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Computer Assisted Microphone Array Design (CAMAD)

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ABSTRACT

The basic aim of Microphone Array Design is to create microphone array recording systems with smooth seamless or 'Critically Linked' segment coverage of the surround sound field. Each configuration must take into account the interaction of the many design parameters, with the specific coverage of each segment that is required. The difficulty in the manipulation of these many parameters is one of the major obstacles in developing a wide range of microphone arrays that meet the needs of each particular sound recording environment. This paper will outline the various parameters that need to be taken into consideration and explain the basic approach to developing MATLAB based software that gives a clear and unambiguous display of all the salient characteristics needed to achieve a stable and reliable microphone array, no matter the number of channels involved, or the type of directivity pattern chosen for each microphone.

1. INTRODUCTION

The design of a few good stereophonic microphone arrays has been achieved in the past by trial and error, associated with multiple listening tests. This is a long and tedious process, however, as I have shown in previous papers, using the intersection of the physical characteristics of the microphones, with the psychoacoustic parameters of the standard listening configuration, one can design very quickly a wide range of arrays that fulfil the needs of the sound recording engineer to adapt to specific sound recording environments.

The representation of the characteristics of these arrays in the form of Stereophonic Recording Angle diagrams is relatively simple operation as shown in two previous papers (AES preprints 2072[(1)] and 2466[(2)]). This approach is however impossible within the present context of development of higher order microphone arrays that are needed to meet the needs of the ever increasing reproduction formats – Stereophony (2 channels), Quadraphony (4 channels), Multichannel (5 Channels), Blu-ray (7 channels) … and who can say how many more are being developed, or how many will appear in the future. Indeed we may be called upon to create recording tools for any number of channels and configurations, that are not based on a specific commercial media format, but simply using a basic file download process - this is probably the end of an era of hardware based media.

Computer Assisted Microphone Array Design (CAMAD) is therefore imperative if we wish to exploit the multitude of different microphone array configurations that are possible in the context of the single main microphone array approach to sound recording, and its associated intrinsic pure high audio quality.

Even though formats for height information are at present being proposed, there is little sign that there is any deep understanding or adherence to the psychoacoustic principles or mechanisms[(3)] involved in height perception. For this reason, design procedures in this paper are only considered with respect to the horizontal reference plane, passing through the axes of the microphones, or as projected onto the horizontal reference plane.

2. THE DESIGN PROCESS

2.1 No of channels and directivity

The design process starts obviously with the selection of the number of channels required. Although the order of priority of the following steps is not obligatory, it is usual to then select the microphone directivity pattern that seems the most suited to our needs, taking into account the frequency response curve associated with each directivity response. This new software is designed to accommodate arrays designs using any number of channels (at present from 2 to 8 channels) with any of the standard first order microphone directivity patterns.

2.2 Basic geometrical layout

The next stage is to design the basic geometrical layout of the microphone array – initially it must obviously be determined by the distance and angle between each of the microphones,. These two parameters will already condition the limits of each specific segment coverage, and its position within the compass rose.

2.3 Steering tools

Introduction of MPO (Microphone Position Offset) will modify the geometrical layout whilst permitting a considerable degree of steering of the segment coverage. The introduction of Electronic Level Offset (ELO) and Electronic Time offset (ETO) will enable us to steer the segment coverage of each segment of the microphone array, with the aim of eventually obtaining perfect Critical Linking, but, this time, without affecting the geometrical layout.

2.4 Critical Linking

The manipulation of these five basic parameters (distance, angle, MPO, ELO & ETO) will influence the width and orientation of the coverage, and therefore the linking of each segment one to the other. The design process is therefore a matter of juggling each of these parameters to obtain the desired segment coverage and, as near as possible, perfect Critical Linking.

The limits of segment coverage are determined by the intersection of the physical and psychoacoustical parameters for each segment. This is the same basic equation as was used in the designing of stereophonic arrays. It is essential therefore, either to survey the evolution of the intersection point, or at least to warn the user when the intersection is problematic. This was one of the fundamental instability problems in the software developed for some previous AES papers (AES preprints 4997[(4)] and 5157[(5)]), and was one of the reasons why this previous software was never made public. This new software, on the other hand, allows us to visually monitor the intersection characteristics, and generates a warning message when intersection is becoming near impossible.

2.5 Angular distortion and loudspeaker configuration

Localisation geometry is highly dependent on the loudspeaker configuration that is associated with a specific microphone array. In general the equal segment loudspeaker layout produces the best overall Angular Distortion characteristics; however the sound recording engineer must of course take into account the specific loudspeaker location associated with each media context, either as a 'standard' layout, or in developing new and improved loudspeaker configurations. This new software shows the various loudspeaker configurations that are considered standard for a particular context, according to everyday practice (or known institutional recommendations), but it also shows, where possible, some other loudspeaker configurations with specific improvements. The choice of layout will also be conditioned by the presence or not of visual information – home cinema being the obvious illustration, where the screen will condition the perception of the front sound stage - in this case surround information becomes less of a priority. Whereas, in pure audio recordings, the surround sound aspect of reproduction is normally considered much more important.

3 The GUI screen presentation

This software uses one main screen to control all the microphone array design parameters for each channel selection configuration. An example of an equal segment 7 channel array is shown in Figure 1. It is divided into three zones:

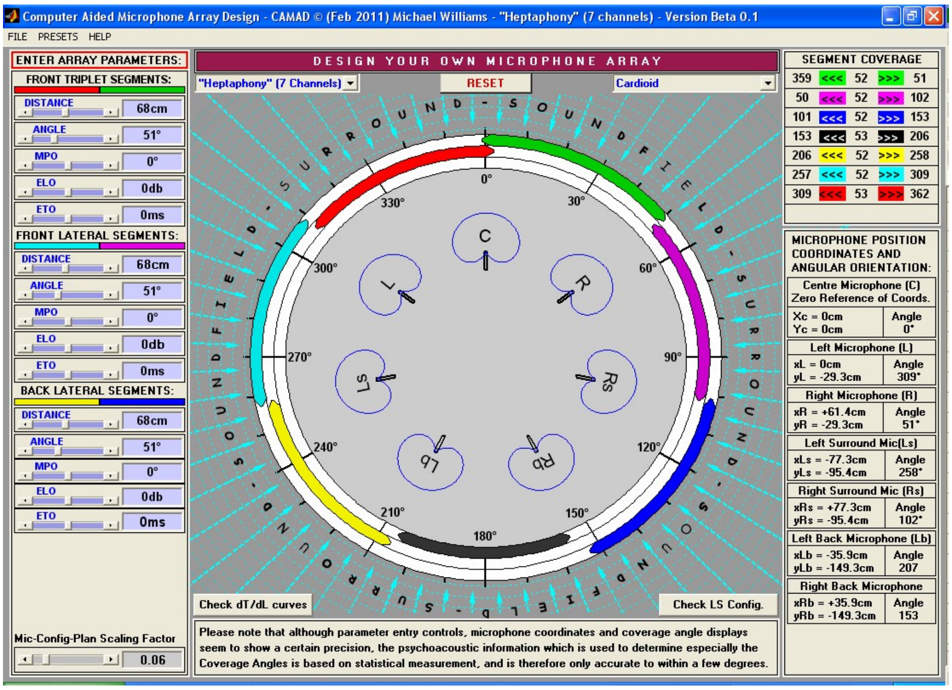

FIGURE 1 – OPENING SCREEN FOR A 7CH EQUAL SEGMENT ARRAY

- On the left hand side, an interface for entry of microphone parameters for each of the symmetrical segments

