Ambisonics – An operational insight

Chris Daubney (IBA)

Investigations made by the IBA into the various proposals for surround sound systems were in two parts, theoretical and practical. This article describes how it was necessary for the IBA to make recordings of suitable test material for the practical investigations. Two vehicles were equipped as a mobile surround sound recording control unit. Several constraints which affected the design of the equipment are explained, and a full description is given of the equipment used, including the comprehensive sound mixing desk. The associated microphone and monitoring techniques also are discussed.

ABOUT three years ago, the IBA decided to undertake a review of the various proposals for quadraphony which were then current; it did this, not because quadraphony was an immediate and vital neccessity for the infant ILR network, but because it wished to guard against international adoption of any standard which was less than optimal and which, if adopted, would be very difficult to change.

On a point of nomenclature, it soon became apparent that quadraphony was not the right word. There is no necessary connection between the achievement of reproduced sound coming from all around the listener and the use of four, and only four, channels of information to carry the sound or for only four loudspeakers to reproduce it. In the absence of a more evocative and accurate name, the IBA decided to style it 'surround sound'.

In essence, the IBA review was of two parts:

- a theoretical one, to determine how each system provided the clues for human ears and brains to perceive the surround effect, and how well the compatible stereophonic and monophonic signals were derived; and
- a practical one, to see how each system fared with all the different styles of programme material and production technique.

Early in the process of making the practical review it became obvious that, with the commercial material then available, the practical part was almost impossible because no programme material, recorded by the same artists in the same environment, but in the different systems, was obtainable. As this comparison was of fundamental importance, the IBA decided to undertake the necessary recordings. While a lot was achieved by using classical 'overall' mic techniques, only very superficial work was possible in the field of popular music. This was because the facilities of the mixing equipment were very limited due to the time available for completing the work. However, the investigation would not have been complete without a detailed look at the problems of popular music recording, as well as a further look at drama, documentaries and 'classical' music. Therefore, the IBA decided to build a more extensive and permanent mobile recording facility.

This article reports on the design of the mobile unit and on subsequent experience from the operational viewpoint, also giving some details of the engineering designs involved.

Design philosophy

Several constraints affected the design of the equipment:

- surround sound is a developing subject, and changes in parts of the system were likely;
- the main involvement of the IBA was in conducting an investigation into systems;
- by the nature and structure of ILR, the IBA transmitters are supplied with programmes by programme contractors; transmissions of suitable programmes, made by use of different surround sound techniques, are possible only with the help and co-operation of the contractors;
- as the programme contractors are separate companies, each has its own philosophy and own experience of OB equipment design and layout;
- the time for design and construction and the availability of suitable vehicles was limited.

The keynote of the design, therefore, was flexibility.

During the early theoretical considerations, it emerged that the system known as 'ambisonics' sponsored by the British National Research Development Corporation, covered the subject much more broadly than did most of its competitors; the various features were evolved from a theory of hearing, and the design covered the entire chain from mic to loudspeaker.

As the ambisonic design appeared to offer a system founded on a theory rather than on empirical guess work, the IBA decided to pursue the majority of its work along the ambisonic line.

In essence, ambisonic technology may be summarised as follows.

For precise central decoding of surround sound in one (horizontal) plane, three discrete channels of information are required. (One additional channel would allow such decoding to include height.) There is no constraint on the number of mics which can be used, nor on the number of speakers, provided that these are more than three. The system is divided into four sections, each with its own format:

A-format: microphones—covered later in this article;
B-format: studio equipment;
C-format: signal transmission;
D-format: decoding and loudspeaker signals.

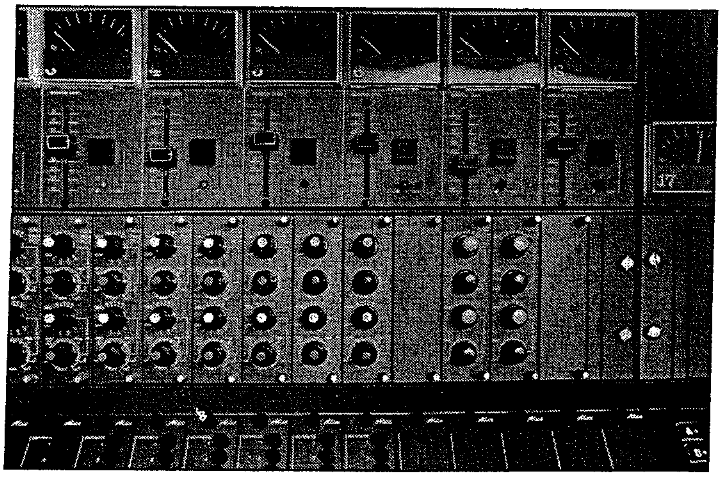
The sound mixing desk operates primarily in B-format, though C-format signals are derived in it for transmission, and the monitoring decoding is in D-format.

