# MULTICHANNEL SOUND RECORDING PRACTICE USING MICROPHONE ARRAYS

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Using the process of Multichannel Microphone Array Design (MMAD), an almost infinite number of microphone configurations can be chosen to suit the needs of a particular sound recording situation. The basic characteristics of Front, Lateral and Back Segment Coverage, together with the process of Segment Offset used to obtain Critical Linking, are part of the main MMA Design process. However many other selection criteria must be considered in order to satisfy specific operational preferences or to obtain the optimum choice of a microphone array adapted to a particular situation. This paper will present an analysis of a range of different selection criteria to assist the sound recording engineer in choosing a selection of suitable microphone array configurations for his particular requirements.

## 1 THE LOUDSPEAKER CONFIGURATION - FROM STEREO TO MULTICHANNEL

### 1.1 Angular Distortion

The basic characteristics of a two channel or stereophonic sound recording system are determined with respect to the position of the loudspeakers and the listener during reproduction. In the Standard Listening Configuration (Figure 1a) for two channel stereophony, the listener is placed at the summit of an equilateral triangle, the loudspeakers being positioned at each extremity of the base of the triangle and directed towards the listener.

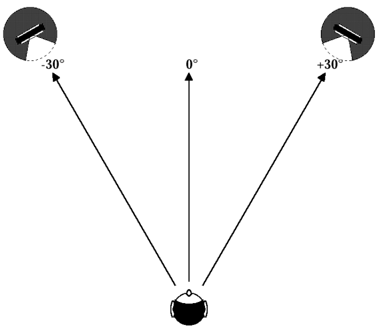


Figure 1a - Standard Stereophonic Loudspeaker Configuration

The recommended Standard Loudspeaker Configuration (Figure 1b) for Multichannel Sound (ITU-R BS.775-1) has first of all added a central loudspeaker between the left and right loudspeakers without changing the angular size of the main sound stage with respect to stereo reproduction. In the search for Multichannel quality reproduction of music, sound engineers often find themselves at a loss to know how to use this centre channel effectively.

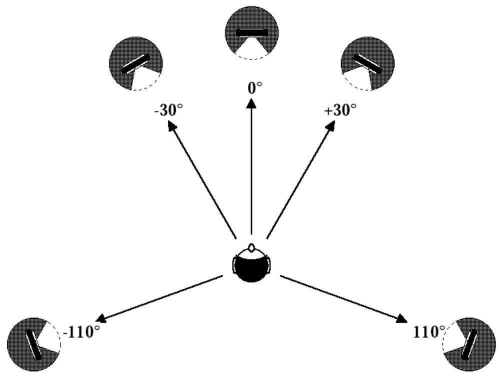


Figure 1b - Multichannel Loudspeaker Configuration

It would be an error to consider the central loudspeaker as a simple addition to the usual left/right sound stage. On the contrary, it has fundamentally changed the characteristics of perception within this main front sound stage. It is now necessary to consider two contiguous segments of 30°, one segment determined by the left front loudspeaker and the centre loudspeaker, the other by the centre loudspeaker and the right front loudspeaker. Therefore the characteristics of the front elements of any microphone array, must now be determined specifically with respect to this loudspeaker configuration. The main advantage of these smaller reproduction segments is that the reproduction of the front sound field is considerably more linear, however there is also a small improvement in the stability of the front sound stage.

The introduction of two loudspeakers placed behind the listener at about 110° to the left and right of the front central axis as specified in ITU-R BS.775-1, not only opens up the possibility of sound reproduction of lateral reflections and reverberation in both the lateral and rear segments, but above all allows us to exploit the continuous sound field reproduction in front of the listener, thereby considerably extending the traditional 60° front sound stage.

Unfortunately the slight increase in stability of the front sound stage is compromised by the addition of an inherently unstable rear segment. In other words, the small improvement in the amplitude of the "sweet spot" created by the use of three front loudspeakers is counteracted by this highly unstable back segment. However energy levels in this back segment can be so low as to render this effect almost negligible. It is one of the factors that the sound engineer must take into account in the Multichannel Microphone Array Design process.

The basis of the process of Multichannel Microphones Array Design (1)(2) is the division of the sound field into individual segments, where each segment is treated as a separate entity, the listener being able to differentiate between each segment as defined only by the loudspeakers on either side of the segment. The left front segment being determined by the left and centre loudspeakers, the right front segment being defined by the centre and right loudspeaker, whereas the pair formed by the left and right hand front loudspeakers does not seem to enter into the equation. Similarly for the lateral and back segments. If this were not the case we would have conflicting virtual images generated by every pair-combination of loudspeakers and the result would be complete audio chaos!

**1.2 Segment Reproduction Linearity**

The linearity of reproduction in each segment in the recommended Multichannel Listening Configuration is mainly determined by the angle of reproduction within each segment. The different segments covered by each respective loudspeaker pair vary from 2 segments covering 30° each at the front, two segments of 80° on the sides, to one segment of 140° at the rear.

Although the recommended listening position for stereophonic reproduction is defined as the summit of an equilateral triangle, the loudspeakers being positioned at each extremity of the base of the triangle, most people will have experienced listening to stereophonic reproduction when positioned either too far away from the loudspeakers, or too near to them. It is an interesting experience to analyse the characteristics of localisation especially with respect to Angular Distortion of the reproduced sound for each listening position, either nearer or further away from the recommended equilateral triangle position whilst still remaining on the central axis. This situation is shown in figures 2a to 2e, where the original sound source is represented by the letters A to K and the resulting reproduction characteristics shown with the listener seeing the loudspeakers at 30°, 60° 80° and 140°.

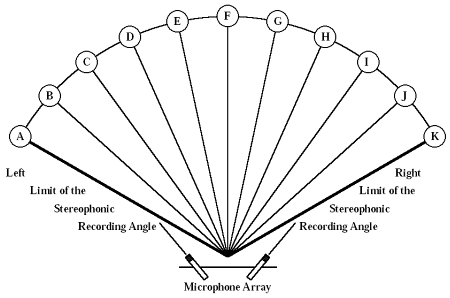


Figure 2a - Position of the sound sources A to K w.r.t. the limits of the SRA

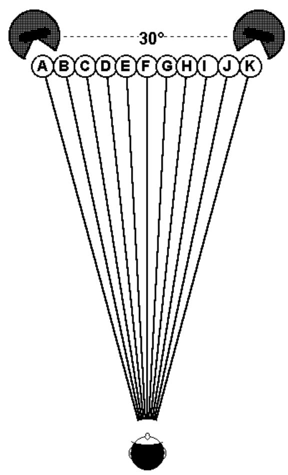


Figure 2b - Loudspeakers at 30°

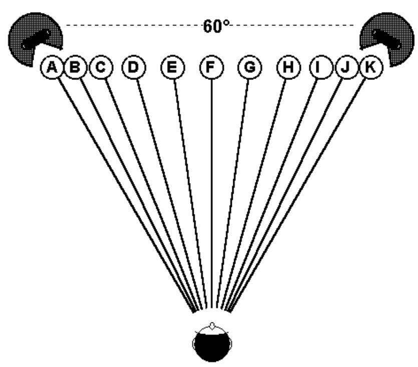


Figure 2c - Standard Stereophonic Loudspeaker Configuration at 60°

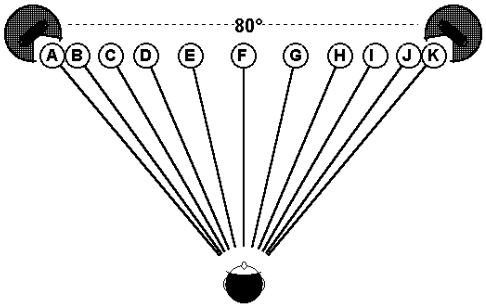


Figure 2d - Loudspeakers at 80°

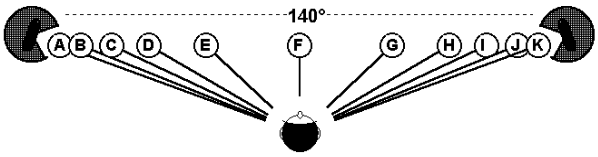


Figure 2e - Loudspeakers at 140°

Careful analysis of the localisation characteristic shows that the further one is away from the loudspeakers, the better is the Angular Distortion characteristic of reproduction, i.e. the reproduction between the loudspeakers becomes seemingly more linear. On the other hand increasing the angle between the loudspeakers produces a marked increase in Angular Distortion of the reproduced sound source, perceived as a crushing of sound source localisation towards the loudspeakers and a corresponding spread of the central sound image.

