ADVANCES IN LINE ARRAY TECHNOLOGY FOR LIVE SOUND

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In recent years, the line array loudspeaker system has become the dominant player in the touring sound industry. Line arrays are currently perceived to offer significant benefits over horizontally arrayed clusters, including a more consistent frequency response over the audience area, increased high frequency throw and reduced set-up time. This paper attempts to offer an insight into why the line array principle has been applied to live sound and explain some of the factors to be considered in implementing a practical design.

HISTORICAL OVERVIEW

Line arrays are not new. The principle of configuring a vertical column made up of closely spaced loudspeakers has been around for decades, principally implemented as column loudspeakers for use in the reverberant environments of churches and railway stations. Line arrays increase directivity in the vertical plane and produce a narrow vertical beam whilst the horizontal coverage remains the same as for a single device.

The line array principle is described in Olsen's 'Elements of Acoustical Engineering', first published in 1940. Olsen showed that the directivity of the line array increases with the length of the array, provided that the distance between the drivers is smaller than the wavelength being produced.

In live sound, the vocal columns of the 1960's were unable to compete with the screams of pop audiences and by the mid 1970's, live sound systems had evolved to use multiples of high efficiency horns to achieve the high SPL's and bandwidth required.

The horns were often configured with the lows, mids and highs in separate boxes. This meant that the bass bins could be blocked together to increase low frequency coupling and the mids and highs could be stacked vertically as line arrays to narrow their vertical coverage angle and increase throw (Fig. 1).

During the 1980's and 90's, this acoustically effective arrangement was largely replaced by horizontally arrayed clusters of identical 3-way boxes with the bass, mid and highs all housed within the same enclosure.

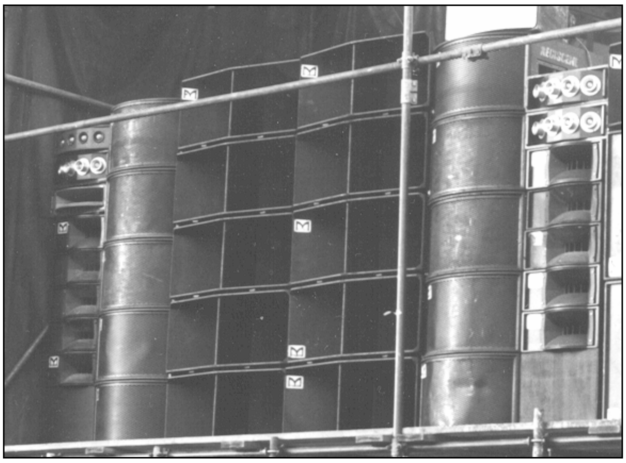


Fig. 1 The Sound of the Seventies

This arguably retrograde step was predominantly caused by a desire for conveniently packaged systems that could be flown over the stage rather than stacked. Horizontally arrayed clusters abandoned the notion of coupling adjacent elements acoustically in favour of an individual "point and shoot" philosophy. The main problem with these clustered systems was that the interference caused by multiple sources could cause big variations in the frequency response over the audience area.

Since there was no HF coupling between adjacent elements, the throw of the system was predominantly dependant on the performance of a single HF device and delay systems were nearly always needed to bring back the high frequencies beyond 50 metres. Also, as a consequence of unintended coupling at low frequencies, the lows and low-mids would build up, tilting the overall frequency response of the array downwards as the array increased in size, even if the individual box had a flat frequency response to begin with [1].

LINE ARRAY FOR LIVE SOUND

From the early 1990's, line array principles began to be re-applied to the problems of live sound – this time in a format that was more conveniently packaged and easy to fly. Line array is now the main technology in live sound reinforcement and is perceived to offer significant benefits over horizontally arrayed clusters - such as a more consistent frequency response over the audience area, increased high frequency throw and reduced flying time.

Fig. 2 W8L Line Array Enclosure

Fig. 2 shows a current Martin Audio W8L 3-way line array loudspeaker, which is normally flown in columns of up to 16 enclosures (Fig. 3). It is a 3–way system with a 1 x 15" bass horn, 2 x 8" mid-horn and a 3 x 1" HF horn. The efficiency of each band for a 1W input is 106dB for bass, 108dB for the mid and 113dB for the high. Crossover points are 220Hz and 2.5kHz.

