Line Arrays

This document is an attempt to give a little more light about a type of sound systems that, although already for some years they have been used actually, follows surrounded by certain confusion. Confusion that in addition has been increased by the search of differentiation of the manufacturers, that try to convince of the excellence of its particular solution to the problem, has still increased plus the doubts about the valid criteria at the time of designing a Line Array.

In this document a review by the solutions will be done of diverse manufacturers, ending the solution that Beyma has designed to facilitate the accomplishment of this type of systems.

L'Acoustics

Although the theoretical principles of the acoustic line sources (Line Arrays) are known from the years 60[1,2], was not to the 90's that the Dr Christian Heil tried to take to practice this type of sources. The main *theoretical advantage* of this type of systems is the attenuation of 3dB when doubling the distance, in comparison with 6dB of a spherical or omnidirectional source. Thus it would have to be easier to project sound to long distances. This question always has been a serious problem at the time of covering great audiences with a good level of acoustic pressure without causing interferences between the boxes that compose the more classic sound systems.

Without entering excessive details, the Web site of L'Acoustics http://www.l-acoustics.com/anglais/retdgb.htm a pair of white papers can be downloaded for free [1,2], in which the basic principles are marked that must fulfill the sound system to obtain *partially* this effect of linear source. And when saying partially, I mean that indeed it is *physically impossible* to actually obtain this effect for all the audible range of the human being, with the technology available nowadays. Which one is to try *to come near* to the maximum to this theoretical linear source by means of individual sources of sound, adapted for each frequency band.

A first criterion that this article explains is the critical distance of separation between near field and far field. In the near field it is where the effect of Line Array takes place. The SPL level diminishes 3dB whenever we doubled the distance. In far field, the source already stops being a line to be considered like a point source, radiating therefore with a diminution of 6 dB with the double of the distance. A formula appears to calculate this distance based *on the frequency*. This one is a key concept, that is to say, the critical distance is different for each frequency:

$$\boxed{d_B = \frac{3}{2}\cdot F\cdot H^2\cdot\sqrt{1-\frac{1}{(3\cdot F\cdot H)^2}}}$$

being d_B the critical distance or of edge from which the transition between near field and far field takes place. F is the frequency in kHz and H is the height of the linear source in meters. Of this formula several conclusions can be extracted:

1) Near field for low frequencies does not exist. Using this formula, it is possible to be verified that for a linear source of H = 4m will be radiating in far field always for smaller frequencies of 80 Hertz.
2) For frequencies over 1/3H the growth of the zone of near field (d_B) increases approximately of linear form with the frequency.
3) The relation between the height of the linear source and the critical distance is quadratic, instead of linear.

For that reason, the near field can be extended much for the high frequencies.

Due to the separations between the diverse sources that form the acoustic boxes, these separations will produce patterns of interference in the zone of coverage in the vertical plane, giving origin to secondary lobes outside the zone that theoretically would have to cover the linear source. These lobes can have as much level as the main lobe. This causes a loss of SPL in the zone to cover and can produce serious destructive interferences between the main lobe and the reflections of the secondary lobes from ground and walls.

If we defined the separation between the centers of the loudspeakers like STEP, and the vertical height of the loudspeaker like D, to reduce to these lobes to –13,5 dB with respect to the main lobe [3], is had to fulfill:

a) Or the relation between D and STEP is greater than 82%, following this formula and being ARF=D/STEP:

$$\boxed{ARF \geq 0.82 \cdot \left[1 + \frac{1}{4.73 \cdot (N+1)}\right]}$$ being N nº of sources

b) Or the frequency band for which the transducer is used would have to be limited to a frequency F, so that the separation between the centers of sound sources STEP is minor half of the wavelength λ of this maximum working frequency F:

$$F < 1/6 \cdot STEP \text{ Or } STEP < \lambda/2$$

The difficulty is based in fulfilling this criterion for the section of high frequency of the box. That’s the reason why of the diverse solutions of the manufacturers: to be able to radiate through long and narrow grooves, that leave the smaller possible space between the adjacent vertical sources. So that a compression motor, with mouth of circular exit, broadcasts a flat front of wave, that does not produce interferences between sources, curvature S of this front of wave must fulfill [3]:

$$\boxed{S < \frac{\lambda}{4}}$$

That is to say, as an instance, not to have interferences between adjacent compression drivers, the curvature of the wave front that is generated in the mouth of the connected waveguide to each motor must be smaller than $\lambda/4$. For 16 kHz, this curvature is of 5 mm. If this is not fulfilled, we will have strong irregularities in the near field, as well as great lateral lobes in the vertical coverage of the line source formed by these components.