Despite the considerable difference in listening positions, the limits to the segment of the original sound field that is reproduced between the loudspeakers, remain practically the same, i.e. the Stereophonic Recording Angle - SRA does not seem to change. In addition, both the Angular Distortion and the SRA seem to be independent of the rotational position of the head in these different listening positions.

In the context of the recommended Standard Configuration for Multichannel Sound, we can therefore expect a much more regular distribution of the sound sources within the main sound stage of the front triplet of loudspeakers. Angular Distortion will be more predominant within the side segments, and even more so within the back segment. This is of little consequence in the reproduction of lateral reflections and reverberation, but becomes more difficult to integrate into realistic total surround sound reproduction, where a more even distribution of loudspeakers would be advantageous.

**1.3 Localisation Resolution**

This angular distortion characteristic must not however be confused with the precision with which we can localize sounds in various positions around the head. In the natural environment, we are capable of localizing sound sources to within a few degrees in front of the listener, and to a lesser extent behind, whereas localisation becomes progressively more uncertain to each side of the listener. It is quite remarkable that this "natural" characteristic of localisation resolution is also apparent in the multichannel listening configuration. The front loudspeakers are, after all, situated in the front zone of good localisation resolution, whereas the rear loudspeakers are placed within the very poor localisation zones on each side of the head. In spite of this, the listener seems to be conditioned by the "natural" listening characteristics of localisation around the head, and not the specific position of each loudspeaker, even though the sound image is of course, a virtual image created within each segment by each loudspeaker pair.

Here are the main characteristics for each segment (also shown in Figure 3) :

- the two front segments, each at 30°, show good linear localisation and maximum localisation resolution,
- the side segments at 80° show Angular Distortion approximately the same as with the Stereophonic Listening Configuration (4° to 9°), but progressively poor resolution to the sides,
- the back segment has very pronounced Angular Distortion, but with reasonably good resolution of localisation

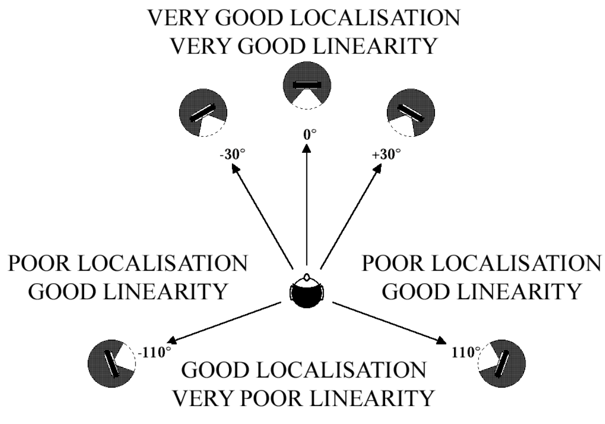


Figure 3 - Localisation and Angular Distortion in the Multichannel Listening Configuration

The type of microphone array used to cover each segment will also influence the characteristics of angular distortion in the various segments. As with stereo, the balanced use of both “time difference” and “intensity difference” in the design of the microphone array enables us to reduce the amount of angular distortion to around 4° to 5°, see ref (3)&(4).

This obviously will apply also to the Multichannel Microphone Array Design process. However there is still considerable debate as to the influence of “time difference” or “intensity difference” dominance in stereophonic recording systems with respect to the characteristic of localisation resolution. It will come as no surprise to hear that the same situation applies to the front and rear segments of a multichannel array.

It remains to be seen if a predominance of either “time difference” or “intensity difference” in the lateral segment coverage will have any noticeable effect on localisation resolution.

## 2 THE SELECTION CRITERIA

The fundamental properties of each of the reproduction segments in the listening area must be borne in mind during the process of initial choice and optimisation of a Multichannel Microphone Array in a specific recording context.

Using the technique of Multichannel Microphone Array Design (1)(2), the sound recording engineer is faced with the difficult task of selecting from a large range of Multichannel Microphone Arrays, those that may suit his particular needs with respect to a specific sound recording session. Much of the difficulty lies in determining a set of selection criteria that mainly concern the different parameters of the reproduced multichannel sound field, but can also take into account some of the more practical limitations, such as the availability of suitable microphone support systems, or the type of mixing and recording equipment used.

The main set of criteria that need to be considered are as follows :

- Front Triplet Segment Coverage
- Lateral Segment Coverage
- Back Pair Segment Coverage
- Microphone Array response above and below the reference plane
- Use of only Microphone Position Offset to obtain Critical Linking
- Availability of operational Time Offset adjustment to obtain Critical Linking
- Segment reproduction linearity with relation to loudspeaker placement
- Quality of localisation in the different segments
- Proportion of Time Difference to Intensity Difference in each Segment Coverage
- Reduction in acoustic cross-talk between array segments
- Availability of microphone support systems
- Choice of microphone directivity versus frequency response

Each of these criteria can have an impact on one or more of the characteristics of the reproduced sound field or, on the suitability of a specific microphone array to the job in hand.

### 2.1 Size of reproduction of the main sound stage

In the great majority of sound source situations that we are called upon to record, the sound source occupies a limited sound stage in front of us, while the surrounding sound field is made up mostly of early reflections and reverberation. Although we are obviously interested in reproducing the total surround sound field as realistically as possible, due to the basic limitations of any microphone recording system in the reproduction of sound perspective, we must give priority to the reproduction of the main sound stage.

As with any sound recording situation, the position of the microphone array is of primary importance in the perception of overall sound perspective. The microphone or microphone array will need to be nearer the sound source to obtain the same sound perspective compared with perception by normal hearing of the actual sound source. The perception of distance is, as usual, mainly a function of the ratio of direct to reverberant sound, and therefore influenced by the directivity of the microphones used in the system. For a given ratio of direct to reverberant sound, a microphone array using omnidirectional or hypocardioid microphones must be placed nearer to the sound source compared to a cardioid or hypercardioid microphone array.

The vertical position of the microphone array must first of all be chosen so as to obtain the desired sound perspective between the different rows of the orchestra, as shown in Figure 4.

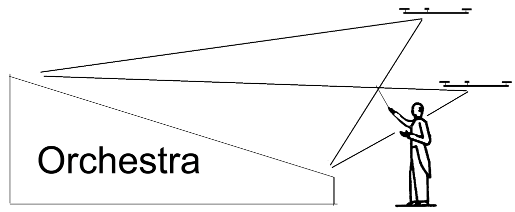


Figure 4 - Vertical position of the microphone array

It is from this subjectively chosen position of the microphone array that the angle for Front Triplet Coverage will be chosen, thereby determining the size of the reproduced sound source. Although this would seem to be a rather simple decision to make, it is of fundamental importance in the perception of the direct sound image. We can choose to limit the reproduction of the direct frontal sound source to the front triplet of loudspeakers, or on the contrary take advantage of the multichannel surround sound environment using the side segment reproduction, and « envelope » the listener in the sound source as much as we consider desirable.

Although the complete use of the side segments for direct sound is not advisable due to the progressively worsening localisation characteristics on the side, it is nevertheless possible to exploit the main sound stage of sound reproduction further than the « 30° + 30° » defined by the front three loudspeakers. An angular spread of the sound source of about 10° or 20° into the lateral segments is quite possible, with the added advantage of a better feeling of envelopment of the listener. In practice this obviously means that we need to use a Multichannel Microphone Array that allows us to fully exploit the recording and reproduction characteristics of the lateral segments with the added advantage of better reproduction of lateral side reflections. Unfortunately up to now our listening habits have been conditioned by the limitation of the stereophonic sound stage to 60°. However it still remains to be seen as to precisely how much more spread of direct sound is desirable with a wide sound source, or on the other hand whether it is necessary to stay within the front triplet 60° for realistic reproduction with the smaller sound sources.

