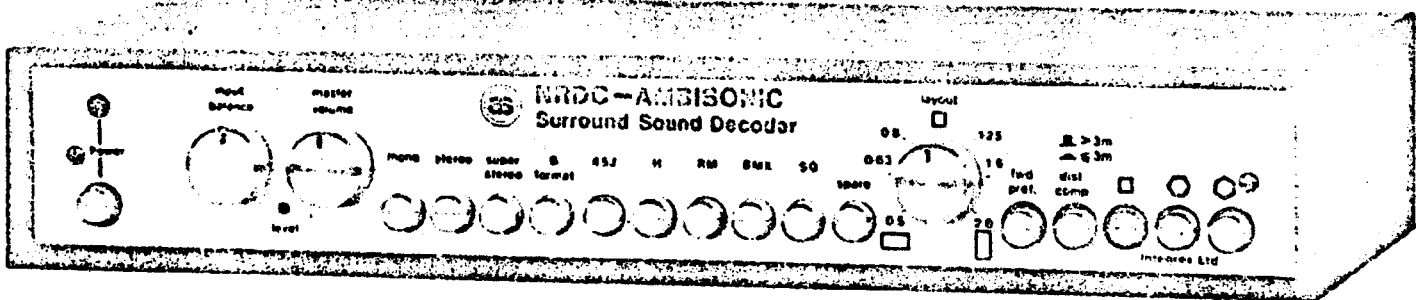# NRDC-AMBISONIC
Surround Sound Decoder

The first ever kit specially produced by Integrex for this British NRDC backed surround sound system which is the result of 7 years research by the Ambisonic team. The design was pub - lished in "Wireless World" July and August 1977.

The following ten push button selected encoded formats are decoded to Ambisonic format (4 or 6 speakers).

Mono	Matrix H (BBC)
Stereo	R.M. (from which QS is derived)
Super stereo (widestage)	BMX
B format (3ch input)	SQ
45J	Spare (future development)

Additionally, push button selection is provided for speaker layout as follows:-

Forward preference

Distance compensation $\rightarrow$ or $\rightleftarrows$ 3 metres

Rectangular layout (with variable control for 1:2 to 2:1 rectangles)

Hexagonal layout (0°)

Hexagonal layout (90°)

The 2 channel input amplifier has a maximum sensitivity of 20mVrms and is internally switched in approximately 10dB steps up to 1Vrms.

Linear phase 3 pole filters follow the input pre-amp to greatly reduce the possibility of slew rate limiting near overload conditions. The signal processing IC's are quad low noise high slew rate types.

Optimum operating signal level indication is given by a front panel LED. Mismatch in level of the input signal of up to $\pm$ 3dB may be compensated by the Input Balance Control.

The Master Volume Control adjusts the level of all 4 or 6 output signals. NB A minimum of only 3 power amps are necessary for full decoding to 4 loudspeakers.

The B format 3 channel input is not affected by the above controls.

The internal operating level and output level of the unit is 300mVrms. All outputs are low impedance and are short circuit protected.

Surround sound decoders

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[1-4]. 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*[1, 5] 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 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[6, 7] and NHK[8] 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 wavefront 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
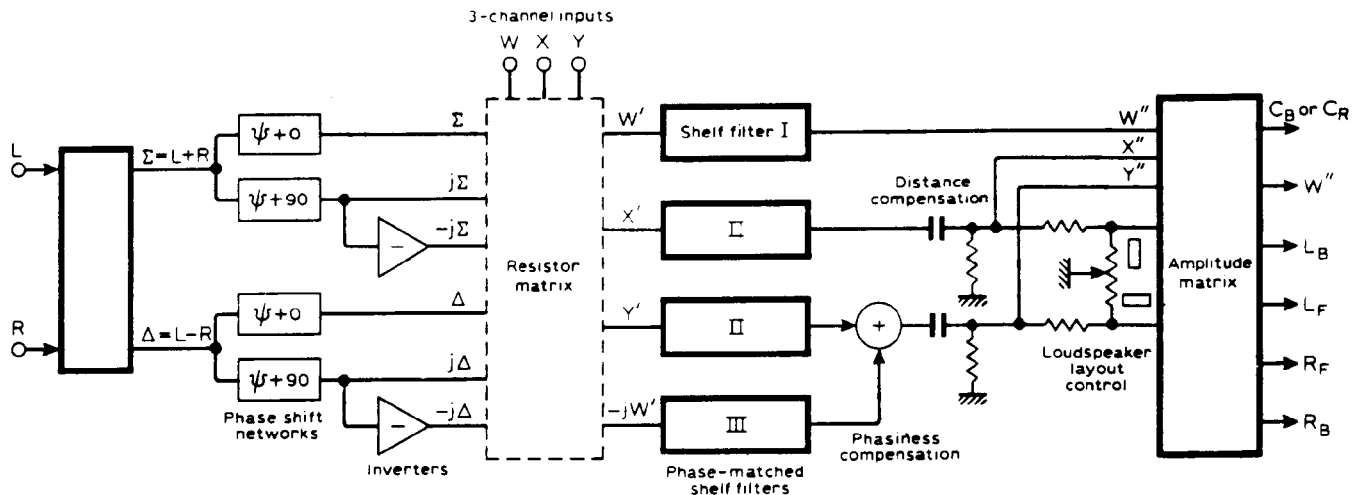
3

3-channel inputs
W X Y

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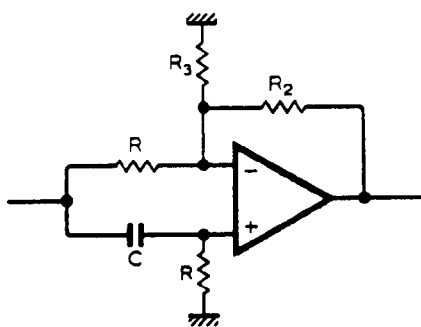
Fig. 2. Phase-compensated shelf-filter circuits allow frequency-dependent decoding. Conventional RC shelf filters would cause unwanted phase differences between signal paths.

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[9].

