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Design and Optimization of High Directivity Waveguide for Vertical Array

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ABSTRACT

Vertically arrayed Loudspeaker Systems have become widely used in several fields of applications: concert sound, large scale systems, corporate events and so on. In this kind of systems the design of a proper Acoustic Waveguide is a key point for the system's performances. An Acoustic Waveguide, properly designed for this purpose, should be optimized for several features at the same time: acoustic loading properties, proper driver-throat matching, minimum internal reflection, low distortion and, most of all, proper wave front curvature optimization for good array-ability. An example of a practical approach to the design, dimensioning and optimization of acoustic waveguide will be shown through loudspeaker system designing principles together with computer simulations and measured final results.

1. LINE ARRAY AND VERTICAL ARRAY

1.1. Introduction

Recent developments in loudspeaker arraying techniques applied to sound reinforcement, showed the effectiveness of rational array design approach for large scale applications. The commonly called Line Array Systems have been populating the sound reinforcement market everywhere in the last 10 years. Immediate interest for this kind of systems arose because of a relevant demand for performance improvements in large scale applications like stadiums, concert sound, special events and, generally speaking, sound reinforcement applications where long-throw sound projection capacity was required together with even coverage and consistency over a wide area. In the last few years, the use of Line Array systems have been virtually expanded to any possible Sound Reinforcement application. But, what it is commonly referred as “Line Array” it would be better to consider it a Vertical Array of elements that very seldom are configured like a perfect straight line. More likely, this Vertical Array of loudspeaker elements is configured in a curved shaped array that should be, case by case, optimized to fit the shape of the particular venue and for the audience arrangement. Almost anyone of these Line Array systems has its specific computer aiming software, dedicated to the configuration and optimization of the shape of the vertical array. Practically speaking, this approach to Sound Reinforcement has become widely diffused and in many cases virtually a fashion. Very often the use of a Vertical Array system it is not even properly suitable. There are many applications where point source systems would be a way better choice for sound reinforcement, but the sound reinforcement market have shown very often to be fashion driven. From the beginning of the 90's, the studies and the experiments held by Christian Heil and his development of the V-Dosc system brought a completely new approach to sound reinforcement for large scale applications using vertically arrayed full range, multi-way loudspeaker systems: here began the Line Array era.

1.2. A new design approach

The key point that comes along with this approach has to do with the general design process of the loudspeaker systems. With very few exceptions most of the loudspeakers systems were designed, since then, using something like a Bottom-Up logical approach. The use and the knowledge of arraying techniques was very limited. The design methodology, since then, it was very often oriented to design a loudspeaker system, a module, optimized in order to work properly by itself and then arrayed together in multiple unit, undergoing the interferences between them and trying to minimize them with few little precautions like working on horn’s directivity control and trying to improve the mechanical suspension systems.

A new loudspeaker design approach came with the Line-Array. The real focus for the engineers became the designing of a loudspeaker array, as a whole. This is a very rational approach and it is more likely a kind of a Top-Down designing process. The good thing that came along with these new ideas, in fact, it was that the designing engineer realized that the large scale sound reinforcement situations would certainly require the capacity of producing large acoustic power, and consequently, a large number of transducers would be operating and contributing to it at the same time.

The rational management of the interference between adjacent units is the new approach. Considering the reciprocal distances between adjacent transducers imposed by their physical dimensions, the interference between multiple units is positively used in order to create the desired amount of array directivity, and this can be made successfully for those frequencies that have a wavelength that can sum properly with the assigned driver spacing.

So, designing the array just like thinking to a single large speaker that uses a large number of transducers operating together at the same time: this is the aim the designing problem will consist on satisfying some very important requirements:

- The large number of transducers used for each band are generally arrayed in the vertical plane. They should correctly couple together delivering proper acoustic power summation controlled directivity properties and no relevant side lobes in the plane of the array.

- Each frequency band should need a dedicated array of transducers and all the necessary band-dedicated arrays should be geometrically arranged together in order to coexist in the same array.

- In order to have correct coupling and directivity control from any array of transducers, the physical dimension of each transducer should be optimized in relation to the wavelength of the highest frequency they are supposed to reproduce.

- The highest frequency that an array of transducers can properly reproduce is the limit where side-lobes, in the array plane, rise over the -12/-10 dB limit that is the reasonable maximum pressure level that a side lobe can have without compromising the general directivity of the array (Fig 3 and 4).

- For arrays of transducers dedicated to the reproduction of high frequency components, transducers cannot be miniaturized under certain dimensions. In this case the use of inherently directional sources, disposed as close as possible, should raise the array's high frequency limit. A proper waveguide device should be used in order to get the appropriate directivity, free of side lobes, up to desired HF limit.

And the designing of this waveguide device is the main focus of this paper.

It can be clearly seen that such a new approach to loudspeaker systems design turn the engineer attention to the designing of a big array of transducers, optimized like if it was a single loudspeaker system and trying to find the way to divide it in several modules in order to be easily transported from one location to another and to be properly shaped, day by day, with the optimal curvature in order to fit the specific audience area requirement in which is being installed.

Generally speaking, this approach have been applied into the vertical plane, while the horizontal plane is being covered by the horizontal directivity properties of the single modules by them. This approach happens to be very practical and it can be easily applied to the horizontal plane instead of the vertical. The relevant thing is that, as a matter of fact, it greatly simplifies the multiple modules arraying problems.

