LINE ARRAYS: HOW THEY WORK

Some differences with Conventional Systems

Line arrays have been specially designed so when several units are combined vertically the whole system behaves as one single sound source. Generally speaking that's what makes them different from the conventional speaker systems. The horizontal configuration of the box, the distance between speakers, the cut-off frequencies and the design of the highs section are the features that allow for this behaviour.

On the Mids and Lows sections of line arrays there are two specific characteristics with which we may start this discussion. The first, which can be noticed immediately, is that the transducers are vertically assembled in one line, as close as possible to each other. The second of these characteristics cannot be seen, but arises in any discussion about line arrays. It is the crossover frequency of the speakers. As we will see later on, this is an important part of the design of the system and cannot be modified without negative effect on the behaviour of the whole system. These two characteristics have a lot in common and can be explained at the same time. However, for clarity purposes, we have preferred to introduce them separately. And so… to the theory!

The ideal characteristics of a sound system.

There are two acoustic characteristics which would be desirable in a sound system. The first would be the capacity of controlling the vertical directivity, so we can send acoustic energy only where it is needed, where the public is sitting, which would avoid sending it where it can be the cause of problems (like the ceiling of the theatres), or be wasted when we are outdoors. The second desirable characteristic would be that the different individual sound sources sum coherently, behaving altogether as one single sound source, yielding a uniform sound pressure level along the distance. In the next sections we will explain in detail why these two characteristics are desirable and how they can be achieved.

Directivity control. Why?

In fig 1 we can see an example of a system with a poor vertical directivity control. The boxes are vertically assembled and as close as possible to one another. However, the transducers that reproduce the example frequency are at a distance from each other due to the vertical design of the box. The lobe that can be seen on top of the polar response will cause a strong reflection, which will reach the audience area, delayed in time, compared to the direct sound. This will certainly affect the intelligibility.

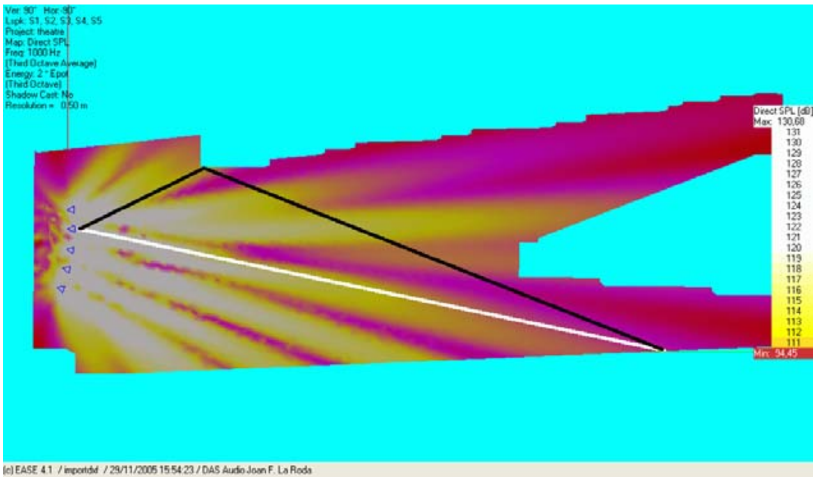


Fig. 1. System with a poor vertical directivity control.

In fig 2 we can see a system with good vertical directivity control. The image shows clearly that the energy is concentrated where the audience is.

Fig. 2. System with a good vertical directivity control.

Directivity control. How?

The vertical coverage angle needed to cover the audience area usually isn't very large. How can we configure a system if we want a narrow vertical coverage? There is a general principle which states that the smaller the distance between speakers of an array of boxes, the narrower the coverage becomes – and - of course the frequency plays a role here too.

We use this general principle in line arrays in order to get a narrow vertical coverage by assembling the transducers in a vertical line as close as possible to each other. The horizontal coverage will be that of one single speaker.

To illustrate this general principle we are going to see an example of how coverage gets narrower as the distance between speakers becomes smaller. I would like to point out here that this can be applied to any system, be it a line array or not. This is not a specific line array phenomenon. The physical laws that apply to line arrays apply also to any other speaker.