In this article also the benefits comment to slightly curve the Array, since greater area of vertical cover is obtained, with mean level more constant than with a totally flat source.

At the time of curving the individual boxes that conform the linear source, the limits for this angle depend on the minimum range to which it is going away to locate a listener d_{MIN}, the separation between sources STEP and relation ARF. As the worse case occurs for the maximum working frequency, seizure like 16 kHz in the article, we can calculate a $STEP_{MAX}$ maximum, that would be:

$$\boxed{STEP_{MAX} = \sqrt{\frac{d_{\min}}{24 \cdot ARF}}}$$

Once we have this data, we can calculate the maximum angle between boxes α_{MAX}:

$$\alpha_{MAX} = \frac{1}{24 \cdot ARF} \cdot \frac{1}{STEP} \cdot \left[1 - \left[\frac{STEP}{STEP_{MAX}} \right]^2 \right]$$

Therefore, whichever greater it is the minimum range to which it is going to be the first row of public which we want to cover with the Line Array, greater will be the rake between boxes that we pruned ourselves to allow. This angle is the one of separation between boxes in *the frontal side of the boxes*, that is to say, pivoting the box behind. This it is the system used by L'Acoustics. Many other systems already use the inclination by the back part of the enclosure, giving trapezoidal form to the box and assuming a Maxima inclination between the boxes.
We can summarize then, the criteria used by L'Acoustics in 5 main points:

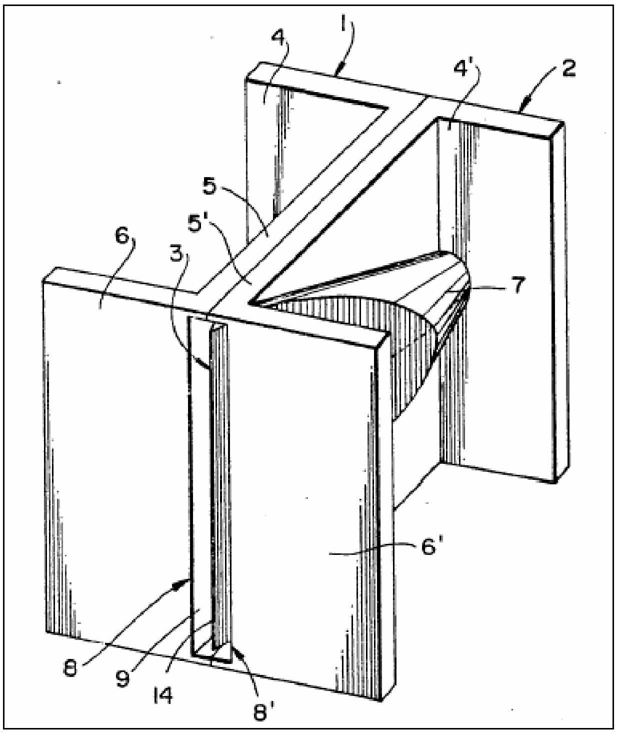
1) Or the sum of the individual areas of flat sonic radiation covers more than 82% of the vertical dimension of the linear source, or
2) Separation STEP between the acoustic centers of the individual sources is minor of 1/6F, is to say less than $\lambda/2$ of the highest frequency of work.
3) The deviation with respect to a totally flat front of wave must be smaller than $\lambda/4$ of the highest frequency of work (being 5mm for 16kHz).
4) In a curved source, in addition to the previous criteria, one is due to fulfill that the product between the rake and the distance to the public of each individual element of the system ($\alpha \cdot d$) is constant.
5) The maximum angle of inclination between boxes would not have to surpass:

$$\alpha_{MAX} = \frac{3^{\circ}}{STEP}$$, being STEP in meters

These 5 criteria are those that conform the concept of the Dr. Heil of "WST" or "Wavefront Sculpture Technology".