If the Front Triplet Coverage is larger than the sound source then the reproduced sound image will be reproduced within the front triplet of loudspeakers. This is to some extent similar to the reproduced sound stage of a stereophonic two channel recording as shown in Figure 5. If however the Coverage is just smaller than the sound source, then the extremities of the sound source will be reproduced by the adjoining lateral segments, thereby producing somewhat improved listener envelopment as shown in Figure 6. If the Front Triplet Coverage is much smaller than the sound source, then the extremities of the sound source will be reproduced well into the adjoining lateral segments as shown in Figure 7, and listener envelopment will be much improved.

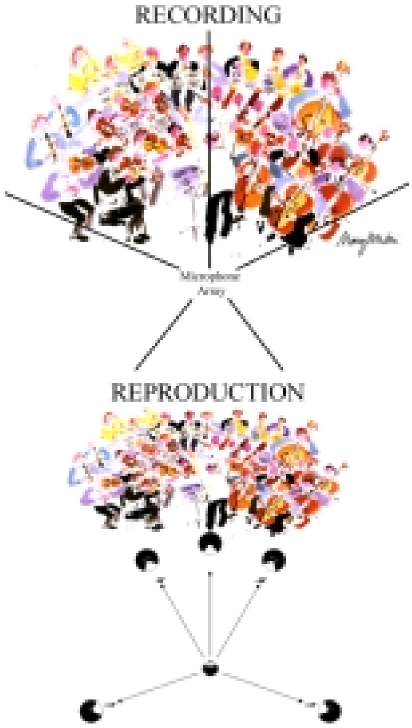


Figure 5 - Front Triplet Coverage
is larger than the sound source

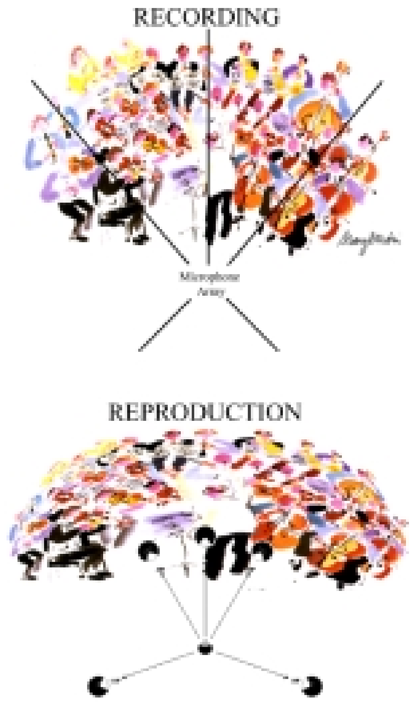


Figure 6 - Front Triplet Coverage
is just smaller than the sound source

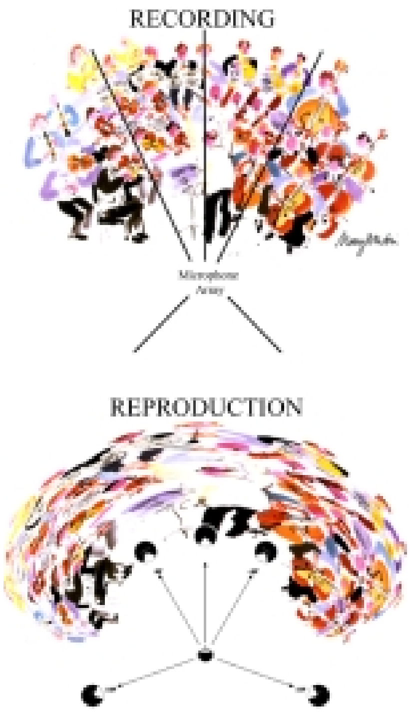


Figure 7 - Front Triplet Coverage
is much smaller than the sound source

The approach is obviously different in the case of a completely surrounding sound source, where one needs to record the natural sound environment of, say a forest. Also in the case of a certain number of musical works written specifically with a view to creating a surround sound environment, the microphone system is, by definition, placed in the middle of the surrounding sound source. In this type of situation the smooth reproduction of the sound field means that each sound reproduction segment covered by a pair of loudspeakers should correspond approximately to the same segment of the original sound field. But the realistic reproduction of sound perspective depends entirely on our ability to place each sound source at the required position and distance with respect to the microphone system.

In a paper (5) presented at the 91st AES Convention in 1991, the author described three specific microphone arrays suitable for complete surround sound recording using four, five and six channel recording systems. However "natural" realistic continuous sound field reproduction could only be obtained with these systems by the use of a loudspeaker configuration using equal segmentation of reproduced sound field. The basic requirement of this type of array is that the angle between the microphones is the same as the angular placement of the loudspeakers. The distance between the microphones is chosen so that the Stereophonic Recording Angle of each segment is also the same as the angle between the microphones. For a 4 channel system we must therefore use a 25cm/90° square array of cardioid microphones, a 39cm/72° pentagon for a 5 channel system and a 53cm/60° hexagon for a 6 channel system, as shown in Figures 8 to 10.