Fig. 3 Flown W8L Line Array

The key to the increased high frequency throw of a line array for live sound is due to:

1. Smaller distances between each element - either horn or direct radiator, and

2. Much flatter wavefronts produced by those elements.

Put simply, if more output from each element adds together constructively with its neighbours, then more output will be available from the whole system or array. To illustrate this, the difference between the three closely spaced horns and three 30° horns spaced 1m apart (typical for a cluster) is shown in Figs. 4 and 5.

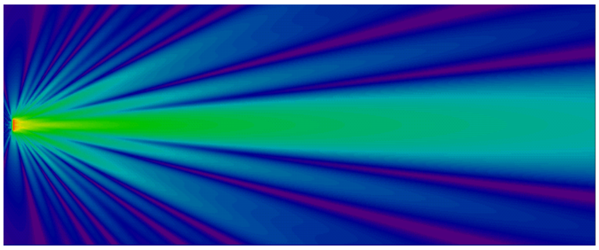


Fig. 4 W8L 3 x 1" HF at 8kHz, vertical dispersion

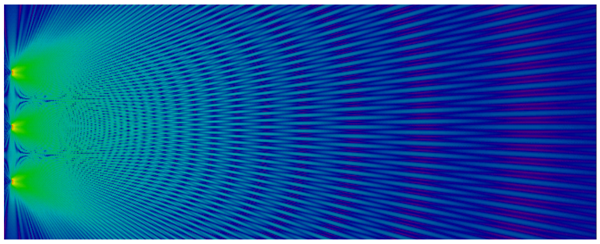


Fig. 5 Three 30º horns 1m apart, 8kHz

STRAIGHT LINE ARRAYS

In straight line arrays, this increased directivity in the vertical plane can result in coverage angles of less than 1° at high frequencies. Whilst this narrow beam might be suitable for aiming voice announcements in transportation facilities, it is of little practical use in live performance applications where few members of the audience would be in a position to benefit.

Much has been made of the notion that a straight line array produces a cylindrical wavefront with an output that falls off at 3dB per doubling of distance rather than the 6dB associated with a spherical wavefront that diverges in both planes. The increased throw by extending the "cylindrical" nearfield out to a greater distance has been promoted as one of the key benefits of line array technology.

There are two problems with this notion in practical systems. The first is that a cylindrical wavefront would require a floor to ceiling column in a typical stadium to cover both the floor and the highest seat.

The second is that only an infinitely tall line could produce such a cylindrical wavefront at all frequencies. A practical line source with a finite length will only approach "cylindrical" for a certain distance after which it will disperse in the vertical plane. Theoretically, with a continuous array 3m high the transition from 3dB to 6dB will occur at the following distances:

100Hz	500Hz	1kHz	5kHz	10kHz
1.3m	6.5m	13m	65m	130m

Whilst perhaps of academic interest, this cylindrical effect is of little practical use as it only really comes into play at high frequencies. If we remember that in a straight line array the vertical coverage may be less than 1°, it is clear that only a very few members of the audience could ever benefit from such a narrow beam.

CURVING THE ARRAY

To achieve the wider vertical pattern required to cover a typical audience area, line arrays for live sound are nearly always physically curved in the vertical plane. Adapting the line array principle from theoretical straight arrays to practical curved arrays has important implications for the acoustic design, physical deployment and electronic control of practical line arrays for live sound.

Firstly, we need to determine exactly what shape of curvature is necessary to achieve the desired directivity for a particular venue. Secondly, we need to determine the appropriate wavefront curvature of individual elements to avoid either too much overlap interference when the array is flat or gaps when it is highly curved.

Since these questions are much too complex to be answered by simple reasoning alone, a computer model was developed incorporating the acoustic and electro-mechanical characteristics of each individual low, mid and HF element and with each element driven by a virtual crossover. The phenomenon of air absorption of high frequencies over distance was also taken into account.