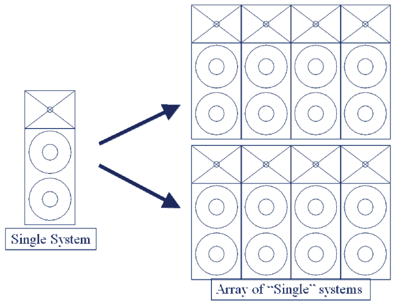

Figure 1 - Traditional array design with a bottom up approach. Design of a loudspeaker system and arraying it in multiple units

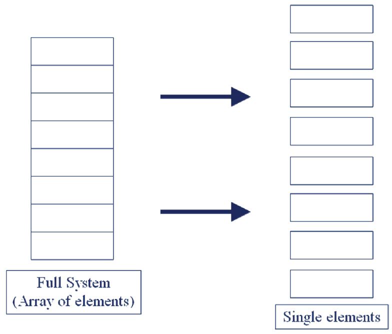

Figure 2 – New design approach: designing an array like if it was a single big speaker and then divide it in modules

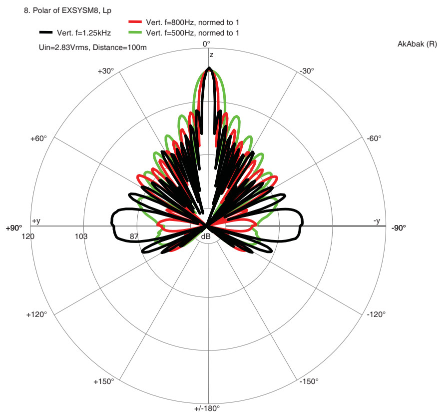

Figure 3 – Polar pattern from an array of loudspeaker units working at frequencies *below* the maximum frequency limit.

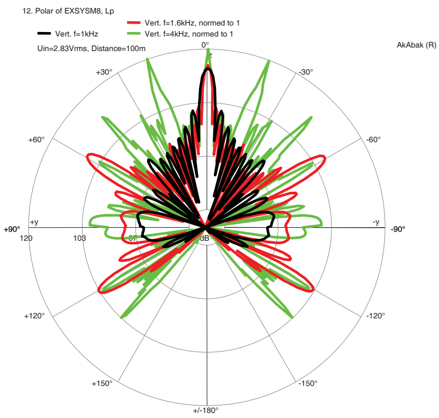

Figure 4 – Polar pattern from an array of loudspeaker units working at frequencies *over* the maximum frequency limit.

1.3. Line Array or Vertical Array

The idea to use the line array technique, a straight vertical line of sources, yields to create a loudspeaker system with very high directivity behavior. This is an essential characteristic if long throw sound projection capacity is an important requirement. Moreover, for a perfect straight and very long line of sources that are operating below the upper frequency limit, the sound field radiated from it can be well approximated by a cylindrical radiation. This makes the sound waves appear to travel easier at relevant distance. The reason for this is because the cylindrical wave propagation from a line source makes the pressure at some point to be proportional to $1/r$, instead of being proportional to $1/r^2$ like it normally happens in wave propagation from a point source or a spherical source. And this takes place almost in a region that is relatively close to array and has an extension that depends from the reproduced frequency and from the array length. This phenomenon looks like the sound pressure falling is 3dB per doubling of the distance from the array instead of 6dB per doubling of the distance like happens from spherical or point sources. This behavior have been interpreted like the Line Array has the capacity to project the sound way more far from the array with reduced losses. In reality the things work differently because what we are really obtaining from this kind of loudspeaker array is that we can cover an audience area, from front to back, evenly. Each finite length Line Array, in fact, over some distance from it, shows a propagation behavior that is very close to the point source. Every sound source, in fact, at some large distance it can be considered like a point source. No matter how big it is. At some distance it will be a point source. And, as it was pointed out by a brilliant paper that Douglas J. Button presented at the AES convention in 2002 [1], and if we think to the idea of moving toward the array, we will find that pressure will increase by 6dB per halving of the distance but, at some point (whose distance from the array depends on the frequency and on the array length), we will find that the pressure will start rising only 3 dB per halving of the distance. In the near field of the array, in fact, the sound field is highly destructive because of the massive interference, and high frequency sound waves, having short wavelength, is more influenced by this huge interference. This give use the feeling that in the near field of a Line array, the HF energy tends to remain relatively constant and stable while walking closer of away from the array.

But one question seems to be reasonable in this case: do we really need a Line Array? Is it really suitable to fit the required front to back coverage for any typical audience area? The answer is NO approximately in 99% of the cases, because we do rather need to give a curve shape to the array in the most of practical situations. Proper and even coverage of our typical audience areas require the whole system to be constant coverage in the vertical plane, and sadly, a Line Array it doesn't show constant coverage at all. The total directivity over the entire bandwidth increases, in fact, over and over as the frequency rise showing a coverage angle that changes continuously with the frequency. In most cases, a perfectly straight array would be completely useless.

For our real-world applications, in most cases, we need to articulate the curvature of the array in order to create the necessary constant coverage behavior. For this reason, we can certainly asses that we do not need Line Array in the most of large scale sound reinforcement applications. What we really need in those cases is a Vertical Curvilinear Array of sources. (As shown in pictures from 5 to 10).