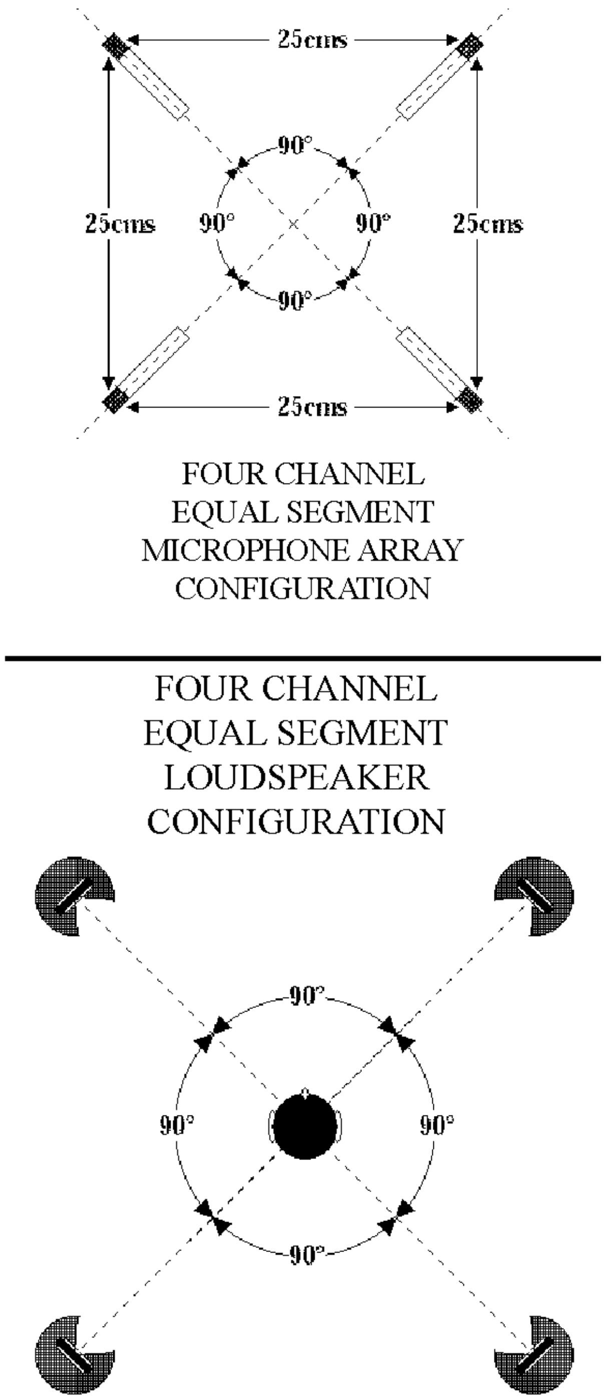


Figure 8 - The Four Channel Square Array

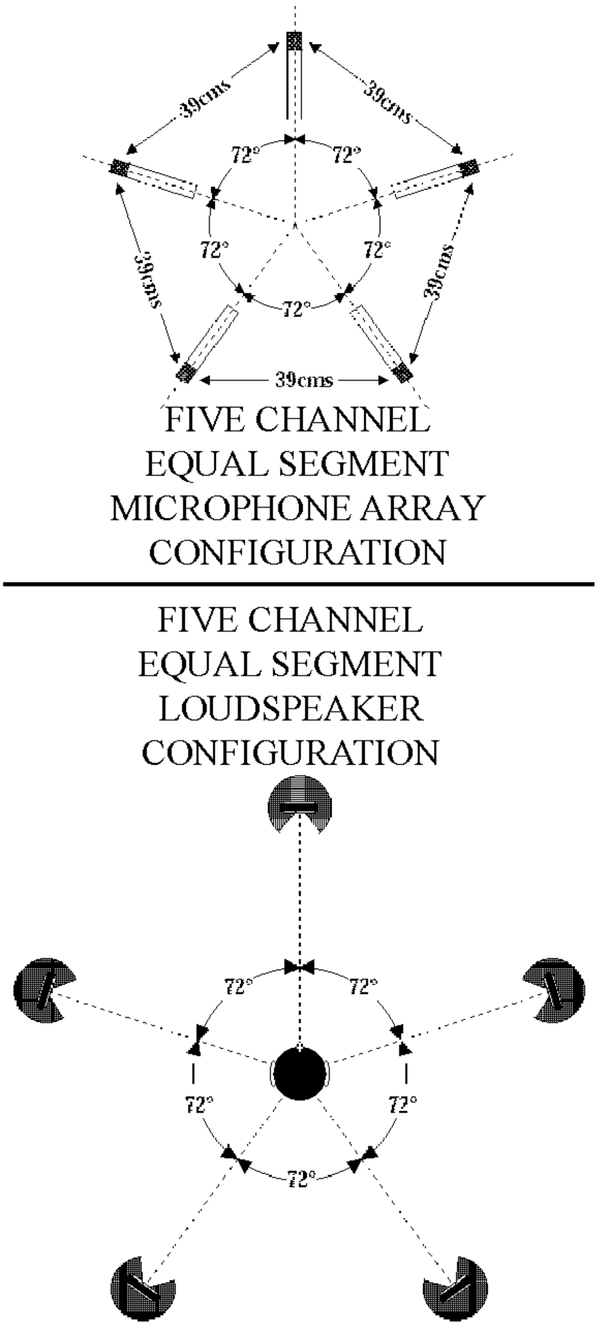


Figure 9 - The Five Channel Pentagon Array

The same approach was also developed in this paper for surround sound microphone arrays using either hypercardioid or hypocardioid microphones. It is evident from this paper that coincident microphone systems (using 1st order directivity patterns) cannot supply the necessary information for this type of continuous surround sound coverage.

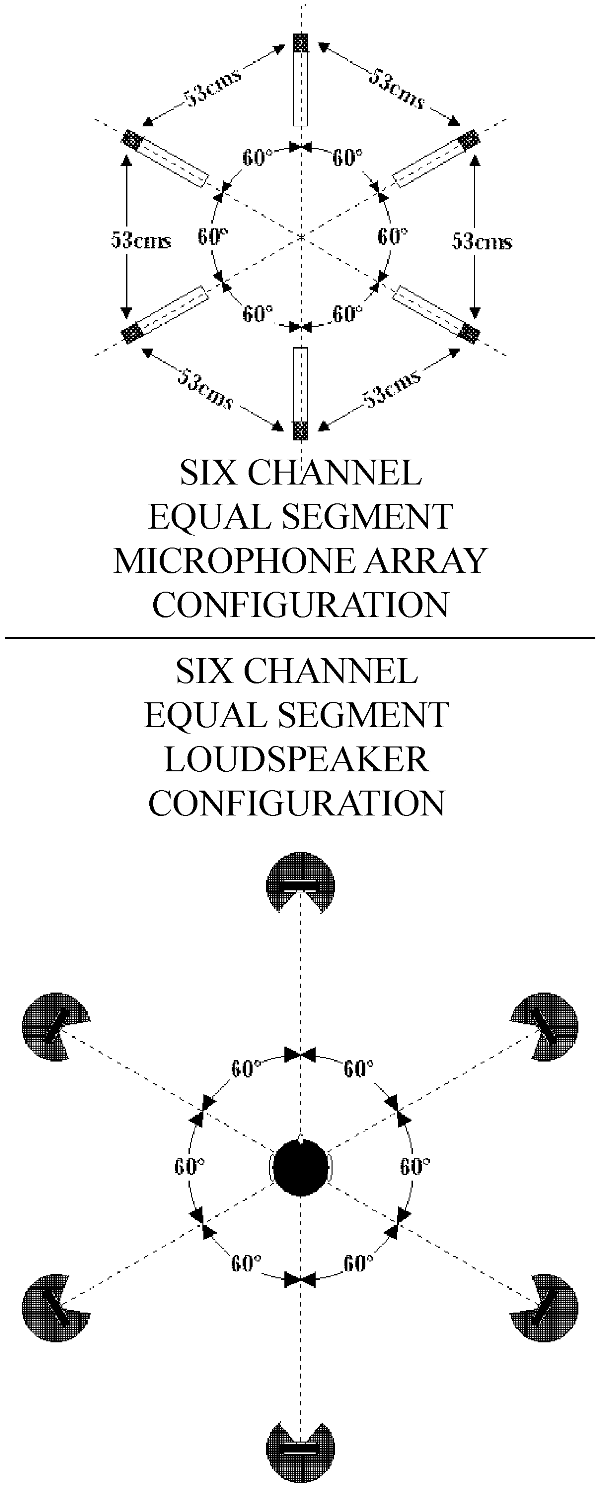


Figure 10 - The Six Channel Hexagon Array

With the present recommended multichannel loudspeaker configuration, it is almost impossible to produce a microphone array that will cover the same segments as those reproduced by the front, lateral and back segments of the multichannel loudspeaker system

(30°, 80° & 140°). In particular, in order to produce a front coverage of « 30° + 30° », the spacing of the front triplet of microphones will need to be so wide as to render completely impossible any lateral and back segment design with critical linking i.e. continuous (or "seamless") reproduction of the surrounding side and back sound field. Correct front coverage microphone arrays can be obtained, but at the expense of neglecting the correct coverage for either the lateral or back segments.

However, as demonstrated in the multichannel listening sessions at the 108th Convention in Paris with the recording presented by Williams & Le Dû, correct coverage can be obtained by using only a 4 channel system, i.e. by eliminating the use of the centre loudspeaker channel. The central microphone of the microphone array presented at these listening tests was, in fact, purely a dummy (used in this case to avoid any visual prejudice in the judgement by the listeners). This type of four channel recording system is nowadays quite often used to record ambient sound for cinema multichannel diffusion or as a reverberant pick-up system in the music recording industry.

The Five Channel Pentagon Array can be a very satisfactory multichannel (5 channels) microphone array when recording the traditional sound source in front of the microphone system.

### 2.2 Early Reflections and the Reverberant Field

Early reflections generated by the floor, ceiling and back walls around the sound source will be reproduced mainly within the front triplet. The lateral segments on the other hand will play a major role in the reproduction of early reflections generated by the surfaces on each side of the sound source. Here again the Multichannel Microphone Array produces a considerable improvement in the restitution of these reflections, as can be judged by the improved feeling of space or volume in the reproduction. Even though the lateral segments suffer from a reduced resolution concerning the localisation of these reflections, this is a natural phenomena and is similar to natural hearing. Of course improvement in localisation in the lateral segments will follow any eventual rotation of the head as with natural hearing, the listener must of course remain in the central listening position.

The reverberant field surrounding the sound source and the microphone array will be reproduced, as a continuous virtual sound image, by each of the segments in proportion to the Coverage of each angular segment. The microphone array response above and below the reference plane will also play a major role in the distribution of the reverberant field, not only as a virtual sound image in each reproduction segment, but also unfortunately as monophonic sound condensed on each of the reproducing loudspeakers where overlapping of the segment coverage occurs, as shown in Figure 11 and 12.

