REAL WORLD LINE ARRAY OPTIMISATION

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1 INTRODUCTION

In a previous publication[1] we demonstrated that the inter-cabinet splay angles for a line array loudspeaker can be determined with the aid of direct search optimisation methods. This method was shown to provide a useful degree of abstraction from the often complicated elemental interactions. Rather than improving the array design through manual trial and error cycles, users could specify properties of the sound field which were translated into objective functions used by the optimisation routines. The major limitation was the radiation model, it was shown to be only valid at high frequencies.

Recent work[2] has improved the radiation model such that it can be relied upon over the entire audio frequency interval. In this work we start by applying numerical optimisation to the original splay angle problem of the earlier work and now include array height as an optimisation parameter. We then extend the scope of optimisation by examining the potential for including elemental transfer functions as parameters.

2 SPATIAL OPTIMISATION

All the results obtained are for a complete 2D section aligned with the array through the venue including the non-audience areas. Audience sections are sampled at a higher spatial resolution and the results are displayed on an 'index plot'. The index plot displays the SPL (relative to the average value at the mix position) over frequency at every point on the section enabling a rapid appraisal of the entire result. Contours are drawn with steps of 6dB. Also marked on the index plots are some dotted lines; black ones correspond to coverage start or stop positions and the blue one represents the mix position. Similarly coloured dots on the venue view diagram show where these positions are on the section.

The optimisation settings determine the influence of a particular property of the sound field during optimisation. High values of the 'Leakage' setting will tend to focus output into the audience areas, 'Smoothness' favors a flat frequency response. 'Profile' influences how the overall level varies with distance, this shape is determined by the levels at the coverage extents relative to that at the mix position. The values of the optimisation settings are displayed in the title of each index plot.

The array[3] used throughout this paper is made up from 16 small full range (65Hz - 20kHz) devices, each element being 115mm high with a choice of splay angles between 0° and 5° in 1° steps. The venue has two connected audience areas, the first is flat extending 20m from the array and the second on an incline around 30m from the array.

The optimisation procedure follows that of the earlier work except that all arrays are 'polished' as a final step. This 'polishing' step takes the best convex array found from a generally very rapid optimisation where array shape is constrained as the starting point for an optimisation that operates directly on the splay angles. Arrays determined in this way often produce surprising array shapes. The splay angles and array height for each case are shown on the venue plot figures.

2.1 Splay Angles

Given a fixed mounting height, how does the shape of the array change for different sound field objectives? We examine the 'profile' property first and set the target levels (relative to that at the mix position) at coverage stop and coverage start to both be 0dB. This means we desire no change in level as we walk from just in front of the array right the way to the furthest seat. We can see from Figure 1 that this is quite a difficult objective to meet and that response flatness has suffered.

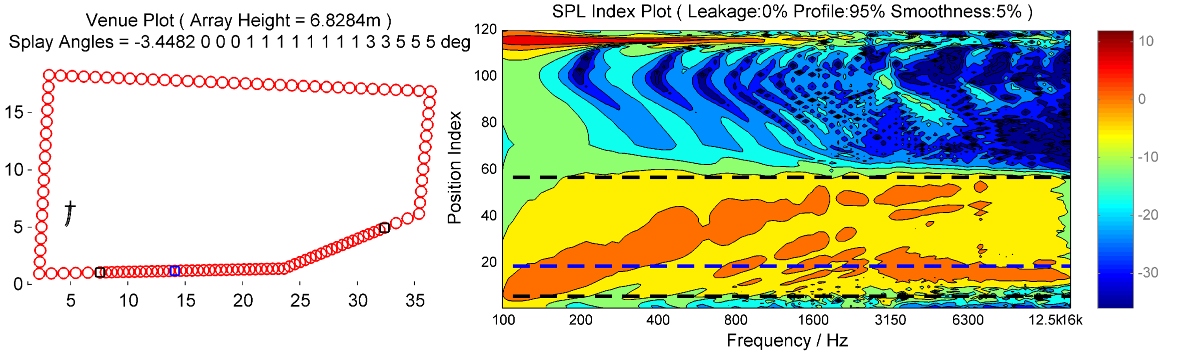

Figure 1 *Fixed Height, High Profile Setting*

For the results shown in Figure 2 we set the 'Smoothness' property much higher. The 'profile' property has suffered, the array now produces higher SPL at coverage start than at coverage stop. However, this array has flatter responses throughout the venue and a more realistic level profile over the audience areas.

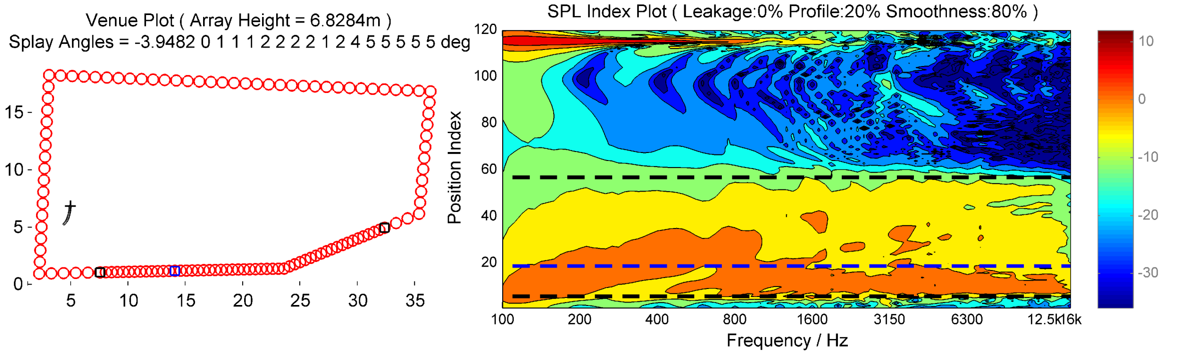

Figure 2 *Fixed Height, Profile with High Smoothness Setting*

We can achieve even flatter responses through the venue but at the cost of a reduced 'throw', this is shown in Figure 3 where the 'smoothness' property is set at 100%. Although the responses are now impressively flat at most positions in the venue, it is perhaps a little too quiet at the back.