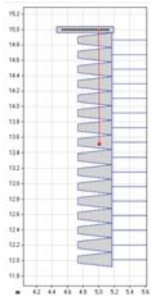

Figure 5 - A line array of loudspeaker elements arrayed on a straight line

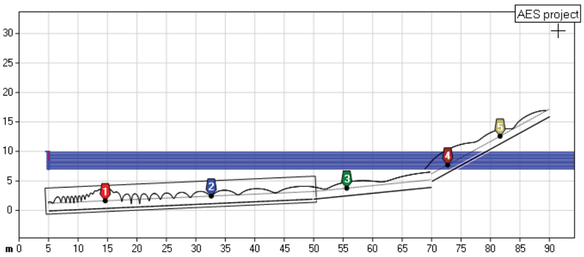

Figure 6 - Example of application of a straight line to a typical large-scale arena

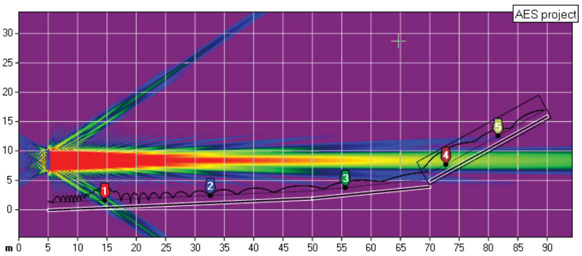

Figure 7 - Vertical dispersion obtained from a straight array calculated for the vocal range. It is not suitable to be applied to the arena example.

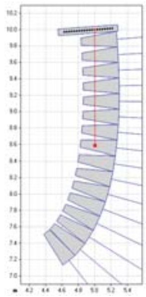

Figure 8 - Proper array curvature to address the vertical coverage requirement into the arena example

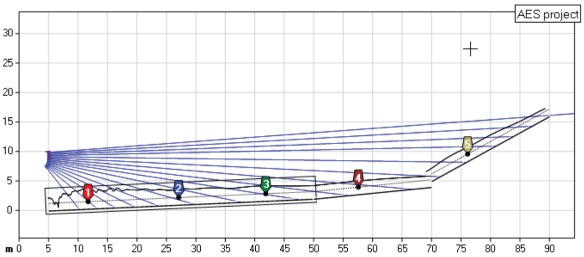

Figure 9 - Vertical system configuration relative to the arena example

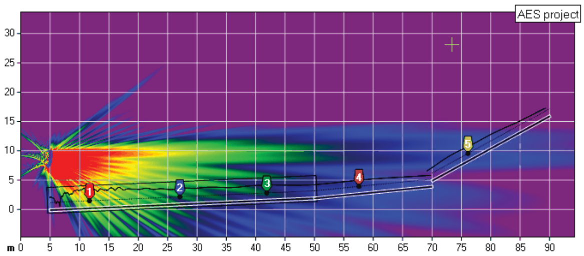

Figure 10 - Vertical dispersion obtained from the curved vertical array

2. LINE ARRAY AND VERTICAL ARRAY

2.1. Directivity requirements for the HF sources

At high frequencies (where wavelength are short) the dedicated transducers, if it has low directivity, would need to be very much miniaturized to let the relative array to be able to sum properly. If proper summation is required (for example) up to 16 kHz, the transducers should be no larger than 2 or 3 centimeters or so. What generally happens is that the transducer assembly that may have the necessary performances in terms of high efficiency and very high output at high frequencies to be used in a large scale line array system, cannot be miniaturized enough to be disposed in array at a reciprocal distance that is short enough to pursue the desired performances in terms of proper HF summation. Generally speaking, the transducers miniaturization could be an effective solution only designing a line array for applications where high output is not required. As already stated, the solution to overcome this problem consists in using HF sources with a very high inherent directivity. This approach tries, at the same time, to array the HF units as close as possible vertically, in order to minimize destructive interference in the HF band. The high directivity characteristic will tend to virtually eliminate those destructive interferences in order to have an even and predictable coverage at high frequency from the vertical array of units.

This is the reason why several designers have been spending a lot of efforts to find proper solution to increase, as much as possible, the directivity properties of these HF devices. One of the most effective solutions happens to be the combination of a HF compression driver with a specifically designed waveguide.

2.2. Waveguide and wave front

How directional should be the waveguide that we are going to use for this purpose? Should it have other additional properties?

The acoustic waveguide to be used in a Vertical Array should have, in the direction of the array's plane a directivity behavior that must be as high as possible. And, in order to get the desired directivity properties by itself and, most of all, while combined with other unit there are 3 very important conditions that must be satisfied:

1. The sound pressure must be evenly distributed at the waveguide mouth

2. Along the direction of the array's plane, the sound pressure arrival time at the waveguide mouth should be almost the same in any point of it. This means, that the wave front curvature, if measured along the array plane, should be as limited as possible.

3. The source size, in the perpendicular direction to the array's plane, should be as small as possible, in order to be able to widen the horizontal coverage as much as possible. A practical value that will allow the waveguide to easily reach the coverage angle of +/- 50° @ -6dB is a mouth dimension that should not be wider than 20mm.

With these input data, the "solution" to the problem seems to be necessarily some kind of narrow waveguide device that will transform the circular emission coming out of a HF compression driver into a narrow rectangular slot that will be about 20mm wide, addressing the radiation in the horizontal plane, and 100 to 150mm high in the array plane (the vertical plane) in order to get there the necessary directivity with the source physical dimension as well.