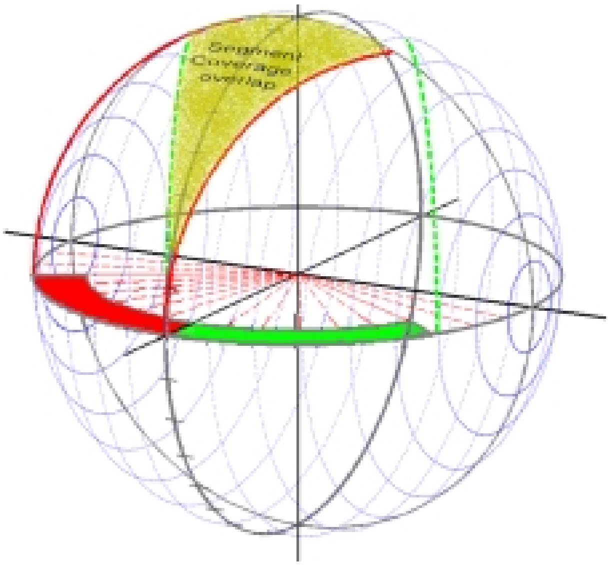


Figure 11 showing overlap above the reference plane in the Front Triplet Coverage

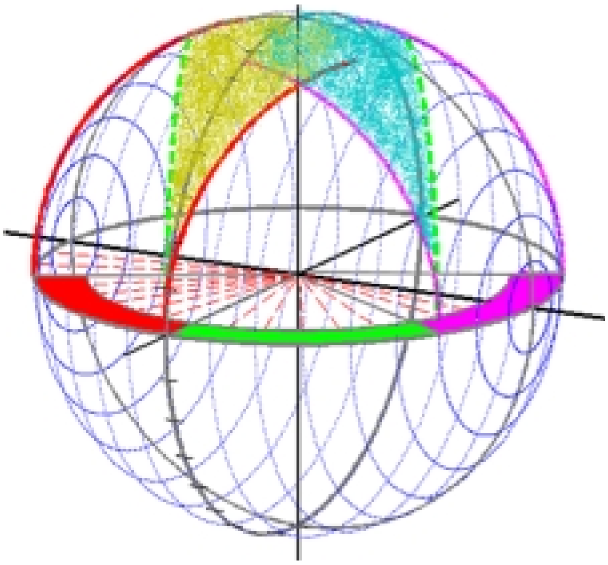

Figure 12 showing the front triplet and one lateral segment of a MMA showing overlap of Segment Coverage in the upper hemisphere

Multichannel Microphone Array Design using Critical Linking certainly allows us to obtain a continuous virtual sound image in the horizontal reference plane. This can be considered as extending to about 20° or

30° above and below the reference plane as shown in a paper (6) presented at the 112th AES Convention in Munich. However gradual segment coverage overlap will increase outside these limits, the resulting sound image of reverberation being "condensed" onto each of the loudspeakers in the reproduction system as a monophonic source.

In stereophony this effect is very pronounced, as all sound outside the specific Stereophonic Recording Angle will be reproduced as monophonic sound on either one or the other of the loudspeakers. A Multichannel system, with its continuous surround sound image in and around the reference plane, is therefore a considerable improvement, however there still remains a small component of monophonic sound "condensed" onto each loudspeaker.

The correct restitution of the whole field of early reflections is the major factor contributing to a real feeling of space or volume around the listener. Perception of timbre in the direct sound is also considerably enhanced through the reproduction of a rich early reflection and reverberant field. This characteristic has been little understood in stereophonic recording where the stereophonic sound stage cannot obviously do justice to all the surrounding early reflections and reverberation. Multichannel, on the other hand, is well adapted to quality reproduction of this aspect of the original sound field and this should be exploited to the full. The actual energy corresponding to the early reflections and reverberation is in fact very small, and one is not always immediately conscious of the importance of its subtle contribution to the overall sound field in a multichannel recording. Careful consideration however must always be given to the choice of suitable acoustics for a recording, in order to enhance the quality of the early reflection and reverberant field, both for the benefit of the musicians and for the quality of the recording.

### 2.3 Microphone Position Offset, Electronic Time and Intensity Offset

As shown in a previous AES paper (1), the introduction of Offset into the MMA design process is essential to produce Critical Linking (sometimes called "seamless" reproduction) i.e. reproduction of the complete sound field without overlap or holes. The use of Electronic Time Offset will depend on the availability of operationally usable delay on the sound desk or in the workstation. However as shown in Figures 13 to 16, where the values are taken from a paper (7) presented at the 110th AES Convention in Amsterdam, many Multichannel Microphone Arrays exist with good Critical Linking using only Microphone Position Offset. These arrays are evidently a useful practical solution in the more simple recording environments.

In general Electronic Time Offset (ETO) is to be preferred over Electronic Intensity Offset (EIO). The use of EIO leads to either a reduction in level of the front triplet of microphones for negative EIO or a reduction in the back pair level for positive EIO in the lateral segments. The EIO can be useful in certain circumstances, but in general ETO is to be preferred.

In all the arrays shown in this paper and in previous papers on MMAD, Microphone Position Offset (MPO) has been used to achieve critical linking in the front triplet. An exhaustive explanation of the theoretical and practical application of MPO, ETO and EIO can be found in references (1) & (2).

### 2.4 Proportion of Time Difference to Intensity Difference in Segment Coverage

In the field of stereophonic sound recording, most sound engineers have some form of preference for using arrays with Intensity Difference or Time Difference as the predominant localisation function. On the other hand, if a minimum of Angular Distortion is desired, the use of a balanced combination of Time and Intensity Difference is preferable. The situation is similar in Multichannel arrays, except that it is not possible to obtain perfect Critical Linking using coincident microphone systems as part of the array design process. However either Time or Intensity Difference dominance can be catered for, at least in part of the array design. In fact Intensity Difference dominance in the front triplet design is absolutely necessary if the sound source is relatively close the microphone array, as shown in Figures 15, 16, 19 & 20. In some cases the "wingspan" of some arrays can easily reach a couple of metres, but it is also possible to choose a front triplet configuration which only has a "wingspan" of about 30 to 40 cm in all.

A good "rule of thumb" is that the sound sources should be no nearer than about 5 times the distance between the microphones. This of course restricts us in the choice of suitable microphone arrays for a specific recording situation. With a small group of musicians such as a quartet or quintet, according to the acoustics of the recording studio or concert hall, the musicians may be no more than 1m50 or 2 metres from the microphone array. A selection of arrays must therefore be chosen with a maximum "wingspan" for the front triplet of no more than 60cm (30cm for each segment). Whereas the recording of a symphony orchestra, where the front row of musicians will probably be around 3 metres away, it is possible to use arrays with a total "wingspan" of up to approximately a metre, as shown in Figures 13,14, 17,18 & 21.

### 2.5 Reduction in acoustic cross-talk between array segments.

A high level of acoustic cross-talk already exists in the stereophonic listening configuration. The right ear will hear both the right hand and the left hand loudspeakers, the only difference being a small delay and corresponding intensity reduction on the signal coming from the left hand loudspeaker. And visa versa for the left ear. The exact values can be determined from the HRTF for the individual listener.

The result is that both Intensity Difference and Time Difference values have to be more or less doubled in order to obtain the same results as those produced with binaural reproduction (i.e. with headphones). An Intensity Difference of only 7db will produce a maximum displacement of the virtual sound image towards the left or right headphone, whereas an Intensity Difference of 15db is necessary to produce the same effect with loudspeakers (i.e. displacement of the virtual sound image up to the limit of the loudspeaker position). The same situation also applies to Time Difference information, where about 0.7 mS produce maximum displacement in binaural listening, whilst 1.12 mS is needed with loudspeakers. However despite this high level of cross-talk, stereophonic localisation can be clear and precise, especially with musical instruments having a short decay time and a high proportion of transients. The existence of cross-talk therefore does not necessarily mean dispersion in the localisation of the sound image.

A clear distinction must be made between the characteristics of localisation obtained with musical instruments with a high proportion of transients, and that obtained during the steady-state or long time decay radiation. Comparison between a stereophonic recording of bells with a long decay time, and "bambou blocks" with high transients and a very short decay time will show clear and precise localisation of the bambou blocks, and very poor localisation of the bells, except of course during the transient radiation of the initial impact. The same situation also applies when listening to the original sound sources, our hearing is in general much more finely tuned to the localisation of percussive sounds with a high transient content as opposed to steady-state or long decay time sounds.

The impact of acoustic cross-talk in a multichannel sound reproduction system will also differ considerably between sounds with a high content of transients and those with a long decay time. In general microphone arrays using the MMAD process will show no degradation of localisation due to acoustic cross-talk during transients. However in some cases disturbing effects will be noticed during the long decay times experienced with some sound sources. If the microphone array design is such that it produces similar level radiation from more than two loudspeakers, then a "comb-filtering effect" will be produced during decay, which becomes even more disagreeable when there is slight movement of the head. This effect can be attenuated by designing the orientation of the microphones in adjacent segments so that the directivity pattern of each microphone reduces to a minimum the amplitude of interfering signals.