In B-format, the signals are:

W—the pressure signals from the mics irrespective of direction;
X—the front-to-back 'figure-of-eight' velocity component signals from the mics;
Y—the left-to-right 'figure-of-eight' velocity component signals from the mics;
Z—the top-to-bottom 'figure-of-eight' velocity component signals from the mics.

The Z component is omitted for

Beneath a number of the multitrack PPMs are housed two of the special surround-sound facilities. To the left are shown some of the combined panning (through 360°) and spreading (diffusing) controls; each module contains two separate panpots and their associated panpot mixture selector. To the right of the panpots are two modules housing four Waltz controls. These allow for a previously balanced surround-sound field mic to be rotated about the listener by as much as 360°.



one-plane (horizontal) surround sound. This arrangement of signals is chosen because it is rugged and any interchannel errors manifest themselves in the least unacceptable form as symmetrical image displacements.

C-format signals are a linear encoding of the final mixed B-format signals:

L—left signal for stereophonic compatibility;
R—right signal for stereophonic compatibility;
T—third channel to allow more accurate horizontal decoding;
Q—fourth channel to carry height information.

The Q component is omitted for one-plane (horizontal) surround sound.

In principle, D-format signals can be derived for any number of loudspeakers greater than three, from either B- or C-format inputs; the speakers are not constrained to being in a square layout. In the IBA mobile unit, four speakers are used, but decoding to six and eight speakers has been tried for certain other experiments.

With the above in mind, it was decided to use the modular 'black box' approach, ie, to provide as many different sorts of facilities in as large quantities as possible, preferably with each having individual inputs and outputs, so that the various styles of sound operation could be explored with the same equipment.

Design detail

Two vehicles were used; one housing the mixing and monitoring equipment and the other the multitrack recorders. The second van also acted as a cable tender.

Single channel sources

Ultimately, for horizontal surround sound, every source has to 'acquire' the three channels of information which characterise its volume and location. Unless, as in the case of the *Soundfield* mic, the output signal from the mic is already in the 3-channel form, all mic sources have to be processed to derive them. The same is true of synthesised sources.

Monophonic channels

There seems no reason to suppose that, with the advent of surround sound, the popular music world will suddenly change its technique of multimiking, etc. Therefore, the desks for such will have to contain a large number of conventional mic channels which can amplify and control the individual sources. They must also feed multitrack tape recorders and artificial reverb devices, feed or insert other processing devices and provide controlled line level outputs of each source for panning and mixing. In the IBA mobile, there was room for only 20 monophonic channels (and four monophonic groups) after space was allocated for all the other facilities. This provides sufficient channels to cope with small 'pop' sessions and is thought to be adequate to allow exploration of the majority of different styles of programming—albeit in a small way.

The line level outputs of the monophonic sources need to be panned so that the sound appears to come from the desired direction in the soundfield. This is effected by use of special panpots.

Panpots

Essentially, a mono signal is fed into a panpot and the three signals (W, X and Y) are derived therein. W is directly proportional to the amplitude; the relative gains and phases of X and Y provide the directional information. Specifically, if the mono input signal is M, then the B-format signals have levels as follows:

$W = M$
$X = 2\,M \cos\theta$
$Y = 2\,M \sin\theta$

where θ is the azimuth of the desired sound source direction measured anti-clockwise from centre front. Thus, as the signal is panned around the circle, its position is determined by the relative values of W, X and Y. In practical terms it would be ideal to provide 360° panning on one control. However, in the time available, the required sine/cosine pots to achieve this could not be obtained. So, in the IBA desk, it is necessary to select on a switch the quadrant required and then to pan on a separate control to the precise position within the quadrant. For flexibility, it would have been useful if the input and output of each panpot could have been available separately; however, with 20 inputs, there are 60 outputs and the jackfield bay was already 34 rows high.

Thus, bearing in mind that at some stage in the proceedings the 20 sets of three outputs had to be combined as part of the final mix, it seemed prudent to achieve some of the mixing in the panpot modules. As a consequence, every panpot can be routed to one of four panpot mixtures and the outputs of the panpots are available only via these mixtures. However, it is possible to monitor the operation of each individual panpot.