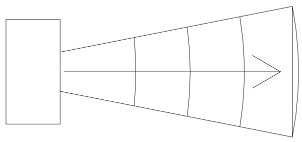

Figure 11 - Schematic view of the conical waveguide

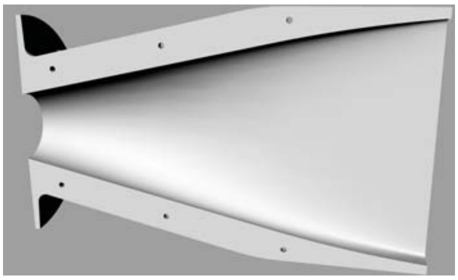

Figure 12 -Inner view of the conical waveguide

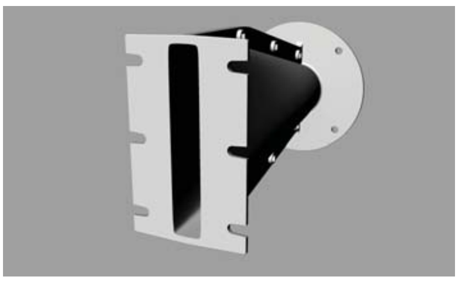

Figure 13 - A 3D model of a hypothetical narrow conical waveguide

And so doing, this waveguide will probably be a narrow conical device that will give to the output wave front some sort of curvature because of the time arrival of the wave front along the vertical slot. What it is very important to understand, at this point, is how much curvature in the wave front can be accepted while still having good summation properties from an array of those waveguides?

If we are attempting on creating a line array of sources using a line of these waveguides, the most desirable characteristic that we can pursue is having no curvature at all. A line array of HF sources that will deliver sound waves with perfectly flat wave front that means to not have any time arrival difference along the output slot would be the perfect solution. This, in a theoretical perfect case will mean summation in the array plane without any significant side lobe. But we have to remember that a good achievement, for us, it will already be good enough if there will be side lobes that will not exceed the approx. -12dB limit if compared to the on axis radiation.

Mark Ureda in a very illuminating paper from year 2001 [2] he demonstrated that in a Line Array of HF devices with a given waveguide mouth and a given wave front curvature, it the difference between the actual curvature of the wave front and the hypothetical flat wave front will not exceed ¼ of the wavelength of the highest frequency intended to be reproduced, the sound radiation from this Line Array will be still very close to the radiation of a perfect Line Array of perfectly flat sources because the side lobe will remain under the -12dB limit if compared to the on axis radiation. Another very important condition in order to get the proper results, as much important as the wave front shape, it is the real amount of radiation area from the waveguide exit slot and how evenly the pressure is distributed on it. This aspect plays a very important role in the array dispersion stability and in minimization of unwanted side lobes.

This means that there is no need to do wave front re-shaping with complicated and sophisticated waveguide devices. It would be sufficient to design narrow conical waveguide deep enough and with an exit slot that is not too high in order to not overcome the acceptable wave front curvature limit.

Moreover, is good to remember that we are not generally interested in designing Line Array because we have already demonstrated that perfectly straight Line Array will not address any of the standard audience coverage requirements.

If we analyze a certain number of different applications for a Vertical Array and we will try to calculate an average curvature that is more frequently used while designing the array curvature we would see that for large scale installations where the array must address a long throw application the average curvature between adjacent elements will be ranging around 1 or 2 degrees. For moderate size applications like theaters and small arenas, 5 or 6 degrees will be more likely the average curvature between adjacent elements.

This tell us that a Vertical Curvilinear Array should be designed and optimized for being used while articulated in a curved shape. We can also understand, very easily, that in these conditions, the Ureda curvature requirements for the HF waveguide could be also relaxed a little bit because we are generally not interested on generating straight Line Array summations between the units.

Moreover, flat wave front for waveguide to be used in articulated curvilinear arrays would not even be really suitable for the purpose. In fact, as much as summing curved sources in a Line Array may lead to non-coherent summation and relevant side lobes if the necessary conditions are not respected, also the use of perfect flat wave front sources will be troublesome if used in a Curvilinear Array.

2.3. General approach to the problem

The various solutions that have been approached from many designers in order to address the problem of creating a waveguide intended to be used in a line array.

The majority of these different approaches uses some ingenious ideas in order to make some sort of wave front reshaping in order to transform circular HF driver output to a rectangular flat wave front at the waveguide output slot. Most of those ideas will attempt to divide the driver output in several parts and direct them in several different geometrical path of equal length that will sum together at the waveguide output with very similar time arrival in order to reduce or eliminate the wave front curvature. Other approaches attempt to do this with completely different ideas like the use of single or multiple reflections into some parabolic surfaces that are located in the inner part of the waveguide.

Most of those ideas relies on the assumption that the HF driver output is a perfect flat, isophasic wave front, generated by a pistonic movement of the diaphragm and perfectly summed at the driver output by an appropriate phase plug.

In reality, most of these assumptions are not really satisfied, but this goes beyond the purpose of this paper. It could be, by the way, an interesting direction of development for future studies on this matter.

2.4. Our approach to this problem

In the purpose of designing a waveguide to be used in a Vertical Array we decided to use the easiest approach to problem. The easiest way to do a waveguide for this purpose it was to design it like a simple, narrow, conical shaped waveguide. In order to address general purpose application of this device, we wanted to extend the high frequency limit to 18kHz in order to overcome the general requirement for large scale application that can set at 12kHz the reasonable limit at HF, but we also wanted to design a waveguide device that can be flexible enough to be used in several different array applications. Getting to a frequency limit that high, by using just a simple conical shape there would be a consequent drawback that would make the waveguide to be about 30cm long and this would not make it a “flexible to use” device.