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[1] 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 Σ = L + R and difference Δ = 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° and 90° phase shift networks, and the 90°-shifted signal is also phase inverted to yield a −90° phase shifted signal in each case. The
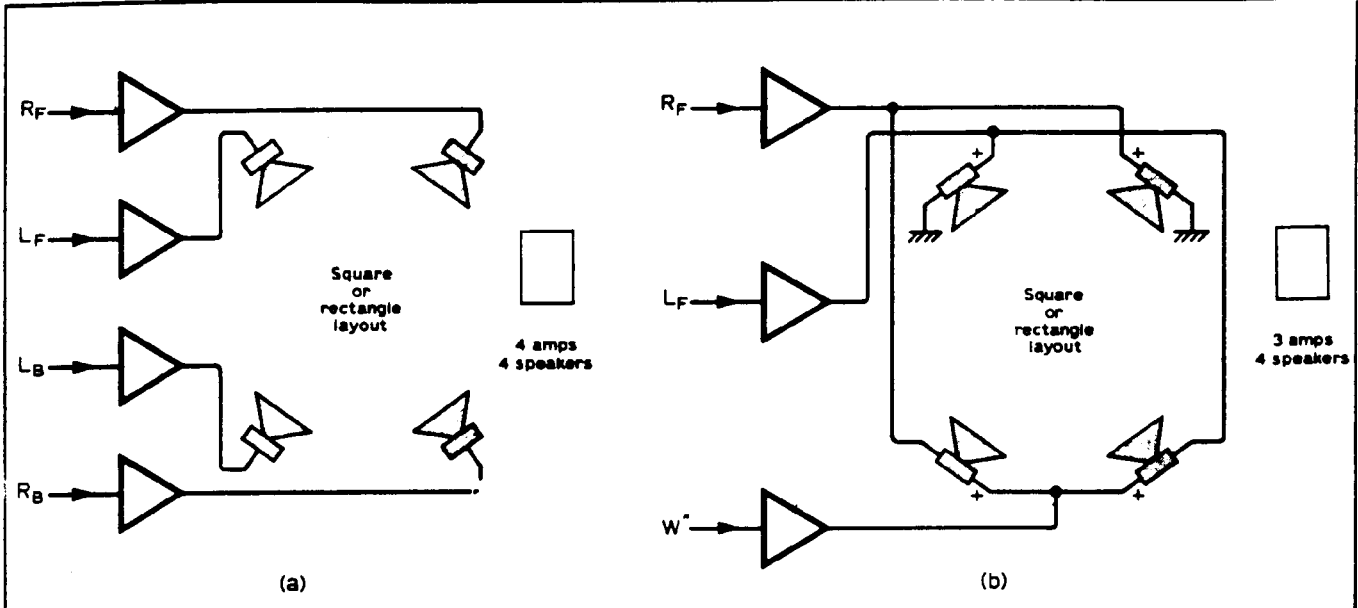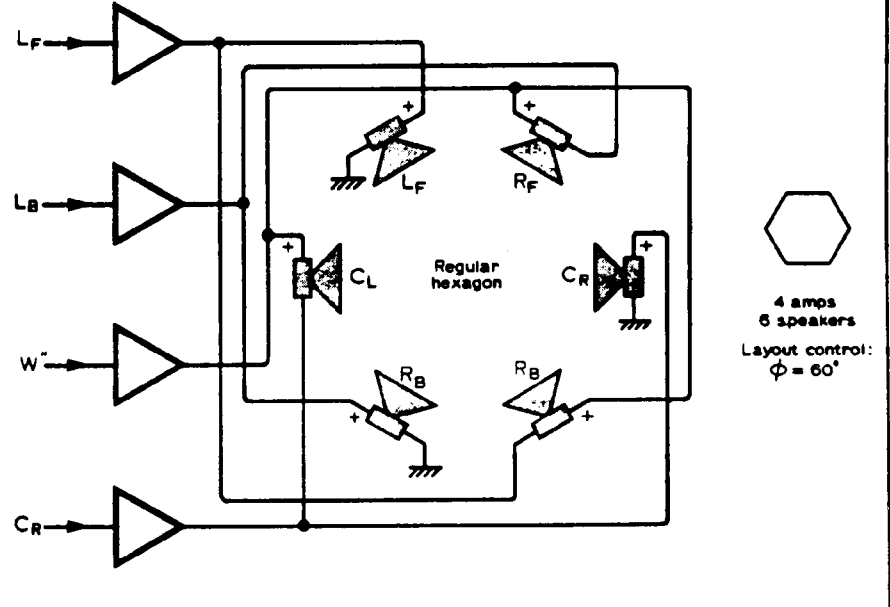
(a)

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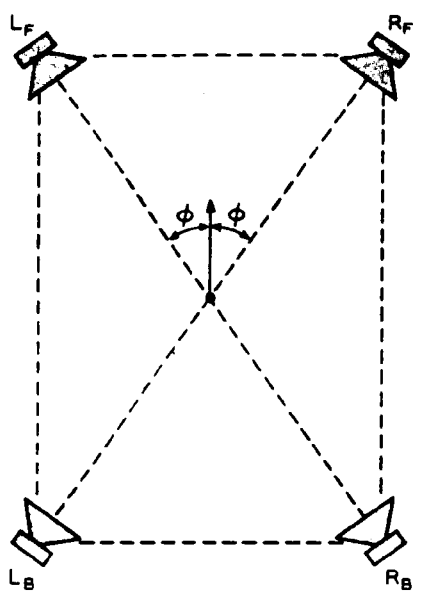
Fig. 3. Methods of feeding four loudspeakers using four amplifiers (a), and three amplifiers (b). Speaker terminals marked + are the positive-phase terminals in each case.

Fig. 4. Compensation is provided for non-square layouts. Angle φ 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 φ used in the equations of the output matrix.
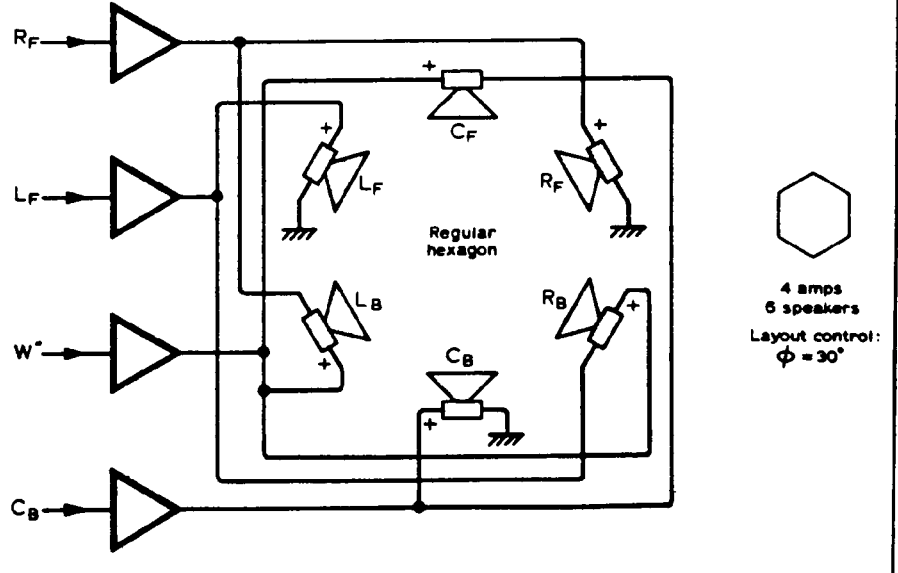
phase-shifters used are high precision types, as the ears are capable of hearing very small errors in localization (as little as 2° in real life). Previous "quadrophonic" decoders have not required such high precision mainly because they gave in any case a poor decoded effect due to sup-optimal design.