This is obviously a limiting factor in the use of omnidirectional microphones in the design of a multichannel array. However using cardioid microphones the interference is already very much reduced, and even more so if the orientation of each microphone has been considered especially with respect to reducing cross-talk into the adjacent segments. The use of hypocardioid microphones needs much closer attention to cross-talk interference to obtain satisfactory results. More psycho-acoustical research still needs to be done in this field to know the magnitude of perceived interference generated by cross-talk in the different microphone configurations.

### 2.6 Microphones Support Systems

The availability of a suitable microphone support system is a very practical restriction in the use of certain microphone array configurations. The use of the "Lorraine Cross" system with a central beam with transversal crossing bars can become very cumbersome when large "wingspan" arrays are required. In this case, the tendency is to use configurations that can be fixed onto a single transversal bar with a minimum of extension arms (normally used for the centre microphone only). Array design must therefore be restricted to microphone configuration where the distance between the front triplet of microphones and the back pair of microphones is reduced to within 20cm as shown in Figure 20. In paper (7) a number of designs were shown which would satisfy this restriction.

However to fully exploit the MMA design process, the author has found it necessary to develop a microphone support system that allows much greater freedom to use any microphone array system described in the Multichannel Quick Reference Guide (7), whilst still retaining a reasonably compact size (comparable to a camera tripod), for transport of the system once folded. This microphone support system will be the subject of a Poster Session presentation by the author at the 24th AES International Conference.

### 2.7 Choice of microphone directivity versus frequency response

In stereophonic sound recording there is considerable freedom to choose any first order microphone directivity for a dual microphone array and exploit the advantages of it's corresponding low frequency response. In multichannel microphone arrays this choice is considerably more restricted, but can to some extent can be integrated into the design process. It is also possible to use a combination of directivities with a single microphone array. This may be desirable to reduce cross-talk within the system, but also can be used to facilitate the design of certain large "wingspan" front triplet combinations, whilst still maintaining Critical Linking in the rest of the system.

Cardioid microphones have a certain degree of roll-off in the bass frequencies which sometimes leads to the preferred use of omnidirectional microphones, despite the considerable increase in Angular Distortion in the reproduced sound image. The use of omnidirectional microphones as the basis for Multichannel Microphone Arrays is not recommended due to the high degree of acoustic cross-talk that occurs. However it is possible to add an omnidirectional microphone in the centre of the array, with any of the cardioid arrays described, using it to cover the bass frequencies only i.e. below about 100c/s depending on the specific cardioid microphones that are being used. The "omni" signal can either be added to each microphone channel or transmitted as a separate signal for the bass frequencies. This is somewhat similar to the reproduction systems using a single woofer for the bass frequencies in stereo or multichannel reproduction. Obviously no comb filtering effect will occur if the distance between the central omnidirectional microphone and each of the cardioid microphones in the array is well below the half wavelength at the crossover frequency.

## 3 SELECTED OPERATIONAL MULTICHANNEL MICROPHONE ARRAYS

Amongst the multitude of microphones arrays that are possible, using different Front Triplet Coverage, Lateral Segment Coverage and Back Pair Coverage, here are a few selected arrays which correspond to certain criteria that have been developed in this paper :

- Figures 13 to 16 : Arrays with no Electronic Offset required
- Figures 13, 17 & 21 : Arrays with Front Triplet Coverage of 50° + 50°
- Figures 14 & 18 : Arrays with Front Triplet Coverage of 60° + 60°
- Figure 15 : Array with Front Triplet Coverage of 72° + 72°
- Figures 16 & 19 : Arrays with Front Triplet Coverage of 80° + 80°
- Figure 20 : Array with Front Triplet Coverage of 90° + 90°
- Figure 21 : Array for mounting on a single transversal bar (FTC = 50° + 50°)

As shown in detail in reference (1), positive Electronic Time Offset means that the Back Pair of microphones is delayed by the requisite amount with respect to the front triplet. Negative Electronic Time Offset means that the front triplet of microphones is delayed with respect to the Back Pair.

Most of these arrays have been selected from AES preprint 5336 (7) where arrays were originally specified in the form of tables of microphone coordinates and orientations. As this form of presentation has proved to be too cumbersome, a CD-ROM containing a full set of plan diagrams of arrays, and some other useful documents can be obtained from soundsscot@aol.com. A web-site is also under preparation at www.soundsscot.com

REFERENCES :


(1) M.Williams & G.Le Dû, *Microphone Array Analysis for Multichannel Sound Recording,* 107th AES Convention in New York (1999) preprint 4997

(2) M.Williams & G.Le Dû, *Multichannel Microphone Array Design,* 108th AES Convention in Paris (2000) preprint 5157

(3) M.Williams, *Unified Theory of Microphone Systems for Stereophonic Sound Recording,* 82nd AES Convention in London (1987) preprint 2466

(4) M.Williams, *The Stereophonic Zoom*

(5) M.Williams, *Microphone Arrays for Natural Multiphony,* 91st AES Convention in New York (1991) preprint 3157

(6) M.Williams, *Multichannel Microphone Array Design: Segment Coverage Analysis above and below the Horizontal Reference Plane,* 112th AES Convention in Munich (2002) preprint 5567

(7) M.Williams & G.Le Dû, *The Quick Reference Guide to Multichannel Microphone Arrays Part 1 : using Cardioid Microphones,* 110th AES Convention in Amsterdam (2001) preprint 5336


*Note : I would like to thank the British artist Mary Woodin for permission to use her painting entitled "Orchestra" in order to illustrate part of the process of multichannel sound recording. I apologize for the sometimes considerable geometric distortion of her original painting, which has been introduced into some of the diagrams in order to illustrate certain aspects of this paper. This distortion is obviously not part of her original work. This and other paintings are published under the name of "Performance Cards".*

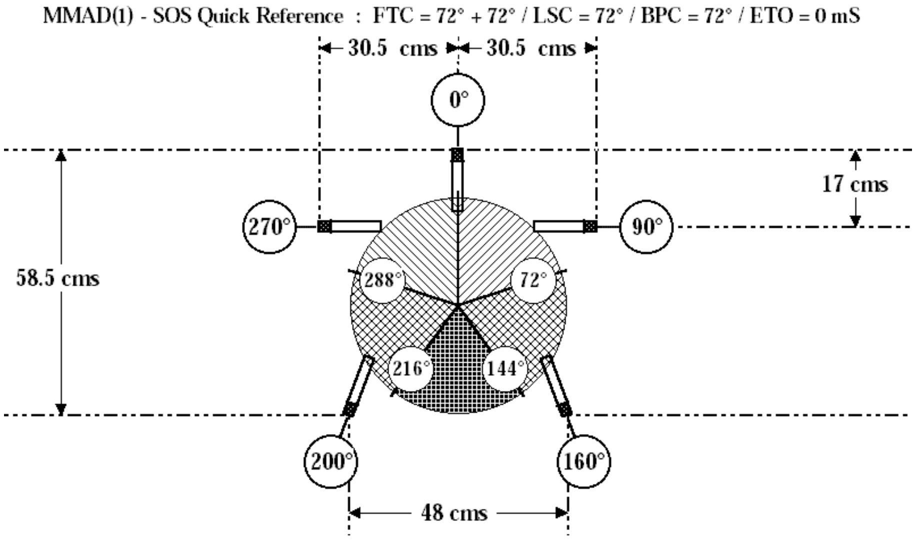


**Figure 13 - Front Triplet Coverage = 50° + 50°**
**(see Figure 17 for improved back segment, but requiring Electronic Offset)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 114° - Back Pair Coverage = 32°;
Recommended minimum sound source distance = 3m - No Electronic Offset needed;
Minimum cross-talk between segments - Balanced proportion of dI & dt in lateral segments.
dt dominance in segments covered by front triplet and back pair