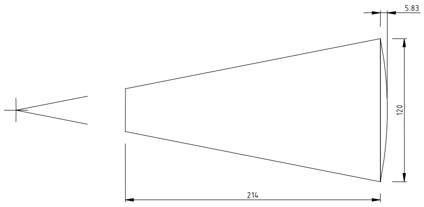

Figure 14 - Reference view of the external dimension of waveguide

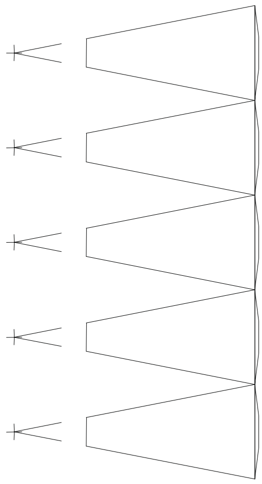

Figure 15 – Hypothetical array of waveguide

So, we designed a conical waveguide optimized for a lower frequency limit that was set at about 14 kHz and consequently reduced the waveguide length down to about 21cm. This is more likely the deepness that many regular horns can have also making feasible an eventual passive filter crossover for a system using this waveguide. Then with a geometrical reshape of the inner part, we introduced just a gentle redirecting of the wave front propagating throughout the waveguide, in order to slightly reduce the wave front curvature and consequently make the waveguide length shorter. The wave front curvature at the output slot will be the one of a longer device. This goal has been achieved with the addition, in the second half of the acoustic path of the waveguide, of few little wings that are supposed to gently change the path of the wave front without sacrificing the acoustic energy in the transmission throughout the waveguide and without any intention of creating any isophasic wave front at the waveguide exit, but just operating a moderate reduction of the wave front curvature in order to rise, a little bit, the HF limit of the waveguide when used in array.

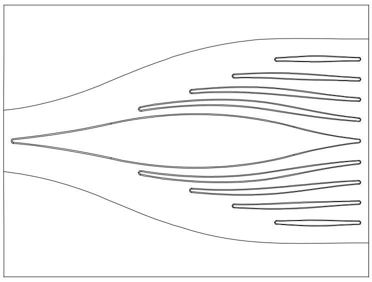

Figure 16 - Inner modification to produce gentle wave front modifications

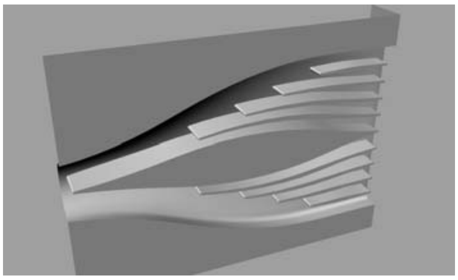

Figure 17 - A 3D model of the inner arrangement of the waveguide

The design philosophy used in this waveguide was focused on minimizing the potential deterioration for the sound waves through it. This was easy to do because the minimal target, in terms of proper array summation, would be reached even though there was not included any specific wave front re-shaping geometry into it.

The gentle development of its inner geometry, in fact, is capable of introducing a very low amount of internal reflections, not sacrificing at all the time response of the HF unit. Area expansion throughout the waveguide path has been managed in order to take the internal sound pressure as low as possible to minimize the distortion while sound waves propagate through it. At the same time the area expansion rate was optimized in order to give proper loading to the HF driver. This will yield very good coupling between driver and waveguide resulting in very high efficiency in the lower band of the driver output, and at the same time, avoiding that huge reflections would occur bouncing acoustic energy back and forward through the waveguide. Additional care was taken while choosing the proper shape of the inner geometry to help the acoustic energy being transferred from the throat to the mouth without relevant losses.

3. RESULTS

3.1. Test and measurements

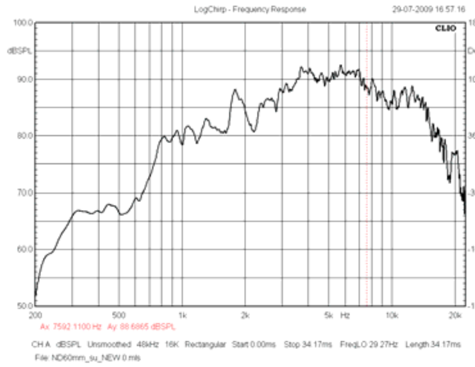

Figure 18 - Frequency response of the waveguide in anechoic chamber

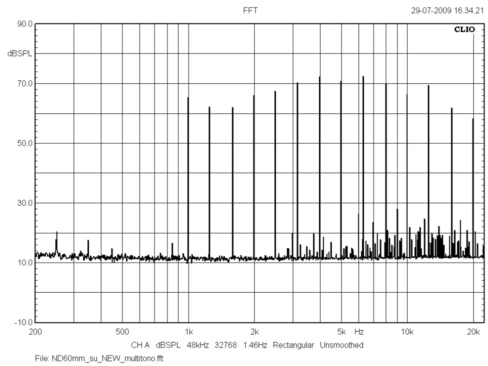

Figure 19 - Multitone analysis from 1 kHz to 20 kHz

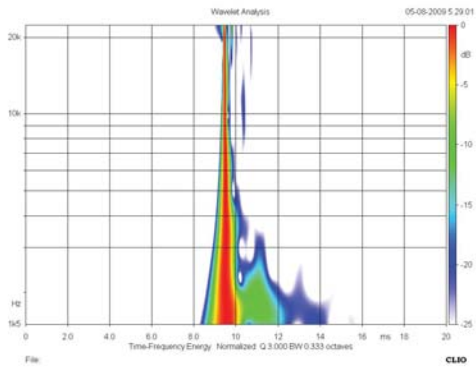

Figure 20 - Wavelet analysis of the time impulse response shows very low amount of internal reflection