- On the right hand side, a display of coverage angles and microphone position coordinates

- A central zone containing the overall segment coverage compass and microphone plan. At the top of this central zone are two drop-down menus for channel & directivity selection, and a general parameter reset. Whilst at the bottom there are two button selections for the loudspeaker configuration display, and the time-difference and level-difference intersection check.

3.1 The main screen
There are 7 main screens used in this software, which correspond to seven different channel configurations:

- 2ch – Stereophony,
- 3ch - Triphony,
- 4ch - Quadraphony,
- 5ch – Multichannel,
- 6ch - Hexaphony,
- 7ch - Heptaphony,
- 8ch - Octaphony.

In each case the opening configuration is an equal segment array (as described in AES preprints 3157[(6)] and 7480[(7)]). Three examples are shown in Figures 1, 2 & 3 for 7ch, 5ch and stereo arrays.

Each of these seven main screens allow us to select the required number of channels for the configuration, and also to switch between different microphone directivity patterns. We can then manipulate all the main design parameters – distance, angle, MPO, ELO and ETO, whilst visualising the coverage angles of each segment.

On the same screen, detailed information is also shown for each microphone position coordinate together with its orientation. A compass rose is used to show the segment coverage, with a microphone plan representation shown in the middle of the compass rose.

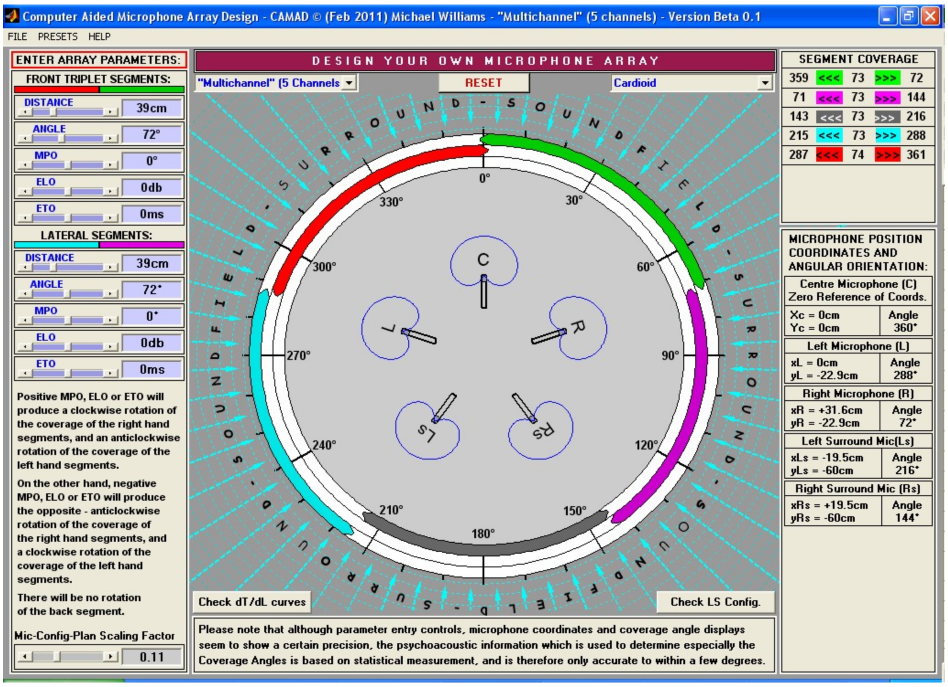

FIGURE 2 – OPENING SCREEN FOR A 5CH EQUAL SEGMENT ARRAY

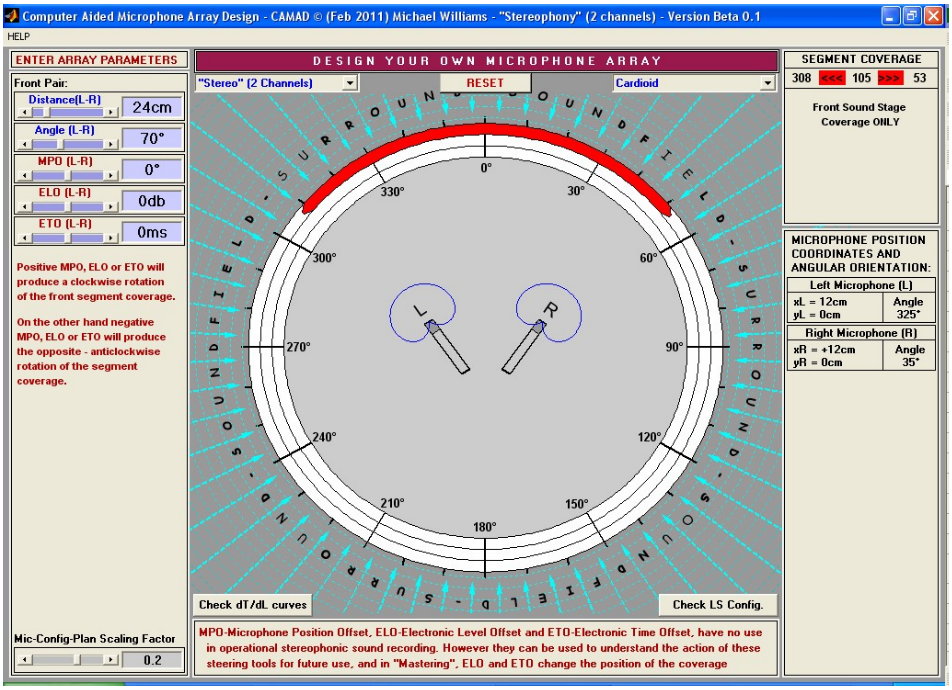

FIGURE 3 – OPENING SCREEN FOR A STEREO ARRAY

A closer look at one of the parameters interface boxes, as shown in Figure 4, shows how each parameter can be adjusted using a slider input or otherwise by direct numerical keyboard input.