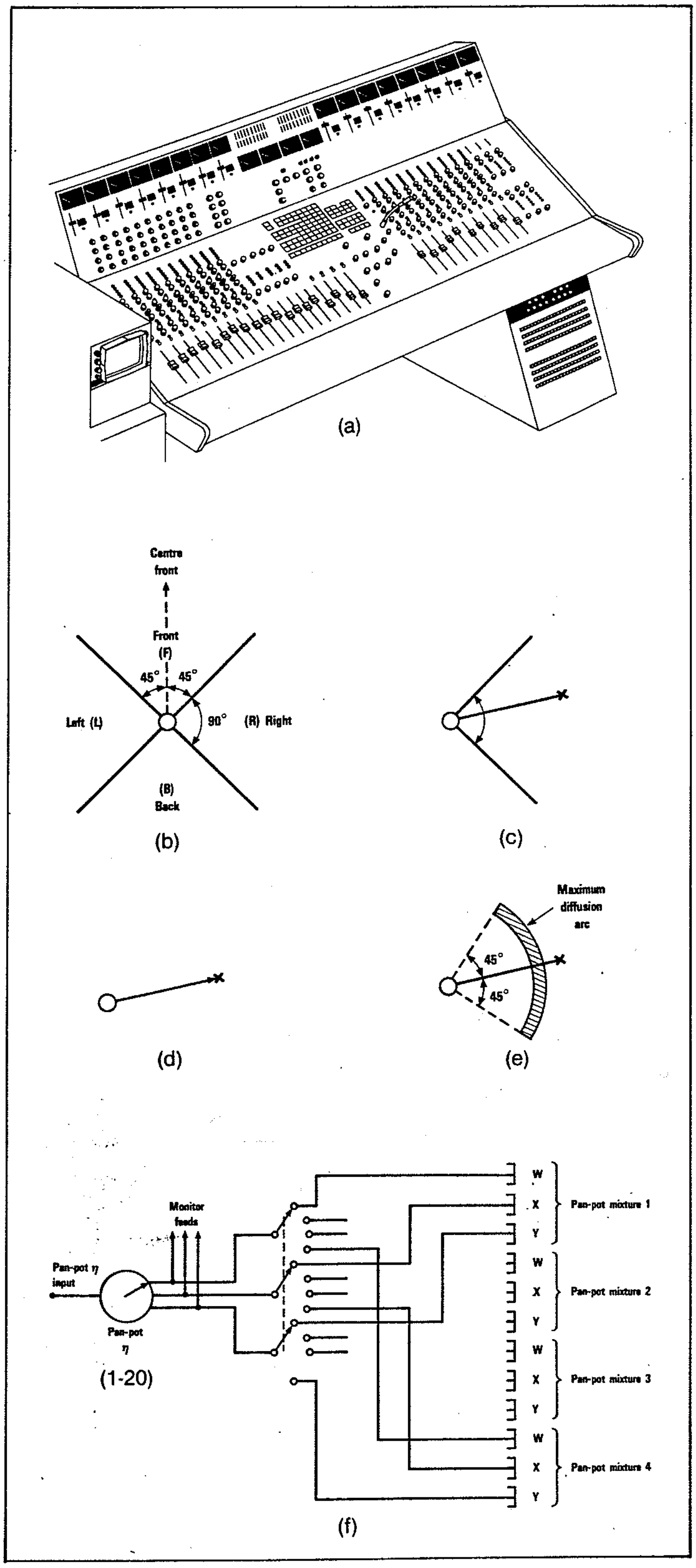
When a surround sound panpot is working normally, the information contained in the three outputs W, X and Y, is sufficient to allow a proper decoding system to produce an accurately localised image. Later in this article will be found a brief discussion on the possible need for deliberately being able to delocalise an image in a controlled way—over a limited range. This facility is available on the panpots.

By use of appropriate networks, it

54 ▶

The main items of equipment are positioned as in (a). The layout was designed in modular form to provide the maximum variety of facilities, in as large quantities as possible, within the available space. The interconnections required for each different style of sound operation are effected on the jackfield.

In the IBA desk, because of a shortage in availability of the requisite components, the panpots are not continuously variable through 360°, but quadrant designation is used first (b); followed by the precise pan (c). Once the required position has been established (d) the sound can be diffused over an angle of and below ±45° on either side (e). Finally, the output of each panpot is available only via one of four panpot mixtures (f).



is possible to spread a panned image by a maximum of ±45° on either side of the precise position. In essence, the diffusion is obtained by panning different parts of the frequency domain to different positions within the chosen amount of spread. The positions are such that a frequency sweep over the usual broadcast audio frequency range (40Hz to 15kHz) would cause the image to swing back and forth six times.