Besides to settle down all these criteria of design, the Dr Christian Heil designed a waveguide to obtain a practically flat front of wave from the circular mouth of exit of a compression motor. This waveguide (Patente US5163167) has a central piece that theoretically compensates the ways crossed by the sound waves of such form that the wavelengths which they arrive when coming out have crossed the same way, being all in phase. This waveguide must also fulfill criteria, as they are:

1) Wide of conduit (space between wall and cogoverning piece of ways) smaller than the wavelength corresponding to the Maxima work frequency.
2) Angle of expansion of the smaller guide of 30º.
3) Difference of ways between routes of the sound wave by the guide smaller than ¼ of the wavelength corresponding to the Maxima work frequency.
4) High of the mouth of exit of the guide of wave greater than the wavelength corresponding to the minimum frequency of work.
5) Wide of the rectangular mouth of exit smaller than the wavelength corresponding to the Maxima work frequency.



WAVEGUIDE OF DR HEIL

The Dr Heil was pioneering in taking to the east practice type of design. Later many more arrived, but Heil's patent date from the 10th of November of 1992. It is the oldest with difference.

Meyer Sound

Meyer Sound has several technical information, all available ones in his Web, among them is one titleholder "Line Arrays:Theory, Fact and Myth", in the section of Sound Lab of his Web site www.meyersound.com[5]. On it, tries to demystify some dark points and incorrect concepts attributed to this type of design of boxes. At no moment it is possible to be read in articles of the Dr Heil who their system is really a linear source, but simply an only valid approach for certain ranks of frequencies, employee of nº of boxes and several factors more already indicated. But the always negative influence of little professional magazines, marketing and "popular wisdom" have obtained that they occur by certain a series of "magical" virtues attributed to these systems, when using totally incorrect and invalid simplifications. For that reason, and evidently also to promote his system, John Meyer it has written several articles that they look for to demonstrate the incorrect thing of many supposed benefits that have these systems.

In order to demonstrate his point of view, Meyer uses exclusive software of acoustic simulation called MAPP. With the aid of this software, one graphically is the lobes to us of which the Dr Heil in his article speaks to all color. Also the omnidireccionality of the LF of the Array can be seen, question that Meyer corrects with an intelligent design with cardioid directivity in a model of his boxes of serious.

The most interesting contribution on the part of Meyer, aside from giving very illustrative numerical examples, consists of remark the attenuation in high frequency produced by the air. It is clear that to project high frequency to long distances of correct form it is made essential to know the standard ANSI S1.26-1995.

Also patent is made the convenience of curving slightly the Array to obtain a more uniform cover in the hearing, something shared by all the manufacturers of

Line Arrays. The used example is not very lucky, to the being with conventional horns instead of waveguides. Simply to demonstrate that accumulating conventional horns we are not going to obtain more than strong interferences and very irregular covers, done by all easily verifiable well-known and in the reality. Also sample that a Line Array without curvature can produce strong lobes of acoustic energy in directions no wished. What it does not indicate is what criteria it is necessary to fulfill to diminish these lobes, question yes clarified by Heil.

In the Web of Meyer, also there are two quite interesting articles. One is in which Meyer shows his particular solution for the high frequencies, the REM or "Ribbon Emulation 6 MAnifold" [ ]. In this article Meyer it argues that, being based on the asymmetry of the air as far as compression and expansion, the shorter the waveguide, the minor must be the second overtone. It contributes as it proves a comparative one between system REM and another one with conventional waveguide. The measurement is taken to 4 m. of the boxes, excessively near distance to contribute something to the behavior in long distances and more real applications. There is not either any comparative measurement of distortion.

