L + R$  and difference  $\Delta = L - R$ , because this leads to simplification of the later parts of the decoder, as well as to a slightly greater tolerance to small component errors. These two signals are each fed to  $0^\circ$  and  $90^\circ$  phase shift networks, and the  $90^\circ$ -shifted signal is also phase inverted to yield a  $-90^\circ$  phase shifted signal in each case. The

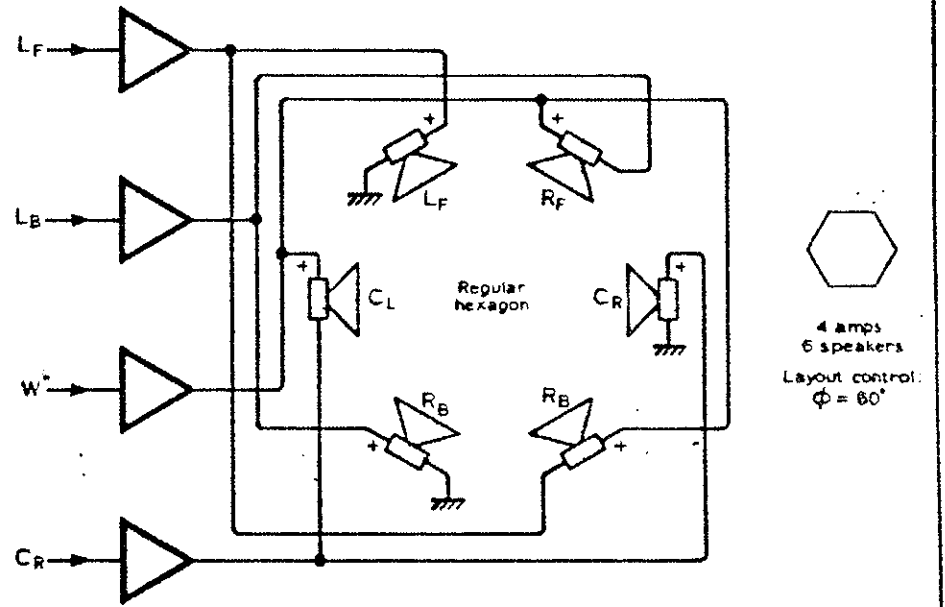
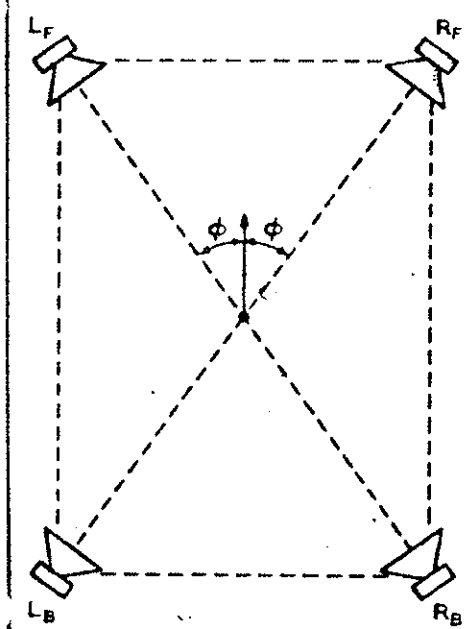
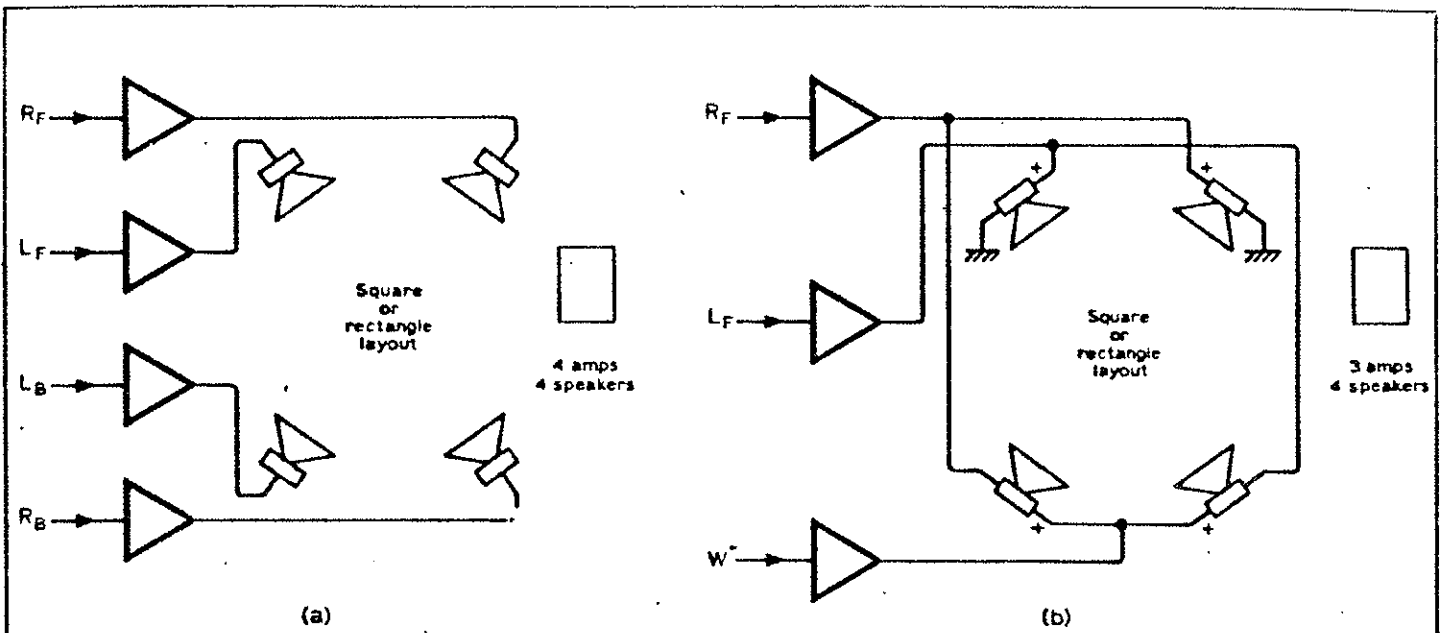
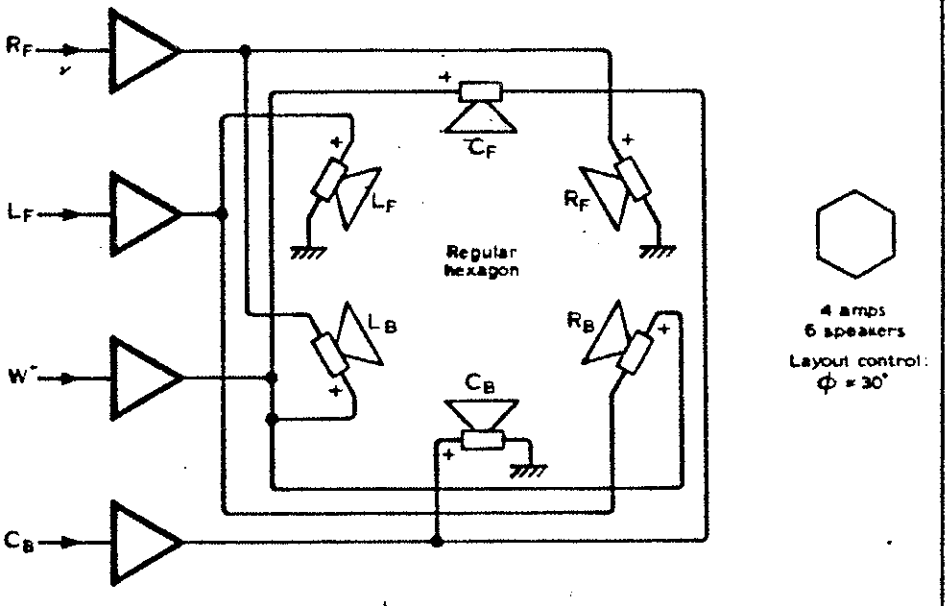


Fig. 3. Methods of feeding four loudspeakers using four amplifiers (a), and three amplifiers (b). Speaker terminals marked + are the positive-phase terminals in each case.