MODEL DEVELOPMENT

Developing the model used to simulate the performance of a curved line array involved:

i) Calculation of the pressure field from a curved source [2] [3] [4] [5]
ii) Compensation for air absorption [4]
iii) Manipulation of array element spatial variables
iv) Simulation of common equaliser functions
v) Measurement of waveshape curvature [5]
vi) Validation

i) Pressure Field Calculation

The most general description of a sound field due to a radiating isolated body is the Kirchoff-Helmholtz Integral Equation [2]. We are currently developing numerical techniques to implement this equation, however, in the meantime we can use an approximation. This method uses the modified Huygens-Fresnel principle, which is shown to give the same total result as the KHIE with the exception of 90° phase shift [3]. The principle states that "*every unobstructed point on a wavefront at a given instant in time, serves as a source of secondary spherical wavelets. The amplitude of the field at any point beyond is the superposition of all of these wavelets.*" The modification is to include a function called the obliquity factor that gives the secondary wavelets directionality. This deals with the reverse traveling wave implied by the spherical secondary wavelets.

$$K(\psi) = \frac{1}{2}\left(1 + \cos\psi\right) \quad \ldots 1$$

where

ψ = angle made with the normal of the primary wavefront

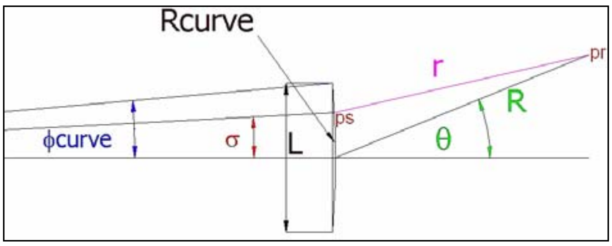


Fig. 6 Pressure Field Calculation

where

Rcurve = radius of curvature of the source
σcurve = ½ included angle of source – gives L
ps = a point on the source
pr = a receiver point
r = distance from ps to pr
ψ = angle that line (ps,pr) makes with source axis
R = distance from center of source to pr

θ = angle that line (centre of source,pr) makes with source axis

The pressure p due to the entire source is given by

$$p(R,\theta,k) = \int_{-\phi curve}^{+\phi curve} \frac{U(\sigma)\cdot K(\psi(R,\theta,\sigma))}{r(R,\theta,\sigma)} \cdot e^{-j(kr(R,\theta,\sigma)+\beta(\sigma))} d\sigma$$

…2

where

$\beta(\sigma)$ = phase along the curve – (set to zero)
$U(\sigma)$ = Amplitude along the curve

$$r(R,\theta,\sigma) = \sqrt{[R\cos\theta + Rcurve\cdot(1-\cos\sigma)]^2 + [R\sin\theta - Rcurve\cdot\sin\sigma]^2}$$

Usually the r in the denominator of the integral is taken outside by assuming far field conditions. We can leave it in since computing is cheap and we are interested in the near field as well as the far field.

With the equation we can sum the outputs of multiple curved sources for a series of receiver points and include additional gain terms and account for air absorption.

$$P(R,\theta,k) = \left| \sum_{n=1}^{NumBox} p(R_n,\theta_n)\cdot H(k)_n \right| - AAtn(R_n)$$

…3

In this instance we can ignore the very small error associated with using R instead of the more precise r in the air attenuation function.

ii) Air Absorption

The procedure for determining the air absorption coefficient in dB/m is described in [4]. The method has been implemented fully to produce the air attenuation function in the above equation

iii) Spatial Variables

In order for the model to be useful the variables need to correlate to practical physical parameters. Each array element was defined in space by the following:

Height of box.
Depth of box.
Length of non-radiating dead space between boxes.
Depth offset of horn mount.
Number of horns per box.
Inter-horn splay angles.
Curvature of individual horn element wavefront.
Height of top box above ground.
Angle of grid.
Inter-box splay angles.
Transfer function H(k) for each module.
Definition of receiver points – plane/polars/paths.