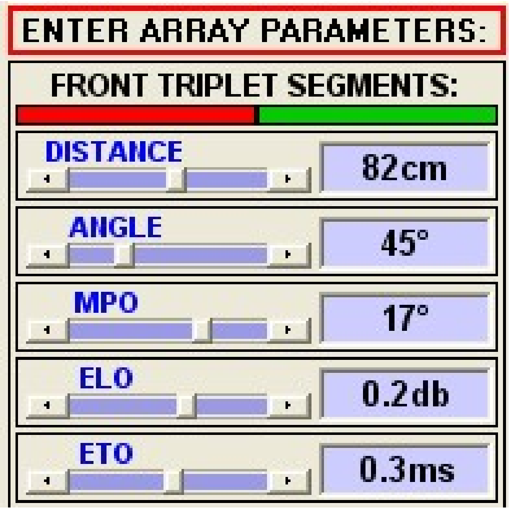

FIGURE 4 – ENTRY OF THE ARRAY PARAMETERS FOR THE FONT TRIPLET SEGMENTS OF A 3/5/6/7/8 CHANNEL ARRAY

The parameter entry values will be applied to each set of symmetrical segments of the symmetrical array i.e. equivalent segments on the left and right hand side of the vertical line of symmetry.

The basic design parameters of MPO, ELO and ETO have been defined and explained in some detail in previous papers – AES preprints 4997(4) and 5157(5) .

MPO, ELO or ETO will produce a clockwise rotation of the right hand front segment coverage, and an anticlockwise rotation of the left hand segment coverage.. On the other hand, negative MPO, ELO or ETO will produce the opposite – anticlockwise rotation of the right hand segment coverage, and a clockwise rotation of the left hand segment coverage.

3.3 Segment Coverage Display

In addition to the segment coverage shown on the compass rose of the central part of the main screen, a numerical display on the top right hand of the main screen (Figure 5) shows the numerical values of limits to the segment coverage and the overall coverage of the individual segments.

SEGMENT COVERAGE				
359	<<<	46	>>>	45
44	<<<	46	>>>	90
89	<<<	46	>>>	135
134	<<<	46	>>>	180
180	<<<	47	>>>	227
224	<<<	46	>>>	270
269	<<<	46	>>>	315
315	<<<	46	>>>	361

FIGURE 5 - SEGMENT COVERAGE DISPLAY FOR A 8CH ARRAY

Coverage displays are shown in a clockwise order. Colour coding is also used to identify each segment with respect to the segment microphone coverage shown on the compass display of the main screen.

Display values have been rounded up or down to integer values so inevitably there can be a one degree approximation in coverage limits. And anyway, even though parameter entry controls, microphone coordinates and coverage angle displays seem to show a certain precision, the psychoacoustic information which is used to determine especially the coverage angles is based on statistical psychoacoustical measurement, and are therefore only accurate to within a few degrees.

3.4 – Microphone Coordinates Display

Figure 6 shows the microphone orientation and coordinates of each specific microphone. Again although coordinate values seem to show a certain precision, every coverage calculation is based on psychoacoustics, so these values can be approximated in 0.5cm steps without much change in the overall coverage angles.

The reference zero for the coordinates is always the centre front facing microphone, except for stereo and quadraphonic arrays, where the zero 'X' coordinate is in the centre between the left and right microphones.

MICROPHONE POSITION COORDINATES AND ANGULAR ORIENTATION:	
Centre Microphone (C) Zero Reference of Coords.	
Xc = 0cm Yc = 0cm	Angle 0°
Left Microphone (L)	
xL = 0cm yL = -31cm	Angle 315°
Right Microphone (R)	
xR = +74.8cm yR = -31cm	Angle 45°
Left Surround Mic(Ls)	
xLs = -105.8cm yLs = -105.8cm	Angle 270°
Right Surround Mic (Rs)	
xRs = +105.8cm yRs = -105.8cm	Angle 90°
Left Back Microphone (Lb)	
xLb = -74.8cm yLb = -180.7cm	Angle 225
Right Back Microphone	
xRb = +74.8cm yRb = -180.7cm	Angle 135
Back Microphone (B)	
xB = 0cm yB = -210cm	Angle 180°

FIGURE 6 – MICROPHONE ORIENTATION AND 'X' & 'Y' COORDINATES FOR AN EIGHT CHANNEL ARRAY

The back microphone 'y' coordinate is always given as the distance between the front 'y' zero reference (or position of the centre microphone) and the position of the back microphone.

4.1 Loudspeaker Configuration Check

The main screen display concerns only the geometrical layout of the microphones and the associated segment coverage of the surrounding sound stage. However these segments will be reproduced within a specific loudspeaker configuration.

Viewing of the standard and non-standard loudspeaker configurations can be selected via the 'Check L.S. Config' button, and are presented in a separate window. Examples of various loudspeaker configuration are shown in Figures 7, 8, 9, 10, 11, 12, 13, 14, 15 & 16. Each segment of the loudspeaker configuration is again colour coded to correspond with the segment microphone coverage shown on the main screen.

Where appropriate this 'Loudspeaker Configuration Check' window allows switching between a number of different loudspeaker configurations that are either recommended by various institutional organisations, or, of interest as experimental listening configurations. However the recommended or 'standard' listening configurations are very often a compromise between demands of various media environments, and can be improved considerably if only one specific context is considered.

The loudspeaker configurations for Stereo (2ch) and Triphony (3ch) are shown in Figures 7 & 8.

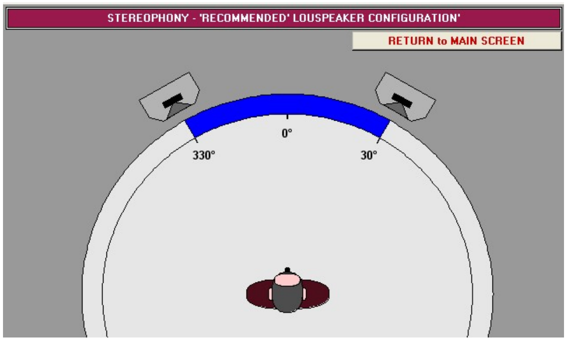

FIGURE 7 – STEREO LOUDSPEAKER CONFIGURATION

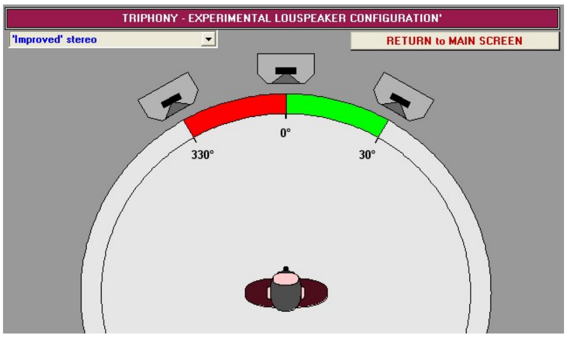

FIGURE 8 – TRIPHONY LOUDSPEAKER CONFIGURATION

These two configurations only allow front sound stage coverage. On the other hand, the opening screen configuration of 4ch - 5ch - 6ch - 7ch - 8ch arrays are initially configured for complete surround sound coverage. But of course it is possible to configure all these arrays so that they can be used also for front & side sound stage coverage. In this case the microphone array is considered to be sufficiently close to the front and side sound stage so that the back segment is without interest except for a certain amount of reverberation pickup.