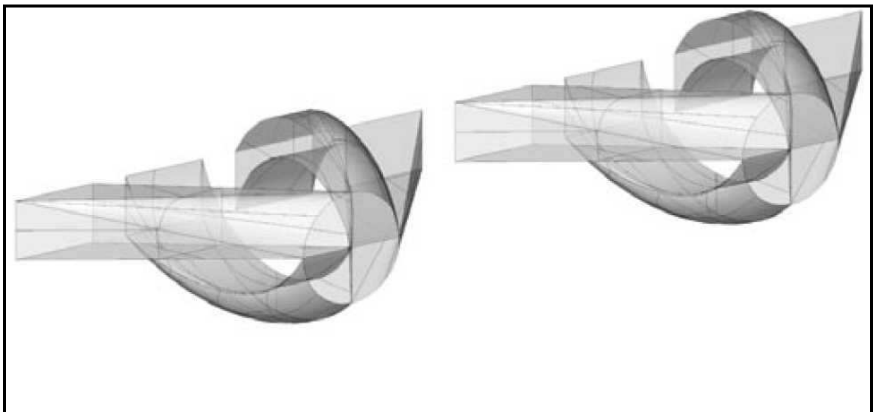


Figure 1: A 3D lit wireframe drawing of two REM waveguides

The problem of the excessively short horns is the impedance of irregular radiation that produces. When a horn is too short and expands of abrupt form, a lack of adaptation between the salient sound wave of the motor and the mouth of the horn takes place. This causes that, in addition to the traveling wave of the motor towards the mouth of the horn, it appears a regressive wave, that "bounces" with this lack of adaptation of impedances and returns towards the mouth of the compression motor, interfering with the salient wave. If we took like example the solution from Meyer for upper middle frequencies, we have a horn of length 3"(75 mm.) in the channels from the mouth of exit of the motor to the horn, that expands quickly. Although the length of the horn is not indicated, does not seem to have more than 3"of length. This lack of length causes an irregular answer in frequency, with pockets and tips in the frequency response, in addition to not controlling the dispersion of the sound. Beranek [1], in the same chapter of which Meyer extracts the information to comment the benefits of his design, also speaks of the minimum length that must have the horn to obtain control of the dispersion from the wished minimum frequency. This length evidently is related to the wavelength, being recommendable that measures at least one wavelength corresponding to this minimum frequency. For a frequency of 560 Hertz, this wavelength is of 24". The length overall of the horn of Meyer does not surpass nor the fourth part of this size. It is clear that the dispersion that can have the design of individually taken Meyer is totally dependent of the compression driver, with very little control contributed by the horn. The nominal horizontal cover of 90º that appears in the technical leaf of the boxes that use system REM is not endorsed by no measurement.

In the specific mission of the Line Array, luckily the important thing is that the boxes are coupled properly, passing the flatness of the frequency response and the uniformity of dispersion of an individual box to background stops most of manufacturers. Therefore although the design of Meyer would not be adapted for its application in individual boxes (in fact, system REM is only used in enclosures for Line Array), in the tactical mission in which it is used, yes can be obtained a good result, as it demonstrates the fact to be a professional system widely used and accepted.

Also there is another point to consider in this design, that is the discontinuities in the vertical direction which they produce the vertical walls of the horns. In the means horn of the model MILO there is a separation of almost 3 inches between the superior part of the radiating surface and the superior edge of the horn. Being the height of the radiating surface of 6,3 inches, this gives a relation us of ARF of little more of 50%. On the other hand, the alternative criterion of separation of $\lambda/2$ between centers d and sources which is not fulfilled either, since this distance comes to be from 12 inches, would approximately indicate a frequency of superior crossing of 600 Hertz. This means transducer covers the zone with 560 to 4200 Hertz.

Of course, the quality of this system of sound, proven cannot be put anywhere in the world in doubt. Everything seems to indicate that the correction produced by system REM is the sufficiently good thing at the time of producing a flat front of wave, of such form that although seems to fail to fulfill the criteria indicated by Heil, the obtained result continues being good.

There is another interesting article, denominated "DSP Beam Steering with Modern Line Array" [7], that basically is a critic to the use of digital techniques to manipulate the dispersion of groupings of loudspeakers. It is a report in which one reports solely the disadvantage and great difficulty of design of this type of systems, with the appearance of back lobes and without any of the advantages of these systems, that are many when they are designed suitably. Like always, there are no good systems and bad, they are different solutions with virtues and different problems. Also in the Line Arrays lobes are created, in this case vertical, that they are not nothing advisable either. While the tool with its limitations is known and a good use occurs him, the result will be the awaited one. But unfortunately still any magical solution without any disadvantage or problematic individual does not exist.

<u>Adamson</u>

Adamson is without a doubt one of the pioneers in this type of systems, after L'Acoustics. With the collaboration of Mr. Earl R. Gedges, of similars characteristic to the development of Heil developed to a device [8]. The differences later rest on in which it uses an adapter to expand the wave front from the mouth of the motor to a rectangular section, and a piece with different lengths from way to end up producing a flat front of wave.

Adamson affirms that its co-governing piece is better to maintain always at least two walls continuously parallel, which seems to be a requirement necessary to be able to maintain a wave front flat.

The second strongpoint of the system of Adamson is the use of a means unit in coaxial form with the compression motor [9].This supposes a considerable improvement of the system in the horizontal dispersion, that would have much more to be consistent and regular that in systems in which the axis of the units of high frequency does not agree with the axis of the means units. This is also the choice made by Renkus-Heinz.

Of course, the complication of construction of the pieces of the system of Adamson is really high, as it is possible to be observed in the drawings of the patents.