Fig. 4. Compensation is provided for non-square layouts. Angle  $\phi$  is set on a layout control.

Fig. 5. Better results can be obtained from four-amplifier, six-speaker regular hexagon decoding. Connections are shown for two hexagon layouts with the angle  $\phi$  used in the equations of the output matrix.



phase shifters used are high precision types, as the ears are capable of hearing very small errors in localization (as little as  $2^\circ$  in real life). Previous "quadrophonic" decoders have not required such high precision mainly because they gave in any case a poor decoded effect due to sub-optimal design.

The six signals are then fed to a resistor matrix, which derives the required combination of these signals to produce the correct pressure and velocity signals  $W'$ ,  $X'$ ,  $Y'$ , for the particular encoding system in use. (For a discussion of this aspect see ref. 1, in particular in connection with its Fig. 2). The resistor matrix used is different for different encoding systems, so that switching is provided for different matrix resistance values. The resistor matrix, which involves no active circuitry, also includes a switched three-channel input option suitable for use with three-channel ambisonic master tapes. In a later article we hope to describe live ambisonic recording for the keen tape enthusiast. These three channel inputs only cost a few resistors and input sockets in the present decoder, and so come virtually for free; in addition, they provide useful test signal inputs for setting-up purposes. We shall give the resistor matrix formulas for the various encoding systems for the signals  $W'$ ,  $X'$ ,  $Y'$  in part 2 of this series. An output  $-jW'$  is provided for phasiness-control in some systems, as described in references 1 and 3.

The frequency-dependent aspects of the decoder are provided by the shelf filters which give one decoding matrix at low frequencies and a second at high frequencies, with the transition centred at 400Hz. Were conventional RC shelf filters to be used, there would be phase shifts between the various signal paths, which would cause quite bad localization errors. For this reason, the shelf filters are designed to give phase shifts identical to one another by making them "all-pass" types. The basic circuit of the phase-compensated shelf filters is shown in Fig. 2. The particular arrangement shown has an input impedance of  $R$  at all frequencies, which means that it is seen by the resistor matrix as a resistive load, suitable for terminating a matrix circuit. The value of  $R_f/R$  controls the ratio of high-to-low-frequency gain of the shelf, and  $R_1$  provides extra h.f. gain to make up the losses of the preceding resistor matrix.

Thus the shelf filters are made to do five different jobs: terminate the matrix circuit, provide gain, give a different matrix circuit at low and high frequencies, give matched phases over the transition frequency band, and give an overall flat frequency response to the decoder at all frequencies.

An additional complication arises because different methods of encoding require different shelf filters in the decoder. In practice, the shelf filters required for BMX, RM, 45J and BBC H

differ only slightly, so that a compromise choice has been made to do all these systems. Decoding mono, stereo and three-channel studio format requires, for best results, a different set of shelf filters, and SQ requires that no shelf filters be used. (SQ decoders cannot be designed to give full ambisonic results; there is a mathematical theorem to this effect. The decoder for SQ provided is, however, less phasey in quality than the SQ designs on the market, and was designed specifically for incorporation into this design. It is not in accordance with CBS Laboratories' SQ specification, but in the author's opinion, it is better than decoders that are.)

The switching of the shelf filters involves equipping the op-amps of Fig. 2 with several filter circuits, which are switched in and out as required.

The outputs of shelf filters acting on  $Y'$  and  $-jW'$  (see Fig. 1) are added to reduce front-stage phasiness, and the velocity signals are then subjected to the RC high-pass distance compensation. This gives us three signals  $W''$ ,  $X''$  and  $Y''$  representing respectively the signal pressure, forward component of acoustic velocity, and leftward component of acoustic velocity heard by the listener. These are fed to an output amplitude matrix, which includes a layout control adjustment to match different shapes of rectangular loudspeaker layout in the room.

The decoder provides six different outputs  $L_B$  (left back),  $L_F$  (left front),  $R_F$  (right front),  $R_B$  (right back),  $W''$  (pressure) and either  $C_B$  (due back) or  $C_R$  (due right), switched. The way these six outputs can be used is itself an interesting story, for they can be used to provide decoding via four loudspeakers in a wide range of rectangle shapes using either four amplifiers, Fig. 3(a), or, remarkably, using just three power amplifiers as shown in Fig. 3(b). The three-amplifier set-up in no way means that there is any compromise in the psychoacoustics of the decoded signal, as precisely the same speaker signals are produced as in Fig. 3(a).

To see this, we first remark that the outputs of the decoder are given by the formulae

$$\begin{aligned} L_B &= \frac{1}{2}(W'' - \sqrt{2} \sin \phi X'' + \sqrt{2} \cos \phi Y'') \\ L_F &= \frac{1}{2}(W'' + \sqrt{2} \sin \phi X'' + \sqrt{2} \cos \phi Y'') \\ R_F &= \frac{1}{2}(W'' + \sqrt{2} \sin \phi X'' - \sqrt{2} \cos \phi Y'') \\ R_B &= \frac{1}{2}(W'' - \sqrt{2} \sin \phi X'' - \sqrt{2} \cos \phi Y'') \end{aligned}$$

where  $\phi$  depends on the setting of the layout control, being  $45^\circ$  for a square layout, and being equal to the angle  $\phi$  shown in Fig. 4 for a rectangle layout. From these formulae

$$L_B + R_F = L_F + R_B = W''$$

so that

$$\begin{aligned} L_B &= W'' - R_F \\ R_B &= W'' - L_F \end{aligned}$$

and it will be seen that the rear speakers of Fig. 3(b) indeed are connected so that the potentials of their "positive phase" terminals relative to their negative phase terminals are  $W'' - R_F$  and  $W'' - L_F$

respectively.

Even more remarkable however, are the four-amplifier six-loudspeaker arrangements possible with this decoder. It has been known for several years that decoders using six loudspeakers are capable of better results than is possible using four, no matter how well-designed the decoder may be. If properly used, the extra speakers give more solid image location over a larger area, with less tendency for the image to hug the loudspeakers than when using four, particularly on difficult waveforms such as audience applause. It has not been possible to market six-speaker equipment; few homes could properly accommodate it, and the market for such special equipment was thus too small to justify manufacture. However, the ambisonic decoding method permits the same decoder and the same four amplifiers to be used for six speakers for the few who can manage it, making this improved form of decoding domestically available for the first time. We emphasize that in no way does the use of four amplifiers imply substandard results: exactly the same speaker signals are given as one would design a psychoacoustically optimized six-amplifier decoder to give.