In addition to these there is frequency, temperature, humidity and EQ. The entire model was implemented in MathCAD for speed of development rather than execution.

iv) EQ

We have replicated the core filter functions of the Martin Audio DX1 to provide data for the transfer function term in eqn 3.

v) Waveshape measurement

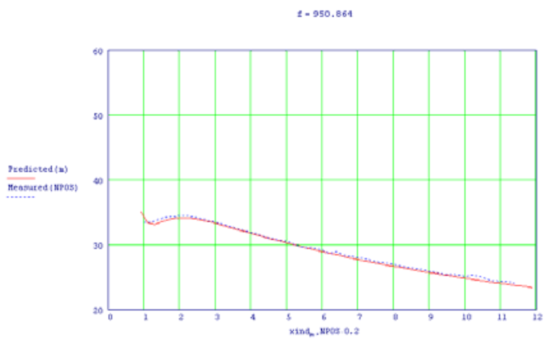
We used a technique based on that found in [5] where the phase distribution at the mouth of the horn was measured with the aid of a small microphone and precise positioning jig. Measurements were made with MLSSA and a program was written to convert the output files into a MathCAD compatible format. The phase data was then unwrapped and displayed in 2 and 3D for analysis. This provides the value of Rcurve in eqn 2.

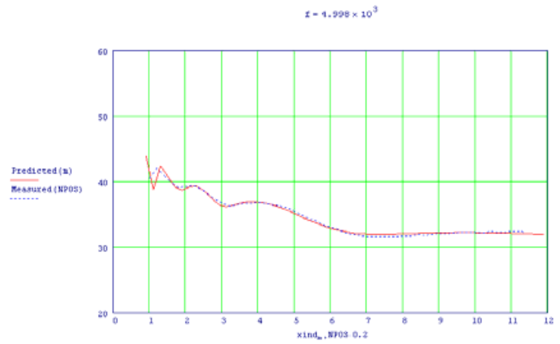
vi) Validation

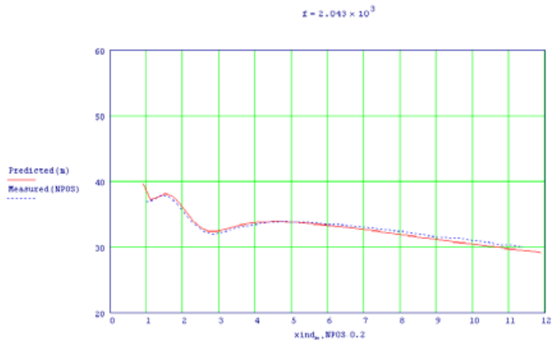
Fig. 7 Model Validation

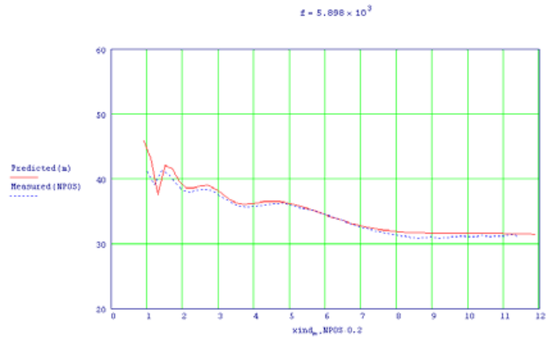
In order to validate the model 12 W8L HF horns were arranged in a line with zero splay between them as shown above. The SPL was measured every 20cm on a 12m path normal to the line positioned at the center.

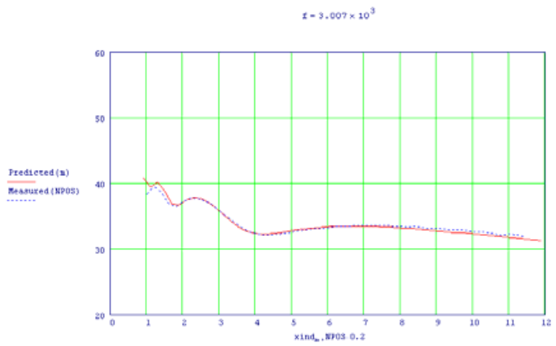
The measured and predicted responses are as follows:

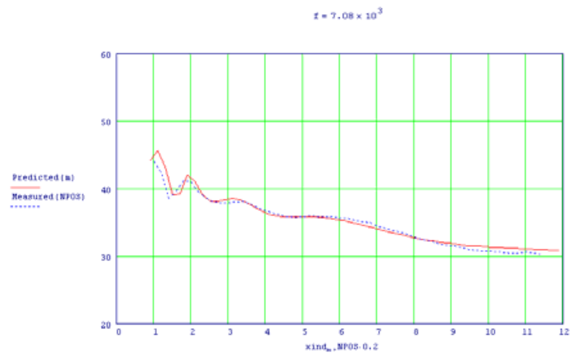


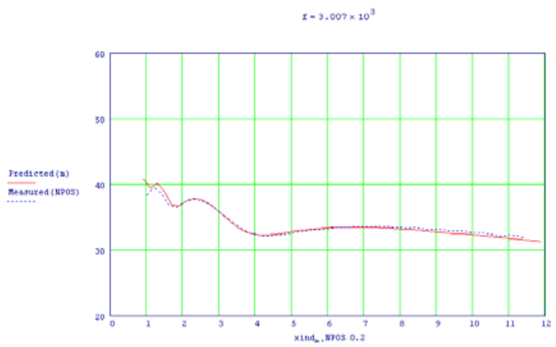


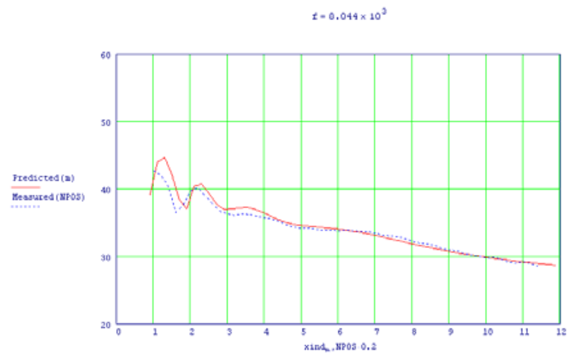












With this model, it was possible to predict the frequency response curves at various points in the audience and use these results to optimise the curvature of the array.

In nearly all cases, the computer model yielded a progressive curvature array profile (Fig. 8), where the curvature increases from top to bottom. This produces a more consistent frequency response from the front rows to the rear seats than often used J-shaped arrays (Fig. 9) having a straight, long throw section at the top and a curved lower section.

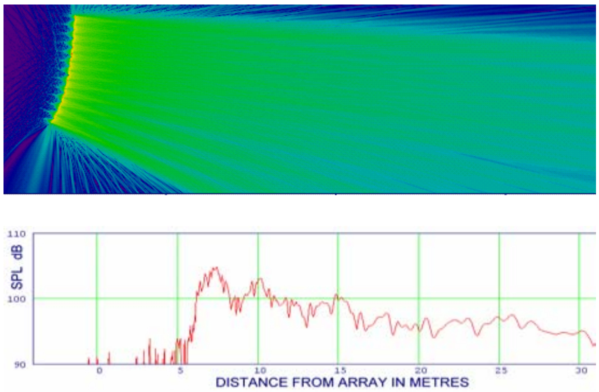


Fig. 8 Progressive Curvature Array

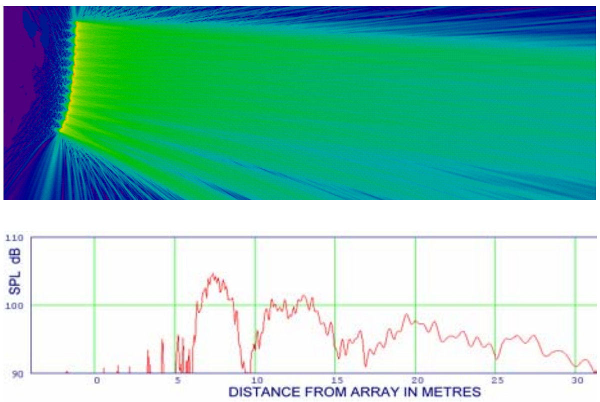


Fig. 9 J-Shaped Array

WAVEFRONT CURVATURE

During the period in the 1990's when the line array was emerging as the main format in touring sound, arguments were put forward that focussed on the need for achieving a flat wavefront from each element.

Whilst this may have merit in a straight line array, a perfectly flat wavefront is not mandatory and can indeed cause problems in curved arrays where the situation is complex and important compromises have to be made. To much wavefront curvature will adversely affect coupling and therefore output at the top of the array where there is typically very little or no splay between each cabinet.

No wavefront curvature will give noticeable high frequency hot-spots where inter-cabinet splay angles are large, typically in the short throw region at the bottom of the array. This is made worse when the hinge point is at the rear of the cabinet and the curved array has gaps at the front.