Spreaders

Later in this article the need for delocalised sources is discussed. The achievement of such an effect is possible in the IBA desk by the use of spreaders; these are an extension of the concept of the spread on the panpots. In the current mono spreader design, there are no external controls, but simply one (mono) input and the three (surround) outputs. The signal swings, not merely over a maximum of ±45°, but right through 360° six times during a sweep of the normal audio frequency range.

An extension of the mono spreader is the stereo version, in which each of the two inputs is rotated in its own network, but the two are then summed to one 3-channel output. At any frequency, the circuitry is arranged so that the two images corresponding to that frequency are 180° apart across the circle. This stereo version was conceived to cater for modern artificial reverb systems which have stereo outputs.

Consideration has been given to a further improvement in the illusion, by using a four input version, but this has not yet been tried.

Three channel sources

Once every source has 'acquired' its three channels of information, consideration can be given to generating the final mixed 3-channel output.

Waltz controls

Operationally, there are many occasions when, having generated a number of 3-channel sources or submixtures, it is desired to rotate one or more of these sets of signals relative to another set to overlay or interleave the respective images. The surround sound extension of the stereo 'offset' facility is the waltz control.

One set of W, X and Y signals is fed into such controls, and from those controls is derived another set. In fact, the W signal, representing the amplitude of the signal irrespective of direction, is unaffected; the X and Y components contain the directional clues, and these are the ones which are changed as the source is rotated. In principle, this function would be provided on one control, such that one rotation of the control provided a complete rotation of the images through 360°. Because of the lack of suitable sine/cosine pots, this facility in the IBA desk is provided by choosing the number of quadrants through which it is required to rotate the images on one control, and then by 'fine tuning' the rotation on a continuously variable control. When the waltz control is set to centre front, there is no rotation of the images; all other shifts are derived relative to this.

Three channel surround groups

Surround group faders are essential to controlling the various 3-channel sources used in deriving the final mix. The desk contains five such group controls, each having a 3-channel input and output. A balance attenuator (in 6dB steps from +18dB gain to −12dB gain) is provided on each channel, but is ganged to one control on each group, and continuous level control is by similar means of accurately matched and ganged VCAs.

Since the *Soundfield* microphone (see below) has high level outputs and contains a Z component, it is convenient to route the outputs of this mic via one of these surround groups. In order to preserve the Z signal (when operationally convenient) at the same level as the W, X and Y signals, one of the surround groups (group A) has four accurately matched channels.

Jackfield and matrix

Because flexibility is the keynote, as many inputs and outputs as possible are made accessible. Apart from an insertion jackfield in the desk pedestal, the great majority of these access points are provided on the main jackfield bay. Because of the need to feed tape recorders etc, as well as the main programme chains, virtually every facility has A and B outputs of every channel. Apart from the designated sources and destinations to and from the matrix, nothing is permanently connected, and the connections with the recording van also are on the jackfield.

The electrical mixing of the signals is effected partially in the panpot mixtures (as previously described) and then in a matrix which is housed in the jackfield bay. The designated sources and destinations of the matrix are shown in **Table 1**.

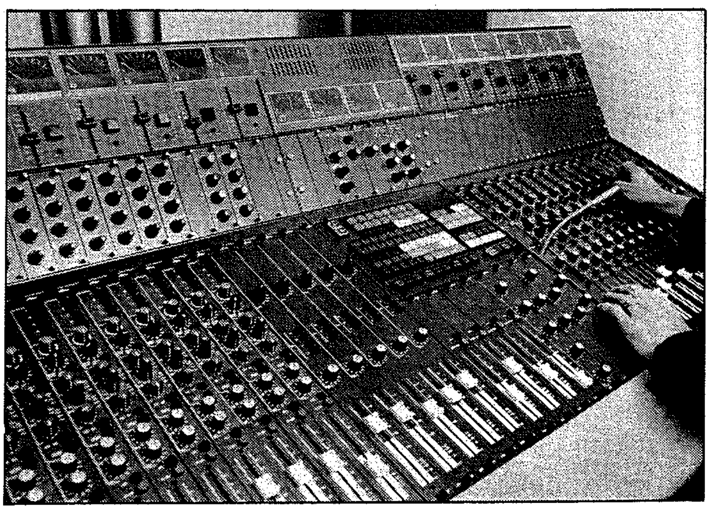
It is possible, therefore, to use the matrix to route signals through the