In fig. 3, 4 and 5 we can see six conventional boxes. In the first image the centres of the transducers are at a distance of two wavelengths, in the next the distance is one wavelength, and in the last one the distance is half a wavelength. It can be seen very clearly that the shorter the distance between transducers, the narrower and more uniform is the coverage.

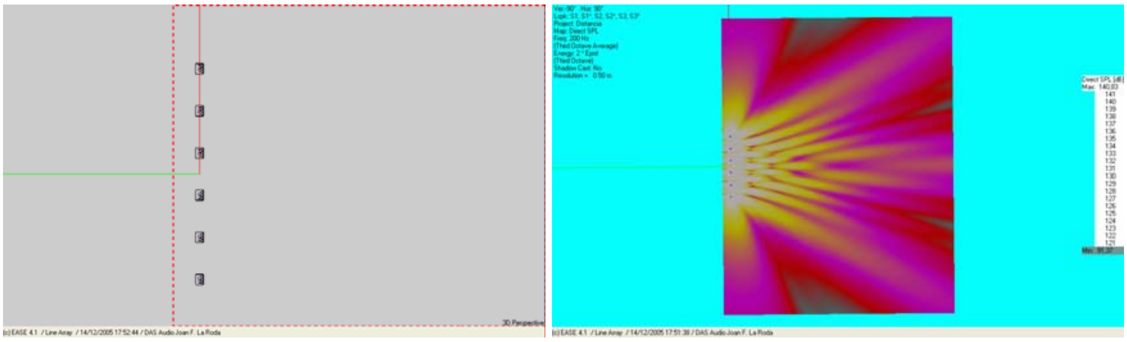


Fig. 3. Vertical polar responses for a distance between speakers 2λ.

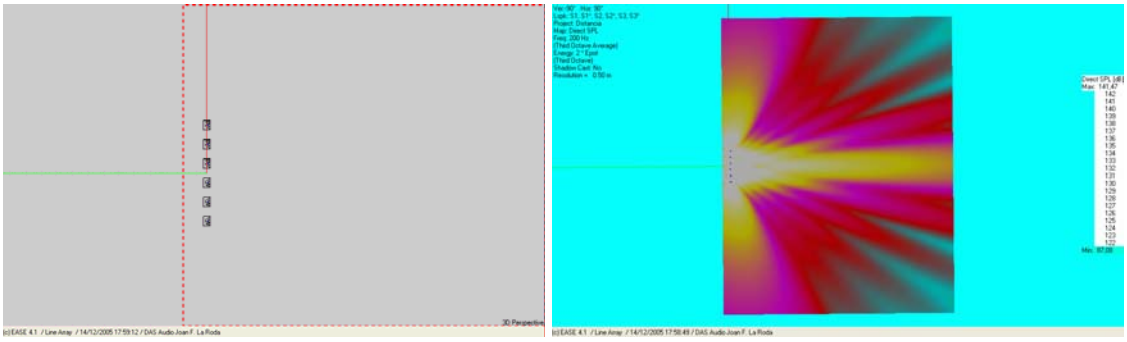


Fig. 4. Vertical polar responses for a distance between speakers λ.

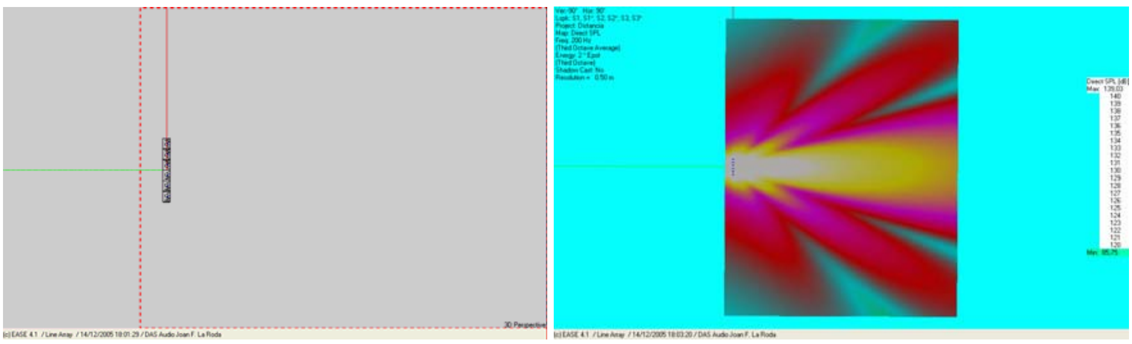


Fig. 5. Vertical polar responses for a distance between speakers λ/2.