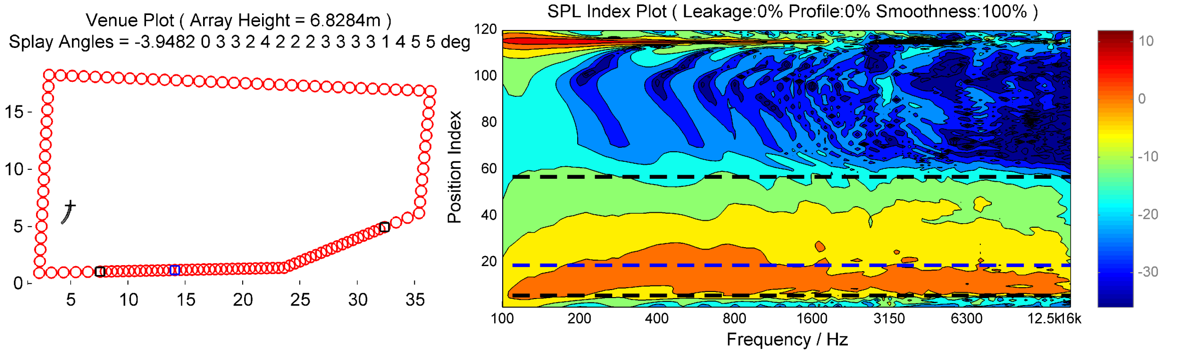

Figure 3 *Fixed Height, Maximum Smoothness Setting*

2.2 Splay Angles and Array Height Combined

What happens if we include array height as an optimisation parameter? The best way we have found to incorporate this extra parameter is via a 'nested' optimisation. The outer level of optimisation is just for the height parameter, at each iteration of the height a convex (array shape) constrained splay optimisation is performed. Once the height value which produces best result is obtained the splay optimisation is re-run and polished as before. In order that the whole procedure completes in a reasonable time (around 3 minutes), the height calculations are performed at a much lower spatial and spectral resolution which is then automatically stepped back up again for the final splay calculation.

An example is shown in Figure 5, here the 'Profile' and 'Smoothness' settings were given equal weighting. The small cross on the Venue Plot indicates where the array was initially manually placed. The array has risen by 1m and the resulting sound field displays a generally flat response and good 'throw'. It is interesting to compare this result with the fixed case shown in Figure 2, the higher array combined with some more emphasis on response flatness has produced a significantly better array, particularly at high frequencies and distant audience positions (higher position index).

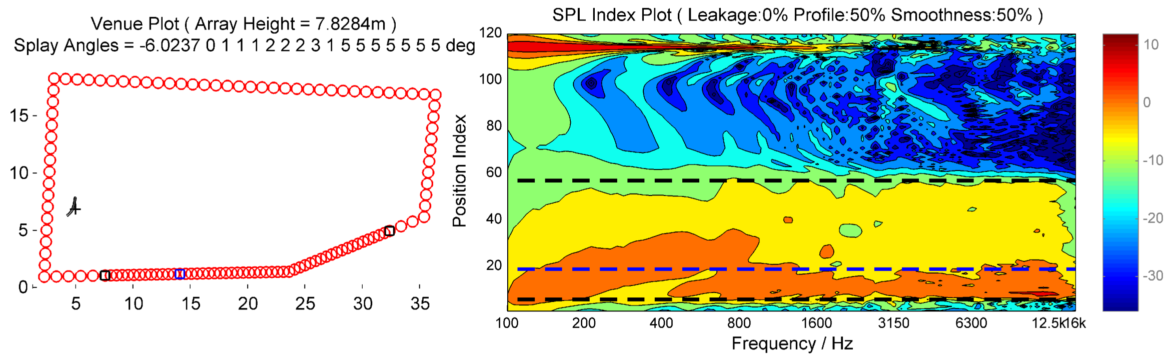

Figure 4 *Variable Height Mixed Profile and Smoothness Setting*

Finally an attempt at an even smoother array. Figure 5 shows that the array has dropped considerably to produce flatter responses throughout the audience areas. However, as in the fixed case the levels at the extremes of coverage are not ideal; too quiet at the back and this time too loud at the front.