56 ▶

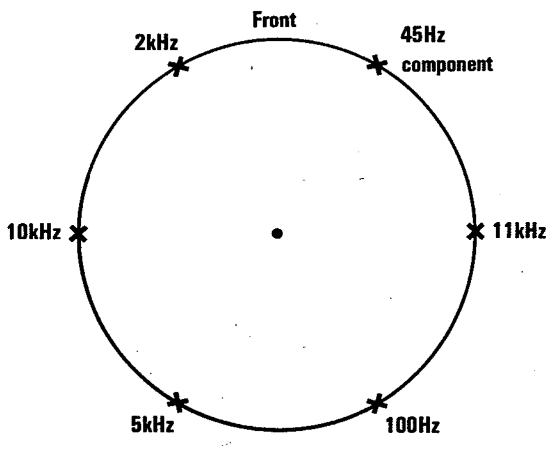
TABLE 1

Source		Signal
1. Panpot mix	1	W
2. Panpot mix	1	X
3. Panpot mix	1	Y
4. Panpot mix	2	W
5. Panpot mix	2	X
6. Panpot mix	2	Y
7. Panpot mix	3	W
8. Panpot mix	3	X
9. Panpot mix	3	Y
10. Panpot mix	4	W
11. Panpot mix	4	X
12. Panpot mix	4	Y
13. Echo (Stereo)		W
14. Echo (Stereo)		X
15. Echo (Stereo)		Y
16. Mono Spreader	1	W
17. Mono Spreader	1	X
18. Mono Spreader	1	Y
19. Mono Spreader	2	W
20. Mono Spreader	2	X
21. Mono Spreader	2	Y
22. Mono Spreader	3	W
23. Mono Spreader	3	X
24. Mono Spreader	3	Y
25. Surround Group	A	W
26. Surround Group	A	X
27. Surround Group	A	Y
28. Surround Group	B	W
29. Surround Group	B	X
30. Surround Group	B	Y
31. Surround Group	C	W
32. Surround Group	C	X
33. Surround Group	C	Y
34. Surround Group	D	W
35. Surround Group	D	X
36. Surround Group	D	Y
37. Surround Group	E	W
38. Surround Group	E	X
39. Surround Group	E	Y

Destination		Signal
1. Surround Group	A	W
2. Surround Group	A	X
3. Surround Group	A	Y
4. Surround Group	B	W
5. Surround Group	B	X
6. Surround Group	B	Y
7. Surround Group	C	W
8. Surround Group	C	X
9. Surround Group	C	Y
10. Surround Group	D	W
11. Surround Group	D	X
12. Surround Group	D	Y
13. Surround Group	E	W
14. Surround Group	E	X
15. Surround Group	E	Y
16. Encoder		W
17. Encoder		X
18. Encoder		Y



On the right-hand side of the nearer set of monophonic channels are the special 3-channel groups. These groups have balance attenuators at the top of the modules which provide equal amounts of coarse attenuation in each of the three channels; the continuously variable control is achieved with the faders driving matched VCAs.



Operationally, there is often a need to delocalise a source of sound—eg, a sound effect such as that of falling rain. The diffusion of the sound is achieved by positioning different parts of the frequency spectrum at different points around the 'circle'; a frequency sweep between 40Hz and 15kHz would cause the image to rotate six times around the 'circle'.

surround groups in a variety of ways to suit operational requirements, and the waltz controls can be inserted in many places.

One of the destinations from the matrix is the B-to-C format encoder.

Encoding, decoding and monitoring

Expressed briefly, the B-format final mix is encoded in the desk and is available on the jackfield bay for line sending in the transmission format.

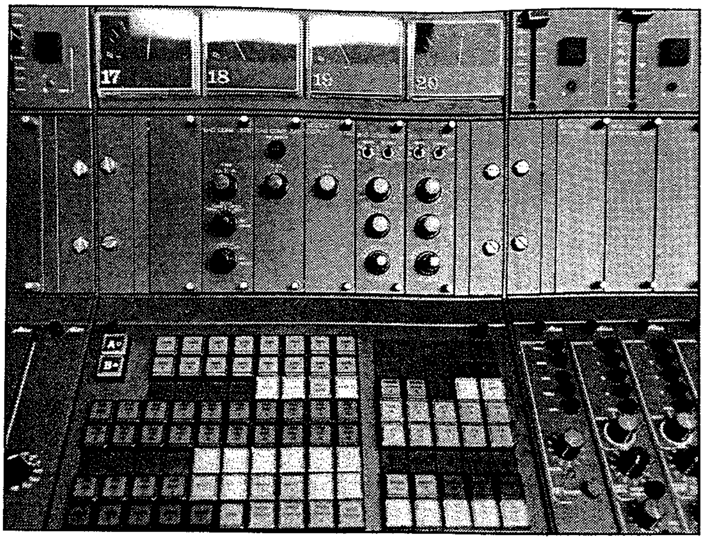
It is absolutely vital in any sound control system, particularly in an experimental one such as this, that the monitoring is the most comprehensive possible. This makes necessary the monitoring panel on the desk and is an additional reason for the extensive jackfield facilities.