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 mastertapes. 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_2/R controls the ratio of high-to-low-frequency gain of the shelf, and R_3 provides extra h.f. gain to make up the losses of the preceeding 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 adjust the outputs of the decoder 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

$$L_B = \tfrac{1}{2}(W'' - \sqrt{2} \sin \phi \, X'' + \sqrt{2} \cos \phi \, Y'')$$
$$L_F = \tfrac{1}{2}(W'' + \sqrt{2} \sin \phi \, X'' + \sqrt{2} \cos \phi \, Y'')$$
$$R_F = \tfrac{1}{2}(W'' + \sqrt{2} \sin \phi \, X'' - \sqrt{2} \cos \phi \, Y'')$$
$$R_B = \tfrac{1}{2}(W'' - \sqrt{2} \sin \phi \, X'' - \sqrt{2} \cos \phi \, Y'')$$

where ϕ depends on the setting of the layout control, being 45° for a square layout, and being equal to the angle ϕ shown in Fig. 4 for a rectangle layout. From these formulae

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

so that

$$L_B = W'' - R_F$$
$$R_B = W'' - L_F,$$

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 emphasise 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

$$C_B = \tfrac{1}{2}(W'' - \sqrt{2}\,X'')$$
$$C_R = \tfrac{1}{2}(W'' - \sqrt{2}\,Y'').$$

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
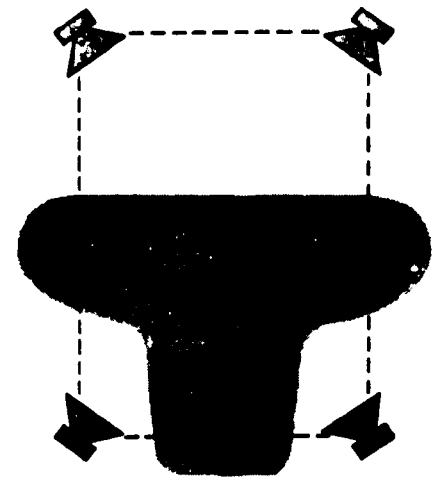

Fig. 6. Approximate listening area for ambisonic decoding (shaded) with a rectangle speaker layout obtained for BMX, 45J, Matrix H and RM systems as well as stereo. Optimal listening is at the centre (X).

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
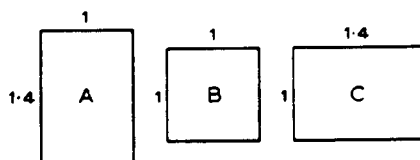

Fig. 7. Most stable front images are provided by arrangement (a), most stable side images by (c), while (b) is a compromise between these extremes.