Fig.3

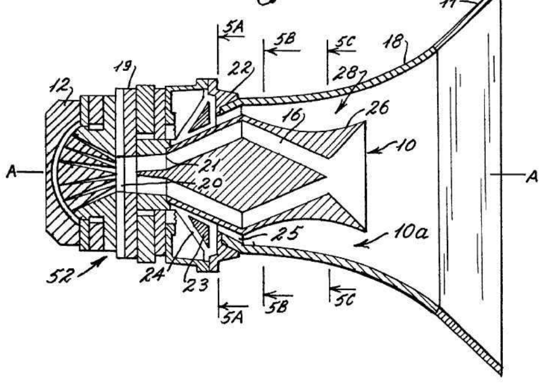
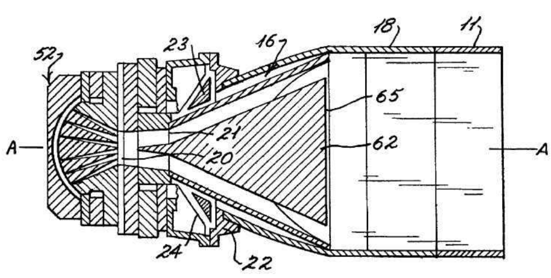


Fig.4



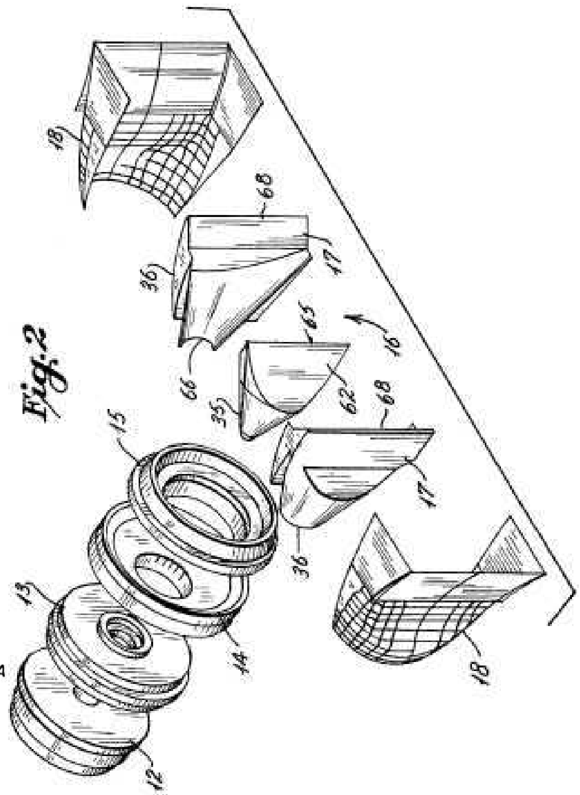
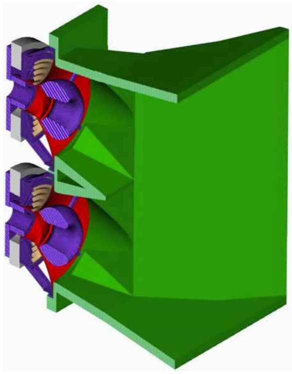


Fig.2

Of these drawings it is included/understood that the system of Adamson has a price consequently.

__Martin Audio__

Martin Audio has made an approach to the Line Array systems different with one philosophy. In the article in that they present/display the process of design of their Line Array [10] (it is possible to be downloaded of its Web), presents/displays photos of some of their first equipment, with all the widened and piled up routes. They have seemed to want to present the image the principles operation of Line Array for years.
What they decide to do is to use a piling up of small motors of 1", with small piled up conventional horns properly. They use numerical calculation and software soon to apply to digital processing in each route and power to calculate and to predict the behavior of the system. The connection of the high frequencies seems to make it without (apparently) no special piece to compensate the difference of ways until the mouth of the horn.
They make much emphasis in the angular importance of suitably the boxes to obtain the coverage of the audience correctly.
Its main innovation also is in the use of two units of 6,5"in horn with a corrector of phase patented by them with ring form, that allows them to create a flat front of wave in the average ones up to 2500 Hertz.
Thus, by means of the software and the horn of means, they are able to apply of effective form its equipment to cover the area with public of advisable form.