Figure 9 is an equal segment Quadraphonic 4ch configuration for surround sound reproduction.

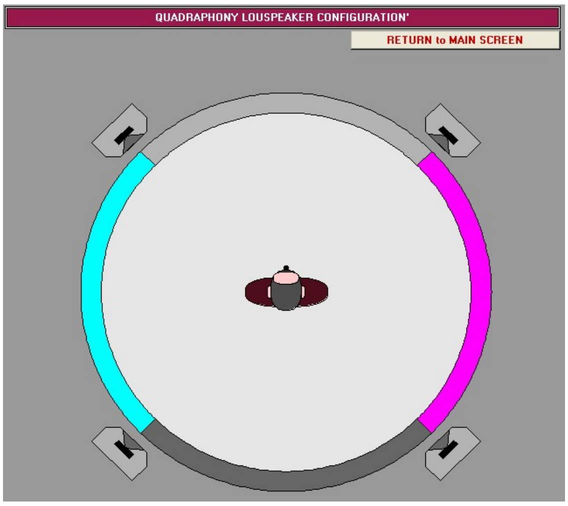

FIGURE 9 – QUADRAPHONIC LOUDSPEAKER CONFIGURATION

Each loudspeaker segment is 90° so a certain amount of angular distortion means that crushing of sound sources towards the loudspeakers is very noticeable. One possibility for a 5ch system is shown in Figure 10.

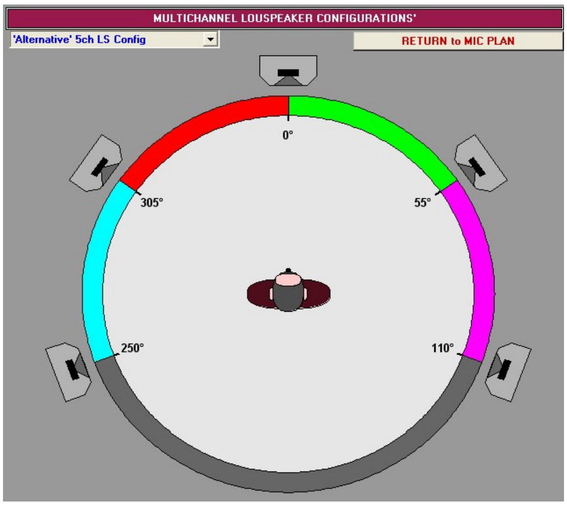

FIGURE 10 – 'MULTICHANNEL' LOUDSPEAKER CONFIGURATION

This is the standard "Multichannel" loudspeaker configuration (Recommendation I.T.U.-BS.775-1). This configuration is reasonably satisfactory as a audio reproduction support to audio-visual media, where the screen is the dominant element, but it is far from satisfactory for surround sound audio-only reproduction - the lateral and back segments suffer from considerable Angular Distortion of localisation (crushing of sound sources towards each loudspeaker).

The lateral segment reproduction can be improved by adopting equal front and lateral segments (see Figure 11 - Alternative 5ch Configuration) or by adopting the all-equal-segment approach (Figure 12) which will improve both lateral and back segment reproduction, but of course somewhat to the detriment of the angular distortion of the front segments.

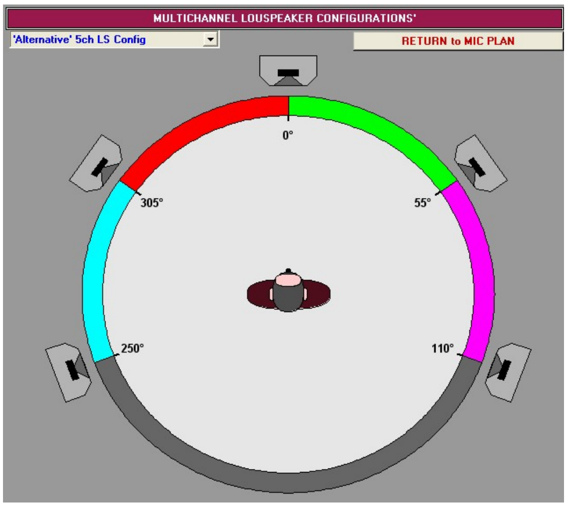

FIGURE 11 – IMPROVED AUDIO-ONLY 'MULTICHANNEL' LOUDSPEAKER CONFIGURATION

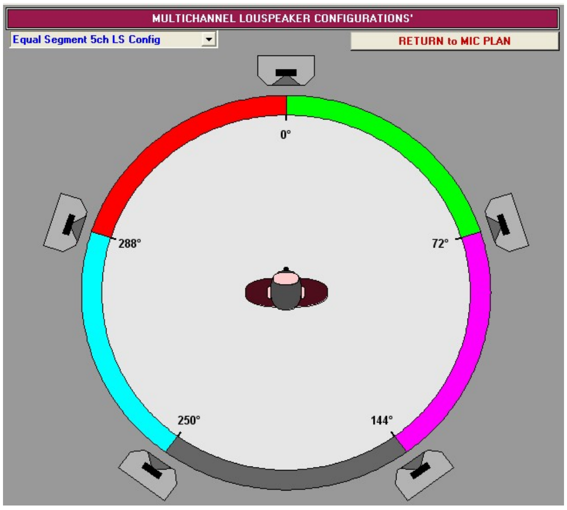

FIGURE 12 – EQUAL SEGMENT 5 CHANNEL LOUDSPEAKER CONFIGURATION

In Figure 13 and 14 simple equal segment six and eight channel configurations are shown which are certainly interesting as experimental configurations but at present there are no commercial products using either of these two formats.