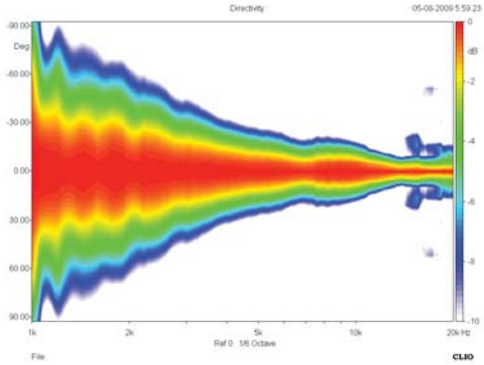

Figure 21 - Polar Directivity Map obtained from the waveguide. First relevant side lobes start to appear at 18 kHz and over

3.2. Dependency from the HF driver behavior

The waveguides for Vertical Arrays are generally described from various loudspeaker system manufacturers like devices that are completely controlling the sound emission from the HF compression driver, dominating almost completely the general device performances.

In order to investigate this topic, we tried to run additional measurement tests also equipping the new waveguide with an alternative HF driver taken from a different manufacturer. The results were interesting because we saw that there are some behaviors that belong to the driver and that are almost transparently transferred through the waveguide. This second driver for example, shows in its impulse responses analysis that there are “packets” of energy that are bouncing inside it, probably between different phase plug slots. This behavior will be transferred outside delivering HF energy that will be affected by some sort of ringing at the top part of the band, and it can be clearly seen from the wavelet analysis in figure number 22.

Some more thing that could be considered even more interesting than this can be seen from figure number 23 where it can be clearly seen how the HF uncontrolled behavior of the driver is very much affecting the performances of the waveguide in terms of directivity performances and in terms of being capable of summing properly when is arrayed in multiple devices.

We have decided to use the measurements performed with this driver to show the waveguide performances when used in array, in order to show the waveguide performances in the worst case of use and to investigate if the target performances are reach also in that case.

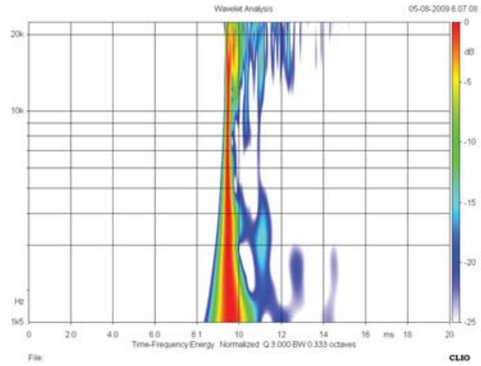

Figure 22 – Wavelet analysis for the alternative HF driver. It can be seen the effect of HF bouncing within the driver itself.

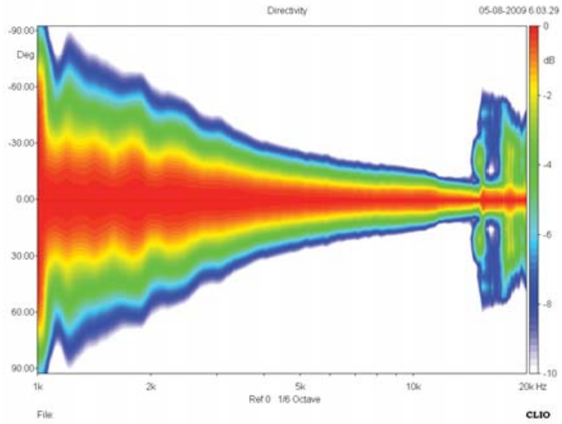

Figure 23 - Directivity polar map measured with the alternative HF compression driver. It can be seen like the HF behavior is heavily affecting also the vertical directivity performances

3.3. Testing the results simulating the array behavior

Testing the result with a Line Array

In order to test the performances of the new waveguide it was used SpeakerLab software from AFMG – Berlin, that is able to run simulations of arrays of any shape build with models that can be created with polar measurements properly collected from it. As already stated, de decided to use the measurements collected with the alternative HF driver because it was showing worst performances, to do the test for the worst case. We first studied the case of a Line Array of 16 elements showing the results at 4 sample frequencies. The chosen frequencies were: 2kHz because is in the mid-band of the waveguide, where the directivity depends from the slot dimension, 5kHz because is already a relatively high frequency where the first small side lobes are supposed to start appearing (but at about -20dB) and then 14kHz and 18kHz have been analyzed where the firs is located within the frequency limits and the second is a little bit over the border.

Figure 24 - Schematic view of a 16 elements Line Array built and simulated with the model created from the waveguide polar measurements.