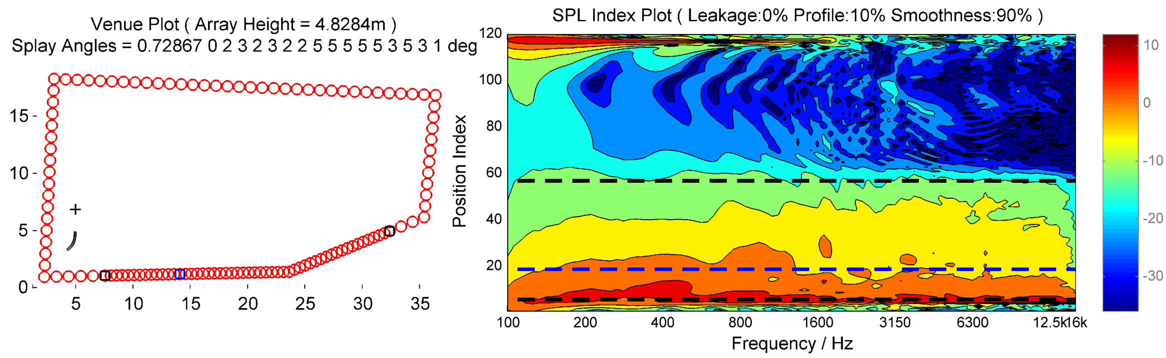

Figure 5 *Variable Height High Smoothness Setting*

3 SIGNAL OPTIMISATION

The idea of adjusting complex elemental driving functions of arrayed transducers to influence the near and far field directional properties of the array is certainly not new, nor is it confined to just radiation or indeed the field of acoustics. Methods to tilt and control the width of beams from an array are well established[4]. Amplitude windows applied to a complex aperture can control the beam shape and side lobe level. We can use traditional window shapes or optimise elemental amplitudes using the Dolph-Chebyshev method to attain a minimum side lobe level for a given far field beamwidth. Linear and quadratically varying phase response along the aperture control tilting and focusing in the near field.

Realisations of this type of array spatially sample the aperture in either a simple uniform way[5] or at more elaborate non-uniform positions[6,7].What these implementations have in common is that the aperture is generally planar, the elemental sources are small compared to the radiated wavelength and elemental source density is high (spaced at half the minimum wavelength for uniform sampling). These characteristics often restrict either the upper frequency limit of controlled radiation or overall output capability and for these reasons such arrays are well suited for background music and/or speech applications.

Line arrays used for live sound reinforcement are required to deliver very high SPL, which generally implies the use of highly efficient horn loaded drivers. Such arrays are composed of widely spaced elements that sample a non-planar aperture, each possessing an increasingly narrowing vertical dispersion with frequency. The narrowing vertical pattern of the elements greatly lessens the amplitude of the grating lobes due to the directivity product theorem[8], but simultaneously limits the bandwidth where electronic tilt of the array is possible. However, as we have seen in section 2 array shape can generate good coverage from uniformly driven elements. This section is not about steering or beam-forming, it seeks to explore how much improvement can be made by individually controlling elements of a high power live sound line array.

3.1 Global Signal Optimisation

Before we look at the unique elemental transfer functions we should make mention of the equalisation applied to the arrays of section 2. Here all the elements are driven uniformly and the question is: 'What is the best equalisation to apply to the system as a whole?'. We could just use the response at the mix position to tailor the spectrum. In doing so we would ignore the spectral balance at every other position. The method we have implemented makes use of the average response shape in the audience areas and a 'house curve'. The house curve is a user defined response which represents a user's ideal balance and is defined with familiar analog type filters (Ramp, Shelves, PEQs etc). Given this target and the average response in the venue it is a simple matter to develop an optimisation that brings the average very close to the house curve. Figure 6 shows a typical house curve and some of the filters used to set it and Fig 7 the resulting global EQ.