Coherent coupling. Why?

Coherent coupling between sound sources will result in a uniform sound pressure distribution along the distance. We can achieve this by avoiding secondary lobes to appear in the vertical polar response. If we pay attention to figures 6 and 7 (to the pictures on the left) we will see how in the first case secondary lobes appear in the vertical polar response, whilst in the second case we can say there’s only one main lobe.

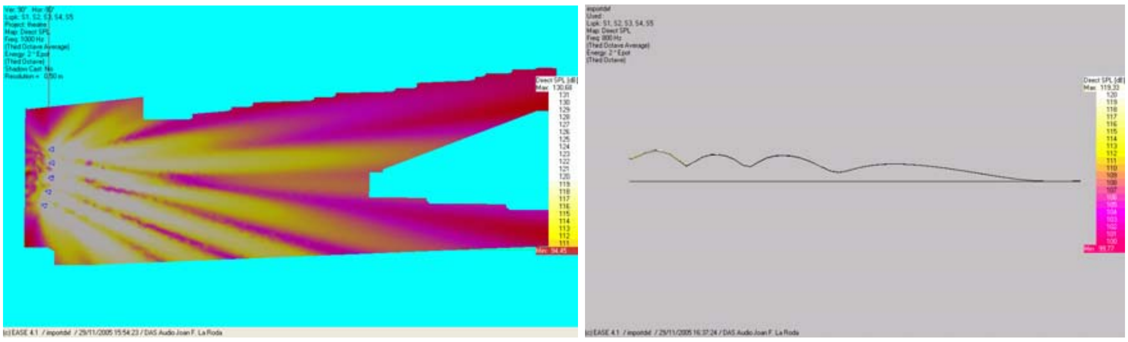


Fig. 6. This system does not yield a uniform SPL distribution along the distance.

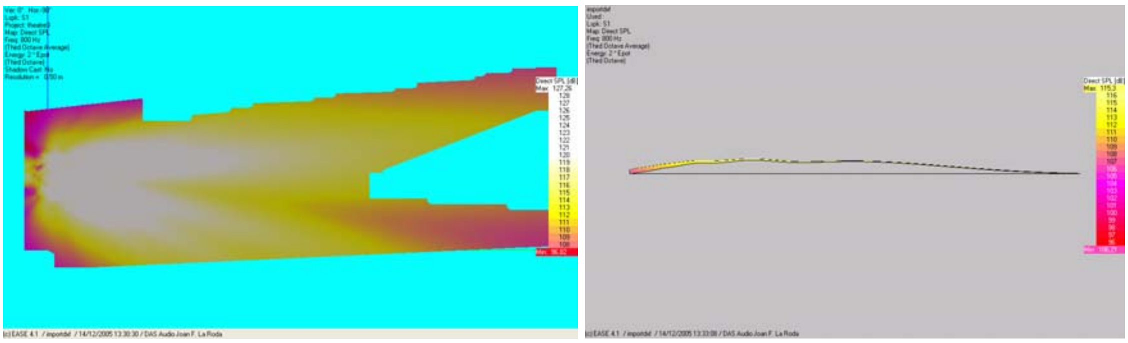


Fig. 7. This system yields a uniform SPL distribution along the distance.

The effect in the pressure distribution along the distance can be seen in the images on the right, which correspond to the SPL on the main floor. In fig. 6 it can clearly be seen that the irregularities in the SPL along the distance fit with the lobes in the polar response. In that case, the frequency balance that can be heard at the mixing console can be quite different from that on another position. In fig. 7 we see that the pressure distribution is more uniform, because there are no lobes in the polar response. A frequency response measurement in several positions will yield more similar results in this second case.

Coherent coupling. How?