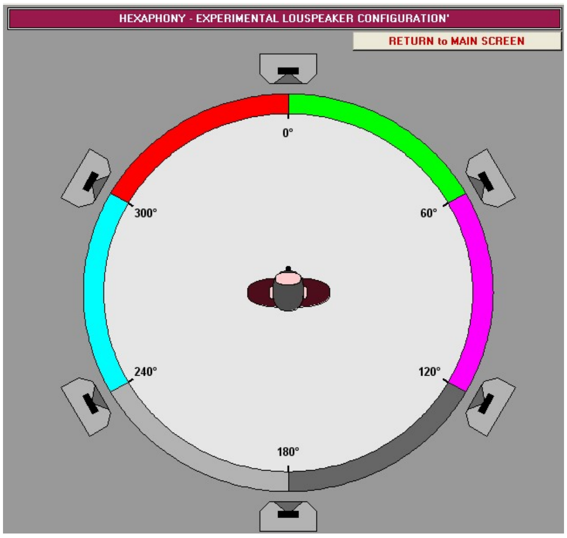

FIGURE 13 – EQUAL SEGMENT 6 CHANNEL LOUDSPEAKER CONFIGURATION

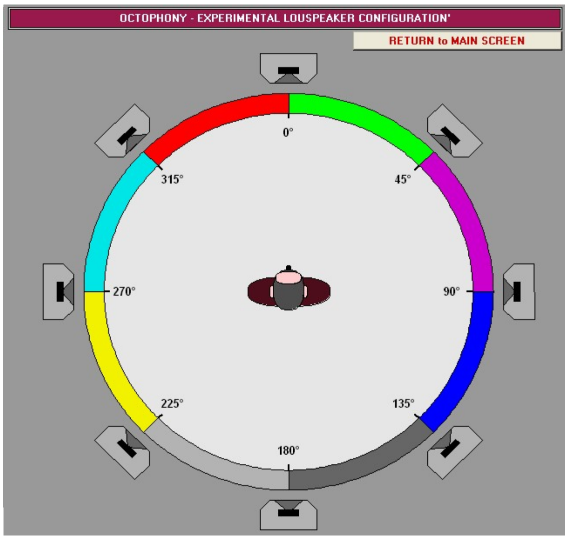

FIGURE 14 – EQUAL SEGMENT 8 CHANNEL LOUDSPEAKER CONFIGURATION

However the 7 channel loudspeaker configuration format, shown in Figure 15, could be considered as the most interesting surround sound format yet to be developed. This configuration is suitable for all Blu-ray media reproduction, either audio-only reproduction or audio-visual projection. It has the additional advantage of being compatible with 7 channel, 5 channel or stereo reproduction.

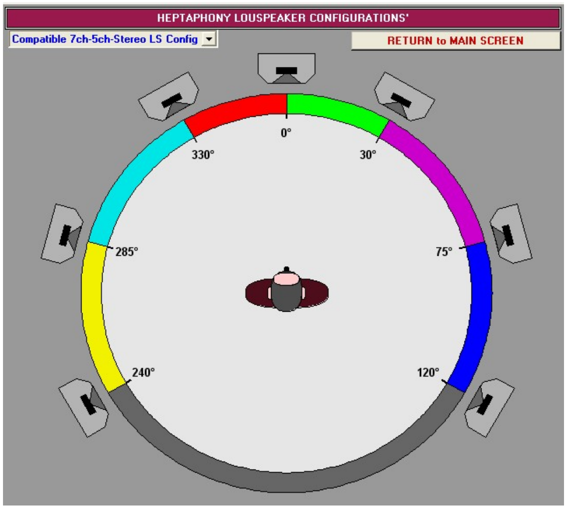

FIGURE 15 – COMPATIBLE STEREO/5CH/7CH LOUDSPEAKER CONFIGURATION

As with the 5ch ITU recommendation, the back segment suffers from considerable Angular Distortion of localisation, which can only be improved by adopting the 7 channel all-equal-segment configuration (shown in Figure 16), again somewhat to the detriment of the angular distortion characteristics of the front segments.

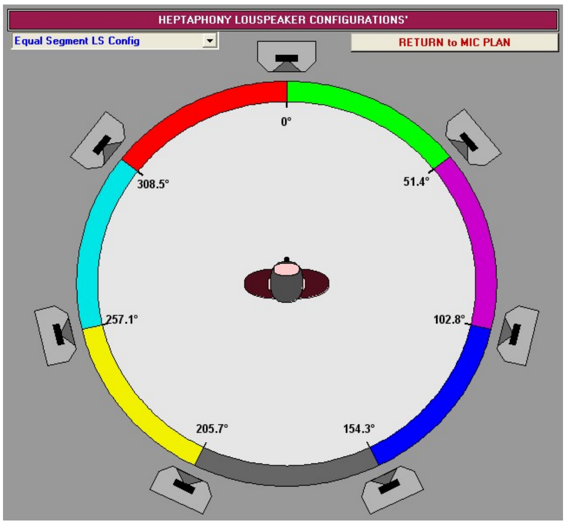

FIGURE 16 – EQUAL SEGMENT 7CH CHANNEL LOUDSPEAKER CONFIGURATION

This is an equal segment loudspeaker array specially suited to audio-only reproduction, but it is not compatible with 5ch and stereo systems due to the wider angles between the front triplet of loudspeakers. Angular Distortion (crushing of sound sources towards each of the loudspeakers) is however relatively small, and is of the same magnitude in each of the seven segments.

In the reproduction of 7ch recordings, whilst an equal segment loudspeaker configuration will inevitably give the best results for an audio-only reproduction, a compatible Stereo/5ch/7ch configuration would seem to be, in general, completely acceptable in both combined video-audio projection, as well as audio-only reproduction.

4.2 Intersection – 'Check dTdL curves'

The intersection between the physical characteristics of the microphones and the psychoacoustic measurements of the loudspeaker listening configuration determines the coverage characteristics of each segment. Much information can be gathered from the inspection of the actual level/time difference curves and their relationship to the psychoacoustic limits with respect to the loudspeaker positions on each side of the segment, the limit situation obviously being when no intersection is possible. If this situation is reached then the program will load the previous working solution thereby allowing the user to choose other parameters values that do allow intersection, and therefore the design process can continue.

Although the steering operators can seem somewhat similar, there is a subtle difference between the action of each offset mechanism. Whilst the MPO will 'pan' the segment coverage one way or the other, inspection of the dT/dL characteristics will show that the sound source localisation is displaced along the fixed transfer function curve. The 'panning' of the segment coverage can seem to be the same with either ELO & ETO but, whilst the position of localisation of the sound source along the transfer function curve will be maintained, the whole of the curve will be translated either along the level-difference axis for ELO, or along the time-difference axis for ETO. Sometimes one type of offset will not produce intersection whereas another type will produce intersection even though the segment steering obtained with both offset functions is the same.

It is therefore again a matter of juggling with each of these parameters to obtain an acceptable result. Figures 17, 18, 19, 20, 21, 22, 23 & 24 illustrate equal 'panning' but with different offset functions.