The visual monitoring facilities consist of 16 PPMs to measure either the record or replay signals from the 16-track tape recorders and PPMs 17 to 20, the feeds to which are described below. The subjective monitoring is from decoded signals fed to four high quality professional monitoring loudspeakers. PPMs 17 to 19 monitor either the B-format signals, C-format signals or, with PPM 20, the D-format signals which feed the loudspeakers. The selection of format for the PPM is made by means of the buttons in the bottom right hand corner of the panel. Unless otherwise latched, the overriding position is C-format because that contains the transmission signals. PPM 20, when not monitoring one of the loudspeaker feeds, can be made to indicate the level of any of the sources on the right hand side of monitoring panel.

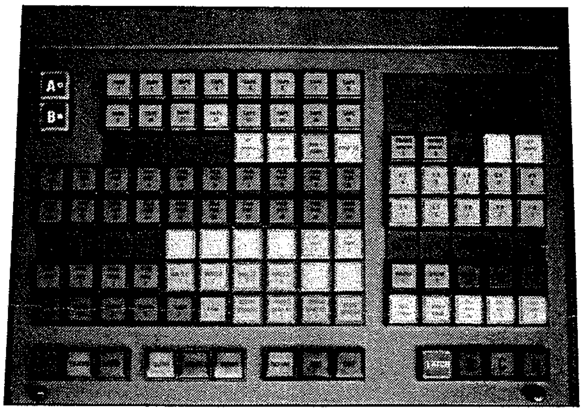
Since most of the signals to be monitored are in B-format, there is a 'B-to-D' decoder to feed the loudspeakers. Any C-format signals (transmission output and radio check off-air input) are first converted from C-to-B (this can be done only if three or more transmission channels are in use). All monophonic signals are artificially encoded for monitoring as centre front signals.

One of the advantages of the ambisonic system is that the decoding arrangements are not related to either the mic technique or the number of channels of information. For central listening, all the information necessary for accurate 'horizontal' surround decoding is contained in the three channels; the decoding can be designed for any number of loudspeakers and can take account of where the speakers are positioned. Recognising that many listening rooms will not allow a completely symmetrical layout of loudspeakers, the decoder provides layout and distance controls which respectively compensate for the speaker positioning and distance from the listener.

In the IBA desk, two decoders are provided, one for the vehicle itself and one for an external set of loudspeakers. The decoders are for four speakers, and the layout compensation allows for a 2:1 rectangle one way to a similar one the other way, with distance compensation (for the effect of different wavefronts) down to 1m.



The transcoding from the mixing format to the transmission format is effected in the module marked B-C. To the right of this is a set of decoders which allow vectorial display of the sound field on an oscilloscope and aural indication on two different sets of loudspeakers, one of which is set in the vehicle. The PPMs can be switched to either B-, C- or D-format signals.



In any sound control system, and particularly in an experimental one such as this, it is vital that the monitoring system shall be very comprehensive. The panel below allows monitoring access to all facilities for objective assessment on the PPMs and for subjective assessment on the loudspeakers.

Initial operational experience

General

After two decades of experience, engineers are still developing new techniques for stereo reproduction; hence, no-one could hope to do more that scratch the surface of the problem of surround sound production in a single year. What the IBA has been able to do is to tap the considerable and diverse talent available to it through the keenness and interest of the staff of ILR stations.

Engineers, sound balancers, music producers, musicians and hi-fi listeners have all contributed useful comments and expertise. Much of the musical world has been examined, though considerable areas of music, documentary material and drama remain unexplored. Many lessons have been learnt, not least of which was to concentrate on simple, non-prestigious programmes in controllable and repeatable environments. By far the most useful experience has been gained from small studio sessions over which a large measure of control could be exercised—mic positions could easily be changed, separation between sources could be controlled, and repeats were possible. The small size of the facilities put its own constraint on what was possible. However, as many 'big' sessions are only extended versions of the sort which IBA facilities would handle, there was never any point in trying a 'big' session when just as much could be learnt from a smaller one.