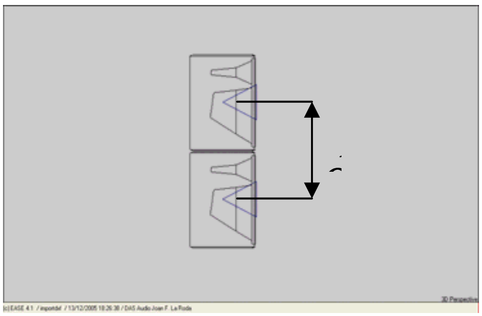
Before discussing how to achieve coherent coupling it's interesting to observe something, which is not mentioned very often. In line arrays the cone speakers are allowed to reproduce only those frequencies at which the polar pattern of each individual speaker component is omnidirectional. We will use this concept later.

In order to find out the coverage of a given speaker according to the frequency we have to study what is known as the k·a factor, where 'k' = 2πf/c and 'a' = the radius of the speaker. When k·a is equal or bigger than 2 secondary lobes appear and the coverage becomes too narrow to be useful in a line array. In table 1 you can see the k·a factor for an 8" speaker at different frequencies.

Frequency	**k·a (8" speaker)**
f = 30Hz	0.105
f = 60Hz	0.211
f = 120Hz	0.421
f = 160Hz	0.562

Table 1. k·a factor for an 8" speaker at different frequencies.

Having said this, to understand how to achieve coherent coupling we will apply again an acoustical principle that we will call "general principle" to emphasize the fact that it applies to any sound source, be it a line array or not. This general principle states that in order to achieve coherent coupling the transducers should be allowed to reproduce only those wavelengths which are large compared to the distance between speakers.



$$\lambda >> d$$

Fig. 8. In order to achieve coherente coupling the wavelengths to be reproduced should be much larger than the distance between the centres of the speakers.

Now you must be wondering precisely how big these wavelengths have to be, compared to the distance between speakers in order to achieve coherent coupling? This question is answered below.

We have said that we don't want secondary lobes to appear in the vertical polar response, in other words we want one single main lobe only. This is the same as saying that we have to design the system so the first minimum of pressure appears at 90º off axis. That way there will be only one lobe, aimed where the audience is. We are going to show an example to understand it. In order to simplify the calculations we will make it with two units only.

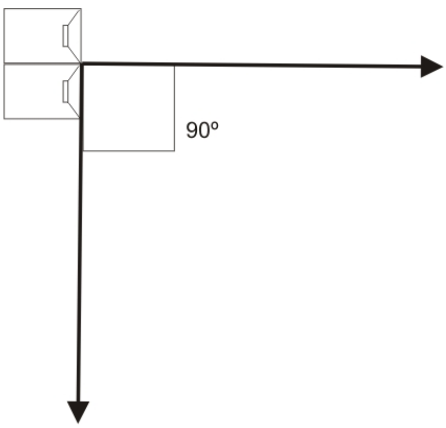


Fig. 9 We will design the system so the first minimum of pressure appears at 90º off-axis.

In fig. 9 we can see two cone speakers assembled in two boxes. On axis both sound waves will arrive at the same time at the listening position, so they will sum in phase resulting in 6dB more SPL than that of one single source, as can be seen in fig. 10.

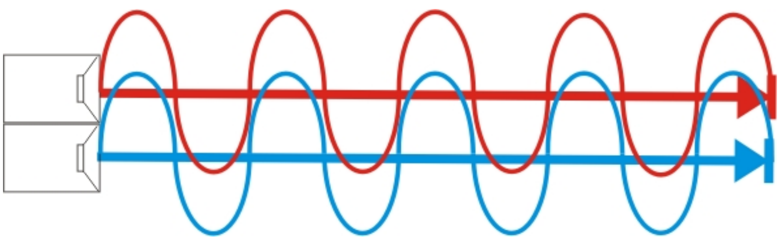


Fig. 10. On axis both sound waves will arrive at the same time, therefore they will sum in phase.

Fig. 11 shows how at 15º off axis there is a path difference $\delta = d \cdot \cos \theta$. In our example, where the distance between speakers is 27cm, the path difference δ is 6.98cm. This path difference represents a phase shift of 46.6º at the example frequency, which is 630Hz. The resultant SPL will be 5.3dB higher than that of one single source.