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

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 "super-stereo" 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 Σ, and the other to handle the difference signal Δ. Input and output signals in the Δ case are given in brackets; the M output path is used only in the Σ circuit and is omitted from the Δ case. The i.c. numbers in brackets are the numbers for the Δ case.
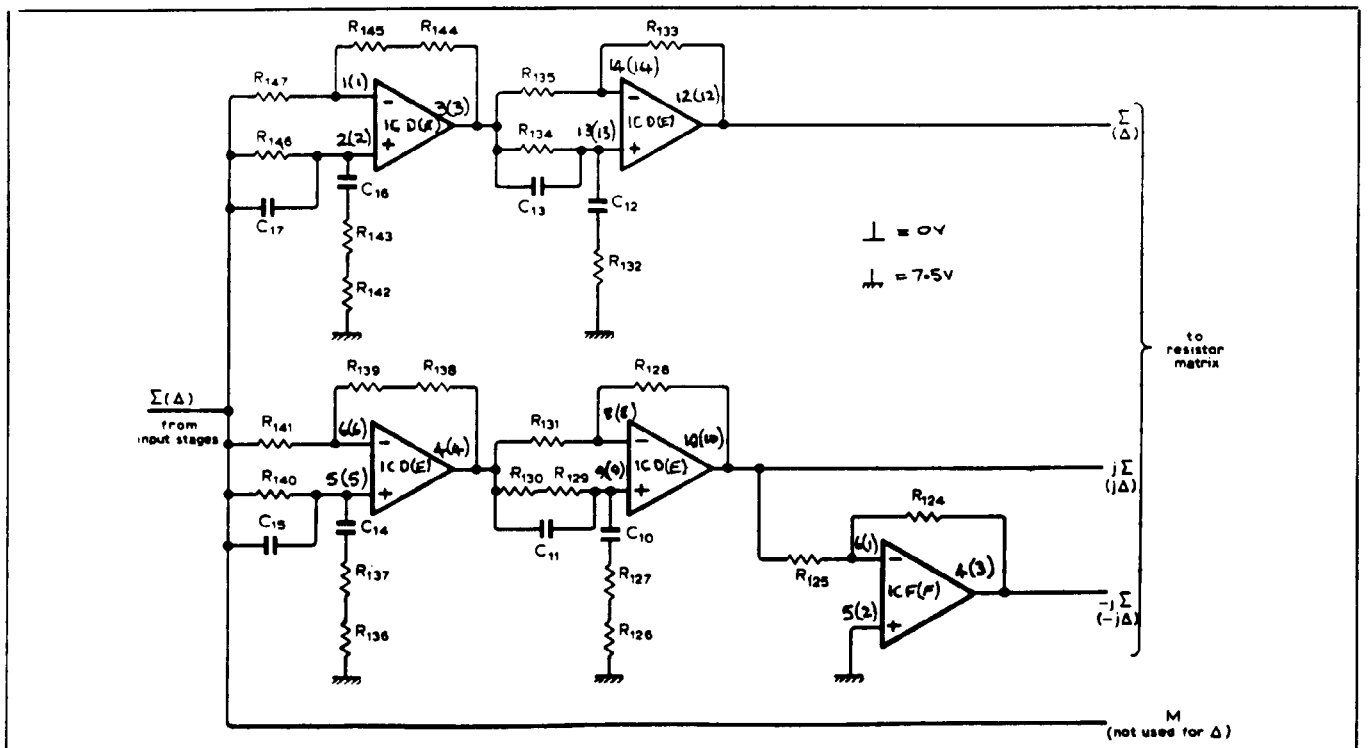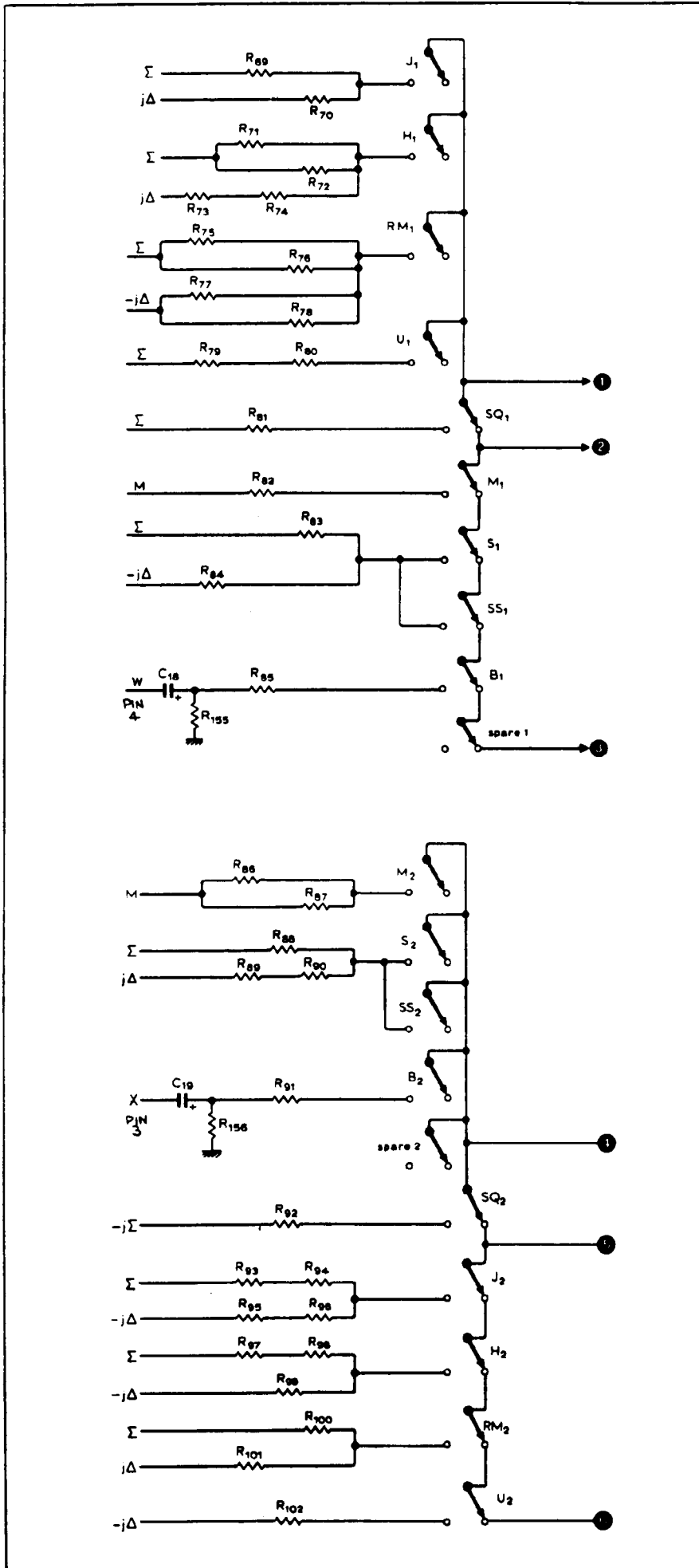
Figs. 2-5. Resistor matrix and switching circuits fed from phase shifter circuits of Fig. 1 and feeding corresponding numbered inputs to Fig. 6. Inputs marked W, X and Y in these circuits are the B-format inputs to the decoder. The spare switches are for future developments. Switch poles with the same letters belong to the same switch, and all switches are shown in the out position.
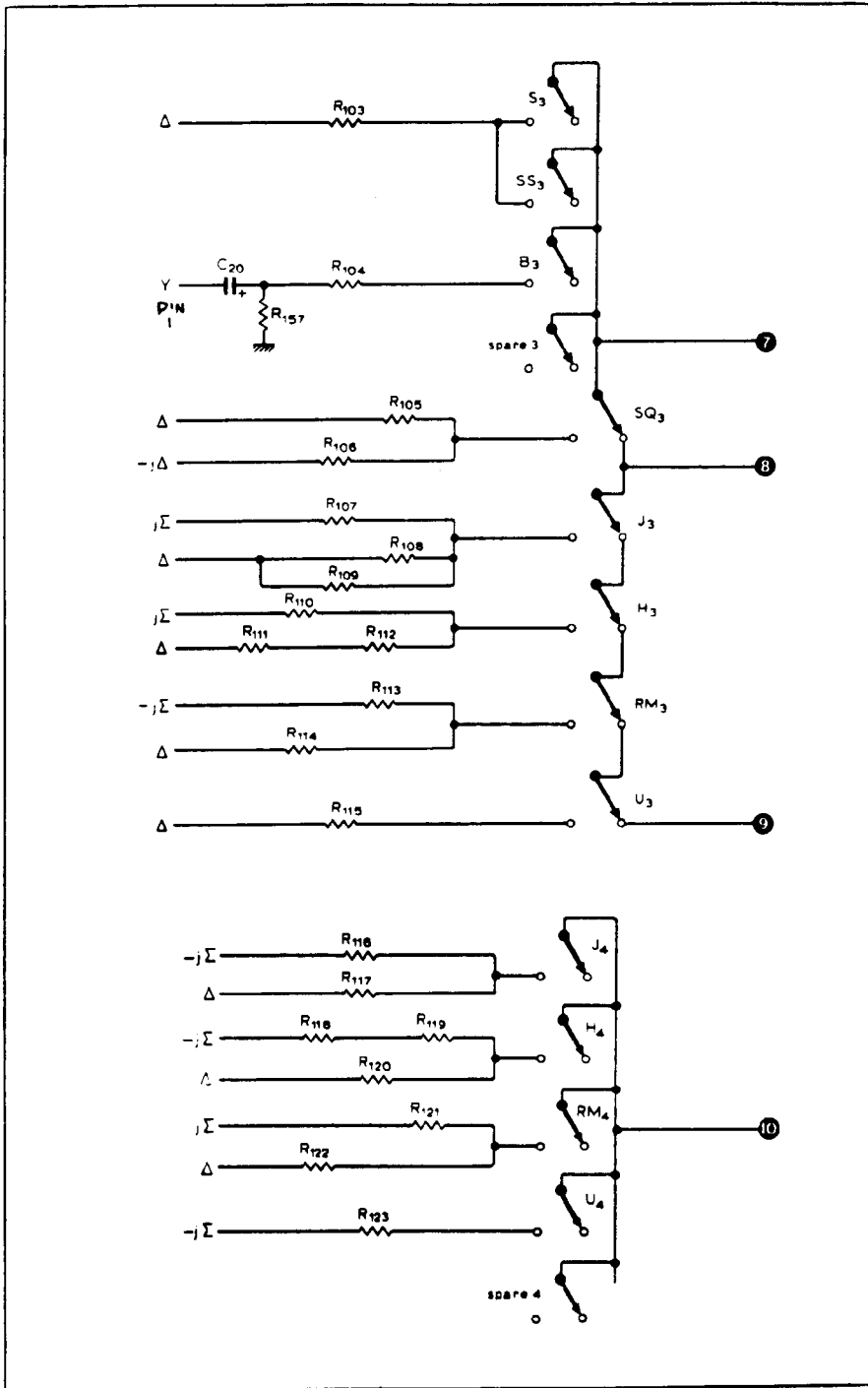
Table 1. List of systems for which decoding is provided. Switch code is the letters by which poles of the switch are indicated in circuit diagrams. Number of switch poles used in each case is listed, although the switches in the available kit have 4 poles each except for the spare position which has 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 × 11ft square layout, and is suitable for distances of <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