The first hurdle to overcome was that of learning to accept reproduced sound coming from all directions. In everyday life one accepts it without question, but artificial reproduction seemed quite different. The IBA set out to find the system which offered perfect naturalness—the counterpart of real high fidelity—and not merely a novel effect which would quickly become tiresome. The developing of aural recognition, and of rightly critical judgement, have also been necessary. It has been obvious, on those few occasions when everything has worked satisfactorily, that a remarkable naturalness of sound is possible. Trying to discover why those few occasions were successful, and why many others were not so, requires working analytically through what has been heard. To this end, use of the multitrack recorders and small sessions has made possible the remixing, in various ways, of the same simple material—sometimes with only very small variations—as a means of making such analyses.

It is a truism that one has to 'learn to listen' when dealing with surround sound. It appears that, although mono and stereo reproductions are quite different from normal everyday sounds, the human ear and brain have learned to accept them. In consequence, when presented with surround sound naturally reproduced, the brain registers surprise.

Perhaps the greatest apparent problem concerns the presentation of the material. One school of thought believes, presumably from an extension of what has been the norm in stereophony, that multimic techniques are, to a greater or lesser extent, essential in all audio material. The other school believes that, with the ability to create around the listener that naturalness of sound which is impossible in stereo, 'classical' mic techniques, use of a point cluster of mics to pick up the sound field at the wanted

58 ▶

position, is all that is needed because the listener's brain will do the rest.

Soundfield microphones

'Classical' stereo mic techniques have involved the use of a 'stereo pair' to provide the fundamental sound stage; the counterpart in surround sound is the *Soundfield* mic.

This mic consists of a cluster of four sub-cardioid mics, on the faces of a tetrahedron, and so orientated that, when projected equally on to one plane, the four capsules point (approximately) in the conventional directions of LB, LF, RF and RB. The capsules have acquired these four designations and, in ambisonic terminology, this is the A-format. However, as the capsules are on the faces of a tetrahedron, three-dimensional sound pick-up is possible. The capsules are spaced as closely as possible; but, in order to attempt (at least to a first order) to produce a true representation of the soundfield at the centre of the tetrahedron, the amplified outputs of the capsules are equalised to take account of their physical spacing and characteristics. At the same time the A-to-B-format conversion is made.

The sound from the mic is critically dependent on the capsule matching, especially as regards the precision of image localisation. One of the acid tests of any surround sound system, and of this design of mic, is the 'walk around' *[in which a person walks round the mic (or uses a surround sound panpot) calling out cardinal directions at various points around the mic, referenced to centre-front—generally referred to as 'North'—Ed]*. The evenness and naturalness of the reproduced sound from a correctly aligned array is a most useful reference against which to make other judgments.

In the classical music use of the soundfield, with the orchestra, etc conventionally arranged, the temptation to be novel and to place the listener in the midst of the orchestra has been resisted and a conventional positioning adopted. The precise positioning of the mic relative to the orchestra depends on the style and period of the music, the size of the orchestra and the acoustics of the studio or hall—in the same way as it does in mono and stereo. Whether the derived stereo and mono is, acoustically, a less or more reverberant sound than that of normal stereo depends on the encoding system chosen for transmission.

The other matter for consideration is whether the mic should be mounted vertically so that the sound field presented to the listener is parallel with the ground, as in everyday experience, or whether the brain is tolerant of a sloping soundfield as conventional slinging of the mic would produce. Because only horizontal surround sound is being decoded, in the horizontal plane of the mic, there is a danger, with the mic set upright near the source of sound, of the source being 'off-mic', and so therefore producing an imbalance in the direct sound in the X and Y components. As the information is not lost, but contained in the unused vertical component Z, it is possible to retrieve the situation by rotating the sound field in the vertical plane. This is easily achieved in a waltz control by overplugging the Y input with the Z component. The current version of the complete *Soundfield* mic which is commercially available has a steering box associated with the mic; this steering box will produce this electrical tilt as one of its facilities.

In most broadcasts or recordings which use a 'classical' overall coverage mic, it is frequently found necessary to supplement the output with spot mics close to certain instruments, etc. Matching the sounds from the spot mics to that of the soundfield leads to certain other possible techniques.

Other mic techniques

Because the spot mics are much closer to their sources (in terms of arrival of the sound at the mic and in terms of the sound perspective) than is the *Soundfield* mic, there is a danger 'lumps' arising in the overall sound. Consideration of the typical distances involved shows that the sound at the output of the spot mics may be as much as 30ms ahead of that at the soundfield. With the advent of purely electronic delays, it is possible to try the effect of delaying outputs of the spot mics as to be time-coincident with that from the soundfield.