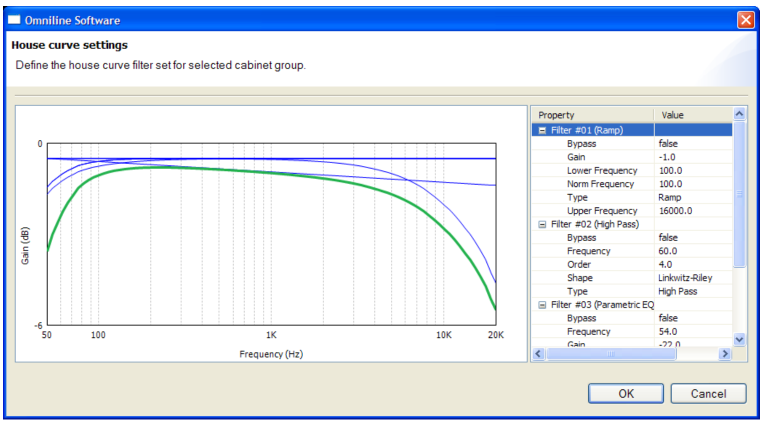

Figure 6 *Typical House Curve*

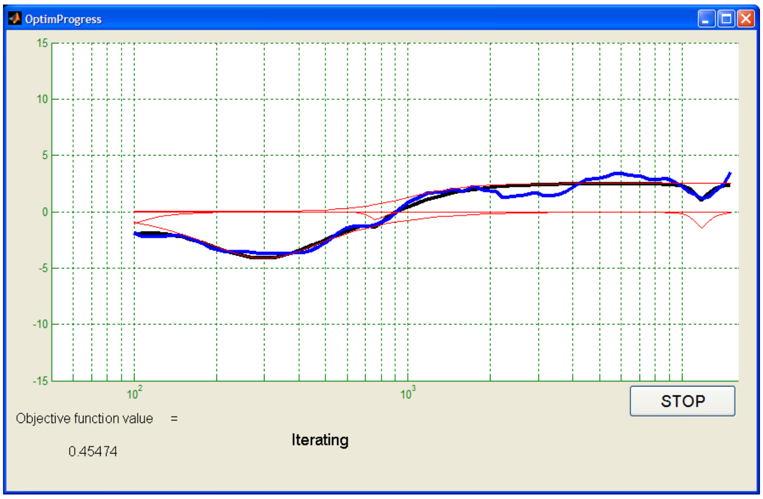

Figure *7 Typical Global EQ (for a 24 box array)*

3.2 Elemental Signal Optimisation

In a previous publication[9] we introduced the concept of 'band-zoning' which attempts to compensate for the effect of high frequency air attenuation. It was shown that subdividing the array into different physical bands; HF(upper, centre, lower), MF(upper, lower), LF(uniform) and applying different signals to each band could achieve a more consistent spectral balance over the venue audience areas. These settings were determined manually for each length and curvature of array – a time consuming task. In this section we extend and automate this approach by allowing each array element to have arbitrary complex signal input. The physical array used in the following examples is the one shown in Figure 4.

3.2.1 Starting Point

For reference Figure 8 shows the index plot and uniformly spaced samples of the frequency response throughout the audience areas with no equalisation.