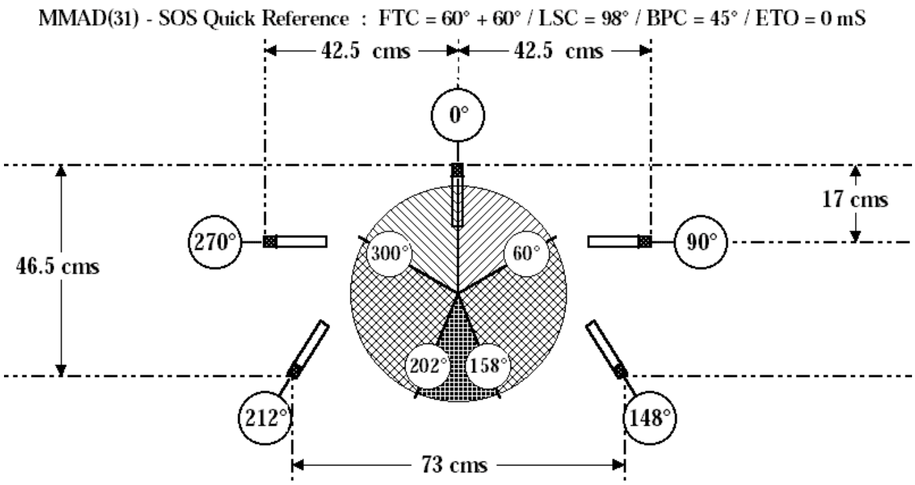


**Figure 14 - Front Triplet Coverage = 60° + 60°**
**(see Figure 18 for improved back segment, but requiring Electronic Offset)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 98° - Back Pair Coverage = 44°;
Recommended minimum sound source distance = 2m - No Electronic Offset needed;
Minimum cross-talk between segments - Balanced proportion of dI & dt in lateral segments.

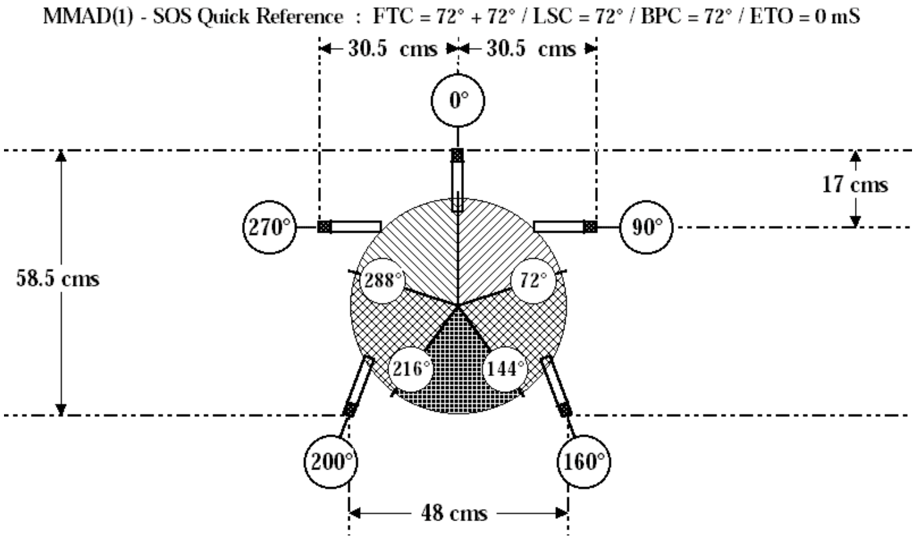


**Figure 15 - Front Triplet Coverage = 72° + 72°**
**(the ideal array for 72° + 72° - no improvement needed)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 72° - Back Pair Coverage = 72°;
Recommended minimum sound source distance = 1m50 - No Electronic Offset needed;
Minimum cross-talk between segments - Balanced proportion of dI & dt in all segments.

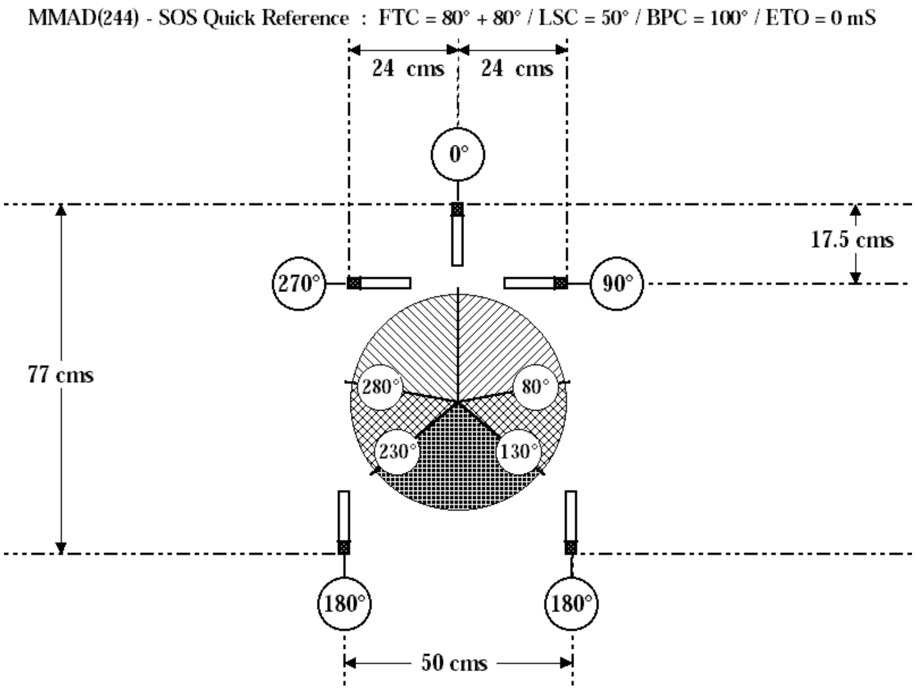


**Figure 16 - Front Triplet Coverage = 80° + 80°**
**(see Figure 19 for improved back segment, but requiring Electronic Offset)**
(a strange beasty, but the only solution at present possible for 80° + 80° without Electronic Offset)
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 50° - Back Pair Coverage = 100°;
Recommended minimum sound source distance = 1m20 - No Electronic Offset needed;
Minimum cross-talk between segments - Balanced proportion of dI & dt in front segments.
dt dominance in lateral segments - dt only in the back pair (maximum Angular Distortion)

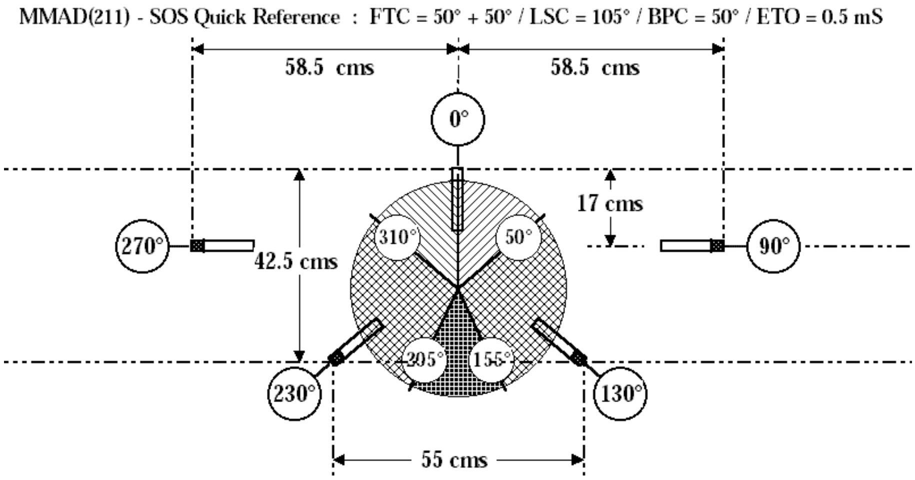


**Figure 17 - Front Triplet Coverage = 50° + 50°**
**(improved version of Figure 13 but needing positive Electronic Offset)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 105° - Back Pair Coverage = 50°;
Recommended minimum sound source distance = 3m - Electronic Offset of 0.5mS;
Low cross-talk between segments - Balanced proportion of dI & dt in all segments.