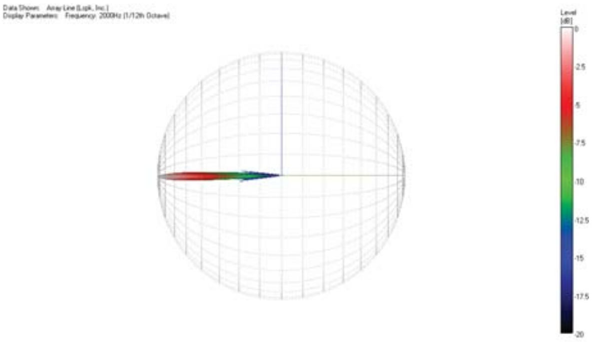

Figure 25 - Array beam at 2 kHz

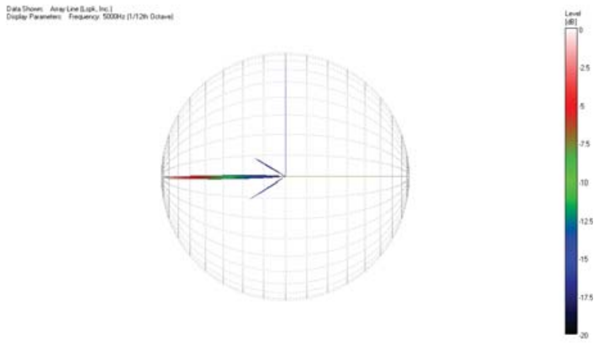

Figure 26 -Array beam at 5 kHz

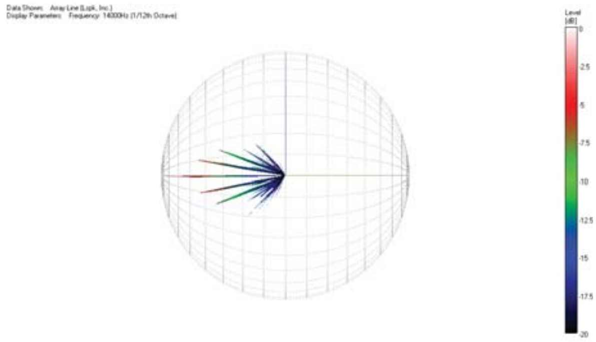

Figure 27 - Array beam at 14 kHz

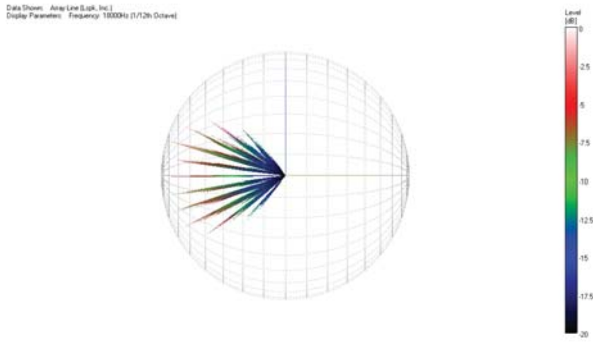

Figure 28 - Array beam at 18 kHz, the HF limit for this waveguide. Expected side lobes and instability can be clearly seen. The use of a better controlled HF driver could greatly improve this behavior.

Testing the result with a Vertical Curvilinear Array

The following test are performed in a similar way but with the array geometry shown in figure 29. This represent a very common application of a Vertical Array articulated to fit a very general case of audience area.

Figure 29 - Schematic view of a 16 elements Vertical Curvilinear Array built and simulated with the model created from the waveguide polar measurements.

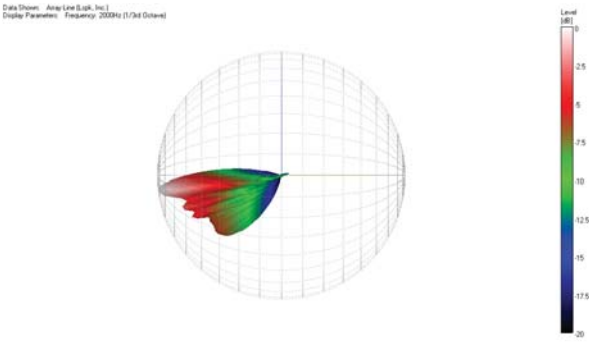

Figure 30 - Array beam at 2 kHz

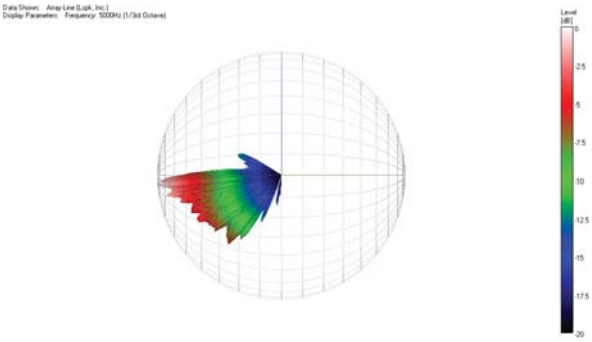

Figure 31- Array beam at 5 kHz

It can be seen that the new waveguide when is used into an articulated vertical array behave correctly at various frequency. Since the lower part of the array have been intentionally curved at large angle (8°, 10° etc…) it can be pointed out that the 12.5kHz lobe is showing some instability just right in the lower part of it (Figure 32). This happens because the design of the new waveguide have been focused on the optimization of it, to have best coupling properties with low angle values. This was the reason why the curvature of the wave front has been reduced as well. The lobe instability at very high frequencies, in the direction where the array has large curvature values, is demonstrating that flat (or almost flat) wave front sources, even though they have very good performances when arrayed in Line Array in multiple units, they are not very effective to address the needing of large curvature array segment. The best way, in fact, to build an articulated Vertical Array it would be using different elements, with different directivity, each one of those optimized to fit various section of the array.