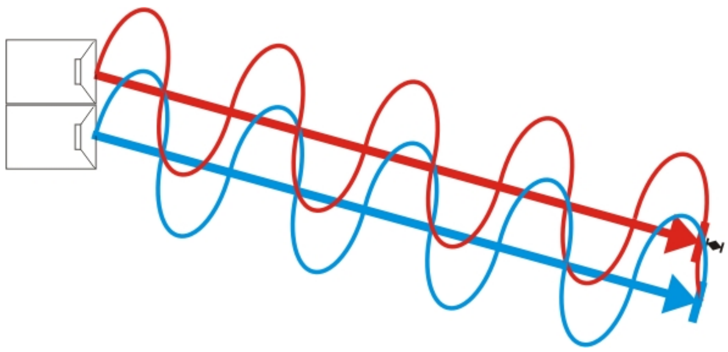


Fig. 11. At 15º there is a path difference, which implies a phase shift between both sound waves.

Table 2 shows the path difference, phase shift and resulting pressure increment for different angles off axis. Two of them are of special interest.

Angle	**Distance increment**	**Phase shift**	**SPL increment**
0º	0cm	0º	6dB
15º	6.98cm	46.6º	5.3dB
30º	13.5cm	90º	3dB
45º	19cm	127.3º	-1.1dB
60º	23.4cm	156º	-7.7dB
75º	26cm	173.9º	-19.1dB
90º	27cm	180º	- Infinite

Table 2. Path difference, phase shift and resulting SPL increment for different angles off-axis.

The first result worth mentioning is that at 45º off-axis. At that angle we can see that the SPL is 1.1dB lower than that of one single source. When the phase shift between two sound waves is 120º the resulting SPL is the same as that of one single sound source. As the difference in phase increases the resultant SPL decreases, being lower than that of one single source, as above.

The second interesting result is that of 90º off-axis. We can see in fig. 12 that both waves are now in exact counter phase or, in other words, the phase shift between them is 180º. This obviously results in total cancellation, because of the 180º phase difference and because the levels of both units at that position are the same. Remember that the polar pattern of each individual speaker component is omnidirectional, as has been said at the beginning of this section.

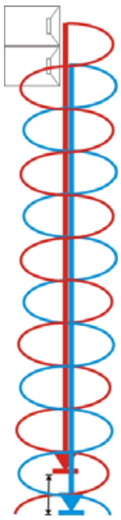


Fig. 12. Sum of both sound waves at 90º off-axis.

In fig.13 we can see the polar response of this system at the example frequency. It can be seen how at this frequency one single lobe appears, and no secondary lobes show up, just as we wanted it.

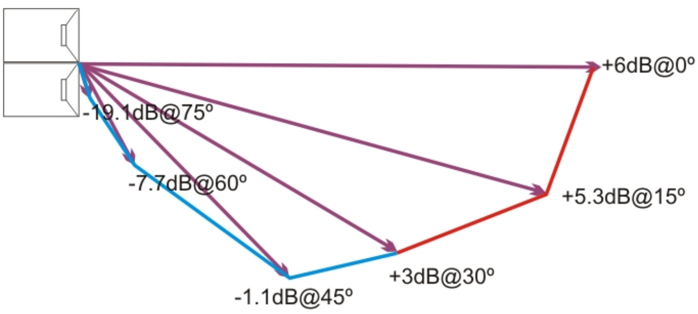


Fig. 13. Resulting vertical polar plot of both units.

The way to avoid secondary lobes to appear in the polar response, which would yield a lack of uniformity in the SPL along the distance, is by limiting the band of frequencies to be reproduced by each transducer to those at which the distance between speakers d is equal or smaller than half a wavelength. The well known equation:

$$d <= \lambda/2$$

The smaller wavelength λ ($d = \lambda/2$) to be reproduced is that of the higher cut-off frequency. The wavelength of the cut-off frequency is larger than d, exactly two times larger. The lower frequencies will be even larger, as stated in the equation of fig. 8,

$$\lambda >> d.$$

This shows why the cut-off frequency should never be modified in one of these systems. We risk allowing the cone speakers to work at frequencies at which secondary lobes appear in their response, messing up the uniformity of the SPL along the distance, which is one of the main benefits of line arrays.