R₁₀₃ · S₃ · SS₃ · B₃ · spare 3 · SQ₃ · J₃ · H₃ · RM₃ · U₃
C₂₀ · R₁₀₄ · R₁₅₇ · R₁₀₅ · R₁₀₆ · R₁₀₇ · R₁₀₈ · R₁₀₉ · R₁₁₀ · R₁₁₁ · R₁₁₂ · R₁₁₃ · R₁₁₄ · R₁₁₅
R₁₁₆ · R₁₁₇ · R₁₁₈ · R₁₁₉ · R₁₂₀ · R₁₂₁ · R₁₂₂ · R₁₂₃ · J₄ · H₄ · RM₄ · U₄ · spare 4

tively high tolerances are necessary for good subjective results, as one is producing a 360° sound stage from the two-channel inputs in place of the 60° 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° phase shift in physically realisable circuits, these shifters consist of two all-pass networks one of whose outputs phase-leads the other by 90°. The design shown is a high-quality unity-gain eight-pole design giving 90° relative phase shift over the frequency range 30Hz to 16kHz, ideally with an error of ±1½°, but with an error of ±3° approximately using 2% tolerance components. A phase inverter at the output of the 90°-lead circuit produces a 90° lagging signal. The phase shifters used for the Σ and Δ signals are identical circuits, except that for the Σ 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 R₁₂₄ to R₁₄₇ and of the capacitors C₁₀ to C₁₇. Seven outputs are provided from the phase shifters, namely M, Σ, jΣ, -jΣ, Δ, jΔ and -jΔ where j indicates a relative 90° 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

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 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 Σ = L + R and difference signal Δ = 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 ±2% tolerance unless otherwise indicated, when a lower ±5% tolerance is adequate. Similarly all capacitors are ±2% or 2½% tolerance unless otherwise indicated. These rela-
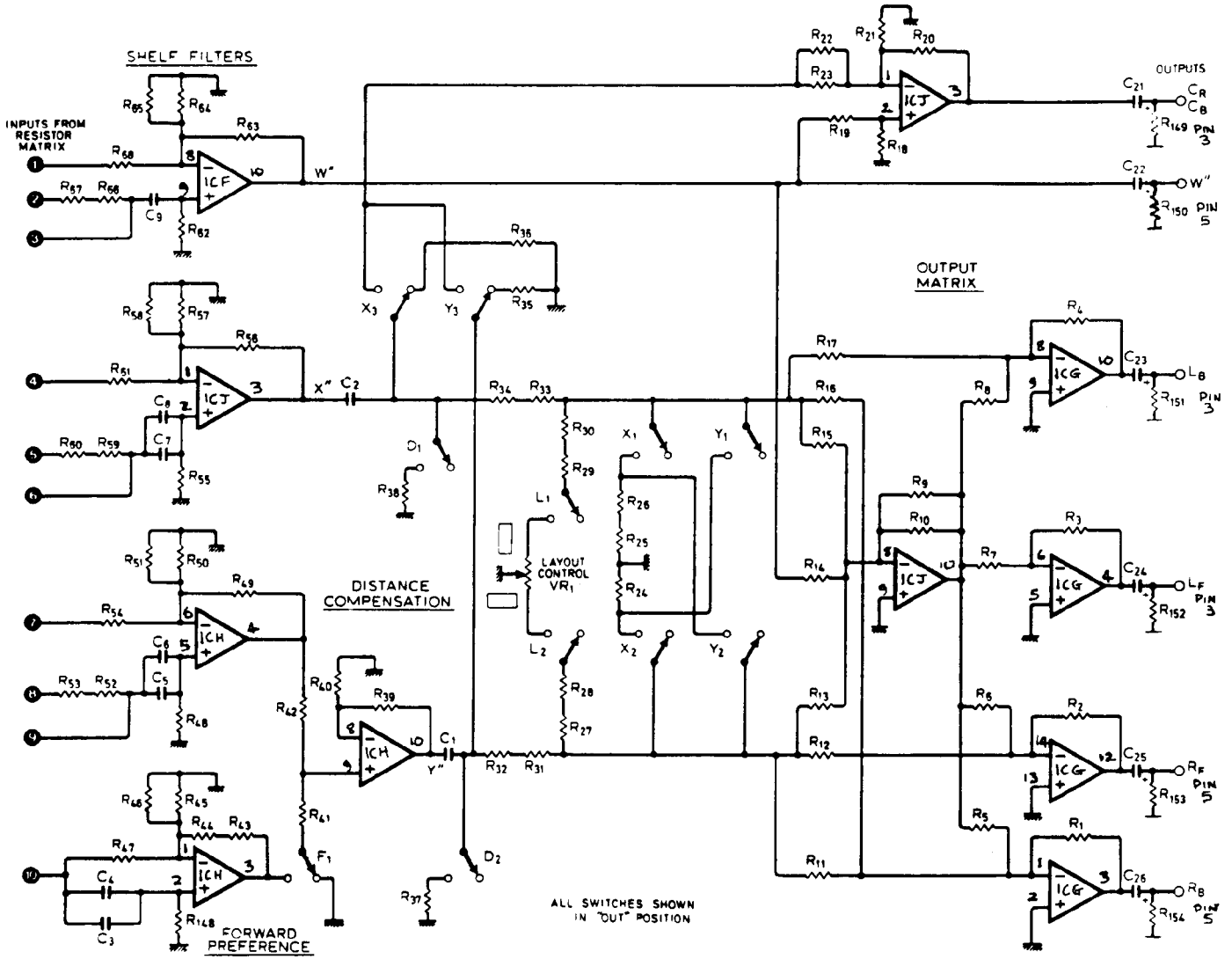
Fig. 6. Shelf filters, distance compensation and output matrices, including forward preference switch, distance compensation switch, rectangle and hexagon selector switches and layout control. Switch poles with the same letter (as in Table 2) correspond to the same pushbutton, and all are shown in out position. See text for selection of resistors R_{28} and R_{29} associated with the linear 5kΩ 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) us used in the input stages to be described in the next article.

Apart from the seven signals M, Σ, jΣ 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 22kΩ to terminate the resistor matrix, and which except in the SQ mode give a phase at their output 90° in advance of their inputs at a frequency of 400Hz, thereby ensuring virtually identical phase responses. The amplitude gains (ignoring phase) of these filters at frequencies much less than 400Hz and at frequencies much more than 400Hz 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 400Hz satisfy the following

Signal path	J, H, RM, U		M, S, SS, B, SPARE		SQ
	l.f.	h.f.	l.f.	-h.f.	both
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 ±2%, except when marked with asterisk when ±5% is adequate.