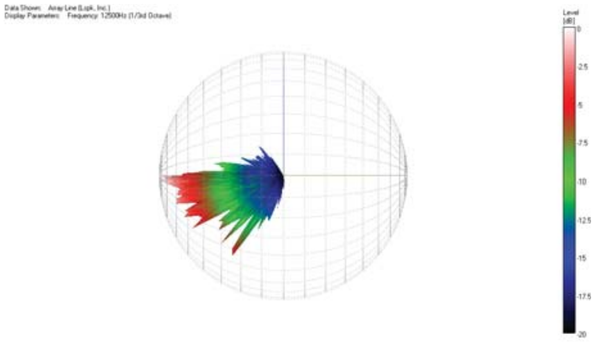

Figure 32 - Array Beam at 12.5k

4. COMPUTER ANALYSIS AND FUTURE DEVELOPMENTS

4.1. Computer aided analysis results.

Some additional results were investigated simulating the 3D model of the waveguide with computer analysis in order to analyze the inner propagation of sound waves throughout the waveguide at some sample frequencies. The results obtained so far were only partially reflecting the acoustic performances that could be measured in the far field while collecting a polar set of data. The reason for this would certainly need to be investigated in depth, and it is probably a field of investigation that could contribute to improve Vertical Array technology. At the same time some hypothesis can be analyzed and more can be understood from future developments. By the way, as already demonstrated by some of the tests that have been performed over the waveguide, there is a large influence of the driver output over the directivity properties of it. The sound waves that propagate through the waveguide are not only dominated by the waveguide geometry but also by the way the driver output in terms of uniformity of the energy distribution and time arrival at his exit.

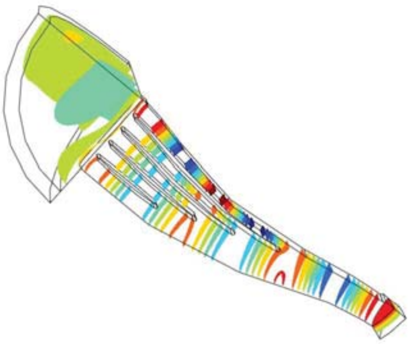

Figure 33 - Computer simulation of inner wave front propagation at 5 kHz where source is a convex dome

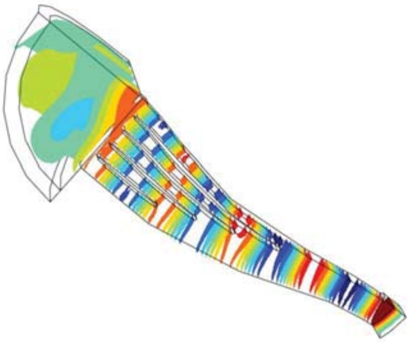

Figure 34 - Computer simulation of inner wave front propagation at 6.3 kHz where source is a convex dome

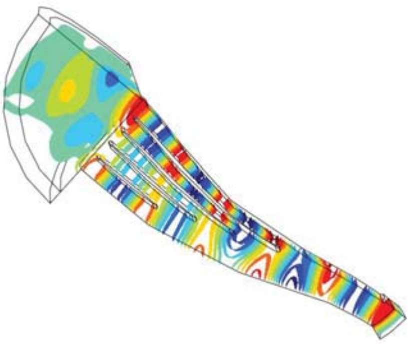

Figure 35 - Computer simulation of inner wave front propagation at 8 kHz where source is a convex dome

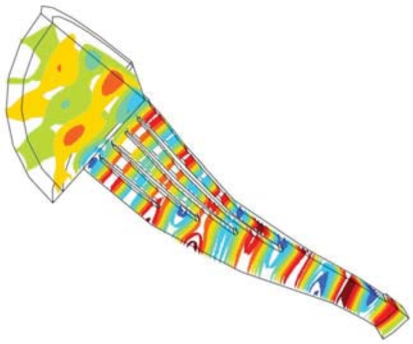

Figure 36 - Computer simulation of inner wave front propagation at 10 kHz where source is a convex dome

Figure 37 - Computer simulation of inner wave front propagation at 12.5 kHz where source is a convex dome

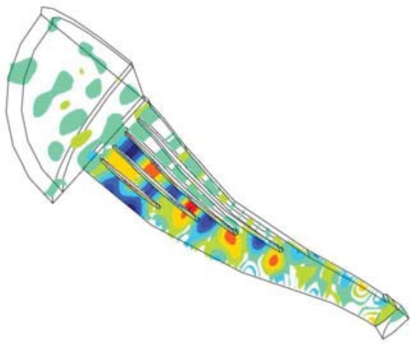

Figure 38 - Computer simulation of inner wave front propagation at 16 kHz where source is a convex dome

More accurate investigations, in fact, should take in count the driver effects in the model. This will drive the waveguide with something that is more closely representing the actual wave front of the acoustic waves that come out from the driver exit. The results at that point should be more accurate than those obtained with a constant pressure and phase flat disc or a constant pressure convex dome.

Figure 39 – Wave front shape calculated at the waveguide exit at 5 kHz. Driving source: convex dome

Figure 40 - Wave front shape calculated at the waveguide exit at 5 kHz. Driving source: convex dome

5. ACKNOWLEDGEMENTS

I'm personally grateful to all the people that, at 18 Sound, gave their contribution to this work. The Transducer Engineering Department for the support obtained with the measurement and the simulations and the company management for all the support for supplying all the tools and prototypes that were necessary to investigate this topic.

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