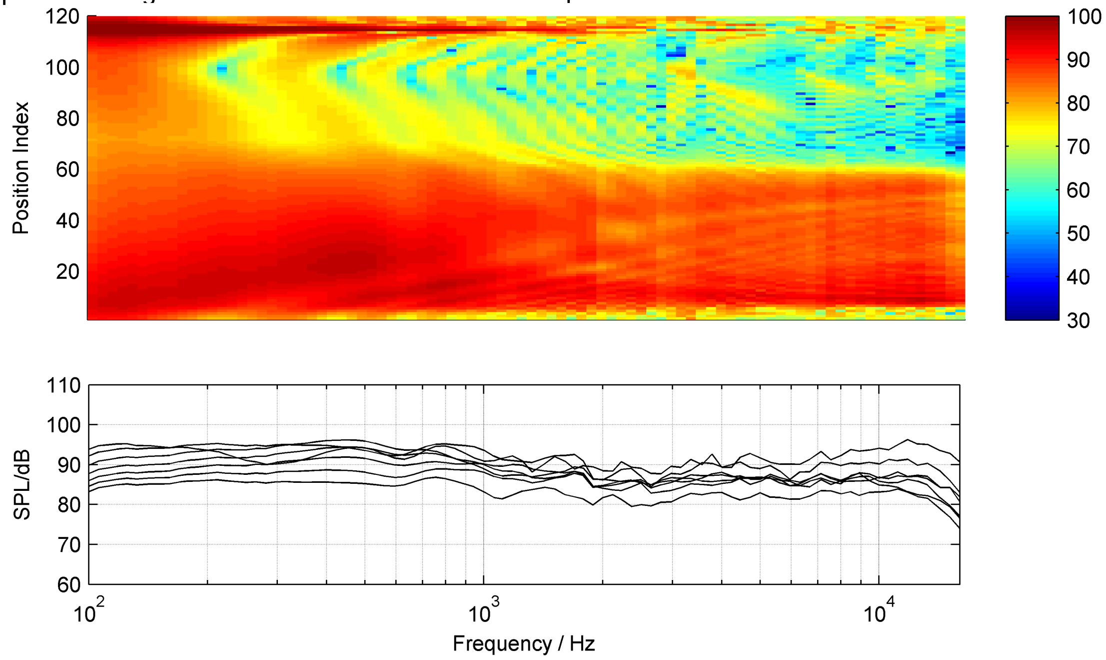

Figure 8 100-16000Hz Starting Point – No Equalisation

3.2.2 Flat Response

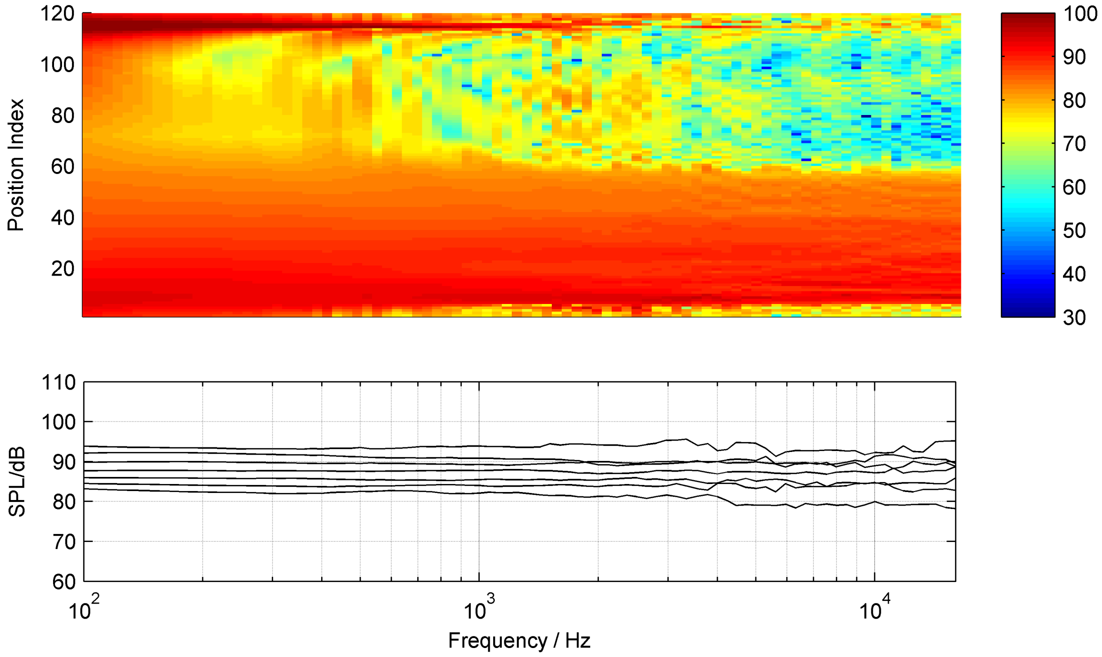

Figure 9 *100-16000 Hz Complex Elemental Signal Optimisation(flatness only)*

3.2.3 Uniform Level and Flat Response

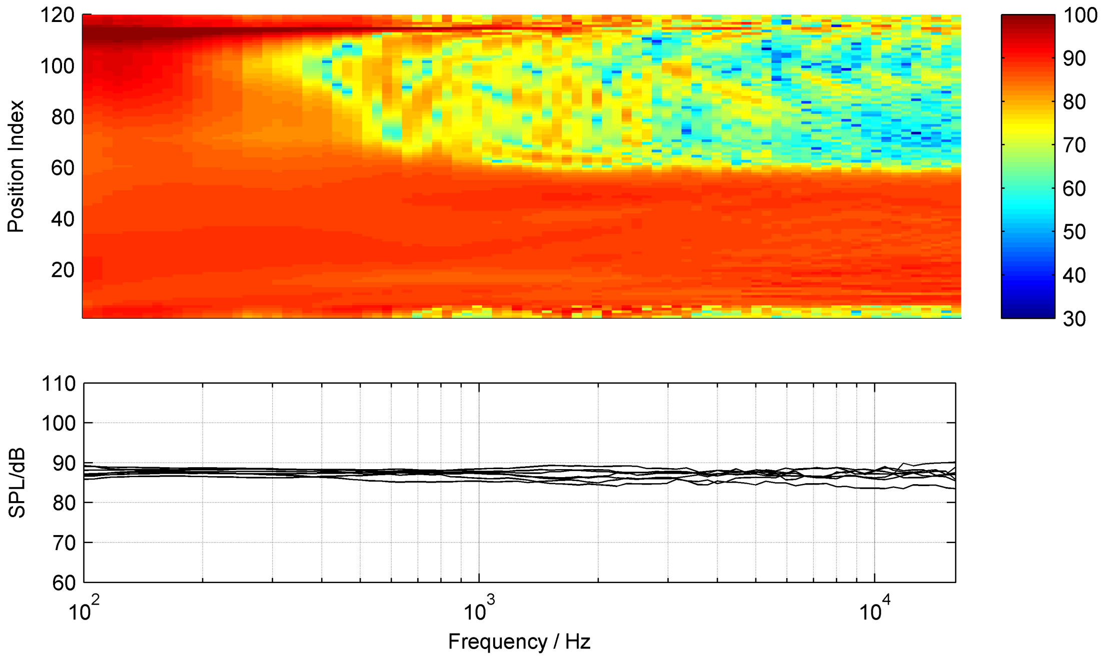

Figure 10 *100-16000 Hz Complex Elemental Signal Optimisation (flatness + profile)*

3.2.4 Minimum Leakage

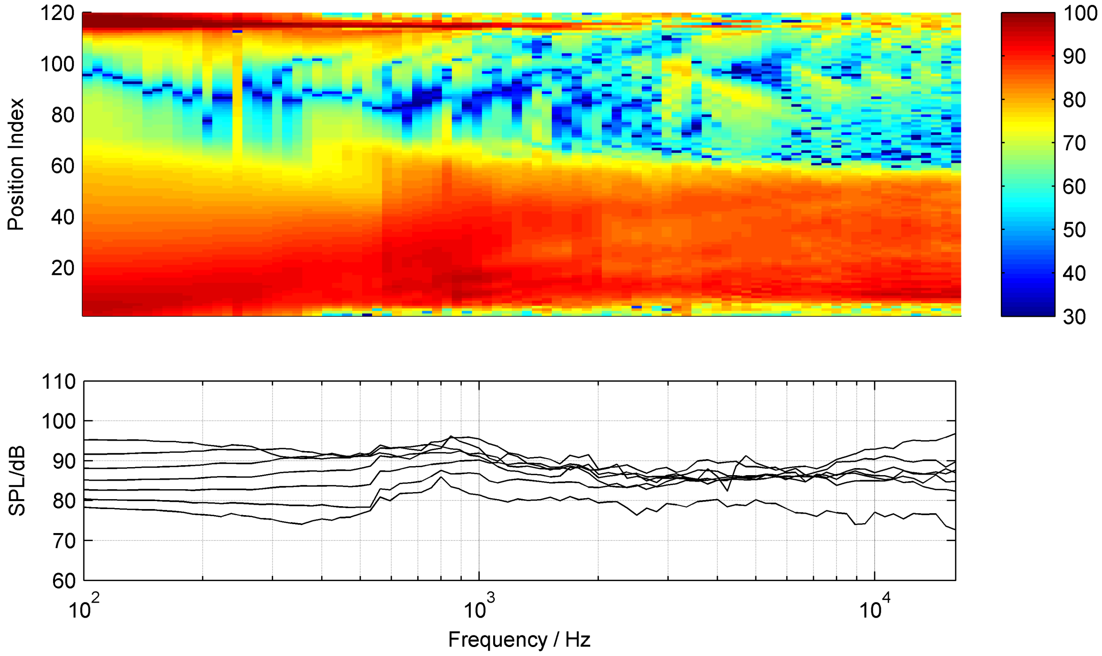

Figure 11 *100-16000 Hz Complex Elemental Signal Optimisation (leakage + v.low flatness)*