R$_{1-4}$	200k	R$_{92}$	91k
R$_{5-10}$	68k	R$_{93}$	100k
R$_{11-13}$	33k	R$_{94*}$	3.9k
R$_{14}$	200k	R$_{95}$	47k
R$_{15-17}$	33k	R$_{96*}$	3.3k
R$_{18}$	27k	R$_{97}$	150k
R$_{19}$	100k	R$_{98*}$	8.2k
R$_{20}$	150k	R$_{99}$	33k
R$_{21}$	110k	R$_{100,101}$	82k
R$_{22*}$	1M	R$_{102}$	68k
R$_{23}$	270k	R$_{103}$	100k
R$_{24}$	6.8k	R$_{104}$	56k
R$_{25*}$	39Ω	R$_{105,106}$	68k
R$_{26}$	2.7k	R$_{107}$	330k
R$_{27,30}$	1.6k	R$_{108}$	56k
R$_{28,29}$	see text	R$_{109*}$	2.2M
R$_{31,33}$	6.8k	R$_{110*}$	1M
R$_{32,34}$	10k	R$_{111}$	56k
R$_{35,36}$	120k	R$_{112*}$	4.7k
R$_{37,38}$	27k	R$_{113,114}$	82k
R$_{39-42}$	47k	R$_{115}$	68k
R$_{143}$	22k	R$_{116*}$	240k
R$_{43*}$	3.3k	R$_{117*}$	2.2M
R$_{44}$	27k	R$_{118*}$	390k
R$_{45}$	12k	R$_{119}$	22k
R$_{46}$	150k	R$_{120*}$	1.8M
R$_{47,48}$	22k	R$_{121,122}$	220k
R$_{49,56}$	82k	R$_{123}$	180k
R$_{50,57}$	39k	R$_{149-158*}$	390k
R$_{51,58*}$	680k		
R$_{52,59*}$	56Ω		

Two off of the following

R$_{53,60*}$	1k
R$_{64,61}$	22k
R$_{65,62}$	22k
R$_{63}$	82k
R$_{64}$	18k
R$_{65*}$	820k
R$_{66*}$	1.1k
R$_{67*}$	1.8k
R$_{68}$	22k
R$_{69}$	100k
R$_{70*}$	910k
R$_{71}$	100k
R$_{72*}$	2.2m
R$_{73}$	430k
R$_{74*}$	8.2k
R$_{76-78}$	270k
R$_{79}$	100k
R$_{80*}$	5.6k
R$_{81}$	91k
R$_{82}$	120k
R$_{83}$	110k
R$_{84}$	270k
R$_{85}$	56k
R$_{86*}$	360k
R$_{87}$	100k
R$_{88}$	68k
R$_{89}$	120k
R$_{90*}$	18k
R$_{91}$	56k

Two off of the following

R$_{124,125}$	47k
R$_{126*}$	1.8k
R$_{127,132}$	22k
R$_{129,133}$	47k
R$_{128*}$	3.9k
R$_{130,134}$	47k
R$_{131,135}$	12k
R$_{136*}$	2.0k
R$_{137}$	33k
R$_{138,144*}$	5.6k
R$_{139,145}$	100k
R$_{140}$	39k
R$_{141-147}$	22k
R$_{142*}$	3.3k
R$_{143}$	47k
R$_{148}$	56k

(PHASE SHIFT)

Two off of the following:-

R168	10K
R169	22K
R170	47K
R171	150K
R172	470K
R173-5	10K
R176*	1K8
R177*	5K6

Capacitors

Tolerance ±2 or 2½%, except when marked with asterisk when ±10% is adequate.

C$_{1-2*}$	680n		
C$_3$	1n		
C$_{4-8}$	10n		
C$_9$	15n		
C$_{18-29}$	47μ	10V low tolerance	

Two off of the following

C$_{10,11}$	470p	C36	820pF	2½%
C$_{12,13}$	100n	C37	120pF	2½%
C$_{14,15}$	27n	C38	1300pF	2½%
C$_{16,17}$	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)

$$W'' = 0.73\Sigma$$
$$X'' = -0.73j\Sigma$$
$$Y'' = 0.73\Delta - 0.73j\Delta$$

R160	22K	
R161	10K	
R162	150K	ITEMS
R163,164	22K	CIRCLED
R165	110K	
R166*	4K7	ARE
R167	68K	
R178,179	27K	AMENDED
R180-3	22K	
R184	6K8	TYPOS
R185	33K	
R186	6K8	
R187	33K	
R188	24K	
R189*	1K	
R190	10K	
R191	39K	
R192,193	82K	
R194*	2M2	
R195*	100Ω	
R196,197	22K	
R198	10K	
R199*	100Ω	
R200,201*	1K	
R202 *	1K8	
C30,31	100uF	3V tant
C32-4	10uF	16V
C35	0.1uF	
C39,41	10uF	
C40	1nF	disc
C42	1000uF	25V
C43,44	100uF	3V tant
Cd (12 off)	47nF	disc

SEMICONDUCTORS

D1-4	1N4001,2,3,4.
T1 (2)	ZTXA21
T2 (2)	ZTXA11
T3	ZTXA21
IC Reg	L131,TDA1415
ICa	741
ICc-j	RC4136; ICb LM381
LED (2)	TL209

SWITCHES and POTS

VR1,2	5K LIN single
VR3	5K LOG.LOG
10bank	I/P selector
5 bank	layout ..
Mains	c/w bracket
Rotary	I/P level

HARDWARE

Main Chassis
Front panel c/w 2 angle brackets.
Transformer
Fuse & Holder
Knobs (3)
5 pin DIN skts (4)
3 core Mains cable and bush
Solder tag, PC pins (4).
Misc screws and nuts
Solid Mahogany Cabinet

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$_{35-38}$ 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$_B$ or C$_R$ described in part 1, and which alter the matrixing coefficients for the L$_B$, L$_F$, R$_F$, R$_B$ 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$_1$ 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 5kΩ within 2%, but this would be extremely expensive and not very practical. Thus two padding resistors R$_{28}$ and R$_{29}$ 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$_1$ has a total resistance of 6.2kΩ. If precision measurement of resistors is not available (and if the circuit is not being built from a kit with R$_{28}$ and R$_{29}$ provided to match VR$_1$), then choose R$_{28}$ and R$_{29}$ to have values identical to within 5% such that the total measured resistance of R$_{30}$, R$_{29}$, VR$_1$, R$_{28}$ and R$_{27}$ 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

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.