The six-speaker connections for two shapes of regular hexagon layout are shown in Fig. 5. The three speakers that are fed in a "simple" manner in each case form an equilateral triangle of speakers; this helps minimize the subjective effects of slight mismatches of amplifier gain. The signals  $C_B$  and  $C_R$  are

$$\begin{aligned} C_B &= \frac{1}{2}(W'' - \sqrt{2} X'') \\ C_R &= \frac{1}{2}(W'' - \sqrt{2} Y'') \end{aligned}$$

Although detailed instructions for calibrating and using the decoder will be given at the end of this series of articles, it is worth emphasizing now that all amplifiers and loudspeakers

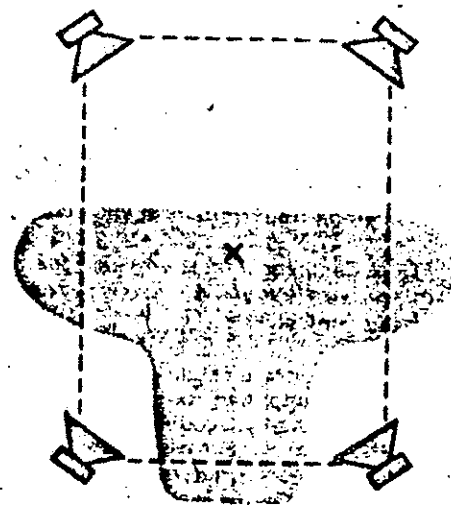


Fig. 6. Approximate listening area for ambisonic decoding (shaded) with a rectangle speaker layout obtained for BMX, 45J, Matrix H and RM systems as well as stereo. Optimal listening is at the centre (X).

must be accurately matched for correct ambisonic results. Unlike "quadraphonic" decoders, both front and rear loudspeakers co-operate to produce sounds in any direction. Thus, for example, the rear speakers provide outputs that help to reinforce the localization of sounds that are reproduced in front of the listener. Thus one cannot try turning the rear speakers up or down in the mistaken idea that the front and rear are independent of one another. When the outputs are not matched, the sound field tends to "fall apart"; in fact turning down the rear speakers often makes them much more audible (as distracting noises at the back) than in a correct balance.

While it is not absolutely necessary to have all power amplifiers of the same make, they should be adjusted for identical gains and phases, and one should check (e.g. by using an X-Y oscilloscope display) that they have substantially identical phase responses over the audio band. If in doubt, identical amplifiers should be used. While identical speakers should be used, a small number of speaker manufacturers (e.g. IMF Electronics) have taken trouble to match the different models in their range carefully, and in such cases different models can be used at front and rear. Again, if in doubt, use identical speakers for best results.

The decoder not only reproduces surround sound from a variety of existing systems, but also handles mono and stereo, using ambisonic techniques to get the most natural possible reproduction (using four or six speakers) from existing records and broadcasts. Except for exceptional stereo material, the decoder does not create "pseudo surround sound", but reproduces stereo over a conventional frontal stage with a subtle enhancement over two-speaker stereo, and without any gimmickry. The mono decode mode reproduces a mono source from straight in front, but the

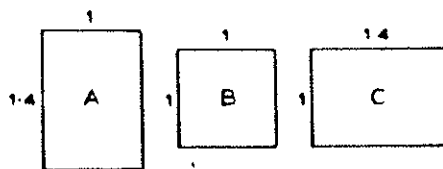


Fig. 7. Most stable front images are provided by arrangement (a), most stable side images by (c), while (b) is a compromise between these extremes.

rear speakers help to lock the image solidly in space behind the front loudspeakers. Neither mono or stereo decode modes enhance badly recorded material, but neither do they degrade it any further. In practice, many ambient SQ classical recordings, such as those released by EMI, will be found to reproduce better in the stereo decode mode than via SQ decode mode, owing to the inherent limitations of the SQ system.

Fig. 6 shows the approximate usable listening area for most decoding modes (excluding SQ) in a typical domestic room using a rectangle layout. The listening area will in practice depend on the loudspeakers used, the room acoustics, the layout shape used, the programme material, and also on the system being decoded. The type of listening area shown has been obtained both with ambisonic recordings made in concert halls, and with commercial "easy listening" (sic) music in the BMX, 45J, Matrix H and RM systems, as well as with stereo material played in stereo decode mode.

It is found that a longer-than-wide layout of four speakers as in Fig. 7(a) gives the most stable front images for non-central listener and the least stable side images. A wider-than-long layout as in Fig. 7(c) gives excellent stable side images for most listeners, although the front stage tends to be drawn over to the nearest speaker. A square layout,

Fig. 7(b), is a compromise in terms of image stability between these extremes. Extremely thin rectangles (whether long or wide) cannot be expected to give good results, although the layout control adjustment will help to minimize the inevitable defects.

Part 2 will give details of the decoding matrices used. Patent rights in circuits described in this and subsequent parts of this article are owned by the National Research Development Corporation. A kit of parts for the decoder will be available from Integrex Ltd - see advertisement.

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## Surround sound decoders — 7

# Multi-system ambisonic decoder

## 2 — Main decoder circuits

by Michael Gerzon, M.A., *Mathematical Institute, Oxford*

The ten systems of decoding provided in this decoder are listed in Table 1. The mono and stereo decoding modes provided are not conventional two-speaker reproduction, which in any case is not a sensible means of reproduction with the first hexagon speaker layout of Fig. 5 last month. Instead they provide a full ambisonic multispeaker reproduction of conventional mono and stereo records or broadcasts, providing a subtle enhancement of first-rate material, but no gimmickry or "pseudo-quadraphonic" effect. The enhancement is not obvious except during extended listening, and a more obvious but still gimmick-free effect over a wider stage is provided by the "superstereo" mode.