Another technique which seems to provide a useful improvement (at least on initial trials) is that of using the spread facility on the surround panpots. When mixing outputs of the spot mics with that of the soundfield, each of them will be routed through a panpot in order to position its output electrically and to derive the necessary three channels of information. Using a very small amount of spread on each panpot (not more than 5° to 10°) seems to relieve the 'lumpiness' quite considerably. Whether combined spreading and delaying would effect worthwhile improvements is a matter for future investigation.

Panpots and spreaders

Apart from the use of the spread facility of the panpots as just described, the facility is a useful way of diffusing certain instruments or vocals, particularly when an ethereal effect is required, or for spot effects in drama.

There is a likely desire to position, say, one instrument in front of another in a balance. Some initial experiments have been tried by changing the relative gains of W relative to X and Y. Certainly, when listened to at a point remote from the central position, the image can be heard to move towards the central listener's head, but the effect seems less positive when heard from that central position. The perspective of the original sound has a significant effect; and the use of spreading and delaying the sound, in addition to changing the relative gains of the three channels, would be worthy of research.

The overall spreaders have found occasional use with instruments, but are very useful in coping with artificial reverb return signals or dramatic effects such as rain and wind, all of which will probably need to be delocalised completely. The design of the spreaders is critical, especially in the way the mid-band frequencies are distributed. This ensures that the dominant pitch of the effects will not become lumpy. Any residual lumpiness can be steered to the best position by a waltz control.

Waltz controls

It seems quite possible that, in the world of popular music, a motorised version of a waltz control might soon be required, so that the sound field can be made to spin; but the manual control is most useful, not only for steering the output of spreaders or the 'tilting' of the *Soundfield* mic, but also for correcting any twist which the *Soundfield* mic might suffer from hanging on a cable. In addition, a *Soundfield* mic can be 'turned around' if, on any particular occasion it is found to sound better with an other than normal orientation. In multimic operations, the waltz control is a quick and convenient means of moving one group of pre-panned instruments (say, the brass section) around the sound stage relative to another group—say, the rhythm section.

Monitoring

The greatest problem encountered so far is with the small size of the vehicle and the consequently much too small listening area. The decoding is perfectly correct only in the middle of the area; and, in the context of the vehicle, this is very small. In addition, the speakers are too close and can produce too intimate an effect for some balances.

The reverb time is well-controlled and low (about 0.2s). When in the central listening position, the imaging seems good. Without losing too much room, this was about the highest RT which could be achieved economically. There is a school of thought which claims that a higher than conventional RT in control and listening rooms leads to improved imaging; but, within the vehicle, testing of that claim was not possible.

Operational checking of facilities is helped by the use of a vector display; by using the Z input of an appropriate oscilloscope in addition to the X and Y inputs, a vector display of the soundfield can be obtained. Panpots, waltz controls etc, are easily checked, some of the patterns produced during programme being very revealing—particularly when the sources of sound are in a highly reverberant building such as a cathedral. Making the soundfield cohesive is much more difficult when working with artificially-generated soundfields. With practice, the vector display can be used as a guide; but care is necessary, firstly to retain concentration on analytical listening to the balance of sound, and secondly to avoid undue interest in the vector display.

Conclusions

Just as colour television might convey greater sense of reality than does black and white presentation, so may surround sound stand superior to stereo and mono audio reception. In what might be regarded as a presentation of sound more natural, and therefore more acceptable to the human ears and brain, may lie the paradox that the listener will be the more critical of it and the more displeased when it is imperfect. When the system is used to convey an existing, naturally-balanced soundfield, the producer and sound balancer might need to work much harder to ensure that the realism is not distorted by the system. For those areas of material for which no preconceived conventions exist, it will be a matter of experience as to what extent, if any, the listening public will accept the multitude of effects which the system is able to offer.

On the assumption that the usable listening area can be made worthwhile without involving unrealistic expenditure on numbers of loudspeakers, the most important underlying question for broadcasters and recording companies is whether consumer demand will be sufficient to render economical any method of surround sound provision.

Time and experience in operational use will undoubtedly bring greater reality as people master the techniques, but will it be worth all the trouble and expense if the listening public is not going to want it, or be prepared to bridge the gap between real life and reproduced sound? ■

Reprinted, by permission, from IBA Technical Review 14, *Latest Developments in Sound Broadcasting*, June 1981.