Correction. P.10.
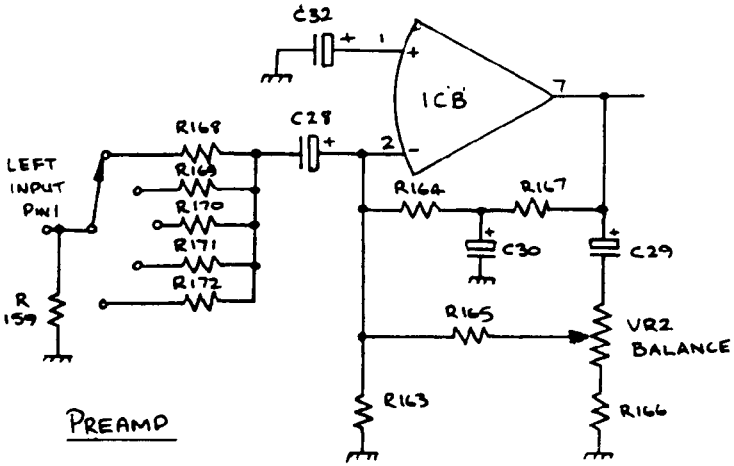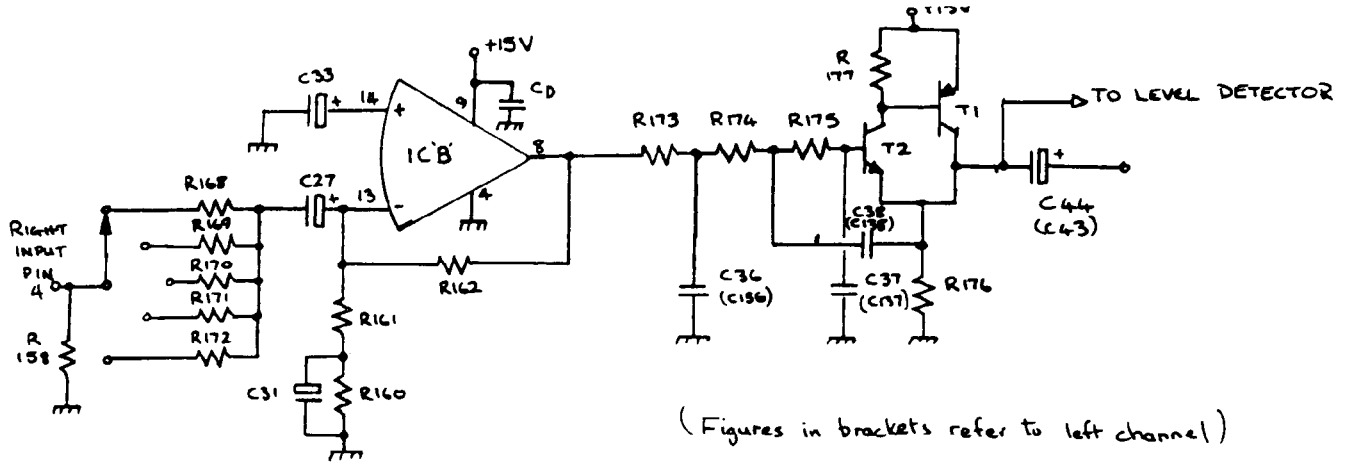
(R7,8 output ICs — pins are 14, 13, 12 and not 1, 2, 3 ———— ·· correct)

(Figures in brackets refer to left channel)

FILTER AS ABOVE

BESSEL FILTER

PREAMP

VOLUME CONTROL

TO PHASE SHIFTERS

LEVEL DETECTOR

MAINS
INPUT

STRAIN RELIEF BUSH

TO FRONT PANEL
POWER ON' LED

FUSE
HOLDER

240V

NST.LI

N   E

BRACKET

LED2
K   A

'LEVEL' LED

MAIN P.C.B.

BRAC

FRONT PANEL

MECHANICAL LAYOUT

Position
on wire

Squeeze
pliers

Push into
hole

FITTING THE STRAIN
RELIEF BUSH

metal

O/P Gnd I/P

IC REG

RED
PLASTIC

K   A

LED

K   A

CATHODE

D1-4

c
b
a

ZTX
underside

SOCKETS

3   1

5   4

2

viewed from
back

3 O   2 O   O 1

5 O   O 4

track view

240V
~

ON/OFF

1A
Fuse

D3

D1

17.2V

D4

D2

A   LED
K

R
202

+ C42

IN   IC
     REG
OUT

COMM

+ C39

C
34

R197

R
196

IC 'C'

C40

R198

R195

+15V

+7.5V

Cp

Cp

0V
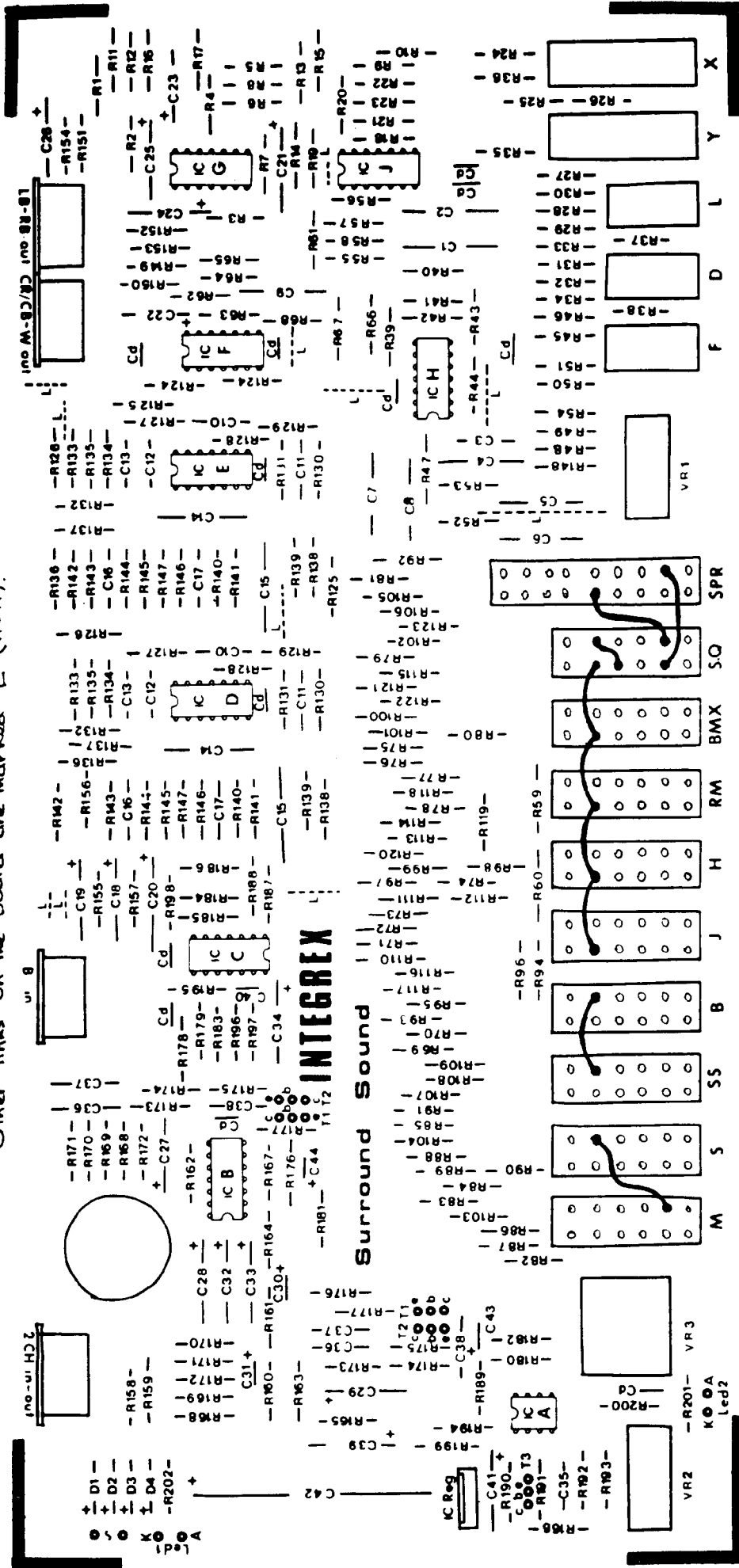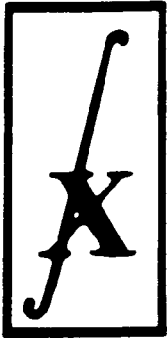
COMPONENT LAYOUT