Superstereo also gives excellent reproduction of many Regular Matrix and QS records with a full 360° stage, although the RM decoding mode is in some ways better optimized for surround reproduction of records in that system. The five decoding modes for recordings made in the System 45J,

**This series of articles describes a decoder capable of decoding all major existing and proposed two-channel surround-sound systems, including the Ambisonic System 45J, SQ, Regular Matrix, BMX and BBC Matrix H. For systems other than SQ, the decoder gives full psychoacoustically optimized results using NRDC Ambisonic decoding technology. In addition, Ambisonic playback of mono, stereo and of three-channel studio-format signals is provided. The decoder is suitable for three-amplifier/four speaker, four amplifier/four-speaker, and four amplifier/six speaker reproduction.**

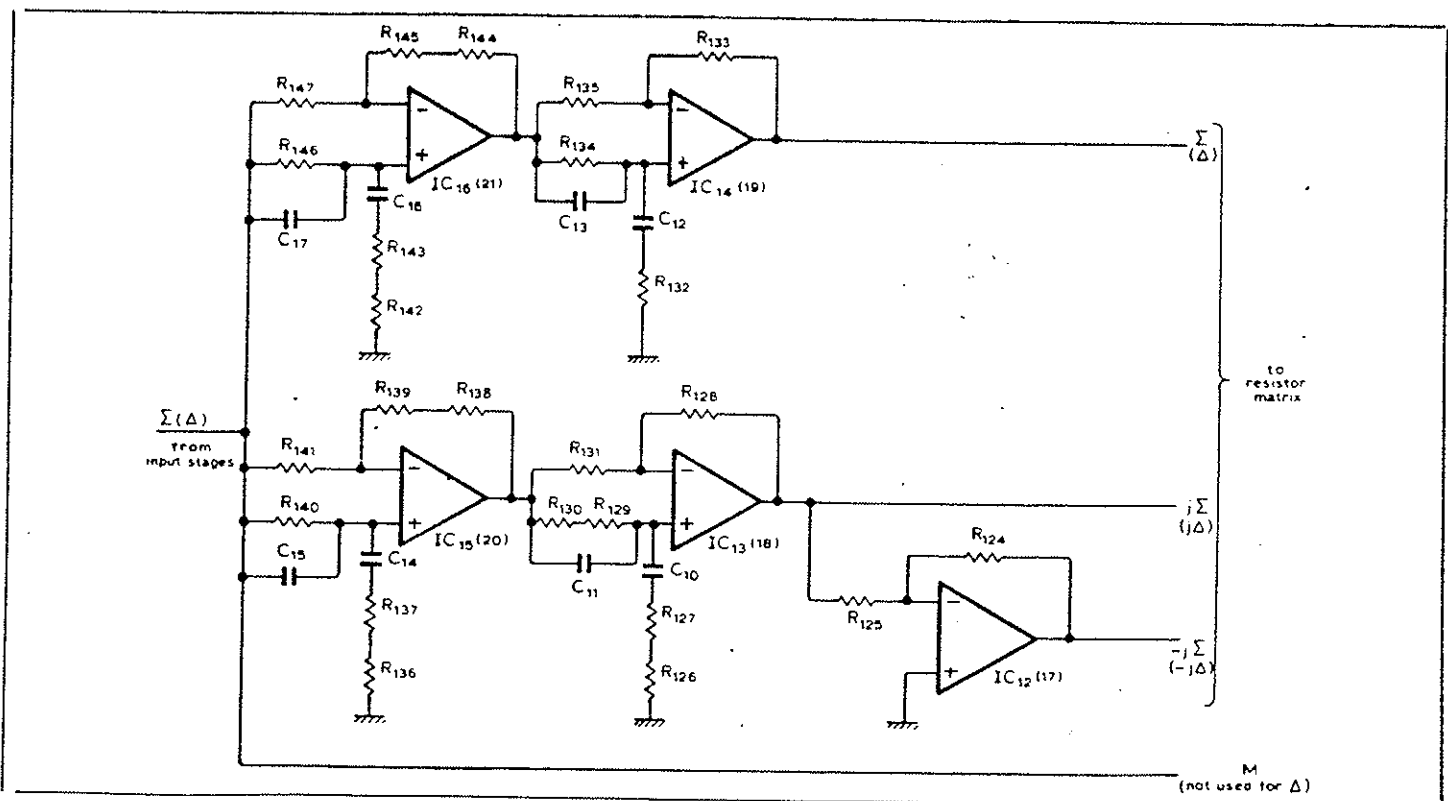
Matrix H, Regular Matrix, BMX (such as Nippon Columbia UD-4 issues) and SQ systems have an obvious purpose, and the B-format mode is intended for studio three-channel recordings in ambisonic B-format. The spare mode, presently unused, is provided to allow for the possibility of the decoder being

updated when three-channel discs or broadcasts in System 45J become available, although it can be used by experimenters to test further decoding ideas.

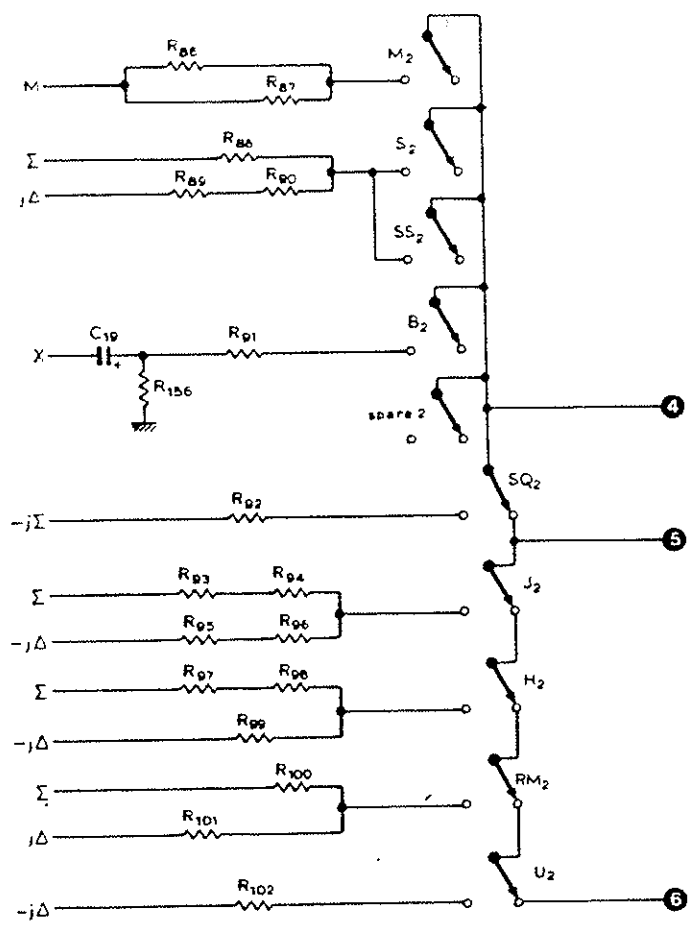
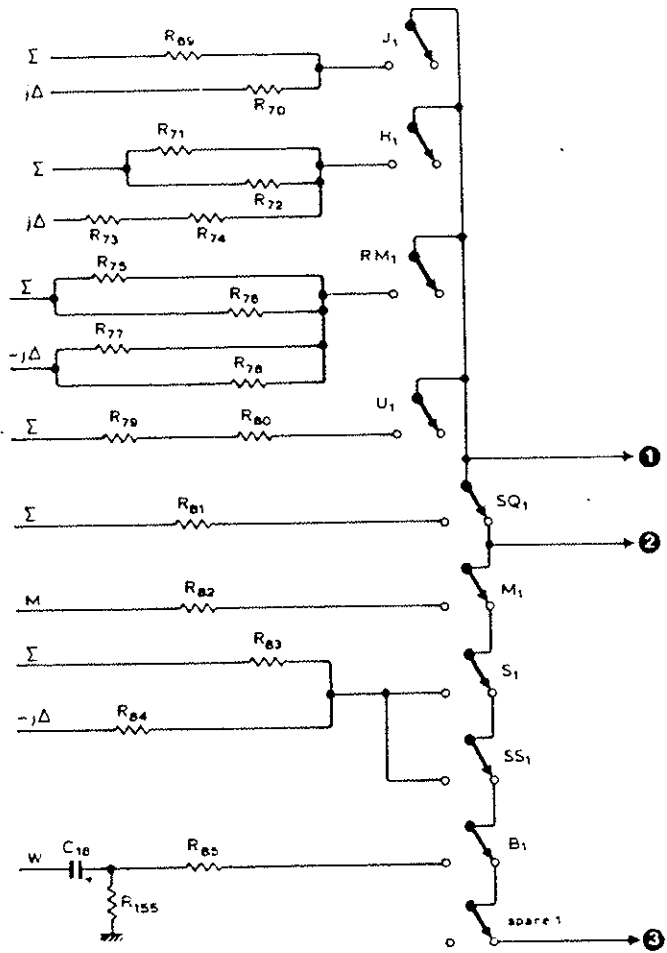
The switching is done by ten interdependent push-button switches; only one switch remains depressed at a time. In the circuit diagrams following, the switches are illustrated in their out position, and the poles of each switch are lettered as listed in Table 1.