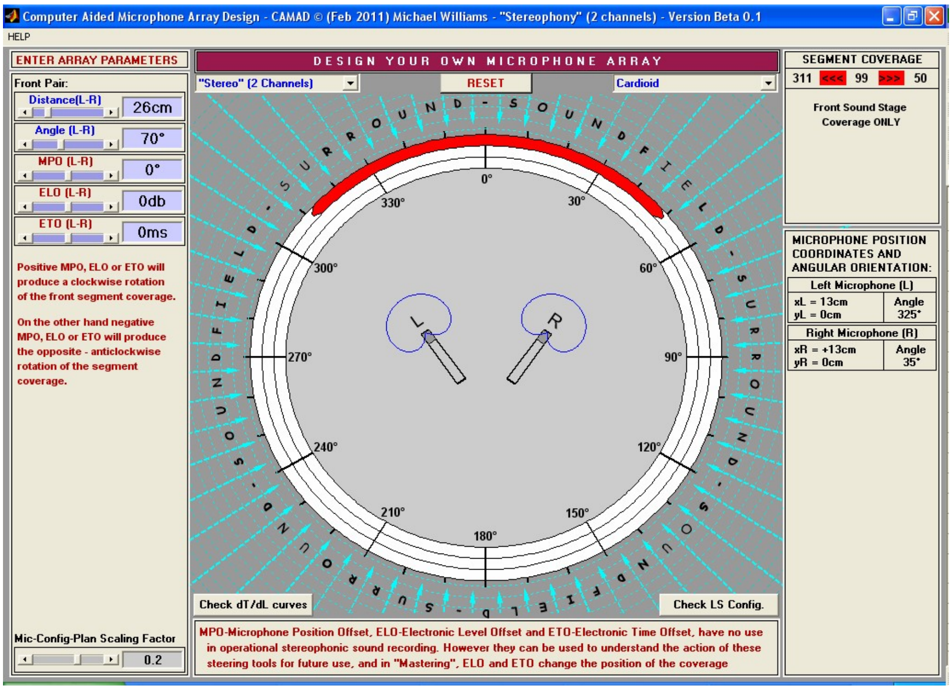

FIGURE 17 – SYMETRICAL STEREO (NO OFFSET)

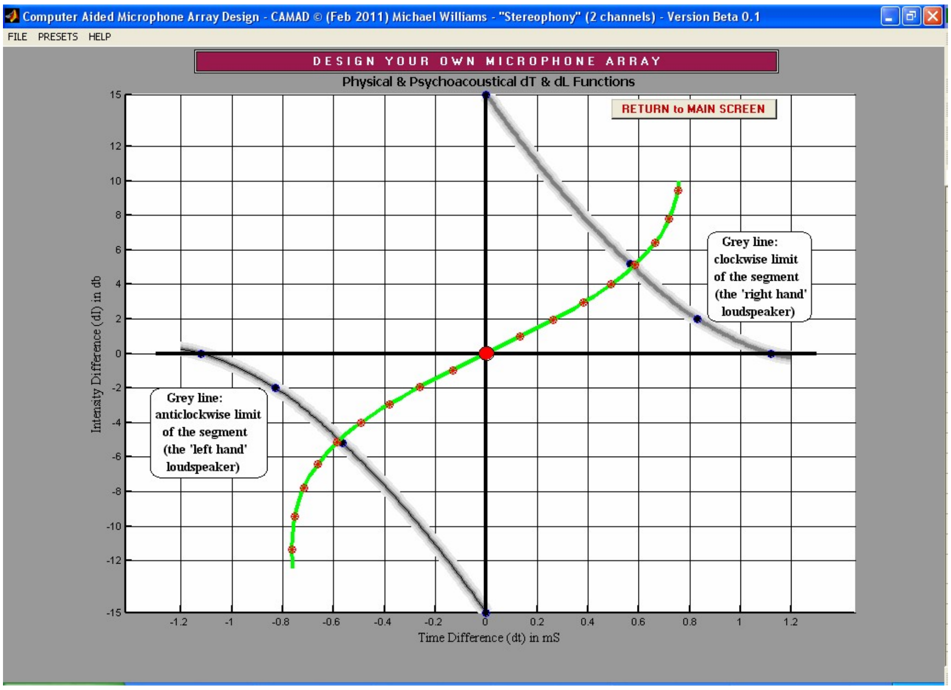

FIGURE 18 – dTdL CURVES FOR SYMMETRICAL STEREO (NO OFFSET)