Note the links which must be made on top of the switch bank

Other links on the board are marked 'L' (11 off).

INTEGREX

Surround Sound

INTEGREX

45J - AMBISONIC DECODER KIT

SPECIAL NOTES

Please check with the parts list that all components have been shipped and are in good order. Any missing or damaged parts should be reported immediately to the address given below.

CONSTRUCTION

4 pins are supplied with the kit and these should be fitted from the track side, tapped in gently with a hammer and soldered. Two fit at the transformer input (marked N) and the other two at LED 1 A & K outputs.
Assembly is simplified if the components are fitted in the following order:-
Resistors - take care not to confuse the red 2% tolerance band with the numerical value
Capacitors - observe polarity of electrolytics
Links - there are 11 links on the board marked (L) - use the wire cut off the resistors and capacitors.
Semiconductors - make sure that they are inserted the correct way round (see pin connection diagram).
DIN sockets - make sure that they are vertical and in line for appearance sake.
Switches - it is advisable to check the switch banks function correctly before insertion as they are difficult to remove once fitted. Take care to push them fully into the board and ensure that they fit squarely otherwise they will not align with the front panel. Having fitted the switch banks connect the links shown on the component layout diagram.
Potentiometers - align the spindles by using the front panel as a template.
LED 2 - with the front panel over the push buttons fit LED 2 and bend the leads so that the LED fits in the hole marked 'Level' on the front panel.

Crop the leads on the track side to avoid touching the chassis. Insert a thin piece of cardboard between board and chassis and fix the PCB in position on the chassis with 4 X6BA screws.

Fix in position the transformer, fuseholder, mains switch and bracket (the latter two using self tapping screws). A solder tag should be fitted under one of the self tappers to take the mains earth lead. Wire up as shown on the mechanical layout diagram. Fit the strain relief bush to the mains cable.

The small angle brackets are now fitted to the front panel with countersunk screws. These brackets are adjustable on the chassis so that the front panel may be aligned with the wooden sleeve cutout.

USING THE UNIT

Connect the signals to be decoded into the 2CH in-out socket (use the tape/source socket on your amp/preamp to do this). The LF,RF signals will be routed back to the same amplifier to provide the left and right front speaker outputs. Other outputs LB, RB, W & CR/CB are available on separate sockets for connection to separate power amps (again via the tape/source socket if available).

To set up the pre-amp gain apply a mono signal and starting with the rotary switch fully anti clockwise rotate clockwise until the 'level' LED just begins to flash. This is the level for optimum dynamic range.

Refer to the article for notes on the use of the control.

SERVICING

In the event that the Kit fails to operate, servicing will be carried out by INTEGREX LIMITED.

The printed circuit board only should be returned.

Please pack adequately - damage in transit (for servicing) is solely the responsibility of the sender.

P.C. boards should be accompanied by a remittance, payable to INTEGREX LIMITED.

of £5.00 inc. VAT @ 12½%
and return pp.

P.C. boards will only be serviced if they have been assembled in accordance with good practice. The decision of Integrex Limited in this matter is final.

In the event of Major servicing being required, the owner will be notified.

Integrex Limited,
Portwood Industrial Estate,
Church Gresley,
BURTON ON TRENT,
Staffordshire.
DF11 9PT.

PERFORMANCE

Input level	20mV → 1V rms, switched internally
Output level	300mV rms
Distortion	⟵ 0.03%
Hum	−85dB ref 300mV
Noise	−74dB ref 300mV
Overload	+16.5dB
Dynamic Range	> 90dB

KIT FEATURES

No test equipment is required. With the exception of the mains transformer, fuse, switch and LED, ALL components are printed circuit board mounted for ease and accuracy of assembly. The P.C. board is fibreglass, drilled, tinned and printed with component locations. Comprehensive instructions are included. (The kit is complete to "the last nut and bolt").

SIZE

The mahogany cabinet size is:

Width	394	mm	15.5 in
Height	69	mm	2.7 in
Depth	173	mm	6.8 in

NOTE

In the event of a significant change in the encoding format of any 2 channel system (e.g. Matrix H), the constructor will be able to obtain the necessary components from Integrex. (normally only a few resistor changes will be necessary).

Ready built units - are available from Integrex.

NRDC AMBISONIC DECODER

Notes on connection to other apparatus.

Since the instructions recommend the use of the TAPE/SOURCE facility on modern amplifirs/receivers, difficulty will be experienced when the operator wishes to use a tape recorder together with the above combination.

The procedure for DIN connections and 4 amps/4 speakers arrangements is as follows:-

1. Connect L and R amplifier/receiver outputs (pins 3 and 5 on socket marked Tape/Source) to the tape recorder input.

2. Connect tape recorder L and R outputs (pins 1 and 4) to the Ambisonic L and R inputs (pins 1 and 4 on socket marked 2CH in-out).

3. Connect Ambisonic LF and RF outputs (pins 3 and 5 on same socket) to the amplifier/receiver L and R input (pins 1 and 4 on tape/source socket).

4. Connect Ambisonic LB and RB outputs (pins 3 and 5 on socket marked LB and RB out) to the second amplifier L and R input (pins 1 and 4) on the tape/source socket.

If the tape recorder does not have an input monitoring facility it will be necessary to switch out the recorder and re-route the amp tape output to the Ambisonic decoder.

Note

Naturally some amplifiers have high level input sockets additional to the tape source socket. The above comments do not of course apply to this type of amplifier.

All Integrex products use DIN standard sockets. To interface with amplifiers, recorders, etc. using RCA phono sockets, DIN to phono leads will be necessary.

GAIN CONTROL

Correct signal level is indicated by a flashing 'Level' LED on the front panel. Gain is set by an internal rotary switch. To adjust, undo underside cabinet retaining screw and push chassis forward out of the cabinet. The switch spindle is vertical and close to the 2CH in-out socket. Rotate as necessary. Be very careful not to touch the mains power supply section.

1.1.78