Can we use the same technique to reproduce the high frequencies?

Clearly not. We have just said that in order to get coherent coupling between sound sources, and to avoid secondary lobes to appear in the vertical polar response, the distance between sources must be smaller or equal to half the wavelength of the highest frequency that is to be reproduced. In the case of the highs, the higher frequency is 20kHz, and the wavelength for 20kHz is 1.7cm. There isn't a high frequency transducer of 0.85cm diameter powerful or efficient enough for professional sound reinforcement applications. Therefore we'll have to search for a different technique to reproduce the highs in our line array.

Ideally, on the Highs, we would want a continuous vertical sound source capable of reproducing an isophasic wavefront. From such a sound source we would get a vertical coverage similar to the one delivered by the assembled Mids and Lows sections of our line array boxes, avoiding secondary lobes given that the whole surface would radiate in phase.

We can make a real system to approximate the ideal one by assembling discreet rectangular shaped sound sources, vertically in a line, as in fig. 14, as long as the distance between them is such that the radiating surface counts at least for 80% of the total surface. It must be taken into account that the discreet radiating surfaces will be separated by the thickness of the enclosure. A restriction regarding the total-surface/radiating-surface ratio is therefore necessary in order to make sure the sum of radiating surfaces approximates an ideal continuous source.

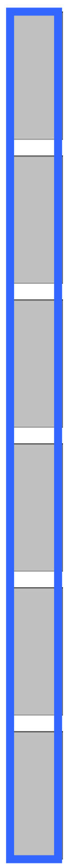


Fig. 14. The sum of discreet rectangular shaped radiating surfaces behaves like one single sound source.

The waveguide.

If we want to obtain a rectangular shaped radiating surface from the round shaped exit of a compression driver we will need a waveguide, which is the key piece in a line array. In the case of the DAS CA28 we use the waveguide in fig. 15.