4 DISCUSSION

The results from section 2 demonstrate that varying array shape gives us a high degree of control at mid and high frequencies. At lower frequencies array length and overall array aiming angle are the dominant means of control. The relationship between mid/high output to the low frequency beam can be determined by splay optimisation; producing either a matched output (flat frequency response throughout audience) or a gently changing balance from front to back. The user generally has a choice between 'throw' and constant spectral balance over the audience. Using both 'smoothness' and 'profile' settings during the optimisation gives the user a simple way of dealing with this compromise.

Where the user has choice over array trim height including this in the optimisation may be useful. Lower arrays can be made to have the flattest response though this produces too high an SPL close to the array and too quiet at distant positions. Changing the shape so that the 'throw' improves produces a higher degree of low to mid/high mismatch than would be the case if the array were higher. Where this can be tolerated low arrays have other benefits such as being louder and having less leakage primarily due to it being closer to the audience area. More often the user can use a very restricted range of trim height due to sight lines or architectural considerations.

The results from the elemental signal optimisation in section 3 display some remarkable responses in the audience area. More strikingly than splay angle optimisation these results demonstrate that the objective functions are working. It is worth noting that the response flatness deteriorates with increasing frequency in Figure 9. We have found this to be dependent on the starting level distribution, beginning the optimisation at 3kHz produces better results. Perhaps the most impressive is the uniform level and flatness result, an audience member walking from underneath the array to the most distant point 30m away would perceive practically no change in spectral balance or level of the direct field. The cost for this performance is an increased amount of leakage particularly at low frequencies, however, Figure 11 indicates that we have the means to trade this performance against reduced leakage.

Both splay only and splay with elemental equalisation are very cheap solutions in terms of unique amplification/DSP channels. The splay only arrays merely requires one channel of DSP and one channel of amplification to achieve good quality broad-band coverage of a venue. Elemental equalisation increases the channel count to 16 but has the potential to produce remarkable results.

5 IMPLEMENTATION

5.1 Software

All acoustic calculation and optimisation was performed in MATLAB[10] with extensive use of the matrix style of computation and the optimisation toolbox. The radiation calculation incorporated element position dependent balloon data[2] and solved problems, like the examples in this paper, in about 1s on a very ordinary laptop. MATLAB code also handled the mechanical analysis of the array using a very precise representation of each element and associated connecting mechanisms.

Sound professionals appreciate easy to use scalable applications as design tools, unfortunately MATLAB is not the best tool to create a rich GUI with. Our traditional approach[11] to building these tools was either to develop the physics inside the interface in for example C++ or to translate the code from the engineer's favourite language into that of the interface. This requires that the engineers are reasonably competent with the interface language and that the programmers are reasonably good at physics. Even when this is the case a great deal of time was spent debugging often minor mistakes in an engineer's code or a programmer misunderstanding something an engineer might have thought obvious.

Our aim for the new software is to separate the physics from the interface. The main program is a Java application that uses an exported library of MATLAB functions written and tested by the engineers. So far, this approach is working well producing rapid development and far less time spent debugging code. Figures 12-14 are screen shots from the new software