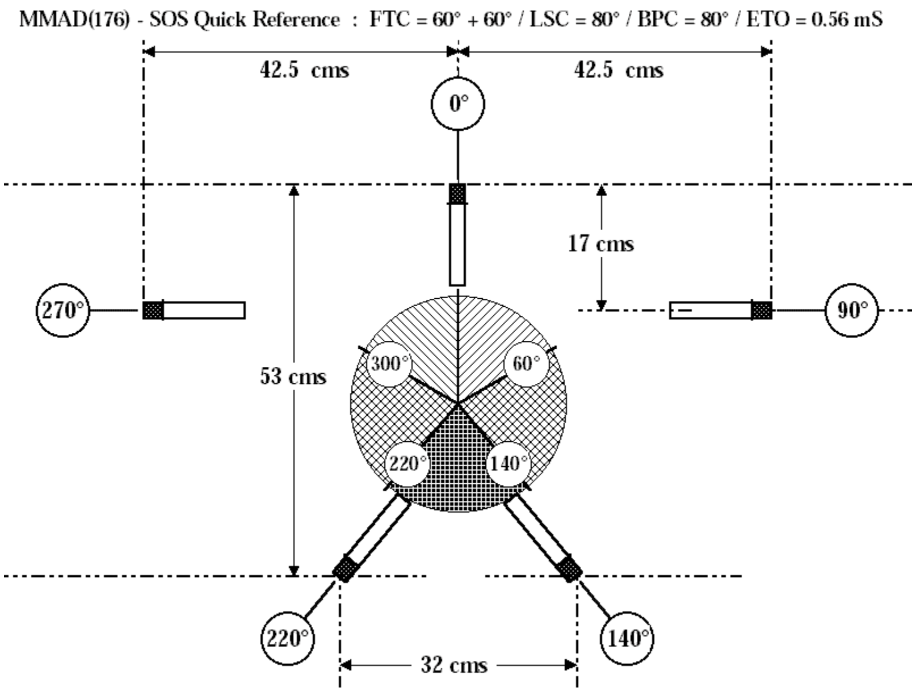


**Figure 18 - Front Triplet Coverage = 60° + 60°**
**(improved version of Figure 14 but needing positive Electronic Offset)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 80° - Back Pair Coverage = 80°
Recommended minimum sound source distance = 2m - Electronic Offset of 0.56 mS
Minimum cross-talk between segments - Balanced proportion of dI & dt in lateral

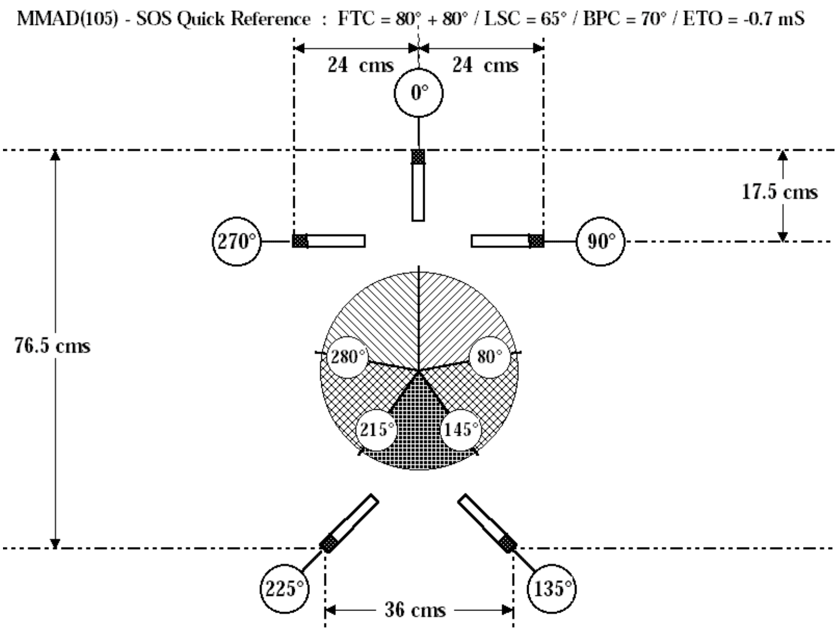


**Figure 19 - Front Triplet Coverage = 80° + 80°**
**((improved version of Figure 16 but needing negative Electronic Offset)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 65° - Back Pair Coverage = 70°;
Recommended minimum sound source distance = 1m20 - Electronic Offset of - 0.7 mS;
Minimum cross-talk between segments - Balanced proportion of dI & dt in front and back segments.
dt dominance in the lateral segments

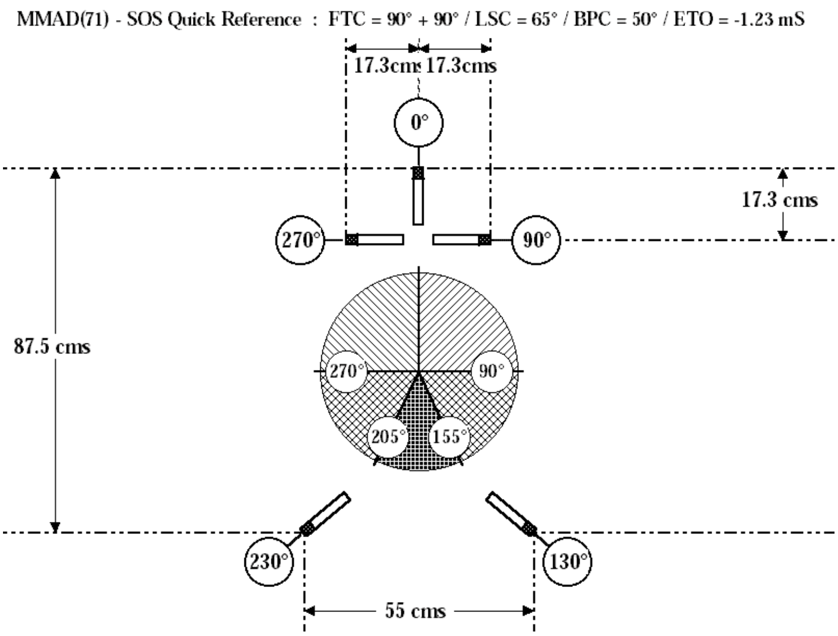


**Figure 20 - Front Triplet Coverage = 90° + 90°**
**((no version possible without Electronic Offset))**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 65° - Back Pair Coverage = 50°;
Recommended minimum sound source distance = 1m - Electronic Offset of - 1.23 mS;
Minimum cross-talk between segments - Balanced proportion of dI & dt in front triplet segments.
dt dominance in the lateral and back segments

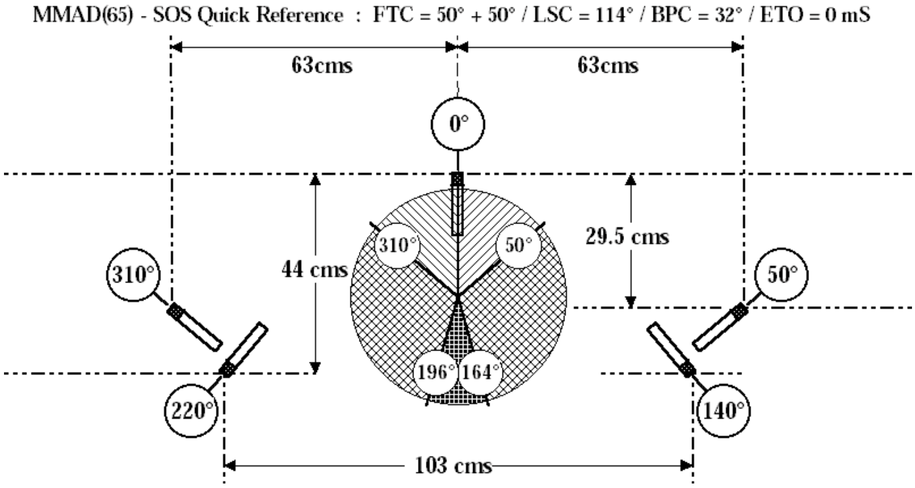


**Figure 21 - Front Triplet Coverage = 50° + 50°**
**(solution for mounting on a single transversal bar with extension only for centre mic)**
Cardioid microphones - Array optimised for recording a frontal sound stage
Lateral Segment Coverage = 114° - Back Pair Coverage = 32°;
Recommended minimum sound source distance = 3m - No Electronic Offset required;
some cross-talk between segments -. dt dominance in the front and back segments
dI dominance in lateral segments