Another criteria advanced for line array calls for the vertical distance between drivers to be less than a wavelength at the highest frequency reproduced. This may be true for direct radiators, but this is one area where the performance of horns and direct radiators can differ: - a horn can be driven by drivers (Fig. 10a) which are greater than one wavelength apart at the highest frequency that they reproduce and still produce a low curvature wavefront, as shown in Fig. 10b.

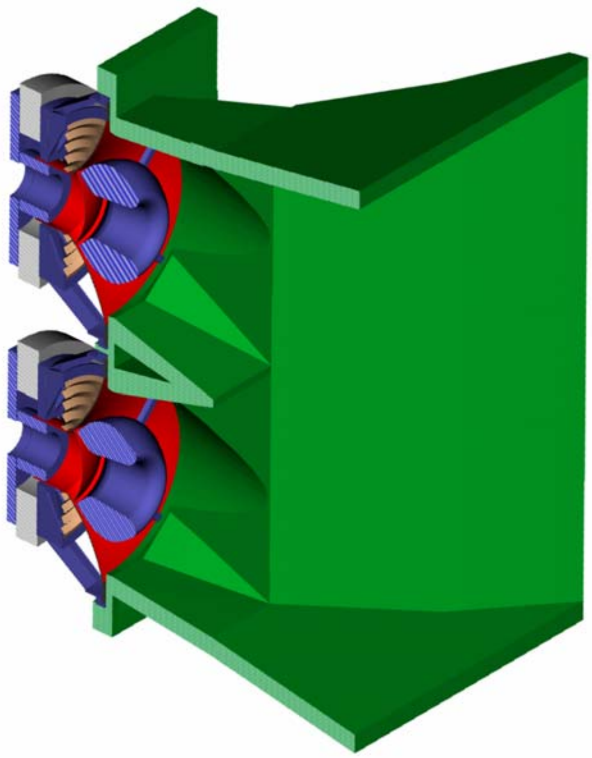


Fig. 10a W8L mid horn

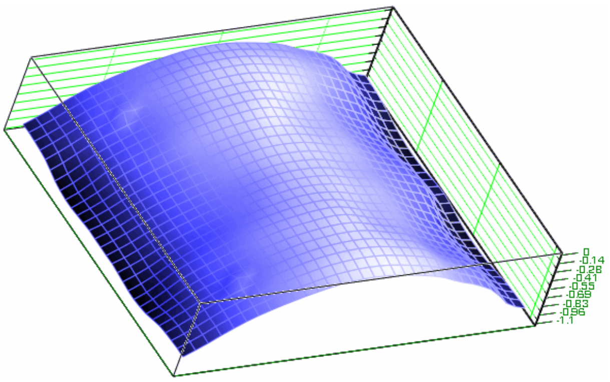


Fig. 10b W8L mid horn – measured wavefront curvature at 2.5kHz

This was confirmed by measuring the phase distribution over the mouth of a W8L mid horn driven by two vertically stacked 8 inch drivers [5]. Results showed that the mid horn produced a low curvature wavefront in the vertical plane up to 2.5kHz, its upper limit. (Note that the curvature from left to right is due to the 90º horizontal dispersion of the horn).

HINGE POSITION

Because they need to be curved, practical line array enclosures are linked by flying hardware with hinges at the front or rear to permit a range of inter-cabinet splay angles – typically from 0° to between 5° and 10°, depending on the particular design.

Not only is it important to curve the array correctly in order to achieve consistent frequency response at any point in the audience, the position of the hinge point plays a significant part. With a hinge point at the front of the cabinet, the spacing between each element is the same, irrespective of splay angle. This is an advantage as the splay angle increases, usually toward the bottom of an array (Fig. 11). With the hinge point at the rear, noticeable drops in output occur towards the upper end of the frequency spectrum when the listener is off-axis of each cabinet (Fig. 12).