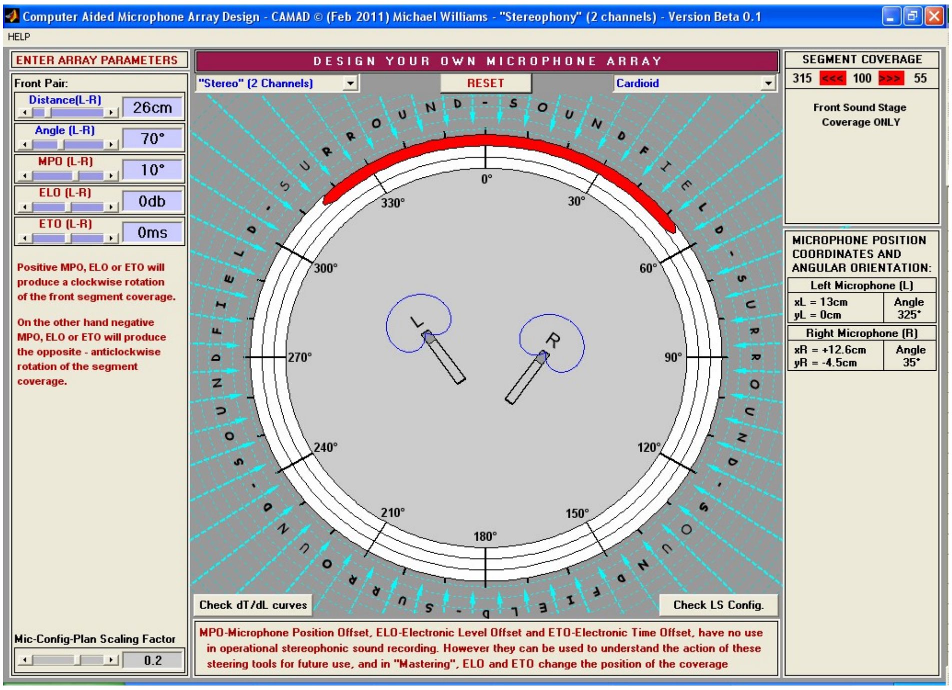

FIGURE 19 – 5° CLOCKWISE STEERING WITH 10° POSITIVE MPO

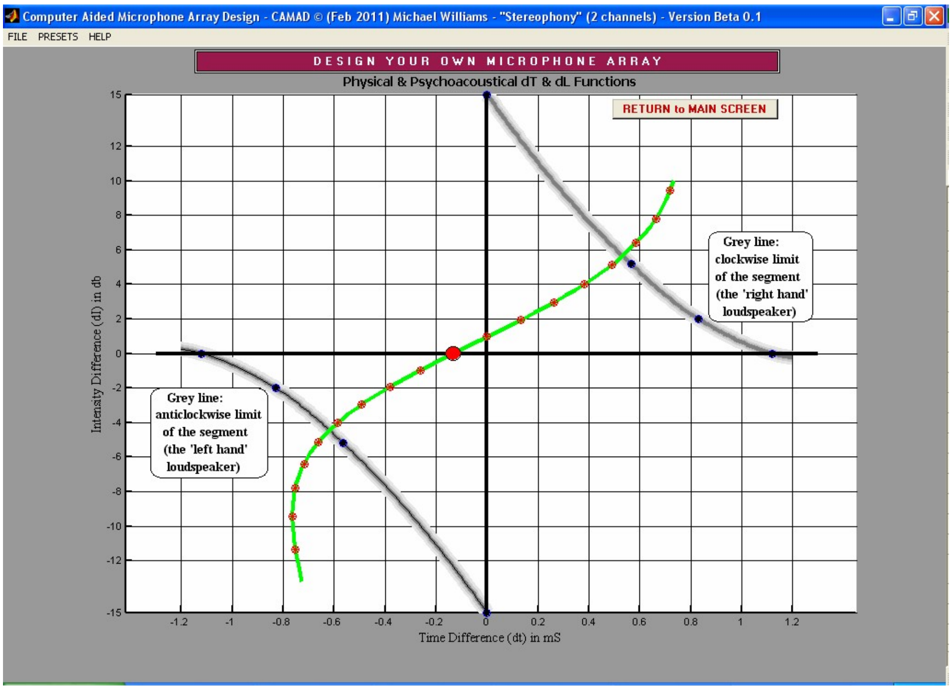

FIGURE 20 – dTdL WITH 10° POSITIVE MPO

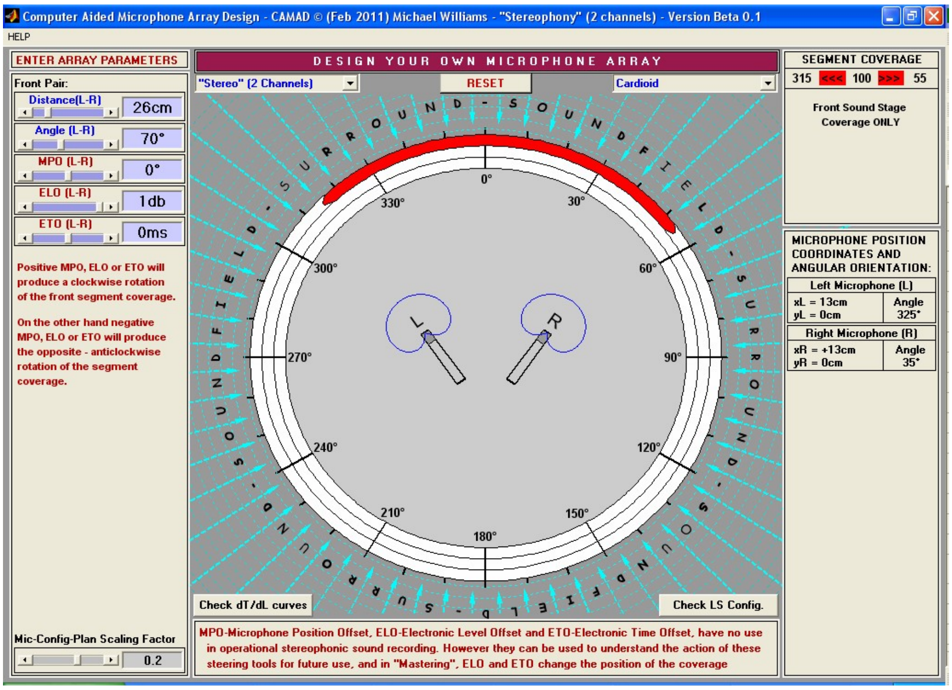

FIGURE 21 – 5° CLOCKWISE STEERING WITH 1db POSITIVE ELO

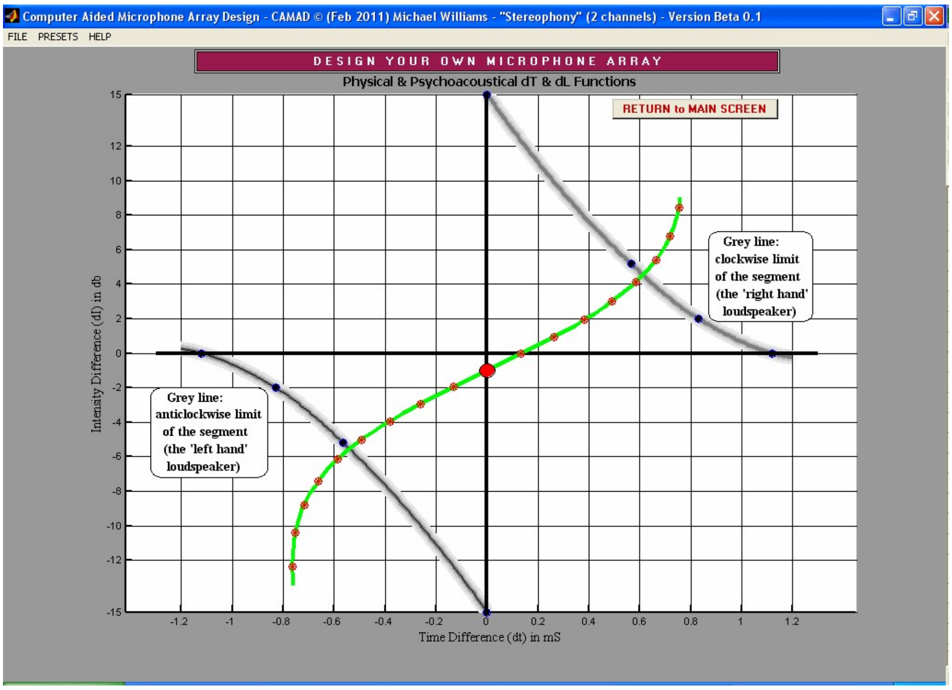

FIGURE 22 – dTdL WITH1db POSITIVE ELO

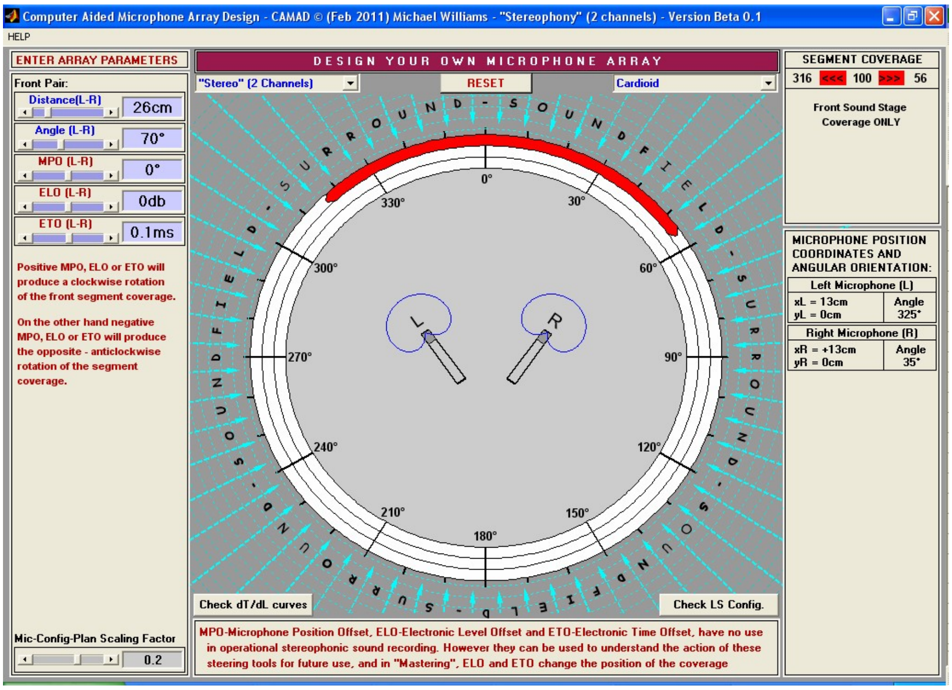

FIGURE 23 – 5° CLOCKWISE STEERING WITH 0.1mS ETO

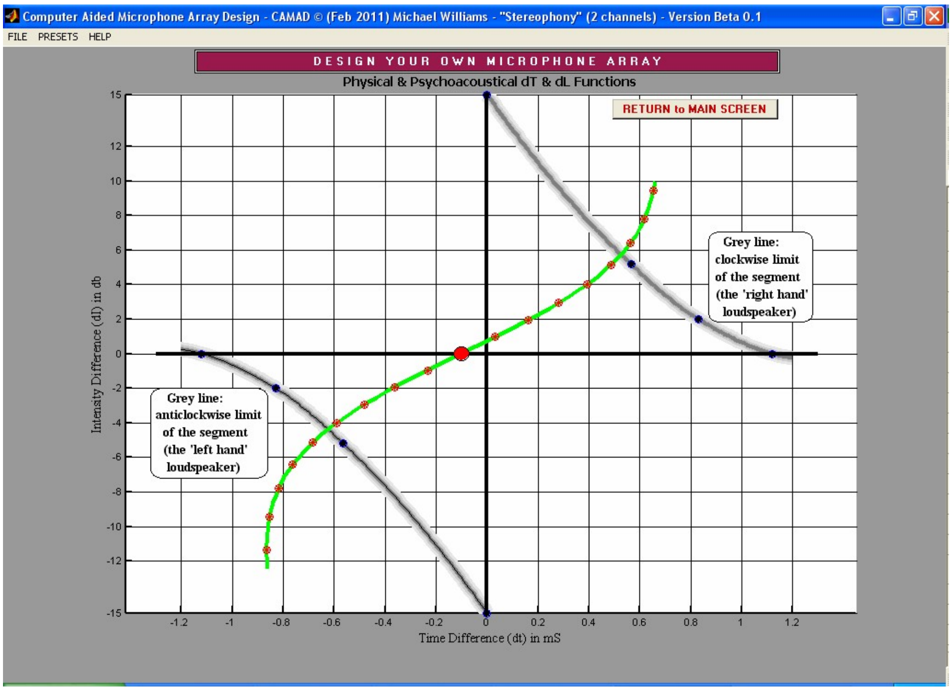

FIGURE 24 – dTdL WITH 0.1mS POSITIVE ETO

4. DROPDOWN MENUS

The usual drop-down menus are provided, plus a 'presets' selection, showing a large number of the known microphone array configurations. Specific user-defined microphone array designs can also be added to this list.

5. DESIGN TIME

Although this type of software allows the operational sound recording engineer complete freedom to design their own specific microphone array, it is far from being a rapid process. If rapidity is the prime factor then the design process shown at
http://www.mmad.info/MAD/Ch_n_cov.htm
is by far the quickest access to a specific design.

These website designs are for the time being limited to a maximum of 5 channels, and as yet, not all the different categories of parameter variants have been described. The advantage of this approach is that access time to a specific design is possible in about 10 sec, a specific choice from the many thousands of microphone array configurations is thereby accessible in a very short time.

Some of these designs were originally published as AES Papers (AES preprints 5336[8] and 6059[9]) where, for practical reasons, the designs were described as coordinates in table form – it would seem that a few users had difficulty in transcribing these coordinates to real designs! This was therefore the motivation for developing a CD-ROM-html approach (later transferred to the above cited website) giving access to a whole range of microphone array designs.

6. PSYCHOACOUSTIC MEASUREMENTS

For those who are familiar with this work, as it has developed over the years, it goes almost without saying that the migration of stereo microphone arrays to higher order arrays is based on the premise that the psychoacoustic measurements of the stereo loudspeaker configuration can be extrapolated to the surround loudspeaker configurations. In practice, this has generally proved near enough to be the case, but it has taken over 19 years for part of this premise to be verified in a little more detail. The work described by Simon and Mason in their AES paper[10] does examine this problem in relation to an octagonal loudspeaker configuration. It is of course limited to an eight channel loudspeaker array, and also does not take