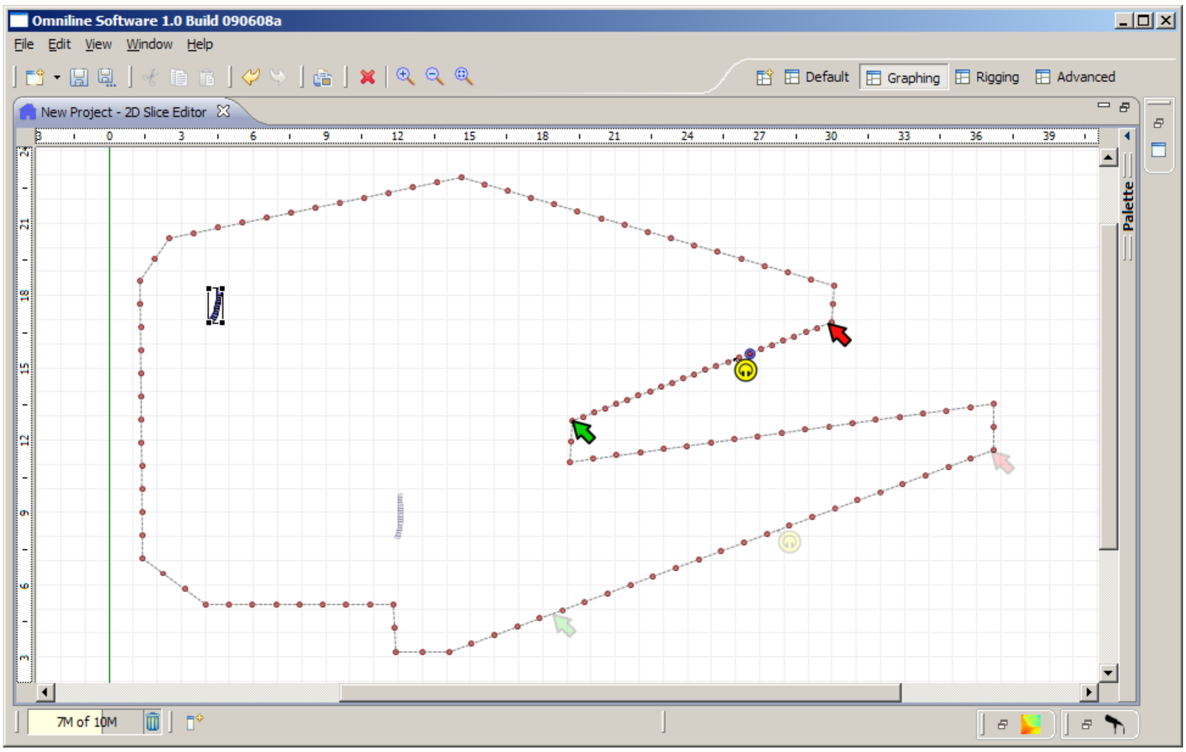

Figure 12 *Venue Entry screen*

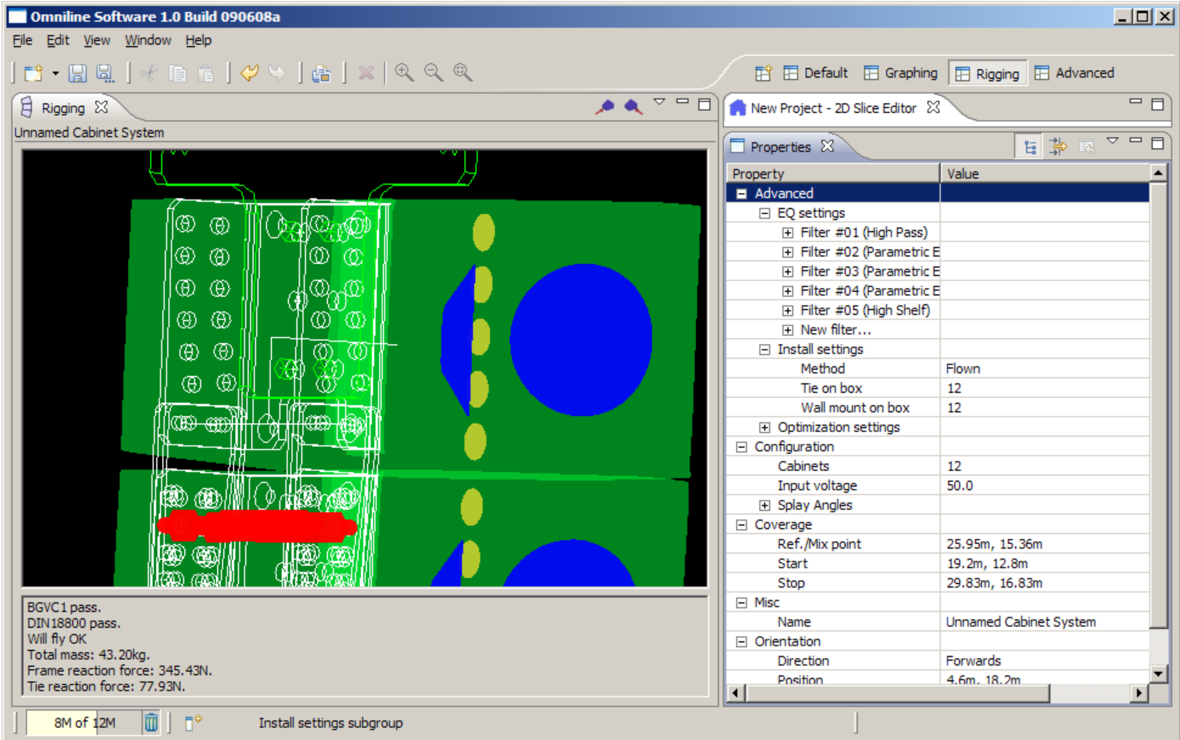

Figure 13 *Mechanical/Rigging Analysis View*

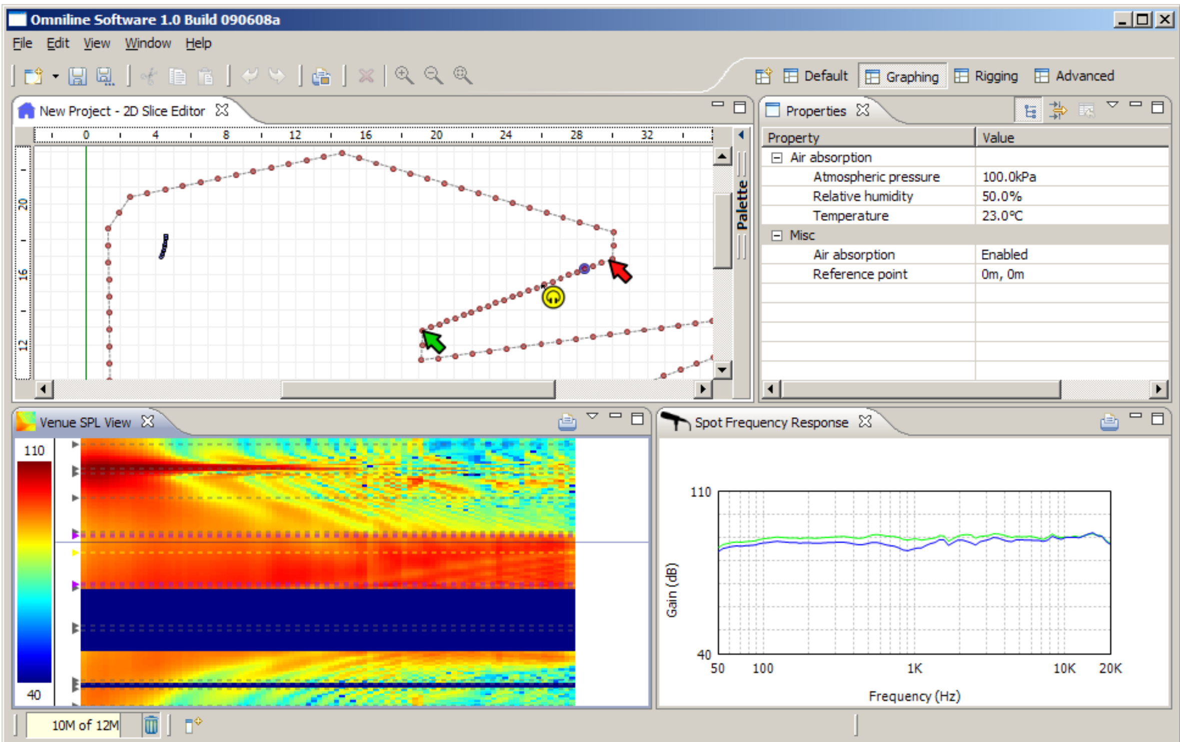

Figure 14 *Graphing/Design View*

5.2 Filters

The complex signals required to produce the uniform flat SPL result of Figure 10 are shown in Figure 15. Clearly combinations of simple analogue style filters are not able to generate such transfer functions with much fidelity. However, particular regions can be identified where phase and/or amplitude are having a dominant effect. This may suggest the path for future work aimed at finding a workable implementation of these transfer functions.

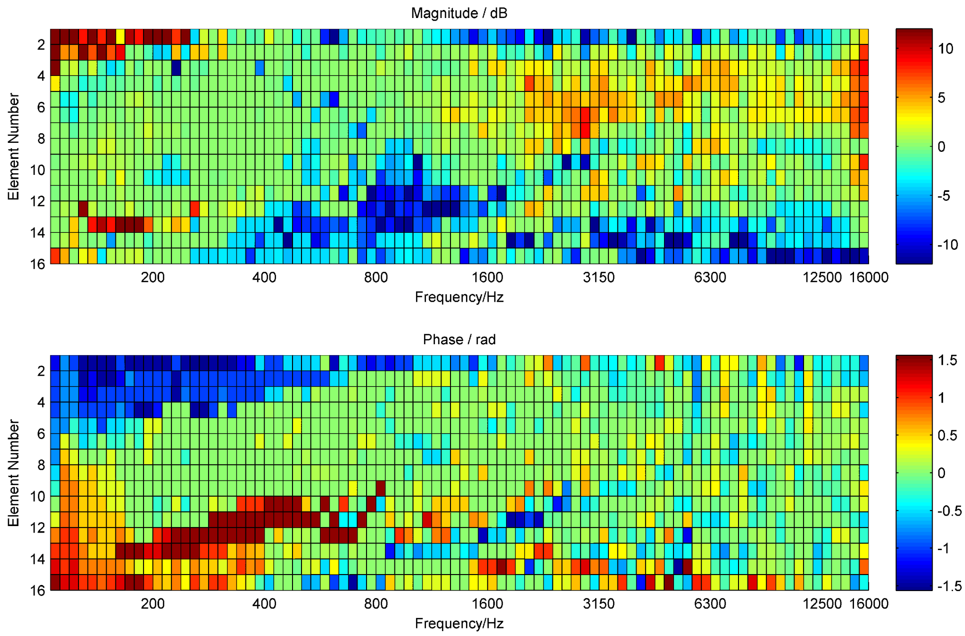

Figure 15 *Elemental signals used to produce the uniform flat SPL result*

5.3 Environment

We have assumed that the environmental conditions are constant. For indoor venues this has more chance of being valid but outdoors temperature gradients and wind are often significant. The radiation model must be improved to take these effects into account. It may be possible to partially compensate for these effects dynamically, based either on a model with environmental sensors or a direct acoustic measurement at relatively distant audience positions.

6 CONCLUSION

A simple method for users to determine the array shape and position has been presented that produces good results over a wide bandwidth. Users may trade various qualities of the sound-field from such simple arrays without the time consuming low level manipulation of splay angles.

To escape the restrictions imposed on performance at low frequencies of the simply powered array, elemental control can be used. Elemental control also offers the potential to produce almost perfectly flat responses throughout the audience with a particular desired level profile from front to back. Compared to the traditional sampled (uniformly or otherwise) planar aperture approach the elementally controlled curved line array can offer higher SPL output and wider operating bandwidth with less and more widely spaced elements.

More work needs to be done improving the radiation model to include environmental factors and in methods to generate the elemental transfer functions.

7 ACKNOWLEDGEMENTS

The author would like to thank Richard Cheney, Lead Developer at Martin Audio, for his work on the JAVA interface.

8 REFERENCES

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