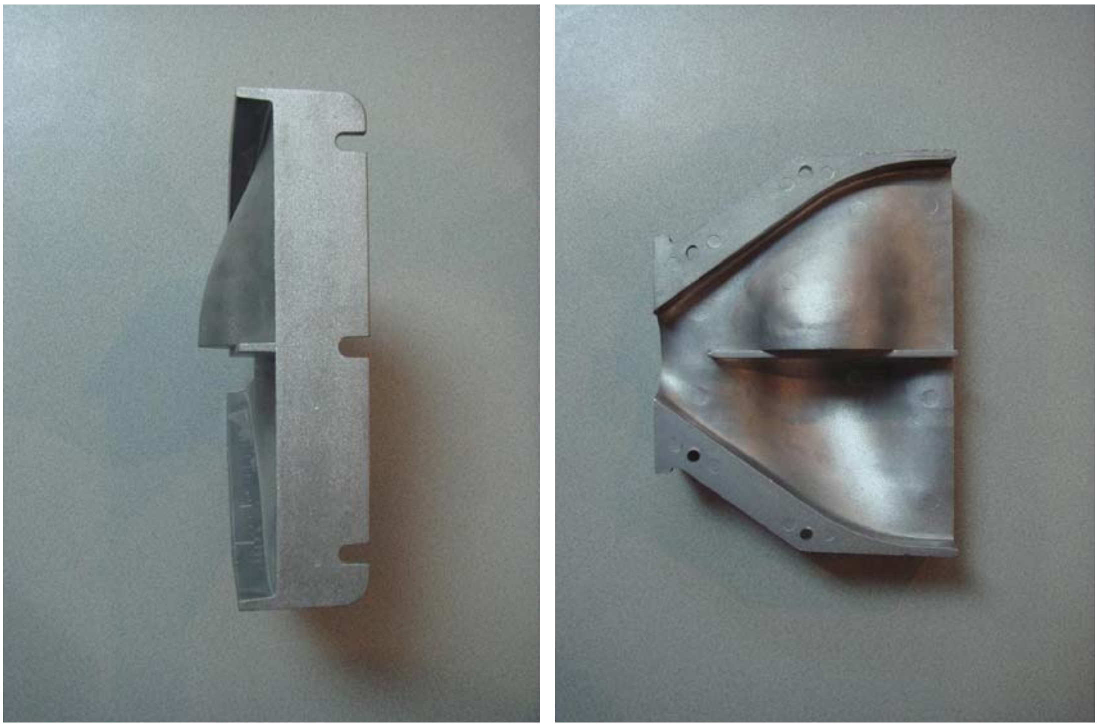


Fig. 15. Half a waveguide seen from the front (left) and one side (right)

This waveguide features a curvature, which is curved more strongly in the centre than it is in the higher and lower parts. If it was not designed this way, the wave would arrive first at the centre and later to the upper and lower parts, resulting in a curved wavefront which wouldn't be useful in a line array because it would produce interference between adjacent sound sources, being the cause of lobes in the polar response.

Fig. 16 shows how the wavefront would be with and without the curvature in the waveguide.

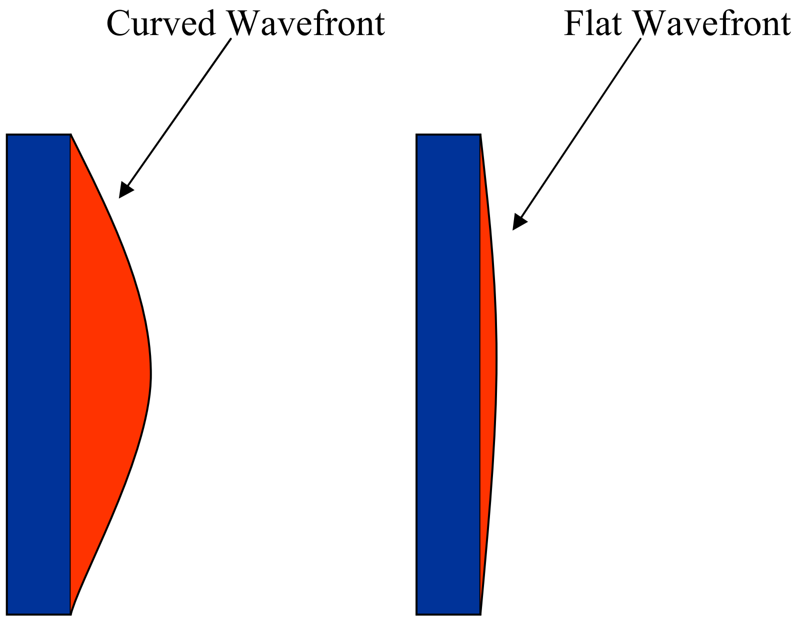


Fig. 16. Wavefront obtained from a waveguide without length correction (left) and with length correction (right)

The waveguide then is responsible for adding an acoustic delay, making the path length the same from the exit of the driver to the centre and to the upper and lower sides of the rectangular radiating surface. That way we get an isophasic wavefront, which will provide a narrow vertical coverage, avoiding interference with the adjacent radiating surfaces as we have already said. Other existing waveguides use a different technique in order to achieve the same result.

Main characteristics of line array systems.

To summarize everything that has been said so far, it would be interesting to highlight the three main characteristics that set apart line array systems (at least those that use direct radiation) from conventional systems. If a line array with direct radiation speakers (no acoustic lenses…) lacks any of these characteristics it can't be said it will behave like a line array. These characteristics are the following:

1.- The cone speakers are assembled in a vertical line, as close to each other as physically possible, which will contribute to the control of the vertical directivity.

2.- The highest frequency a cone speaker will reproduce (the higher cut-off frequency) will be approximately the one that corresponds with the wavelength defined by the equation

$d = \lambda/2$, where d is the distance between the centres of the membranes.

3.- The Highs section will make use of some kind of waveguide which will produce an isophasic, or almost isophasic, wavefront (depending on the design of the system and its application)

Joan La Roda
Audio Engineer

techart1@dasaudio.com
www.dasaudio.com
T. +34 961 340 860
F. +34 961 340 607