into account the recording angles and associated angular distortion generated by different types of microphone array (level difference dominant, time difference dominant or balanced hybrid). But it would be interesting to see this work extended to cover all the other surround sound formats, especially 5ch and 7ch systems. The Octagonal LS Array does also suffer from certain front-to-back perceptual ambiguities associated with its extreme symmetry – the positioning of the left and right side LS at 90° also does not help the perception of the virtual side images. It must also be accepted that the resolution of side images is also a function of our natural hearing characteristics and can therefore never produce the same quality of resolution as in the front (and back) critical perceptual zones.

The conclusion of the authors of this paper[10] is that localisation curves for the side segments can be somewhat different to the front stereo segment, and they recommend that different curves should be used to characterize the microphone array characteristics on the side segments. However "in the absence of such curves, the use of localisation curves measured on a loudspeaker setup having small differences of loudspeaker placement, scaled for the angles of the loudspeaker setup in use, can be acceptable compromise, depending on the precision of localisation required." In my experience scaling of loudspeaker segment angles has little or no effect on the overall coverage angle, but a considerable effect on Angular Distortion values.

I have made a multitude of recordings over the years using 7 channel arrays derived from the same approach as used to develop this software. The psychoacoustic characteristics described in this paper[10] would suggest that a certain amount of extra crushing (exaggerated Angular Distortion) should appear in the side segments. This has certainly not been my experience, but, as with all on-going research work, more detailed experimental work is needed.

If a reliable set of psychoacoustic transfer functions are published at a later date then it is certainly my intention to incorporate these functions into a dropdown menu for the possible choice of psychoacoustic initialization functions for each segment.

7. CONCLUSIONS

This paper should only be considered as an interim report on the subject of 'microphone array design'. The software should certainly facilitate the development of array designs for the individual user, and help widen the operational experience that is needed to validate this approach Further development of CAMAD will try to mirror the research in this field of sound recording as it becomes available.

7. REFERENCES

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