Five other push buttons also provide the facilities listed in Table 2. Forward preference, which is operative only in the 45J, Matrix H, RM/QS and BMX system decoding modes, enables the

Fig. 1. Two of these phase shift circuits are used, one to handle the sum signal  $\Sigma$ , and the other to handle the difference signal  $\Delta$ . Input and output signals in the  $\Delta$  case are given in brackets; the M output path is used only in the  $\Sigma$  circuit and is omitted from the  $\Delta$  case. The i.c. numbers in brackets are the numbers for the  $\Delta$  case.







Figs. 2-5. Resistor matrix and switching circuits fed from phase shifter circuits of Fig 1 and feeding corresponding numbered inputs to Fig. 6. Inputs marked W, X and Y in these circuits are the B-format inputs to the decoder. The spare switches are for future developments. Switch poles with the same letters belong to the same switch, and all switches are shown in the out position.

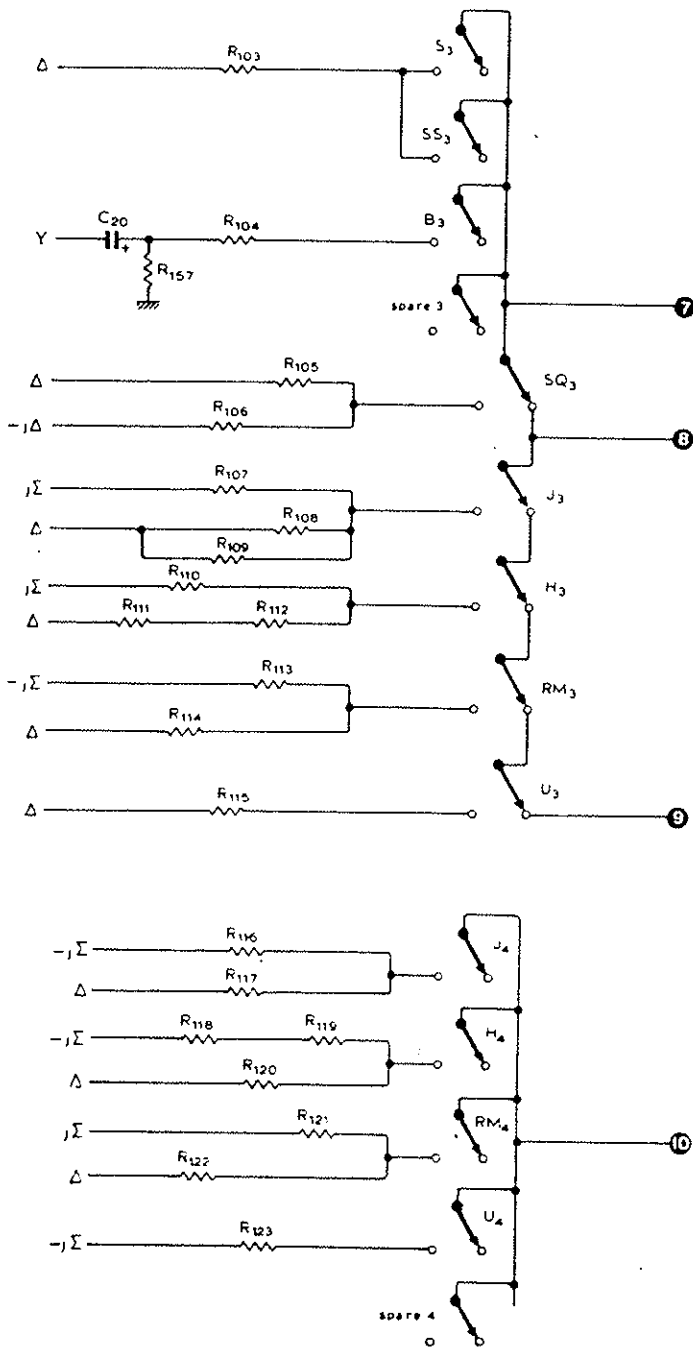
Table 1. List of systems for which decoding is provided. Switch code is the letters by which poles of the switch are indicated in circuit diagrams. Number of switch poles used in each case is listed, although the switches in the available kit have 4 poles each except for the spare position which has 6. The fourth pole of the SS switch is in a part of the circuit to be given in part 3.

Decoding for system	Switch code	Number of poles
Mono	M	2
Stereo	S	3
Superstereo	SS	4
B-format	B	3
System 45J	J	4
BBC Matrix H	H	4
Regular Matrix/QS	RM	4
BMX (UMX/UD-4)	U	4
SQ	SQ	3
Spare	SPARE	(6)

Table 2. Other pushbutton operated facilities. The three switches L, X, Y are interdependent, only one remaining depressed at a time.

Facility	Switch code	Number of poles
Forward preference	F	1
Distance compensation	D	2
Rectangle layout	L	2
Hexagon 2 (part 1)	X	3
Hexagon 1 (part 1)	Y	3

user to choose the decoding mode most suited to his requirements. The in position gives low phasiness for predominantly front-stage material e.g. most classical music and much pop, and the out position gives higher front-stage phasiness but better rear-stage sound quality, e.g. for drama and "easy listening" music. Distance compensation compensates for the effect of the distance of the loudspeakers from the centre of the listening area. The in position is for a nominal speaker distance of 2.4m, corresponding to an 11ft x 11ft square layout, and is suitable for distances of  $\leq 3m$ . The out position is for speaker distances greater than 3m, being nominally exact for a 4m speaker distance. Finally, three interdependent pushbuttons select whether



the decoder is to be used for a rectangular-shaped four-speaker layout, or for the two alternative hexagon layouts, which are wired up as shown in Fig. 5 last month. In the rectangular mode, the shape of the rectangle is compensated for by the layout control potentiometer.