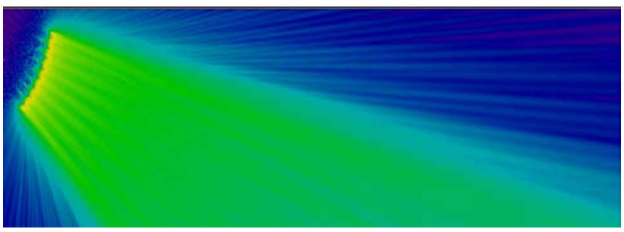


Fig. 11 Front hinge, 8 cabinets at 5.6kHz

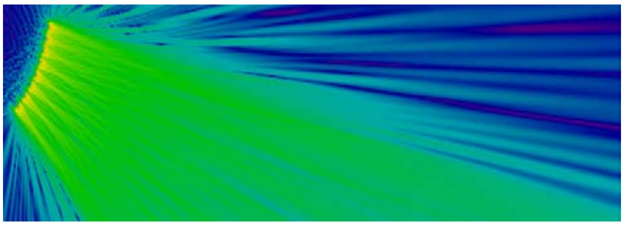


Fig. 12 Rear hinge, 8 cabinets at 5.6kHz

AIR ABSORBTION AND EQUALISATION

Whilst line arrays have greater high frequency output capabilities than cluster based systems, all sound systems are still limited by the phenomenon of air absorption, which is a function of temperature, humidity, atmospheric pressure and frequency (Fig. 13).

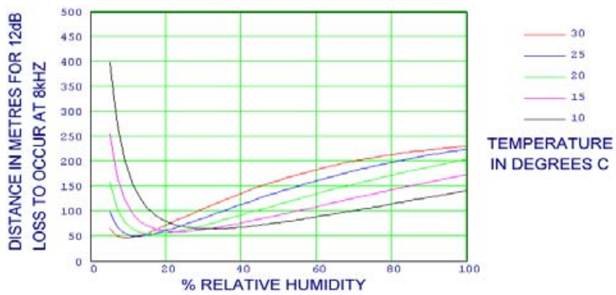


Fig. 13 Temperature and humidity effects

The relationship between these quantities is quite complex but losses always increase as frequency rises and distance from the source increases. Note this effect is in addition to the overall SPL loss as distance increases. For instance, weather conditions can attenuate output at 8kHz by 12dB at a distance of only 50m from the source. On another day the same system could throw over 200m! This is clearly an appreciable effect which needs to be addressed by appropriate equalisation.

To offset the affects of air absorption, progressively more EQ is required as the distance from the array increases. Since air absorption primarily affects high frequencies, it is of most benefit to split the drive to the HF devices into a number of separate channels (typically three) so that optimal EQ can be added to suit the requirements of the short, medium and long throw sections of an array.

By using this “HF band zoning” technique, people near the front do not have to listen to the extra high frequency EQ that the people at the back must have in order for them to hear high frequencies adequately. This simple technique can deliver consistent sound quality over the whole venue (Fig. 14).

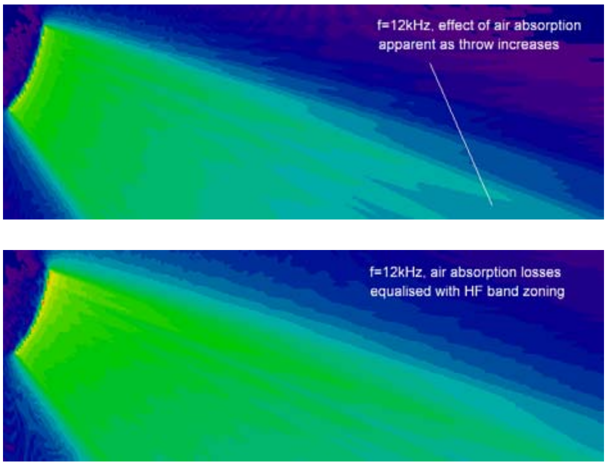


Fig. 14 HF band zoning compensates for air absorption

PRACTICAL TOOLS

In terms of physical and electronic set-up, line arrays are not nearly as forgiving as point-and-shoot systems and we needed to find a way of eliminating guesswork in determining the best array shape, line length and control settings for any particular venue. The progressive curvature rules established by the mathematical model have been incorporated into a proprietary optimisation program, Viewpoint, (Fig. 15) that will automatically optimise the curvature of the array to suit the venue. Designs can be saved to disk and printed out ready to give to the crew assembling the array.