Martin Audio high and mid frequency horns

Nexo and Outline

These two companies, Nexo of France [11] and Outline of Italy [12] use *a similar* system (not identical) in which they use reflectors for the section of high frequency. Explained of a simple form, by means of a philosophy similar to the used ones in the satellite antennas, they use reflex baffles with the center in the mouth of the motor and the used reflex baffle like indirect emitter of the sound. This allows a very compact design of box and that in theory also it is able to emit sound with in front of flat wave. Most problematic to solve they are the problems of diffraction and interference debt to the very abrupt changes of angle. The two companies reached similar solutions although each one gave a really different final solution actually, as it is possible to be seen in the graphs and following photos. Both designs are very novel by the remarkable amount of space that is saved at the time of controlling the front of wave by means of these reflecting surfaces.

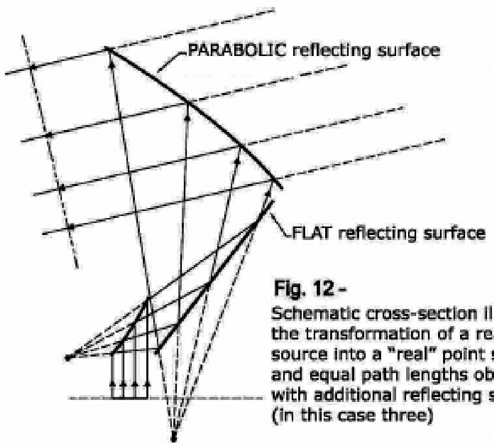
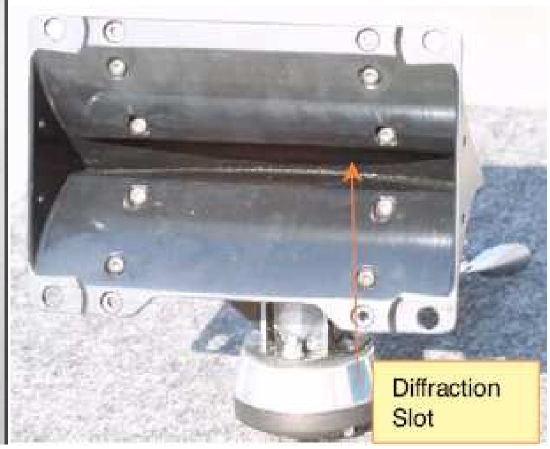


Fig. 12 -
Schematic cross-section illustrating the transformation of a real planar source into a "real" point source and equal path lengths obtained with additional reflecting surfaces (in this case three)

The frontal view:



As much in the Web of Nexo like specially in the Web of Outline, there are available documents to read about his designs. The explanations are enough complete and interesting.

<u>Other Systems</u>

Many other Line Array systems exist like the one of EAW, Renkus-Heinz, Electro-voice, Sound Projects, DAS, JBL, QSC, HK, etc., etc.
In general, all these manufacturers have decided on solutions that in fact are not very complex nor different. Once the systems are understood and known here commented, the others are smaller variations of some of explained here, or its innovation really does not come given by the solution of the high frequency. In addition, this document would become interminable.

Having a clear idea of the basic requirements and essential dices by the Dr Heil, can be known with certain security what systems correctly are designed and which no. What yes it must be clear is that to design a system Line Array is not simple, but not as complex as some great brands have wanted to transmit to the great public.

[1] Beranek, I read ` *Acoustics'*. Mc Graw-Hill. 1954
[2] Olson, Harry F. ` *Acoustical Engineering'*. D. VanNostrand. 1957. Reimpresión by sional Profe s Audio Journals, 1991
[3] Marcel Hurban, Christian Heil and Paul Bauman, ` *Wavefront Sculpture Technology*'>, AES Journal, Vol. 51, n°10.
[4] Christian Heil and Marcel Urban ` Sound multiple Field radiated by sources' AES Convention Paper #3269Article presented AT the 92nd AES Convention, Vienna 1992

[5] Meyer Sound, "Line Array Technical Report"
[6] Perr in Meyer, "The Design and Performance of the REM Ribbon emulation Manifold Waveguider", 2004 February
[7] Perrin Meyer, "DSP Beam Steering with Modern Line Array " 2002 December 1, 2004
[8] Alan Brock Adamson, United States Patent nº 6581719 B2, June 24, 2003
[9] Alan Brock Adamson, United States Patent nº 6628796 B2, June 24, 2003
[10] Bill Webb and Jason Baird, "Advances in Line Array for Technology live sound", October 20, 2003
[11] NEXUS, "GEO Reflective Wavesource Technology", September 16, 2002
[12] Guido Noselli "Butterfly System White Paper", January 2003