To minimize the possibility of constructional errors and to keep the already rather complicated circuit as simple as possible, the circuit has been based on integrated circuit operational amplifiers. Such a construction requires special precautions in terms of signal levels and input bandwidth to minimize the risk of transient intermodulation distortion. The input stages that include these precautions are described in part 3, along with details of the recommended

op-amp types and their connections. The input stages include low-noise input stages, Bessel filters to prevent t.i.d. and slew-rate distortions, preset gain adjustments to cope with a variety of input levels, a sum and difference matrix to produce the sum signal  $\Sigma = L + R$  and difference signal  $\Delta = L - R$ , and a ganged volume control affecting the sum and difference signals rather than left and right, to minimize the subjective effects of small tracking errors between the potentiometers.

The rest of the circuit performs the signal decoding and is shown in Figures 1-6. Resistors are  $\pm 2\%$  tolerance unless otherwise indicated, when a lower  $\pm 5\%$  tolerance is adequate. Similarly all capacitors are  $\pm 2\%$  or  $2\frac{1}{2}\%$  tolerance unless otherwise indicated. These rela-

tively high tolerances are necessary for good subjective results, as one is producing a  $360^\circ$  sound stage from the two-channel inputs in place of the  $60^\circ$  wide stage of stereo. Such a magnification of the size of the sound stage means that errors are also magnified to a degree that the ears can hear faults that would be negligible in stereo. In addition, there are sufficiently many processing stages that small errors can accumulate. If the decoder were designed for a lower quality of directional reproduction, for example in a music-centre application, tolerances could be relaxed.

To minimize possibility of constructional error, resistors or capacitors of the same value have the same tolerance, although a few resistors — some of those used in series or parallel combinations to make up non-preferred values — could be of lower tolerance than stated. For studio and laboratory applications, the  $2\%$  tolerances may be replaced by  $1\%$  tolerances, because no precision resistor or capacitor values in the circuits given deviate from their ideal values by more than  $1\%$ , and most by considerably less.

The sum and difference signals are each fed into a separate phase-shifter stage as in Fig. 1. Because it is impossible to produce an absolute  $90^\circ$  phase shift in physically realisable circuits, these shifters consist of two all-pass networks one of whose outputs phase-leads the other by  $90^\circ$ . The design shown is a high-quality unity-gain eight-pole design giving  $90^\circ$  relative phase shift over the frequency range 30Hz to 16kHz, ideally with an error of  $\pm 1\frac{1}{2}^\circ$ , but with an error of  $\pm 3^\circ$  approximately using  $2\%$  tolerance components. A phase inverter at the output of the  $90^\circ$ -lead circuit produces a  $90^\circ$  lagging signal. The phase shifters used for the  $\Sigma$  and  $\Delta$  signals are identical circuits, except that for the  $\Sigma$  signal only, a path bypassing the phase shifter is also provided (marked M in Fig. 1). Because of the duplication of circuits, two each are needed of the resistors  $R_{124}$  to  $R_{147}$  and of the capacitors  $C_{10}$  to  $C_{17}$ . Seven outputs are provided from the phase shifters, namely M,  $\Sigma$ ,  $j\Sigma$ ,  $-j\Sigma$ ,  $\Delta$ ,  $j\Delta$  and  $-j\Delta$  where j indicates a relative  $90^\circ$  phase shift.

These outputs feed an elaborate switched resistor matrix, shown in Figs 2-5. The elaboration is, of course, a consequence of providing ten different options for decoding. For mono decoding, the signal M is taken from before the phase shifters, to minimize phase distortion. Although by present day standards, the type of low-Q phase distortion produced by the phase shifters is not very audible, it does have some audible effect, and so should be avoided where possible. It is not possible to avoid phase distortion in two-channel surround-sound systems with currently available technology, and the justification for allowing such phase distortion is that the beneficial effects of

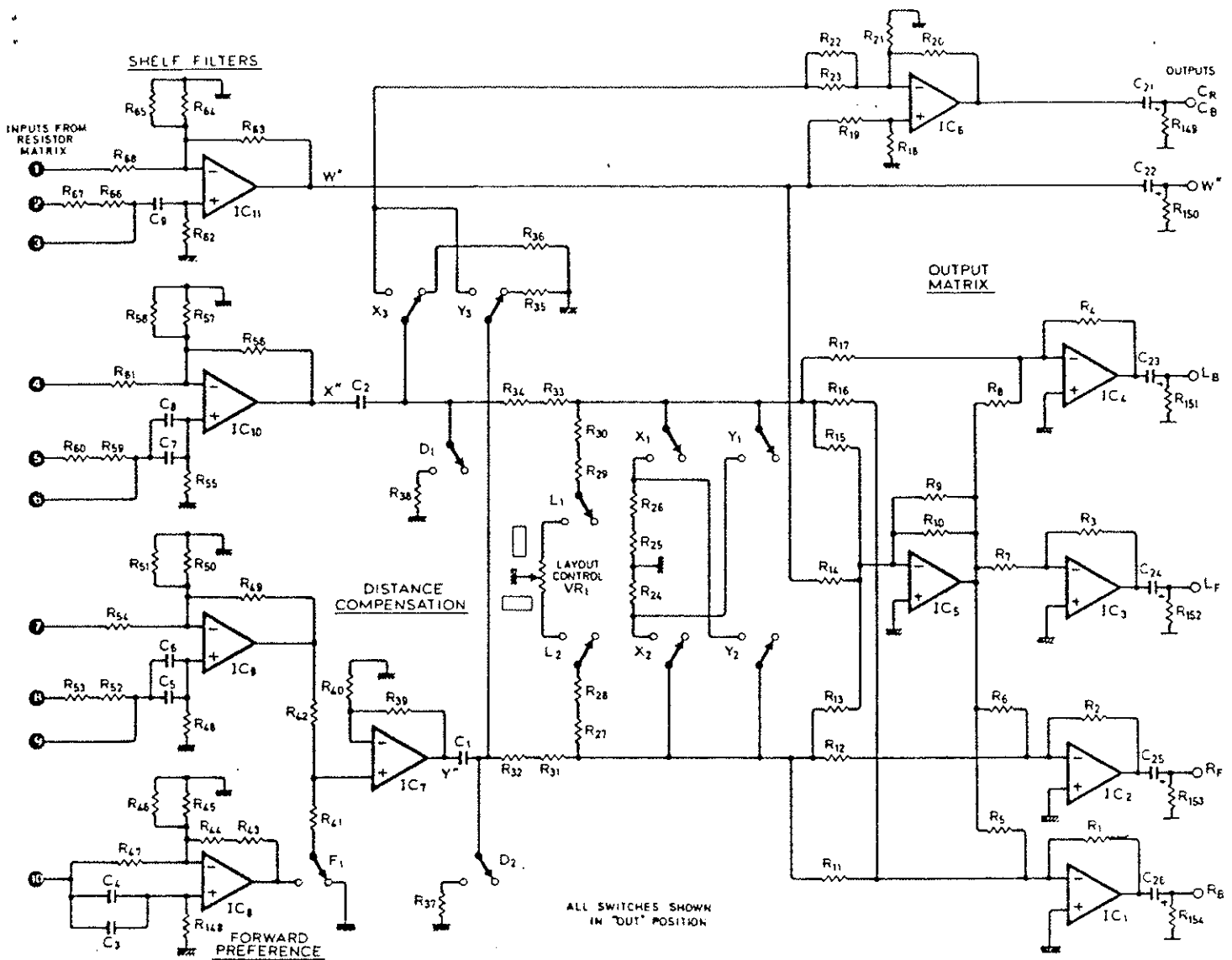


Fig. 6. Shelf filters, distance compensation and output matrices, including forward preference switch, distance compensation switch, rectangle and hexagon selector switches and layout control. Switch poles with the same letter (as in Table 2) correspond to the same pushbutton, and all are shown in out position. See text for selection of resistors  $R_{28}$  and  $R_{29}$  associated with the linear  $5k\Omega$  nominal layout control potentiometer  $VR_1$ . Circuit fed from Figs 2-5, and outputs  $L_B$ ,  $L_F$ ,  $R_F$ ,  $R_B$ ,  $W''$  and  $C_B$  or  $C_R$  feed power amplifiers or "quadraphonic" preamp. Switch  $X$  selects  $C_B$  output and switch  $Y$  selects  $C_R$  output.

surround sound can easily outweigh any small quality losses thus caused.