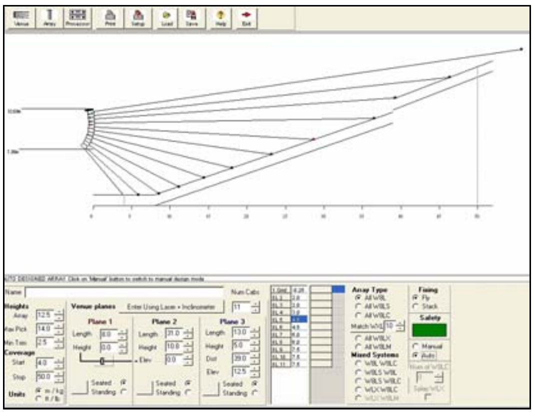


Fig. 15 Viewpoint

Before the advent of live sound line arrays, it was common practice to use a single digital controller preset for a particular loudspeaker system, with users adding their own preferred EQ and crossover tweaks. Whilst the simplicity of this approach may still hold some appeal, line arrays benefit greatly from specific presets that take into account variables such as the line length and degree of curvature.

The mathematical model enables the determination of a family of presets that are optimized for different array curvatures and also take into account the highly variable effect of air absorption. The presets are called up by the Viewpoint program during the array design process and ensure consistent sound quality over the audience whatever the size or shape of the array and atmospheric conditions on the day.

THE WIDER ISSUE

With so much attention devoted to the vertical aspects of line array, the line array story can become somewhat one-dimensional. There is much more to the whole picture than just the performance in the vertical plane. The way that the line array principle is implemented in the individual low, mid and high-frequency elements of the design is of prime importance.

As with any type of loudspeaker, the measured performance and sonic signature depends on the expertise of the designer and their design preferences. For instance, some designs use direct radiators for bass and midrange and others opt for horns. Since some line array designs cross over into compression drivers below 700Hz whilst others cross over above 2kHz, it is unsurprising that line array systems from different manufacturers both measure and sound very different.

In the horizontal plane, most live sound line arrays aim for a 90° coverage pattern. Achieving consistent horizontal directivity across a range of frequencies is a challenge to designers and, with the main focus on the vertical coverage, it is important that design decisions pertinent to vertical criteria do not compromise the horizontal performance of the array. It is particularly important that the frequency response of the system does not change as the listener goes from on-axis to 45° off-axis, but just drops by 6dB in level.

One accepted way of achieving consistent horizontal coverage is to utilise constant directivity horns to define the coverage pattern. The W8L utilises constant directivity horns for both the midrange and high frequency elements to achieve a 90° (-6dB) horizontal coverage above 200Hz. Fig. 16 shows that the horizontal polar patterns overlay well from 200Hz upwards.

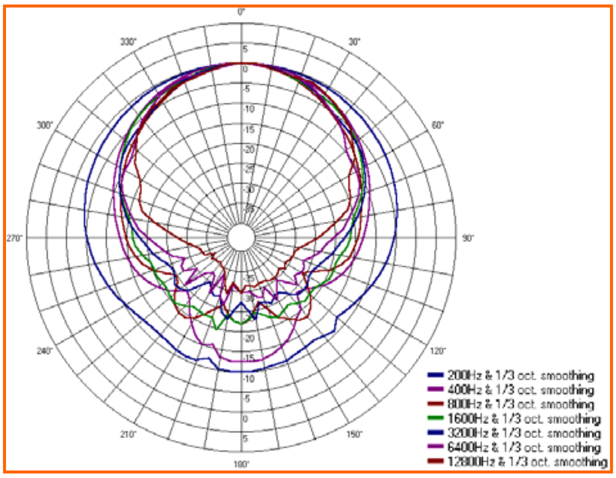


Fig. 14 W8L Horizontal Polars

CONCLUSIONS

The line array is now the dominant player in live sound reinforcement and can offer advantages over cluster-type arrays in terms of a more consistent frequency response, increased high frequency throw and reduced set-up time. Adapting the line array principle from theoretical straight arrays to practical curved arrays for live sound has important implications for the acoustic design of the individual elements and the physical deployment and electronic control of the system. Curved line arrays are complex in nature and benefit from practical, mathematical tools that can help eliminate guesswork and tailor the physical configuration and control settings of the array to the specific venue and atmospheric conditions.

ACKNOWLEDGEMENT

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