The switching shown in Figs 2-5 selects the required matrix resistors and modifies the action of the shelf filter circuitry following to obtain the shelf filter characteristics required for each system. The poles of each push-button switch are marked with the same letter (eg J) as indicated in Table 1, but with a number running from 1 to 4 indicating the pole used. All switches are shown in their out position. For example, switch pole RM4 is the 4th pole of the regular Matrix/QS push button. Pole SS4 (not

shown) is used in the input stages to be described in the next article.

Apart from the seven signals  $M$ ,  $\Sigma$ ,  $j\Sigma$  etc already discussed, another three inputs marked  $W$ ,  $X$ ,  $Y$  also feed the resistor matrix, and are taken from a separate input socket. These three inputs are for studio B-format signals, previously discussed in ref. 1. The ten output connections of Figs. 2-5 are fed to the corresponding numbered points at the input of Fig. 6, which includes the rest of the decoder.

The circuits surrounding the operational amplifiers  $IC_{8,11}$  are the shelf filters, which have a resistive input impedance of  $22k\Omega$  to terminate the resistor matrix, and which except in the SQ mode give a phase at their output  $90^\circ$  in advance of their inputs at a frequency of  $400\text{Hz}$ , thereby ensuring virtually identical phase responses. The amplitude gains (ignoring phase) of these filters at frequencies much less than  $400\text{Hz}$  and at frequencies much more than  $400\text{Hz}$  are shown in Table 3.

Including the effect of these filters and the effect of the summing circuit at  $IC_7$ , a pressure signal  $W''$  and two velocity signals  $X''$  and  $Y''$  are produced at the outputs respectively of  $IC_{11}$ ,  $IC_{10}$  and  $IC_7$  in Fig. 6, that at frequencies well above  $400\text{Hz}$  satisfy the following

Table 3. Amplitude gains of shelf filters at low ( $<400\text{Hz}$ ) and high ( $>400\text{Hz}$ ) frequencies for the decoding systems in Table 1, for the signal paths handling pressure signal  $W'$ , velocity signals  $X'$  and  $Y'$  and phasiness compensation signal  $P$  (through  $IC_9$ ). The  $P$  signal for spare mode is as for J, H, RM and U systems.

Signal path	J, H, RM, U		M, S, SS, B, SPARE		SQ
	l.f.	h.f.	l.f.	h.f.	
$W'$	3.73	5.65	3.73	4.56	3.73
$X'$ and $Y'$	3.73	2.91	3.73	3.23	3.73
$P$	1.38	3.73			

matrix equations for the various decoding modes. For systems J, H, RM, U, the expression for  $Y''$  is the sum of two bracketed terms, the second of which is deleted for the forward preference switch in the out position.

Mode M (mono)

$$W'' = 0.707M, X'' = 0.707M$$

Mode S (stereo) and SS (superstereo)

$$W'' = 0.717\Sigma - 0.291j\Delta$$

$$X'' = 0.717\Sigma + 0.291j\Delta$$

$$Y'' = 0.583\Delta$$

For superstereo, the signal gain is modified in the input stages as well.

Mode B (B format)

$$W'' = 1.288W$$

$$X'' = 0.911X$$

**Resistors**

Tolerance  $\pm 2\%$ , except when marked with asterisk when  $\pm 5\%$  is adequate.

R <sub>1,4</sub>	200k	R <sub>92</sub>	91k
R <sub>5,10</sub>	68k	R <sub>93</sub>	100k
R <sub>11,13</sub>	33k	R <sub>94*</sub>	3.9k
R <sub>14</sub>	200k	R <sub>95</sub>	47k
R <sub>15-17</sub>	33k	R <sub>96*</sub>	3.3k
R <sub>18</sub>	27k	R <sub>97</sub>	150k
R <sub>19</sub>	100k	R <sub>98*</sub>	8.2k
R <sub>20</sub>	150k	R <sub>99</sub>	33k
R <sub>21</sub>	110k	R <sub>100,101</sub>	82k
R <sub>22*</sub>	1M	R <sub>102</sub>	68k
R <sub>23</sub>	270k	R <sub>103</sub>	100k
R <sub>24</sub>	6.8k	R <sub>104</sub>	56k
R <sub>25*</sub>	39 $\Omega$	R <sub>105,106</sub>	68k
R <sub>26</sub>	2.7k	R <sub>107</sub>	330k
R <sub>27,30</sub>	1.6k	R <sub>108</sub>	56k
R <sub>28,29</sub>	see text	R <sub>109*</sub>	2.2M
R <sub>31,33</sub>	6.8k	R <sub>110*</sub>	1M
R <sub>32,34</sub>	10k	R <sub>111</sub>	56k
R <sub>35,36</sub>	120k	R <sub>112*</sub>	4.7k
R <sub>37,38</sub>	27k	R <sub>113,114</sub>	82k
R <sub>39-42</sub>	47k	R <sub>115</sub>	68k
R <sub>43*</sub>	22k	R <sub>116*</sub>	240k
R <sub>44</sub>	3.3k	R <sub>117*</sub>	2.2M
R <sub>45</sub>	27k	R <sub>118*</sub>	390k
R <sub>46</sub>	12k	R <sub>119</sub>	22k
R <sub>47,48</sub>	150k	R <sub>120*</sub>	1.8M
R <sub>49,56</sub>	22k	R <sub>121,122</sub>	220k
R <sub>50,57</sub>	82k	R <sub>123</sub>	180k
R <sub>51,58*</sub>	39k	R <sub>149,156</sub>	390k
R <sub>52,59*</sub>	680k	R <sub>157*</sub>	390k
R <sub>53,60*</sub>	56 $\Omega$		
R <sub>54,61</sub>	1k	Two off of the following	
R <sub>55,62</sub>	22k	R <sub>124,125</sub>	47k
R <sub>63</sub>	22k	R <sub>126*</sub>	1.8k
R <sub>64</sub>	82k	R <sub>127,132</sub>	22k
R <sub>65*</sub>	18k	R <sub>128,133</sub>	47k
R <sub>66*</sub>	820k	R <sub>129*</sub>	3.9k
R <sub>67*</sub>	1.1k	R <sub>130,134</sub>	47k
R <sub>68</sub>	1.8k	R <sub>131,135</sub>	12k
R <sub>69</sub>	22k	R <sub>136*</sub>	2.0k
R <sub>70*</sub>	100k	R <sub>137</sub>	33k
R <sub>71</sub>	910k	R <sub>138,144*</sub>	5.6k
R <sub>72*</sub>	100k	R <sub>139,145</sub>	100k
R <sub>73</sub>	2.2k	R <sub>140</sub>	39k
R <sub>74*</sub>	430k	R <sub>141,147</sub>	22k
R <sub>75,78</sub>	8.2k	R <sub>152*</sub>	3.3k
R <sub>79</sub>	270k	R <sub>143</sub>	47k
R <sub>80*</sub>	100k	R <sub>146</sub>	56k
R <sub>81</sub>	5.6k	R <sub>148</sub>	22k
R <sub>82</sub>	91k		
R <sub>83</sub>	120k		
R <sub>84</sub>	110k		
R <sub>85</sub>	270k		
R <sub>86*</sub>	56k		
R <sub>87</sub>	360k		
R <sub>88</sub>	100k		
R <sub>89</sub>	68k		
R <sub>90*</sub>	160k		
R <sub>91</sub>	8.2k		
	56k		

**Capacitors**

Tolerance  $\pm 2$  or  $2\frac{1}{2}\%$ , except when marked with asterisk when  $\pm 10\%$  is adequate.

C <sub>1,2*</sub>	680n
C <sub>3</sub>	1n
C <sub>4,8</sub>	10n
C <sub>9</sub>	15n
C <sub>18-26</sub>	47 $\mu$ 10V low tolerance

Two on of the following

C <sub>10,11</sub>	470p
C <sub>12,13</sub>	100n
C <sub>14,15</sub>	27n
C <sub>16,17</sub>	1n

$Y'' = 0.911Y$

Mode J (two-channel system 45J)

$W'' = 0.998\Sigma + 0.107j\Delta$

$X'' = 0.374\Sigma - 0.772j\Delta$

$Y'' = (0.132j\Sigma + 0.798\Delta + (-0.295j\Sigma + 0.032\Delta))$

Mode H (BBC Matrix H)

$W'' = \Sigma + 0.219j\Delta$

$X'' = 0.215\Sigma - 1.037j\Delta$

$Y'' = (0.044j\Sigma + 0.736\Delta + (-0.186j\Sigma + 0.041\Delta))$

Mode RM (Regular Matrix/QS)

$W'' = 0.728\Sigma - 0.728j\Delta$

$X'' = 0.515\Sigma + 0.515j\Delta$

$Y'' = (-0.515j\Sigma + 0.515\Delta + (0.310j\Sigma + 0.310\Delta))$

Mode U (BMX)

$W'' = 1.018\Sigma, X'' = -0.720j\Delta,$

$Y'' = (0.720\Delta) + (-0.406j\Sigma)$

Mode SQ (for SQ recordings)

$W'' = 0.73\Sigma$

$X'' = -0.73j\Sigma$

$Y'' = 0.73\Delta - 0.73j\Delta$

These decoding equations include some allowance for maximizing the number of preferred resistor values, and are arranged so as to give substantially the same loudness in all decoding modes.

The shelf filters in the X'' and Y'' paths are followed by a passive RC high-pass filter for distance compensation, with switched resistors R<sub>35-38</sub> arranged so that switch D changes its time constant and the hexagonal-mode switching does not alter the relative time constants in the X'' and Y'' paths. The hexagonal-mode switching activates a summing circuit to derive the signals C<sub>B</sub> or C<sub>R</sub> described in part 1, and which alter the matrixing coefficients for the L<sub>B</sub>, L<sub>F</sub>, R<sub>F</sub>, R<sub>B</sub> outputs. Switches are shown in their out position, and as before the numbering indicates the pole number of the switch lettered as in Table 2.

In the rectangular decoding mode, a linear-law potentiometer VR<sub>1</sub> is switched into circuit to enable the output matrix to be varied continuously so as to compensate for layout shape. This potentiometer should ideally have a total track resistance equal to 5k $\Omega$  within 2%, but this would be extremely expensive and not very practical. Thus two padding resistors R<sub>28</sub> and R<sub>29</sub> are provided so that the total track resistance be brought up to the desired exact value. They should be chosen to have identical values R such that 2R + VR<sub>1</sub> has a total resistance of 6.2k $\Omega$ . If precision measurement of resistors is not available (and if the circuit is not being built from a kit with R<sub>28</sub> and R<sub>29</sub> provided to match VR<sub>1</sub>), then choose R<sub>28</sub> and R<sub>29</sub> to have values identical to within 5% such that the total measured resistance of R<sub>30</sub>, R<sub>29</sub>, VR<sub>1</sub>, R<sub>28</sub> and R<sub>27</sub> in series is the same as that of two 4.7k 2% tolerance resistors in series.

The layout control is connected so that a long and narrow loudspeaker layout involves a setting with the wiper near the X'' end of the potentiometer, and conversely for a short and wide layout. This arrangement may be found confusing because many people feel

(incorrectly) that it should be the other way round. The central setting of the potentiometer corresponds to a square loudspeaker layout, and the end settings correspond to a rectangle whose long side is twice its short side. Calibrations for the layout control are provided in kit versions, and calibration instructions for do-it-yourselfers will be provided later. The equations describing the action of the output matrix and layout control were given in part 1.

Details of the input stages and of recommended op amps are given in part 3.

**Reference**

1. Gerzon, M. A. Ambisonics, Part two. Studio techniques, *Studio Sound*, vol. 17, Aug. 1975, pp. 24-6, 28, 30. Correction *ibid* vol. 17, Oct. 1975, p. 60.

Here are corrections to the Gerzon "Multi-system Ambisonic Decoder" published in Wireless World magazine, July and August, 1977.

Corrections to the phase shift circuits (Fig 1, pg 69, Aug 1977 and m/l on pg 73 of same issue:

Reference Designator(s)	m/l value	correct value
C16	1nF	1nF
C17	1nF	1nF
R147	22k	22.1k
R146	56k	50.339k
R145+R144	105.6k	120.041k
R143+R142 (1)	50.3k	55.772k
C13	0.1uF	0.1uF
C12	0.1uF	0.1uF
R135	12k	12.1k
R134	47k	22.006k
R133	47k	115.860k
R132	22k	47.235k
C15	27nF	47nF
C14	27nF	47nF
R141	22k	22.1k
R140	39k	20.122k
R139+R138	105.6k	120.108k
R137+R136	35k	22.315k
C11	470pF	470pF
C10	470pF	470pF
R131	12k	12.1k
R130+R129	50.9k	23.809k
R128	47k	115.655k
R127+R126	23.8k	50.941k

Notes:

1. R142 on schematic incorrectly listed as R152 on material list
2. Capacitors are chosen to be standard values most closely matching Gerzon values (except C14/15 which were typos)
3. Resistors are either standard 1% (E96) values (where choice is 'arbitrary') or calculated ideal values to 5 or 6 digits of precision.
4. Capacitors should be selected to 1% tolerance (or purchased that way)
5. Resistors should be selected from nearest standard 1% value or made of series/parallel combination to get closer to exact value